IMPULSE ALIGNMENT OF LOUDSPEAKERS AND MICROPHONES

by Tom Lubin and Don Pearson

PART ONE

When the earliest recordings were done there was little if any attention given to the acoustic phase or electrical polarity of the mechancial devices used to record and reproduce sound. Phase and polarity have little significance as long as only one microphone picks up the sound and one speaker plays. When recording left the experimental stage it became possible to mix together more than one microphone. This allowed for better control and balance among the instruments.

With advancements in technology, multi-microphone techniques developed. In not too long a time the recording engineer discovered that occasionally when the outputs of two microphones were combined their summed output level would be less than the output of each one separately. In some cases the cancellation was almost complete and affected all frequencies. In other instances cancellation occurred at certain frequencies only. Thus, electrical polarity and acoustic phase cancellation became observable problems with the increased use of multi-microphone techniques.

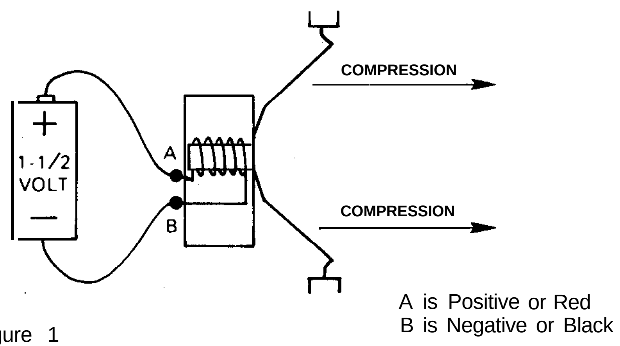
Similar problems existed when monophonic reproduction became stereo. The electrical phase relationship between the two speakers had to be the same. Multi-speaker systems have made the problem of polarity and phase even more critical as each element must be connected correctly. This may not necessarily mean that the electrical polarity be the same for all of the speakers. Acoustic phase cancellation occurs in multi-speaker systems as well, but not until very recently had it been acknowledged, possibly because it is less distinct than the cancellation which occurs when two or more microphones which are picking up the same sound are electrically combined. When our ears mix the signal from two or more speakers what we hear is influenced by the acoustics of the room, and the fact is we have ears instead of amplifiers doing the combining. Before describing a technique for evaluating phase, polarity and a number of other aspects of speaker system analysis, an explanation of polarity and phase should be given.

Polarity and phase are relative terms. Polarity refers to the property that physical quantities have of being greater or less than some reference value that we may arbitrarily designate as the point of reference or "zero." A point on a line may be thought of as being closer to an observer than another point thought of as a reference or farther away than that same reference point. Its position may be described as corresponding to a positive number in one case and to a negative number in another. A voltage may be thought of as being positive with respect to one reference potential and another voltage may be observed to be negative referenced to that same potential. Both voltages may be either positive or negative when referenced to the potential of the earth which, by the way, may not be resting at zero with respect to the universe.

Phase is a term that is implicitly linked to an ongoing time sequence of two or more series of events as observed relative to some common reference point in time. Events that are considered to be "in phase" are events that have their time sequences of increase and decrease occurring simultaneously. Events that are said to be "out of phase" occur in such a way that their increasing and decreasing sequences do not occur precisely together. The measure of the difference in phase is always expressed as a time relation, be it in terms of actual seconds, minutes, hours, etc., or as relative time in terms of increments or fractions of complete cycles of events, such as in units of degrees or radians. It is clear that two or more events may be precisely in ***phase*** with one another while being of either positive or negative polarity. Phase and polarity are related although one is not precisely identical to the other.

Electrical Polarity

Electrical polarity in a speaker is defined in terms of whether the speaker ***condenses*** or ***rarifies*** when it is energized by a



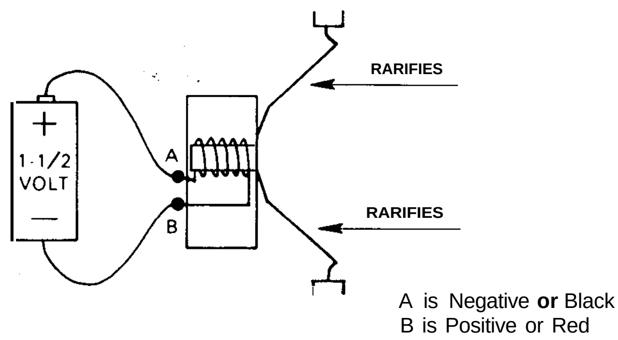


Figure 1

positive pulse (Figure 1). Most manufacturers indicate with colored binding posts a difference between the speaker terminals. Unfortunately, inconsistant or incorrect polarity distinction is quite common, partly due to erratic quality control and the fact that some manufacturers wire their elements opposite to other manufacturers. The polarity of a woofer can be easily determined with a 1% volt battery. When the speaker leads touch the battery terminal, the cone will move in or out. If it moves out the terminal touching the positive side of the battery is positive in that it condenses the air. If the cone moves in then the terminals are reversed. The negative side of the battery is connected to the positive side of the speaker causing the cone to rarify.

Unfortunately, high-frequency speaker elements cannot be checked in this manner because the diaphram movement is so slight and is usually difficult to see since it is usually deep inside the horn.

Crossovers

In systems that use a number of full range speakers the phase

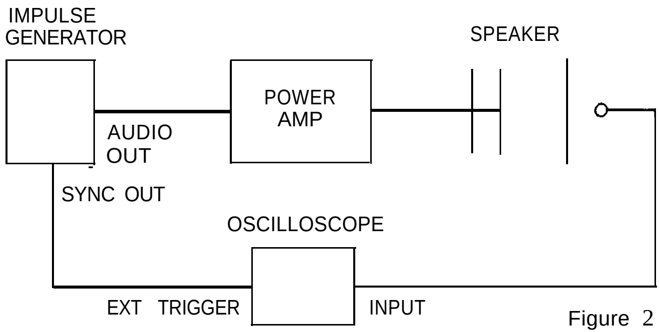


Figure 2

should be the same for all units, however with the use of specialized speakers which reproduce only one area of the audio range, a crossover of some type must be introduced into the audio path. Almost all crossovers introduce some degree of phase shift in order to achieve a sufficiently steep roll off to both sides of the pass band.

For example, let's say we have a three-way network that crosses at 500 Hz and 2,000 Hz. At 500 Hz both the woofer and the mid-range are reproducing a signal that is 3 dB down from their respective full power passband levels. At that frequency both of them are theoretically reproducing an equal acoustic power level, so their on-axis response will sum by 3 dB. If they sum by 3 dB, and they are both down by 3 dB at the crossover point then the system should have a flat response providing all the phases are correct. But what is the correct phase? That's the crunch.

The degree of phase rotation introduced by the crossover will vary from unit-to-unit, but can be considerable. Almost always the phase of the roll-off/roll-on of adjacent bands will rotate in opposite directions at the crossover frequency. The net result will have the mid-range acoustically out of phase with the woofer at the crossover frequency because of the phase shift in the crossover.

In order for the entire system to be acoustically in phase at the crossover point, it may be necessary to alternate the speaker polarity of each adjacent band. Manufacturers of crossovers fail to meet this need as phase reversal switches are seldom provided. The fact that the speakers of two adjacent passbands are electrically out of phase is of no consequence since it is only at the crossover points that they share common information, and must be acoustically in phase with one another.

