THE PA BIBLE

A guide from Electro-Voice on applying our building-block group of horns, drivers, bass boxes, crossovers, and full-range speaker systems.

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Is Electro-Voice Getting into the Publishing Business?

No. But we do have a long history of improving sound for professionals and we are convinced that the next big step for musicians will require studying some advanced techniques developed recently. So, we decided to put together a series of publications for that purpose.

Electro-Voice began over fifty years ago when a couple of men who were not satisfied with the performance of available microphones designed some that were better. Since then, E-V has regularly introduced innovations which have advanced the art of audio.

Electro-Voice brought to the world such developments as noise-cancelling microphones, directional microphones with flat up-close bass response and the first American-made electret condenser microphones.

In the early days of high fidelity, E-V was one of very few pioneering companies who brought about a revolution in sound quality for consumers. Again, in 1973, E-V made the first commercially available hi-fi speaker system employing the advanced theories of A.N. Thiele and computer-aided design techniques. In the years since, this breakthrough in speaker system design has allowed nearly every company in the business to be able to improve their products.

More recently, Electro-Voice introduced yet another phase of speaker design technology which promises to be revolutionary. This is the development of "constant directivity" horns for professional P.A. use. These horns provide the capability to cover the audience area with all frequencies at a constant level. No more bright spots and dull spots.

In 1979, for the unprecedented third year in a row, E-V sound will be used at the prestigious Montreux Jazz Festival in Switzerland. In 1978, E-V sound was used for the "Happening" at the summer NAMM show, and will be used again for a big musical event at the 1979 NAMM.

A growing list of performers are using and endorsing the E-V sound: Journey, Rod Stewart, Marshall Tucker Band, Bob Seger, Yes, Ronnie Milsap – the list goes on and grows.

Many of the most respected audio consultants in the U.S. regularly specify the E-V sound as the one of choice. At Yankee Stadium, the Las Vegas Convention Center, conventions of the Audio Engineering Society and at many more locations, the E-V sound has met the challenge of giving clear, wide-range, high-level coverage to audiences – small, large, and giant.

In keeping with E-V tradition, we want everyone to be able to share our innovations. Over the years, E-V has been a leader in providing instructions for designing and building cabinets, assembling systems, and applying products to the tough jobs. With this booklet, and the series of publications to follow, we hope to show you how to achieve the best sound modern technology and techniques can achieve.

In support of this program, E-V has also developed and will add to a series of components especially designed for you – so you can combine them to create the best of professional sound for your specific needs.

How to Read This Guide

Obviously – if you can – read it *all!* If you just want to know what components you need to provide sound in a small room you can look in the "Basic Approach to System Design," "Small Size Room" section on page 7. Sections 1 and 3 (starting on page 1) will inform you who Electro-Voice is and what this guide can do for you. Section 4 (page 2) gets into the typical problems which can plague P.A. systems and solutions to these problems. Section 5 (page 6) describes three selected room sizes and gives a recommended system layout and connection diagram for each particular room.

What This Booklet Is About

You really don't have to put up with poor quality sound and we hope to show you here the "whys" and "hows" of getting good results within the budget you set for yourself. In the process, we will show you how to go from the typical "packaged P.A." to a flexible, "building-block" system of pro sound components: horns, drivers, bass-box, electronic crossovers, and all the rest. We've tailored our comments to the real world of the performing musician, but our material is helpful in designing high-quality fixed installation sound reinforcement systems too.

In this guide we'll address those basic problems, annoyances and questions that plague every musician at one time or another. Let's try some on for size:

1. You wonder why you still can't go loud enough with your new power amp that has twice as much power as the old one.
2. You've turned all the knobs in every direction but before the system gets loud enough it goes into feedback and sounds like you stepped on your puppy dog's tail.
3. Your PA sounds just plain gross and distorted and you're tired of it.
4. Your favorite recording group sounds great on your hi-fi but you can hardly understand a word on their touring PA.
5. Your rig sounds OK in the local auditorium but won't cut it at the high-school gym.
6. The guests of the management up front are holding their ears but the paying customers in the back say they can't hear.
7. You want a new amp but don't know how big it should be.
8. You're thinking about bi-amping your new speaker but don't know where the lows leave off and the highs begin.

9. You've put a lot of work into getting the words just right and then most people in the audience can't understand them.