Finally, a speaker is theoretically a single point source of sound. With the addition of each speaker to a system, the

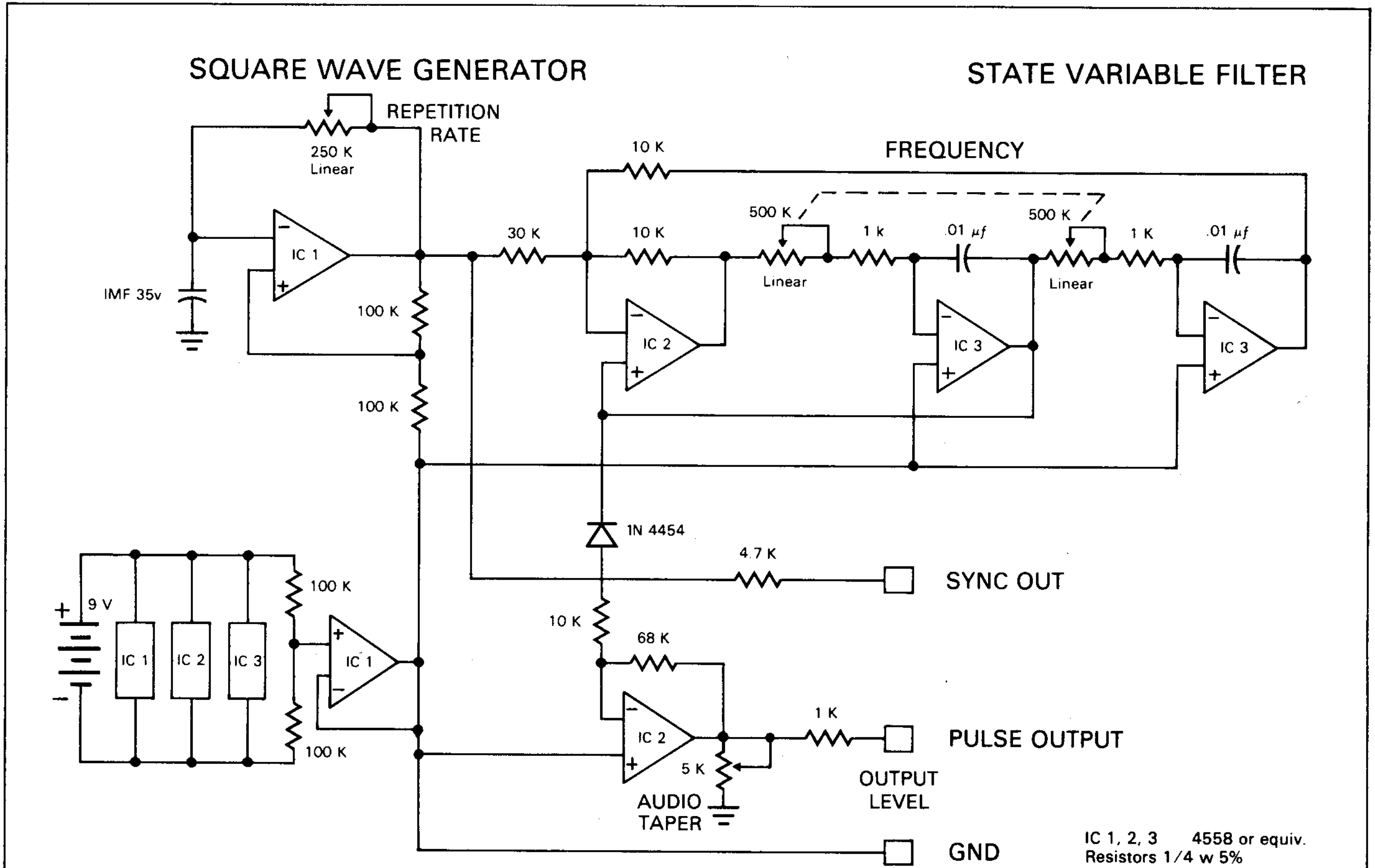

SQUARE WAVE GENERATOR
STATE VARIABLE FILTER
REPETITION RATE
250 K Linear
FREQUENCY
10 K
30 K
10 K
500 K
Linear
1 k
.01 μf
500 K
Linear
1 K
.01 μf
IC 1
IMF 35v
100 K
100 K
IC 2
IC 3
IC 3
1N 4454
4.7 K
SYNC OUT
10 K
68 K
9 V
IC 1
IC 2
IC 3
100 K
IC 1
100 K
IC 2
5 K
AUDIO TAPER
1 K
PULSE OUTPUT
OUTPUT LEVEL
GND
IC 1, 2, 3 4558 or equiv.
Resistors 1/4 w 5%

Figure 3

number of point sources increases. If each point source is not exactly the same distance from the listener then what occurs is an auditory double image. This is particularly apparent at the crossover points. Basically, a single moment in time is generated by all the speakers at the same instant, but arrives at the listener at a number of different times. This causes the intelligibility of the entire system to be lowered.

Clarity is also affected by "out of band" distortion generated by the enclosure. If a cabinet is not adequately braced it will resonate or "ring" substantially. The box can put out almost as much sound as the speaker itself. Likewise metal horns, if not properly dampened, can contribute undesired vibrations.

The solution to many of these problems is fairly simple once an accurate method of measurement is provided. Operating as an independent testing service, Don Pearson and Gary Leo, of *Ultra Sound,* located in Larkspur, California, has developed such a system. With the aid of their computer they provide information on all of the previously mentioned problems as well as a number of other acoustic and electronic parameters relating to the sound equipment used by their clients. A permanent record is kept, in the form of a printout, for these clients who number several very prominent sound reinforcement companies.

A Simpler Method

Ultra Sound has allowed the publication of a simple circuit which when constructed can be used with a microphone and an oscilloscope to measure polarity, phase of a wave, and ringing. Figure 2 is the schematic of an "impulse" generator with variable frequency and repetition rate. Figure 3 illustrates the proper hook-up of the system. Any oscilloscope with an external trigger will do as will any conventional microphone. Ultra Sound prefers a Nakamichi CM 300 fitted with a CP 3 super-omni ½-inch element. The microphone should be placed a few feet from the speaker (the larger the speaker, the greater the distance between the mike and speaker). The speaker's impulse output should be loud enough to mask any room noise.

Before testing can be done, the polarity of the microphone to be used as the standard must be determined. First connect the

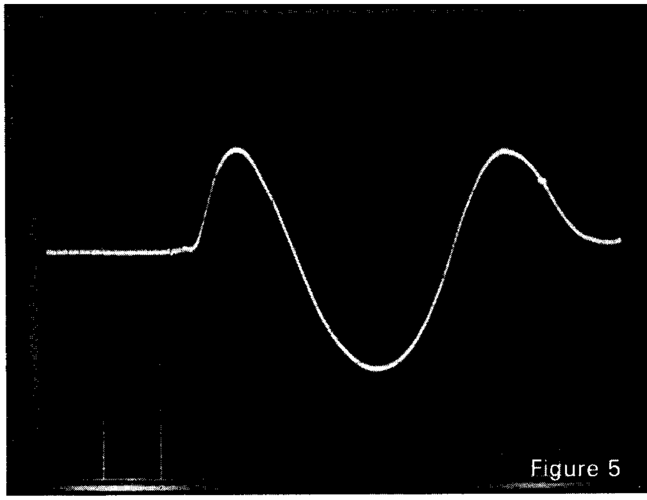


Figure 5

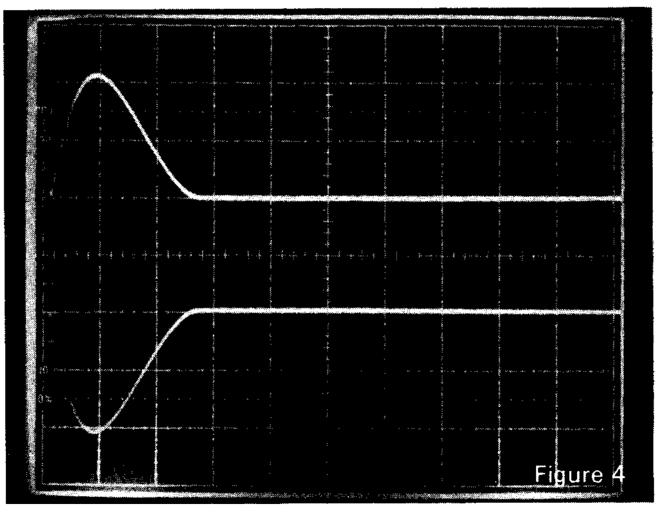


Figure 4

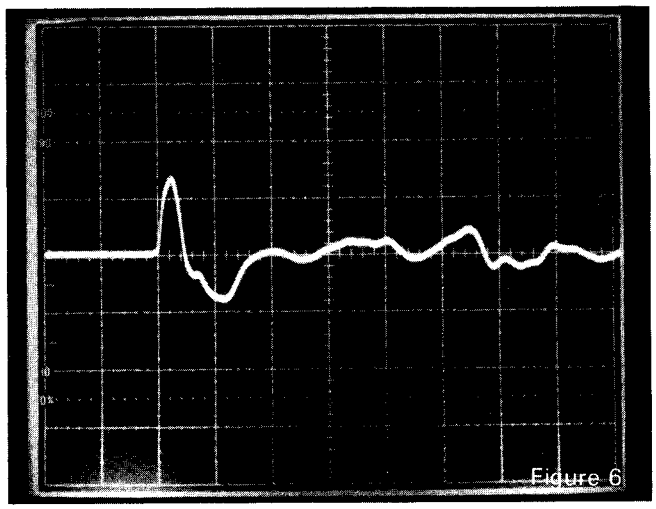


Figure 6

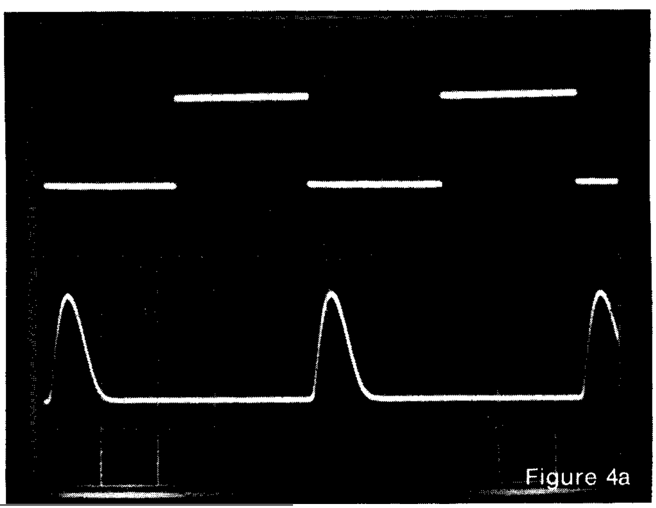


Figure 4a

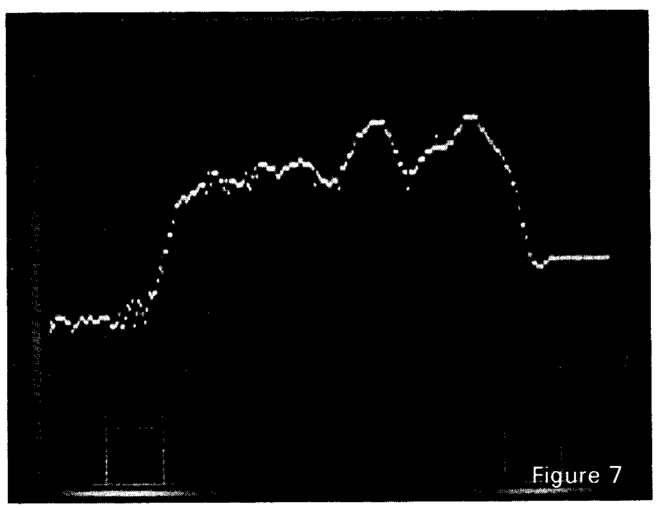


Figure 7

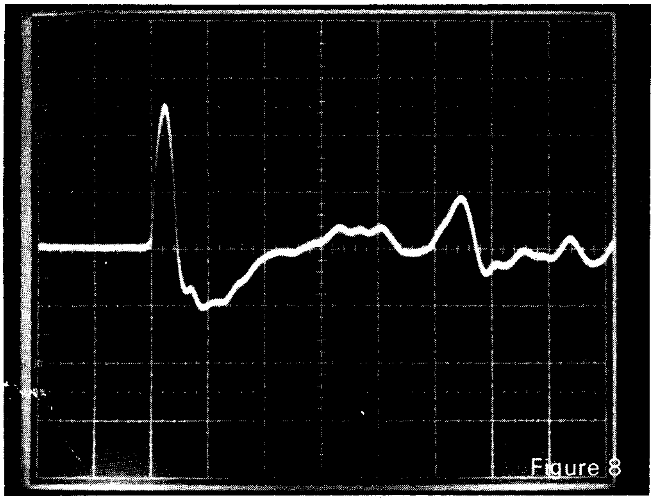


Figure 8

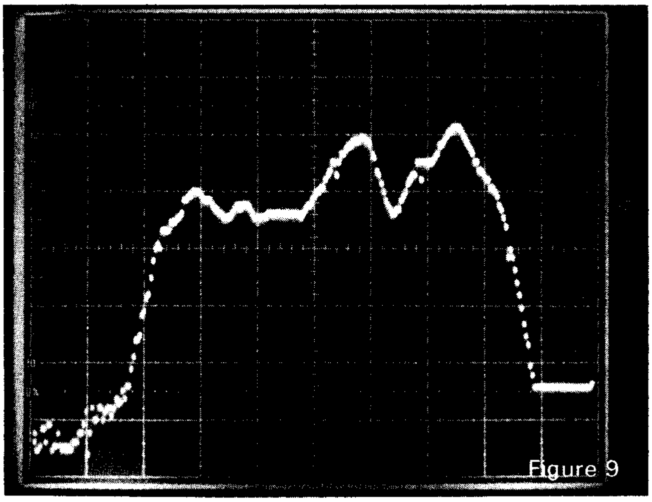


Figure 9

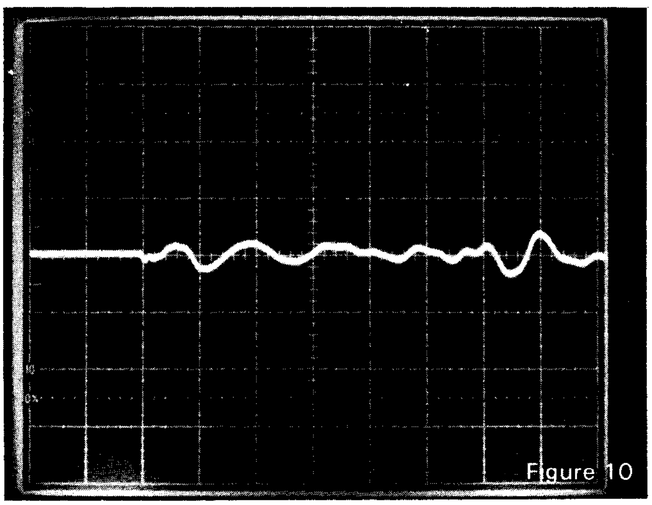


Figure 10

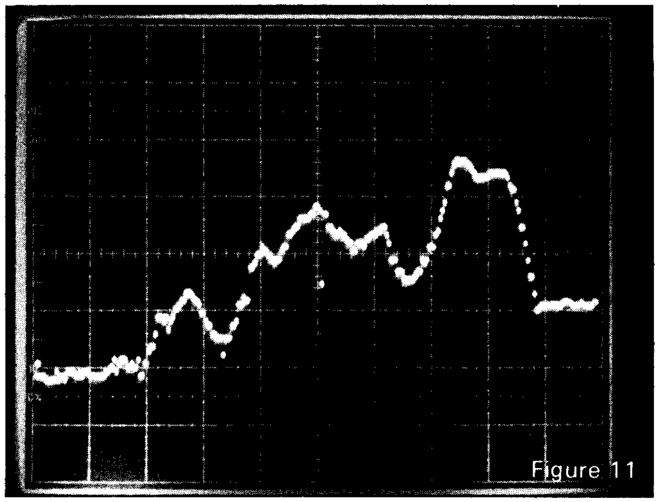


Figure 11

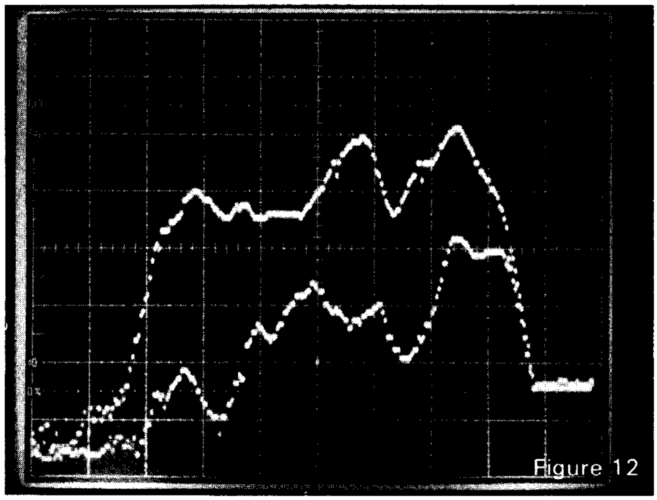


Figure 12

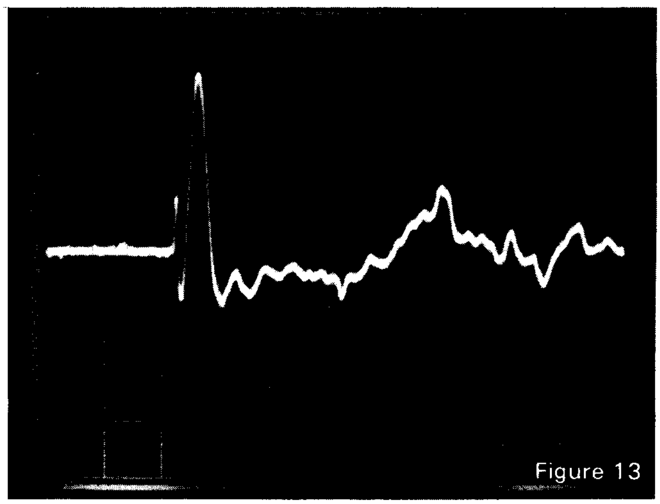


Figure 13

output arriving at the speaker to the input of the scope and observe whether the waveform is positive or negative going (Figure 4). Reconnect the speaker to the amp and connect the microphone to the oscilloscope and again observe the polarity. While observing the scope, adjust it's trigger from the generator so that the screen shows the waveform that first arrives at the microphone and not a later reflection. At the lowest frequencies, it is possible for the first reflection to look very much like the impulse (Figure 5).

Once the relative polarity of the test set-up has been determined and the trigger properly adjusted, testing can begin. Whether the test set-up is positive or negative is not important as long as all the adjustments result in the pulse going in the same direction. With the generator putting out a mid-frequency pulse, Figure 6 shows the type of results expected from a five-inch full-range speaker. The erratic waves after the pulse are reflections from the room where the tests were made. As mentioned, the frequency response (Figure 7) will not change regardless of the polarity in a single speaker system.