To solve the kinds of problems listed above, you first need to know what is wrong with a lot of P.A.'s. There is no riddle or black magic to achieve good sound, but some basic things must be understood and dealt with to reach the desired goal – a professional quality sound system. The next section deals with the major problem areas that have existed since the first sound, and especially since the first sound in a room.

What's Wrong With A Lot of P.A.'s

Here's the basic straight scoop on why many P.A. systems sound so bad, and a first pass at dealing with the problems we just described.

Low-Efficiency Speaker Systems. We should first talk about the term "sound pressure level" (abbreviated "SPL") which your ears interpret as "loudness" or "volume." SPL is almost always expressed in "decibels," abbreviated "dB." dB's are thrown around a lot, without much understanding. When you talk dB's you're always talking the *difference* between two quantities. For example, "100 dB SPL" means a sound pressure level 100 dB above a 0 dB point set at the lowest sound pressure level discernible by the average human ear. You're also talking differences when you say one sound is 3 dB louder than another sound. A 3 dB difference can be heard by most listeners but it certainly won't knock your socks off. It takes something of the order of a 10 dB difference in average SPL to be perceived as a doubling (or halving) of loudness. Yet doubling amplifier power, or adding a second speaker system, gives only a 3 dB increase in output. The same result can be obtained by using a speaker with twice as much efficiency.

Now we can talk about "efficiency." Speaker system efficiency is the amount of sound a speaker system is able to put out for a certain amount of electrical (audio) signal fed in. From this, you can see that high efficiency is good. It means that for a given amount of amplifier power you can get more sound from the speaker system. Efficiency is properly presented by a percentage. For example, a high-efficiency, direct-radiator speaker system (such as the E-V S15-3) approaches 5% efficiency. A good compression driver/horn combination (such as the E-V DH1506/HR60) approaches 25% efficiency.

What all this means is that high-efficiency speaker systems can produce high sound pressure levels. Well designed speaker systems usually incorporate high efficiency as one of their design goals. All Electro-Voice systems from the S12-2 on up are designed to give the highest efficiency possible for their size and type. High efficiency means you can obtain the sound pressure level you want in a room without distorting the power amplifier, which is the next subject.

Not Enough Amplifier Power. Before any speaker system can perform to its highest potential, it must be connected to an adequate amplifier, especially one with sufficient headroom. We don't mean how small a foreign car's interior is! Headroom is the amount of reserve level capability that the amplifier has above the long-term average level your ears hear as "loudness." In live music, 10 dB peaks above the average – of a few milliseconds duration – are common and are continuously going through the system. If the peaks can't get through, the sound will still be as loud but it will sound rough and distorted. This means that if you are playing at a 10-watt average level, you will need a 100-watt power amplifier to pass the peak levels (10 dB higher) without clipping (distorting) the amplifier. When the amplifier does go into clipping, you will know immediately because you will hear it through the speakers. Many times people say their speakers sound bad at medium-to-high levels when in reality it is their amplifier. The speaker has no choice but to reproduce the signal being fed to it whether it is clean or distorted.

This amplifier clipping is also a common cause of speaker failure. When clipping occurs, high-level high frequencies are produced which usually overpower tweeters and midrange speakers and result in smoke and no sound! Therefore, you need to be certain that the amplifier you use has enough power to give plenty of reserve over the average needed to give the desired sound pressure level.

Here's an example to help make the efficiency-versus-power issue more clear. Let's say you are running a 250-watt amp into speakers that are 2½% efficient. You're operating at 25-watts average, so there is headroom for the peaks. But it isn't loud enough (SPL too low). You could go to a 500-watt amp and get 3 dB more; but that's expensive and still not very impressive in additional loudness. You could also go to a 5%-efficient speaker to get the same 3 dB; or you could use a very efficient speaker (maybe 25%) and get really loud but still be clean and have headroom. The point is, speaker efficiency is at least as important as amplifier power.

Poor Frequency Response. Assuming you are using a high-quality microphone, mixer, and amplifier, your speaker system may have poor frequency response. Frequency response is the way a speaker responds (in dB) to a constant input signal swept over the audible frequency range from low bass to the highest treble. Speakers that have varying frequency response will also vary when music is being played through them. This causes the speaker to produce an unnatural, "colored" sound. A speaker that exhibits a flat frequency response over the frequency range it is intended to be used for will sound more natural than one which varies up and down over the same frequency range. A flat response is also desirable to reduce feedback. If a speaker has a large peak in response, the microphone may respond to that peak first and feedback will occur at the frequency where the peak is.