Figure 8 is the impulse measurement of two five-inch speakers with identical polarity. Figure 9 is the response of the pair. Figure 10 has the polarity of the two speakers opposite one another. Figure 11 is the resulting frequency response. Figure 12 is an overlay of Figure 9 and Figure 11. The efficiency of the two curves was maintained so that a direct comparison could be made. The top trace is of the speakers with identical polarity. The bottom trace has them opposite.

Now that the basic procedure and how to read the impulse is understood, let's take a look at a two-way system. Adjust the frequency control of the generator to a low frequency and observe the polarity of the bass speaker. Re-adjust the frequency on the generator to the crossover frequency and observe the polarity of the woofer and tweeter (Figure 13). The small first peak is the tweeter and the second larger one is the woofer. Both

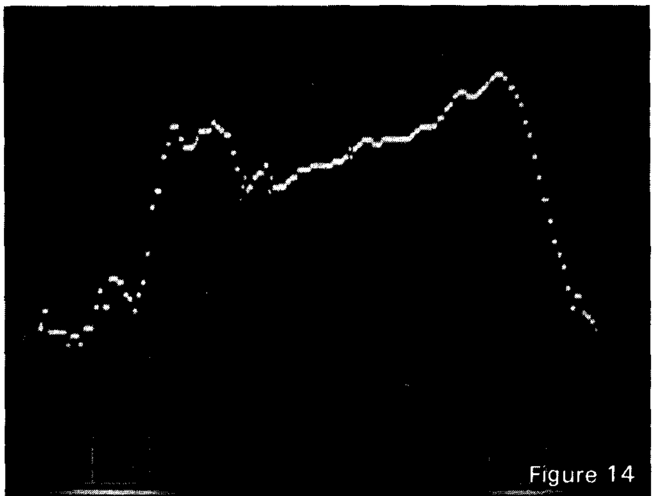


Figure 14

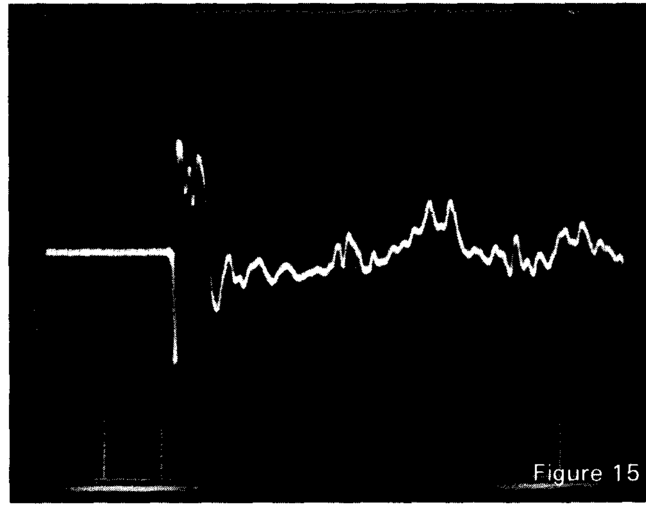


Figure 15

are in the same polarity. Figure 14 shows theresponse. Figure 15 is the same system but with the tweeter polarity opposite that of the woofer. Figure 16 is the resulting response showing a substantial dip at the crossover frequency. Figure 17 again shows a comparison between the system in phase and out of phase response.

The same procedure can be followed with three- and four-way systems. The frequency of the generator is changed to each of the crossover frequencies until the polarity of all the transducers has been determined. If there is more than one driver in any one band range it will be necessary to move the microphone close (one inch) to each driver to eliminate interference from the other drivers in that range.

The effect of band-to-band polarity considerations will result in either a peak or a notch in the frequency response at those crossover points.

Another use of the impulse is to check the polarity of microphones. Once the polarity of the test microphone has been determined, other microphones can be compared to it by connecting each microphone in turn to the oscilloscope. If any of the mikes are out of phase with the majority, then reverse the signal wires on that mike's connector. It should be noted that most European microphones are opposite in polarity to American ones.

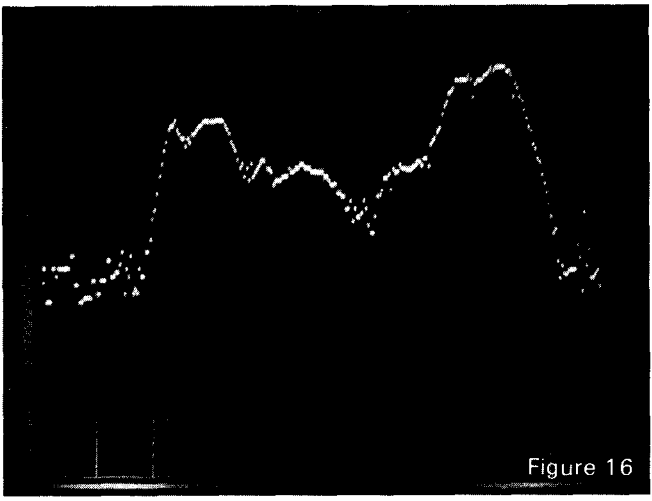


Figure 16

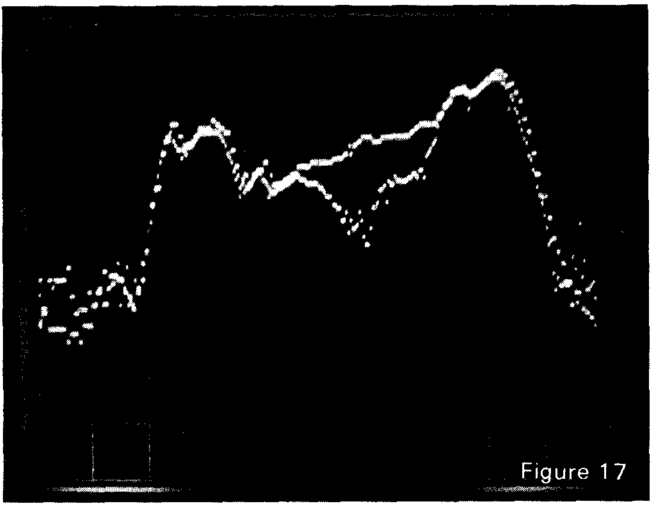


Figure 17

In the second part of this article the impulse will be used to measure cabinet ringing and acoustic phase alignment.

For additional information on this topic Don and Gary have provided the following sources:

"Acoustical Measurements by *Time Delay Spectrometry,"* R. C. Heyser. AES, October, 1967, p. 370.

"Impulse Measurement Techniques for Quality Determination, "in Hi-Fi Equipment, with special emphasis on Loudspeakers. JAES vol. 19, p. 101, 1971, A. Schaumberger.

Applications of impulse measurement techniques to the Detection of Linear Distortion, Alfred Schaumberger. JAES September 1971, p. 664.

Linear Distortion, D. Preis. AES, June 1976.

The Application of Digital Techniques to the *Measurement of Loudspeakers,* J. M. Berman and L. R. Fincham. AES, June 1977, p. 370.

Three Dimensional Displays for Demonstrating Transient Characteristics o/Loudspeakers, Tsutomu Suzuki, Takshi Mor II, and Sumitaka Matsumara. AES, July-August 1978, p. 511.0

IMPULSE ALIGNMENT OF LOUDSPEAKERS AND MICROPHONES

PART TWO

by Gary Leo and Don Pearson

Part one of this article (R-e/p, *December, 1978)* described a method for observing the electrical polarity of speakers and microphones with respect to a known source. It included a circuit and a test setup for making these observations. It then went on to show how to determine the polarity between drivers in a multiband speaker system and how the frequency response would suffer if the drivers were connected inproperly.

ABOUT ULTRA SOUND:
In addition to consulting and design engineering, Ultra Sound offers custom construction of one-of-a-kind electronic projects for sound reinforcement applications and rental of electronic equipment.
A recent project was construction of the six-way stereo electronic crossover used during the final shows at San Francisco's Winterland Arena on December 31, 1978. It was designed for Bill Graham's F.M. Productions as specified by the Grateful Dead. Ultra Sound is supplying a monitor system to the Jefferson Starship, presently rehearsing in San Francisco.
Currently, Ultra Sound is in the process of interfacing its custom test equipment to its computer.
A partial list of Ultra Sound's better-known clients include WAH Sound, Sacramento, California; Starfine Sound, Hard Truckers Speakers, and The Grateful Dead, all San Rafael, California: and Hot Tuna, San Francisco; California.

There are a couple of additions and corrections to Part One of Impulse Alignment. First, picture 4A had no explanation. It shows the signals generated by the circuit shown. The top trace is the signal for the oscilloscope sync input, while the bottom trace shows the signal after it has gone through the bandpass filter (pulse output). When using this circuit, adjust the oscilloscope so that it is being triggered on the negative edge of the pulse.

The circuit itself needs one minor correction. The jumper, connecting the output of Op-Amp #2 to the wiper of the 5 kilohm potentiometer going to the pulse output should be omitted. The final addition is the calibrations on all the frequency response measurement photos and they are identical to those in Figure 3 of this article.

To begin this, the final article on impluse alignment, a few points are in order. If your oscilloscope does not have sufficient sensitivity to view the microphone signal directly, then a microphone preamplifier should be used. It should be pointed out that there is no general rule to follow with regard to consistancy of phase polarity or whether there should be alternating polarity between crossover outputs. This is determined by the taper and slope of the crossover in use.

There are several things that can be done to improve a sound system. First, in a *testing situation,* test the components and make improvements, get rid of rattles and vibrations. Then find the proper alignment and mark the speaker locations relative to each other so that the system can be returned to this configuration every time it is installed.

When the sound system is in use:

1 - Use an impulse test to verify that all of the drivers are connected in their proper polarity, and optimize the alignment. This may be omitted if nothing has changed since the last use or alignment.

2 - Use pink noise and a real time spectrum analyzer to adjust the system for flat response by adjusting crossover output levels and polarity. Then adjust the system equalizer.