Figure 1 shows how sound pressure level varies with frequency at a specified distance in front of the speaker under anechoic (non-reflecting) conditions, which is similar to being outdoors where there are no walls or ceiling to reflect and modify the sound. The best all-around result will come from the speaker with "flat" response. If you want the sound shaped in some way, do it with an equalizer, or the EQ on your board, where *you* control it.

Highs Miss Half Your Audience. Let's say your system uses only a 12-inch cone speaker mounted in a box. Take a look at Figure 2.

At low frequencies sound is dispersed over a very wide angle (in fact, almost omnidirectional). This is because the cone is small in comparison to the wavelength of low frequencies. Wavelength is velocity of sound in air (1130 feet per second) divided by the frequency. So, for example, the wavelength at 50 Hz is 22.6 feet. That is quite a bit larger than a 12-inch speaker. As you go up in frequency, the 12-inch speaker becomes larger than the wavelength and a phenomenon called beaming starts to occur. This is where, instead of having very wide dispersion or coverage angle, you have increasingly narrow coverage. This is why the listeners at the side of the room sometimes can't hear the high frequencies. The sound is dull and unintelligible. Therefore, coverage angle (dispersion) over the range of frequencies involved is an important consideration in speaker system design.

A speaker's dispersion of sound as the listener moves to various angles off the speaker axis is typically shown in a polar response graph such as in Figure 3. Measurements are usually made in both the horizontal (side-to-side) and vertical (up-and-down) planes. Both are shown in Figure 3.