3 - The system is then adjusted to sound good by the person mixing. This is most critical and someone with proven experience will be able to please the greatest number of people. It is not uncommon for a sound system to be hired because of the ability of the personnel operating it rather than its hardware.

Part two will discuss the observation of and problems associated with enclosure resonances and some of the corrective measures that can be taken using the same basic test set-up. It will also demonstrate a method for observing and adjusting the arrival time of the signal from each of the speakers in a multi-driver system.

Common Applications

Figure 1 depicts a typical waveform that should be received at a microphone. Figure 2 shows the impulse response of a woofer in a badly braced speaker cabinet. Note that audible ringing is demonstrated.

Figure 3 is the frequency vs. amplitude response of this woofer/enclosure combination. The peak in the frequency response at 4 kHz is a result of cabinet resonance. The peak in the frequency response plot is due to the resonances that the impulse generated in the enclosure. Figure 3 shows that the acoustic output of the cabinet resonances can be almost as loud as the signal coming from the speaker. The crossover network can be partially responsible for some of the ringing as demonstrated in Figures 4 and 5. Figure 4 is a three-pole Butterworth bandpass (18 dB/octave) filter (800 - 7 k) while Figure 5 is

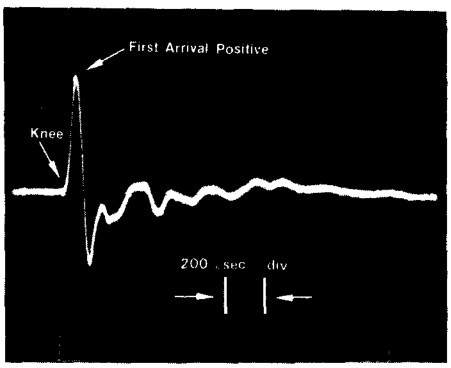


Figure 1. Typical impulse received al microphone. Hororontal scale = 200 microsecond/div.

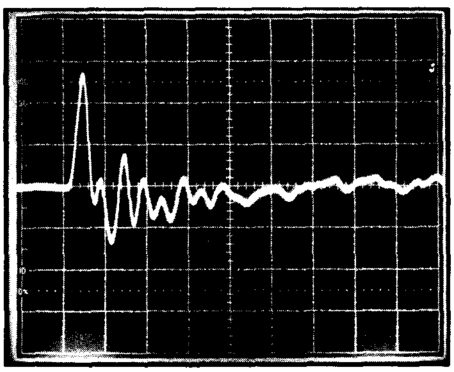


Figure 2. Impulse response of badly braced cabinet.

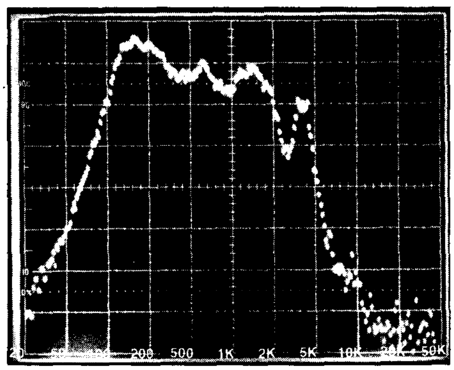


Figure 3. Frequency vs. Amplitude Response of Figure 2.

Figure 4. Impulse response of 18 dB/oct Butterworth bandpass crossover 800-7 kHz.

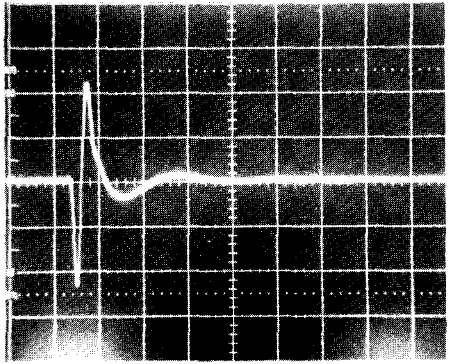


Figure 5. Impulse response of 24 dB/oct Chebyshev bandpass crossover 250-4 kHz.

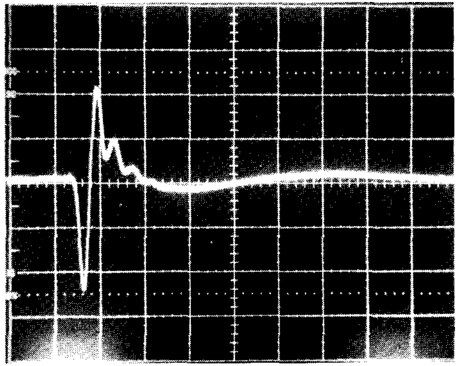
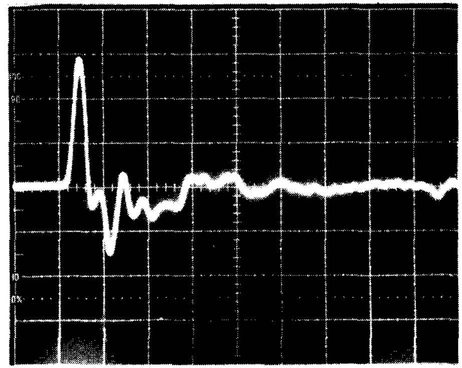


Figure 6. Impulse response of the system shown in Figure 2, after bracing.



four-pole Chebyshev (24 dB/octave) filter (250 - 4 k).

Sweep the pulse output control of the signal generator to different settings and observe the received waveform. The amount of ringing will vary with different settings. The place where the waveform is modulated the most is where the cabinet resonances (ringing) is maximized. Leave the generator set and feel the cabinet with your hands. You should be able to feel the actual panel vibrations. At this point an accellerometer may be substituted for the microphone and placed against the speaker cabinet and walls. The test system is now like a stethoscope for the speaker system.

Much can be inferred by striking the cabinet in different places. You should hear a dull thud. If you hear vibrations, rattles or gong type sounds (extended decay time), then some corrective action must be taken. The air pressure inside the cabinet normally couples the speaker cone to the side walls. As the speaker functions, the cabinet side panels start to vibrate. These vibrations, unlike acoustic resonances, can be different from the passband frequencies generated by the speaker. Acoustic resonances play a part in the ringing problem, since a great deal of power in the bass region will generate sympathetic resonances in an improperly constructed or braced cabinet. Horn-type transducers also exhibit bell type sounds which can be excited with a mallet or similar object.

Enclosure Construction and Bracing

There is no standard procedure for solving the enclosure ringing problem. Bear in mind, though, that if a surface can move it will resonate at certain frequencies. Constructing a properly braced cabinet usually reduces this problem significantly. Ultra Sound has researched the ringing problem with Hard Truckers Speakers, of San Rafael, California, to improve enclosure construction and bracing.