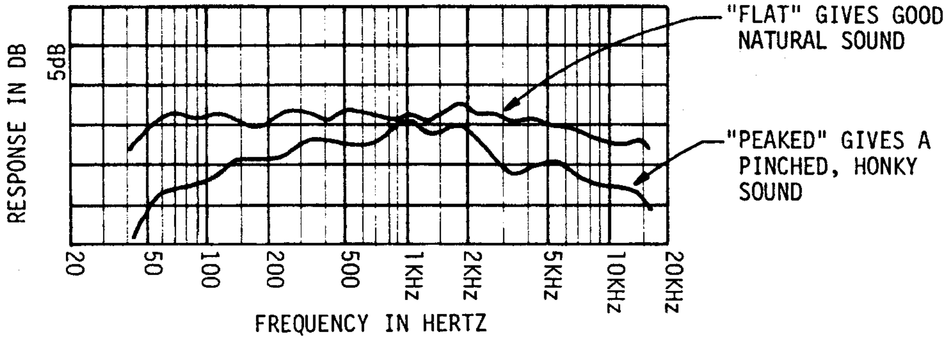


FIGURE 1 – Axial Frequency Response

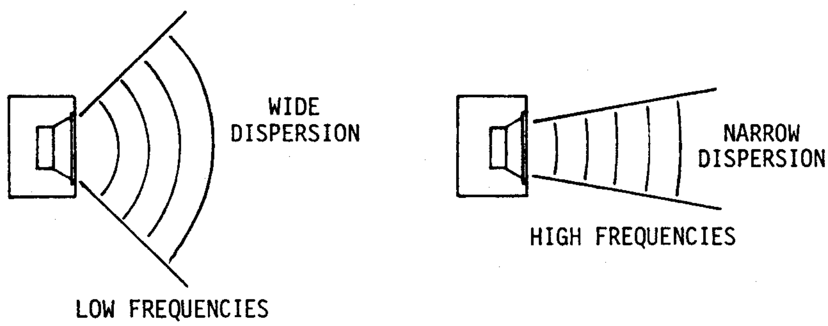


FIGURE 2 – 12-inch Speaker Dispersion

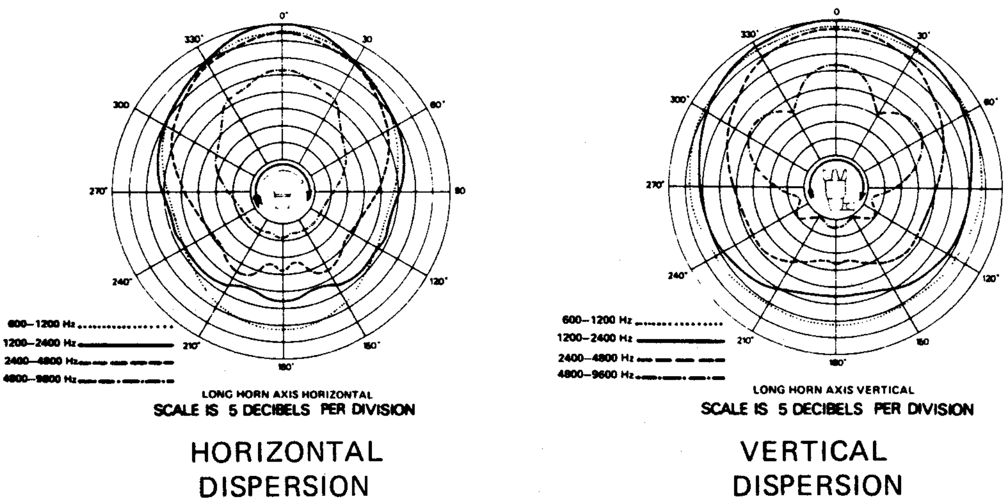


FIGURE 3 – Polar Response, Octave Bands of Random Noise, Horn in Normal Orientation

A typical approach is to feed the speaker a test signal containing all of the frequencies in an octave, like 2400 to 4800 Hz. This avoids the confusing variations of single-frequency measurements. Since the test signal contains all the frequencies in the octave of interest, it has no definable pitch or musicality but, instead, sounds something like the between-stations noise on an FM receiver. Therefore, the signal is called "random noise." The loudness (SPL) of this noise is measured at all points around the speaker, at a constant distance away, and the level is recorded on the polar graph. Note in the example shown in Figure 3 that, in both the horizontal and vertical planes, the 2400-4800 Hz frequencies are about 10 dB louder in front of the speaker than 60 degrees off to the side. Remember, a 10 dB difference in SPL is perceived as "twice as loud." So, with some frequencies "half as loud," some "nearly as loud," and some nearly gone, it is little wonder that people at the side of the room hear poor, muddy sound from a speaker that has not really considered uniform dispersion in the design.

It is convenient to say a speaker has a certain coverage angle (90,° etc.), but if you don't know the dispersion or coverage angle of the speaker for each octave band, you can be misled. Some manufacturers say their speaker has 90° dispersion and that's that. This would be sufficient if it were true for all frequencies. However, in the real world even the best loud speakers only approach this goal. Most differ greatly over their frequency range. Therefore, Electro-Voice supplies not only polar responses but also beamwidth-versus-frequency graphs for most speaker products, as shown in Figure 4. From such a graph you can determine coverage angle for frequencies important to you. In Figure 4 and in our engineering data sheets, we've defined the coverage angle in each octave band as the angle included by the points on the polar response where speaker output is 6 dB below the on-axis response. Although no absolute standard exists, this definition of "coverage angle" or "beamwidth" is often used.