Some of the indications used are as follows:

- Any surface with a span greater than two feet should be braced with a piece of wood across the surface; cross bracing should also connect opposing walls.
- If there are any removable panels, they should be screwed down around the perimeter and also to cross bracing.
- The thickness of the enclosure wall is

Figure **7. Frequency response of the system shown in Figure 3, after bracing.**

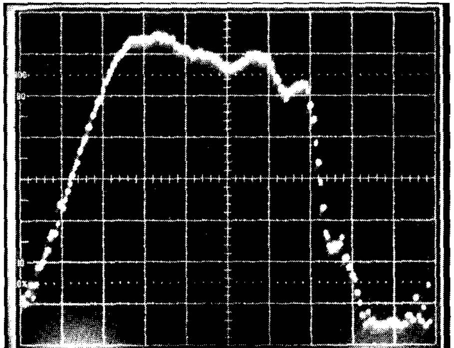


Figure 8. **Arrival time difference — showing both woofer and mid-range pulse. (Difference = 1.6 divisions x 0.5 milliseconds/div = .8 ms or 10.8 inches)**

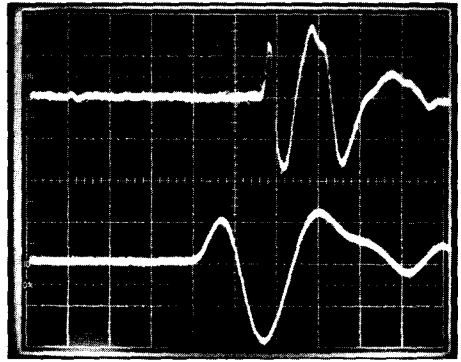
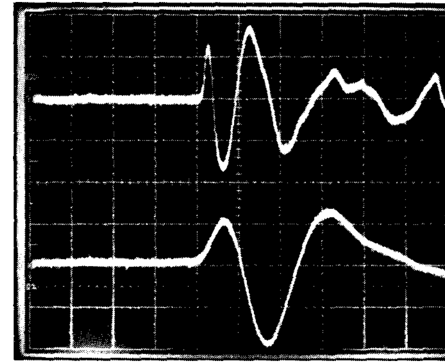


Figure 9. **Same as Figure 8 with speakers adjusted for equal arrival time.**



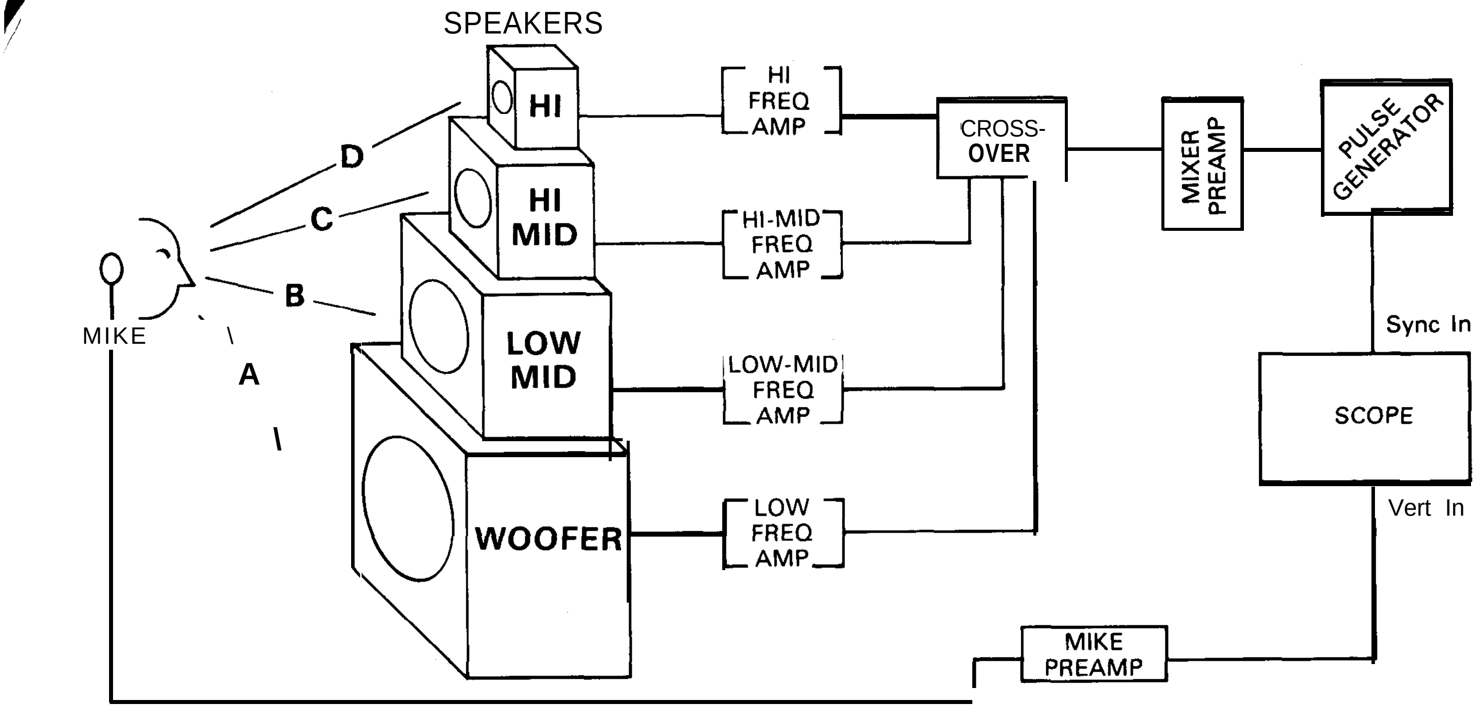
dependent upon the actual size of the cabinet, larger cabinets need thicker walls. In all cases, high quality wood should be used. If plywood is used, only accept wood with plys made of solid laminations with no gaps or filler. Filler, consisting of sawdust and glue, is added to the gaps in cheap plywood. The price of proper materials is much more expensive than ordinary woods. If there are any gaps in the layers, they will start vibrating and breaking up the filler between the plys causing buzzs and other sounds. Hard Truckers only uses imported 14-ply Finnish birch three-quarter-inch marine plywood in their construction and bracing. All of their products make extensive use of these materials and techniques. High quality particle board may be substituted, but it will not be as durable.

- An enclosure made with cheap plywood may superficially resemble a high quality cabinet, but after a short period of time it will sound like a “rattle trap.”
- It is necessary to isolate the speaker from the cabinet with a gasket. The gasket serves two purposes; one is to insure an airtight seal and the other is to provide some shock isolation.
- The impulse response can be dampened by changing the amount of stuffing inside the cabinet. Increasing stuffing should stop the high frequency reflections within the enclosure from radiating out through the port or through the cone material.
- Ringing in horns can also be suppressed to some degree. One approach to the problem involves the utilization of some carpet under-padding (waffled horse hair covered with jute). This material also works well as cabinet stuffing. Hotels and theaters usually throw it away when they replace their carpets. Coat the outer surfaces of the horn with plastic resin or similar adhesive, and press the padding into the adhesive and allow it to harden. Make sure that all fittings and connections are airtight and that any mounting plates or adapters cannot rattle against any other surface.

Figure 6 shows the impulse response of the same system as depicted in Figure 2, except that bracing and stuffing techniques have been applied. Figure 7 is the frequency response of the same system show in Figure 3 after bracing.

Background Information

In the 1920’s, researchers at Bell Labs



TEST SETUP

discovered that the arrival time of the sound from speakers reproducing different frequency components of the same signal was important if high intelligibility was to be maintained. Until recently, these concepts have not been applied in the design of commercially available systems. UREI offers Time Aligned" Studio Monitors (Time Aligned is a trademark of E. M. Long and Associates). Other products are also marketed under various tradenames.

Using the same impulse system setup previously mentioned, it is possible to observe the arrival of the impulse with respect to time. For test purposes, consider a four-way sound system containing an active crossover and a separate amplifier for each speaker component (see diagram).

Preparation Before Testing

In the following discussion, it is important that the crossover be included in the tests for two reasons: first, the crossover prevents out of range signals from damaging the speakers and, second, the crossover exhibits a frequency dependent time delay which needs to be considered in the measurements.

The test microphone should, within reason, be closer to the sound system than the first reflective surface and should be pointed towards the system at approximately ear level. Since we are going to make high-frequency measurements, the microphone needs to be on-axis. If the speaker is on the floor, place some carpet or other absorptive material on the floor between the speaker and the microphone to help minimize reflections.

When a single broadband impulse is injected into the system, it is divided into four separate bandpass outputs by the crossover, and then each output goes to a separate amplifier and speaker (array). The signal that was once a single impulse to the input is now being reproduced by the four separate speakers. If these signals do not arrive at the listeners ears at the same time, the coherence of the sound is reduced. The results may be an auditory double image, a smearing of the sound and a general reduction in intelligibility. This is particularly noticeable in the high frequencies because the psychoacoustic properties of the ear are much more sensitive to mid and high frequencies.

The arrival time of the four signals from the speakers is also controlled by the physical location of each of those speakers.

In the past, many people have lined up the voice coils in a vertical plane. This is a step in the right direction, but not a complete solution. The problem of misalignment may be observed fairly easily. If all of the components are mounted in separate enclosures, moving them forewards or backwards relative to each other will help bring the system into closer alignment.

Most large portable reinforcement systems are built in a modular fashion, so moving individual sections is not too difficult. In the studio, however, the system speakers are usually fixed into a particular place so they are not easily moved.

It should be noted that the speakers should not be tipped or tilted as this movement will alter the polar response. Also, too much movement could put a reflective surface in the path.

Testing Process

The procedure for observing alignment is similar to that used for checking polarity. First, set the generator's output filter to a low frequency setting and adjust the oscilloscope so that the knee or breaking point of the received impulse lines up with one of the vertical lines on the scope graticule (continue to trigger on the negative edge of the waveform).

Now slowly move the generator's frequency control to a higher setting. The amplitude of the low frequency speaker will decrease and the mid-range speaker's impulse will appear and increase in amplitude. This impulse knee will then be either to the right or left of the noted graticule line. If it is to the left, then the speaker is too close to the microphone and needs to be moved back. Conversly, if it is to the right then it needs to be moved closer.

Again move the generator to a higher frequency and reposition the speaker accordingly. Continue until all speakers have been observed and adjusted. If you move any speaker, then you may have to adjust its level to compensate for the increase or decrease in SPL since it is now closer or farther from the microphone.

The aforementioned procedure can also

be used to calculate the misalignment. This is accomplished by observing the relative time interval between break points, then convert the measured time delay to distance. The time delay between the drivers may be found by multiplying the scope horizontal sweep switch setting in milliseconds times the number of divisions separating the impulse break points. Then, since sound travels at 1130 ft/sec @ 70 degrees F, multiply 1130 times the time interval in seconds. The result is the distance of separation in feet.

$$D = CxT = 1130 \times .001 = 11.13 \text{ ft/ms}$$

Where:
D = distance in feet
C = sound velocity (1130 ft/sec @ 70°F)
T = time in seconds

The idea is to get all of the knees to line up on or near the same vertical graticule line. For further adjustment, tune the generator's output frequency upward, and then downward while observing the received waveform. As you follow this procedure you should observe the familiar impulse. This would be a good time to experiment with different polarities between crossover outputs. Reversing polarity will cause a change in the amplitude and the wave shape.

Listening Evaluations

Before you start your listening test be sure to adjust the levels — using pink noise and a spectrum analyzer — on the crossover outputs or on the power amplifiers for your preferred listeners curve. Use these controls as if they were broadband equalizers, which they are.

Recently, many articles have been written on the pro's and con's of impulse alignment. Some have observed that there is no value in making these adjustments and that there will be no audible effect. The authors offer the following experiment which allows the reader to independetly judge for himself.

Using two identical speaker systems, optimize the impulse alignment on only one of them. While you are standing between the two, have someone rattle a set of keys into a microphone which is connected to both systems. The authors have observed that the adjusted side will retain its intelligibililty as you walk towards it while the sound from the non-adjusted side seems to get lost shortly after moving off center.

Other effects the authors have observed are that as you make the original sound more coherent, the reverberation will become less apparent. In the case of stage monitors, the gain level may be reduced slightly due to improve intelligibility, thus effectively lowering the feedback threshold.

One of the worst misalignments measured was of a four-way system and totaled approximately 14 feet. This was a portable reinforcement system whose components consisted of RCA type W horns, a closed box array of 16 twelve-inch speakers, compression drivers on radial and long-throw horns, and ESS Heil Blue Ox Air-Motion Transformers. The misalignment was so great, that no correctiveaction could be taken.

In some commercially-available speaker systems, the alignment is achieved through a combination of driver placement and special delay networks added to the passive crossover. On a multi-amp system, delay would be required at each of the crossover bands prior to the power amplifier. At the present time, Ultra Sound has been unable to locate a commercial delay which operates satisfactorily while achieving the desired alignment. Some of the available delays tested could be used in the lower frequency ranges, but none were acceptable in the high frequency range where "birdies and chirps" from the sampling processes were objectionable. Some of the new generation delays becoming available may resolve these problems.

Another observation that may be made from the impulse is the study of reflections. To illustrate this effect, place an album cover near the microphone and observe the scope while changing the position of the album cover. Now place some foam or other absorptive material near the microphone, between it and the nearest reflective surface. This will reduce some of the reflections. Thus, by manipulating the acoustic environment — that is, by covering or hanging things on the walls or other hard surfaces or by moving furniture — you may be able to minimize or control the reflections arriving at the microphone.

Readers who do not have access to an oscilloscope or circuit construction facilities should still verify that all speakers sharing the same signal are correctly connected with respect to polarity. This applies to all sound systems.

A commercial product is available for determining polarity both electrically and through the air from Sounder Electronics. This device consists of a sending unit and a receiver. The receiver has an internal microphone and various input capabilities. Indicator lights signify whether the received pulse is positive or negative. For further information, contact Sounder Electronics, at 21 Madrona Street, Mill Valley, California 94941. An article showing a hand-held audio phase detector was published in Audio, January, 1978.

In Conclusion

Because every system is made up of different combination of speakers, enclosures, crossover networks and power amplifiers, it is difficult to establish a standardized set of rules to follow with which to "properly" adjust the impulse response of a sound system. The authors' suggestions here are offered as a relatively simple means of observing speaker and microphone connections and placement. Such studies are otherwise limited to those with access to complex and expensive equipment such as spectrum analyzers, wave analyzers and Fourier analyzers.

Many subjects in this discussion were just briefly mentioned since the purpose of this article was to offer a basic measurement technique. It is hoped that interested readers will investigate the references listed in the bibliography that follows. Copies of these references are available at most public libraries.

The important information to be seen from the received waveform is the polarity of the first arrival and its time relationship to a similar first arrival from another speaker. The ripples following the first impulse sometimes can be seen to have a regular, periodic structure. This may be an indication of resonances within the speaker or cabinet. The waveform also contains signals from reflections of other objects near the speaker, microphone, or in the path between them, so an exact interpretation is difficult. This interpretation of the received waveform could be accomplished with Fourier analysis to derive a frequency vs. amplitude plot, for example. ❑

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