Uniform dispersion is one of the most important and most neglect-

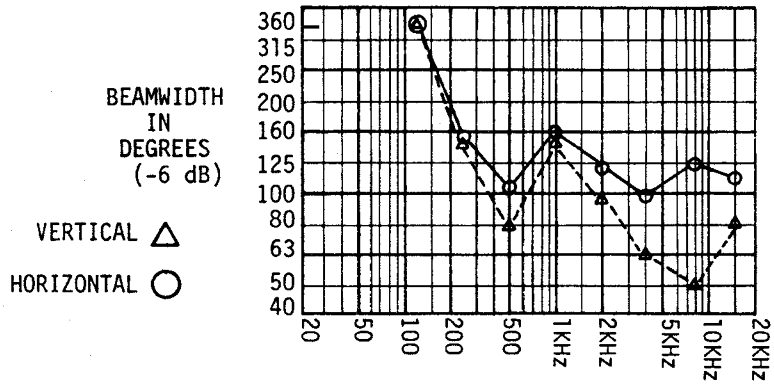


FIGURE 4 – Plot of Coverage Angle (beamwidth) versus Frequency

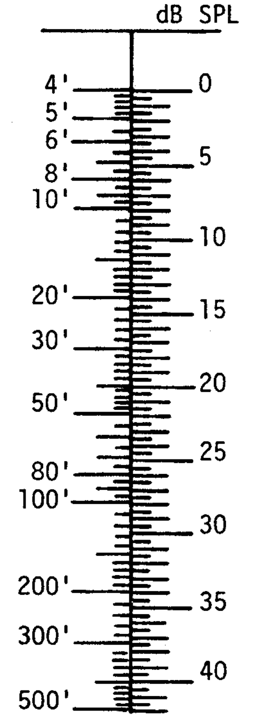


FIGURE 5 – Level Variation with Distance

ed characteristics of a speaker system. Electro-Voice provides coverage-angle data in the form of beamwidth and polar response on all its products, so you can design a system that will put the sound where you want it. To design a speaker system that possesses uniform dispersion, special components are used. For example, the E-V S15-3 is a three-way, full-range system. It has a 15-inch low-frequency driver which is only used up to 600 Hertz, so its coverage won't get too beamy or narrow. Then, from 600 Hertz to 4000 Hertz, a small 6½-inch cone midrange is used, and from 4000 to 18,000 Hertz a wide-angle, constant-dispersion horn tweeter is utilized. (Some horns have a beaming problem similar to a 12-inch cone speaker, for example.) Using separate components designed for operation over their portion of the audio spectrum instead of using just one speaker will generally yield superior overall performance where the application calls for reproducing the full frequency range.

Knowing something about the dispersion angle can help you select speakers for your application. Speakers should be directed to cover the listeners. Viewing the listening area from a desired speaker location, determine what dispersion angle would be needed to adequately cover the listeners without spilling over to the walls in both the horizontal and vertical planes. Once these angles are determined, the correct speaker can be found by consulting E-V engineering data sheets and catalogs.

Double Distance Rule Gets You. You might face a situation where the people in the front row are being blasted right out of their seats while the people in the rear are hardly able to hear. This is because sound heard directly decreases as you move away from a sound source (your speaker system). In a non-reverberant (non-reflecting) environment, such as outdoors, sound pressure level from a simple source will be cut in half (drop 6 dB) every time the distance from the speaker is doubled. This is called the "inverse square law." Figure 5 shows the dB losses to be expected as distance from the speaker is decreased from the four feet used in E-V SPL specifications.

To deal with this "law of nature" you need specialized speakers to project sound to the back of the audience while not hitting the people up front with the extra-high SPL levels this requires at the source.

Room Reverberation Swamps Your Voice. Now you say, "Great, I have a good microphone, mixer, power amp, and an efficient speaker with reasonably flat frequency response and uniform dispersion." But when you use this system, people in the back of the room still can't understand the vocals or really hear the high frequencies. This involves not only the speaker but also the room in which it is operating.

Rooms have a phenomenon called "reverberation." Reverberation is the tendency for sound to continue within a room after the original sound has ceased. Outdoors, in an open field, is considered to be a "non-reverberant" environment, so this continuation does not occur. But, as you are supplying sound to a room, reverberation is occurring. The farther a listener is from the speaker, the better the chance he is in the "reverberant field" and the worse the chance he can understand what is being put out by the speaker itself.

If the listener is close to the speaker, he is said to be in the "direct field" of the speaker. This is where the sound coming directly from the speaker is much higher than the reverberant sound. But, as you move away from the speaker, the sound reflected from the floor, ceiling, and walls gets increasingly louder relative to the sound coming directly from the speaker. This is where trouble begins. In a reverberant environment, there is a point away from the speaker beyond which the "reverberant field" dominates the sound heard. It is interesting to know that the SPL tends to remain constant in the reverberant field, no matter where you're standing in it. Constant SPL throughout the room is, of course, a good thing but when the reverberant field is what's doing it you can get into trouble, as we will soon see.

The distance where the reverberant field begins to dominate is typically 10 to 20 feet from the speaker, and is longest for the least reverberant rooms and the most directional speakers. The distance from the loudspeaker where the direct sound and the reverberant sound are the same level is called the "critical distance." In Figure 6 you can see the two equal sound pressure levels add and become 3 dB higher at the critical distance.

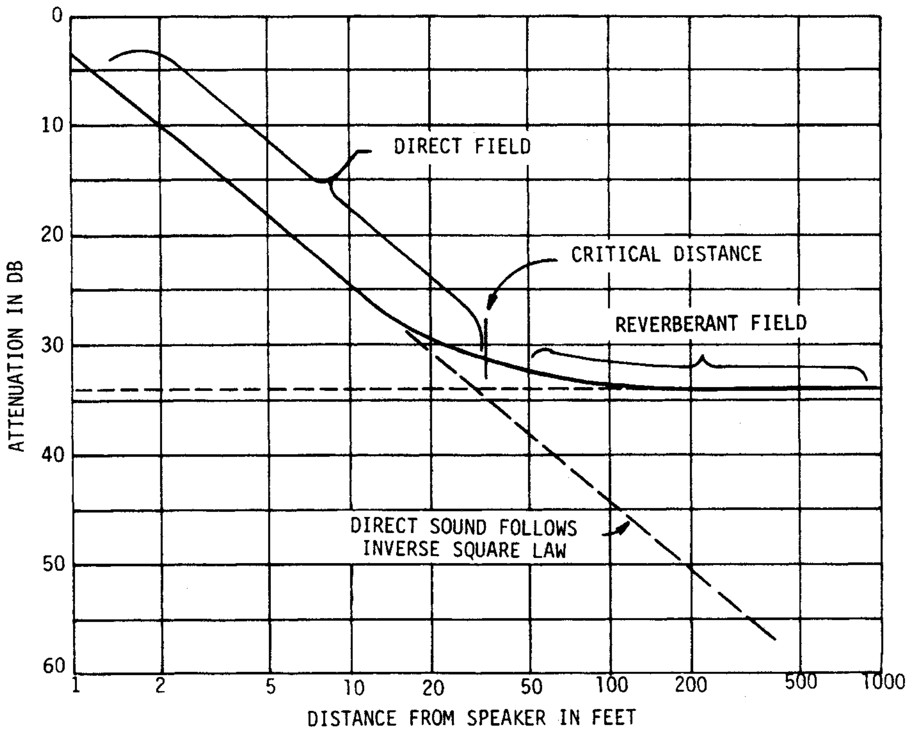


FIGURE 6 – Attenuation with Increasing Distance in Typical Large Room

When you are in the reverberant field part of the room, most of the sound you hear is reflected from the walls, floor, ceiling, etc., and only a small amount comes directly from the speaker. All these reflections cause the sound to reach your ears at slightly varying time intervals, and at a higher level than the direct sound. The result is that listeners in the reverberant part of the room find it very hard to understand what is being sung, or to hear clearly the various instruments being played. The music tends to become a confused jumble of sounds.

The concept of the reverberant field is one of the most important concepts to understand in this whole guide. If we had a way to sound a siren to direct your attention to a particularly important problem which affects P.A.'s in all rooms, it would go off now! When listeners are in highly reverberant rooms or reverberant parts of a room, you're going to need to do something about it. The larger the room is, the worse the problem is. Don't let this scare you – that's why we're making this guide so the problem can be recognized and conquered!

A few methods of dealing with the reverberant field problem would be:

1. *Make the room "anechoic," meaning "no echoes."* This is probably not feasible and would result in a highly modified and bizarre appearing room.
2. *Move outdoors to a big field.* Remember outdoors is non-reverberant, with no walls or other surfaces to reflect sound. This is obviously an impractical solution – especially when it rains.
3. *Select loudspeaker components* which have appropriate directional characteristics for the room. These characteristics are inherent in the dispersion or coverage angle of the components. Ultimately, you would want sound to go only into the area where listeners are, so none is sent to bounce off walls, etc. (Listeners are excellent sound absorbers.) This can only be approached, in actual practice, but the results of paying attention to this factor in system design can be astounding! If you are putting together a system for portable use, by all means try to figure out the system that will conquer the reverberant field problem for most of the rooms in which you typically play. Your audience will love you even more for it.

Let's examine the solution outlined in Item 3, above, in some detail. The reverberant field problem is the basic reason why a single speaker system cannot be the answer to all sound problems, even if it has flat frequency response, high efficiency, uniform dispersion, and big amp driving it. And it's the reason why you need to augment the single speaker with the building-block *components* that E-V has to offer. This is the solution to one of the problems outlined at the beginning of this guide. If you think stacking up several of the kinds of systems you might use in a small room will work well in a large room, you are certain to be surprised and disappointed. This *will* increase the total sound pressure level, but it will probably be unintelligible in most of the room.

Large rooms require both narrower dispersion and higher efficiency than the best single speaker system can offer. For instance, when the listeners in the back part of the room cannot understand the sound because of reflections and reverberation, the solution is to have a *narrow* coverage system that will *aim* more direct sound at the back of the room. (Note: this also addresses the problem of reduction in SPL with distance from the speaker.) It should be noted that wide or narrow dispersion does not denote a good or bad loudspeaker, providing the *system* is designed to provide proper coverage in the room or hall. There are applications where one or the other is needed to best solve a specific sound problem.

Narrow dispersion devices are sometimes referred to as "long-throw." The term "throw" is loosely used to describe how far sound will be clearly projected by the loudspeaker. This is directly related to dispersion. To describe this principle, think of a garden hose with a variable sprayer on the end. The water in the hose is being delivered to the sprayer with a constant pressure (the speaker or driver). The sprayer determines where the water will go (the horn). If you spray a wide pattern, it won't spray very far; but if you clamp down on the sprayer, it will spray a narrow stream and it will spray (throw, project, etc.) a heck of a lot farther. This is exactly what takes place in sound. Most direct-radiating speaker systems are classed as medium-to-wide coverage because they have coverage zones of approximately 90° or wider. However, special devices are needed to generate high SPL, and uniform, narrow coverage angles. These devices are usually horns. It is possible to have horn woofers, horn midranges or horn tweeters. For example, a midrange driver (such as the E-V DH1012) can be coupled to a wide-angle horn (such as the E-V HR90 with 90° side-to-side coverage) for short-to-medium throw, or it could be coupled to a narrow-angle horn (such as the E-V HR40 with 40° side-to-side coverage) for long throw. By their principle of operation, long-throw devices take care of the reverberation problems of medium-to-large size rooms. By raising the sound pressure level of the direct sound at the rear of the room, program material will become more intelligible. The long-throw device is not only used to create higher sound pressure levels away from the stage but also to aim or concentrate the sound on the listeners at a distance away. The direct sound will be kept high relative to the reverberant sound, and – WOW! – you'll be saying, "Can you believe we actually understand the words way back here?"

Now we know how to get good, clear sound to all listeners on paper; but how in the world can you apply this to your specific system and problems? Good question! Obviously, this material cannot give all the answers, but with a clearer understanding of the problem, the E-V engineering data sheets, some research, and a lot of common sense you can usually come up with a system that will get the job done effectively.

The next area is where we will give you some recommendations on total component system design and application.

Basic Approach to System Design

In the discussion that follows, some specific rooms will be selected and systems that could be used in them described. As room size grows, the problems involved in providing adequate sound pressure level plus clear and intelligible sound to the audience increase. This is because larger rooms usually require more acoustical output from the loudspeakers than smaller ones. (Crudely speaking, a doubling of the volume of a room of given absorption characteristics means about twice as much acoustic output would be needed to maintain a given sound pressure level in it.) It is also because more of the volume of a large room will be in the reverberant sound field of a given loudspeaker, making the generation of clear sound for far away listeners more difficult.

The range of room sizes that will be discussed extends from about 10,000 cubic feet (about three times the volume of a typical home living room) to 30,000 cubic feet, to 90,000 cubic feet. These room-size increases make a *very* big difference in how much acoustic power must be injected into the room by the speakers to get a given SPL in the reverberant field of the room. Let's look at an example where we will assume, for simplicity, that the same speaker system would work in both the small 10,000-cubic-feet room and the large 90,000-cubic-feet room. If a 100-watt amp could get 100-dB average level in the small room, the big room would take 10 times the power – or 1000 watts – to get the same 100 dB. What you would really get, of course, is a puff of smoke and a depleted bank account.

How much acoustic muscle you need is also heavily dependent on how loud you want to play your type of music. The specific room and system examples tell you the maximum loudness you can expect but you may not need that much. Figure 7 shows the long-term average sound pressure levels typical of various musical (and a few non-musical) situations. Peaks of a few milliseconds' duration will typically be about 10 dB above the average levels shown.

Normal talking at one foot is about 70 dB. A level of 120 dB is painful to most human ears. A room level of 90 dB would usually

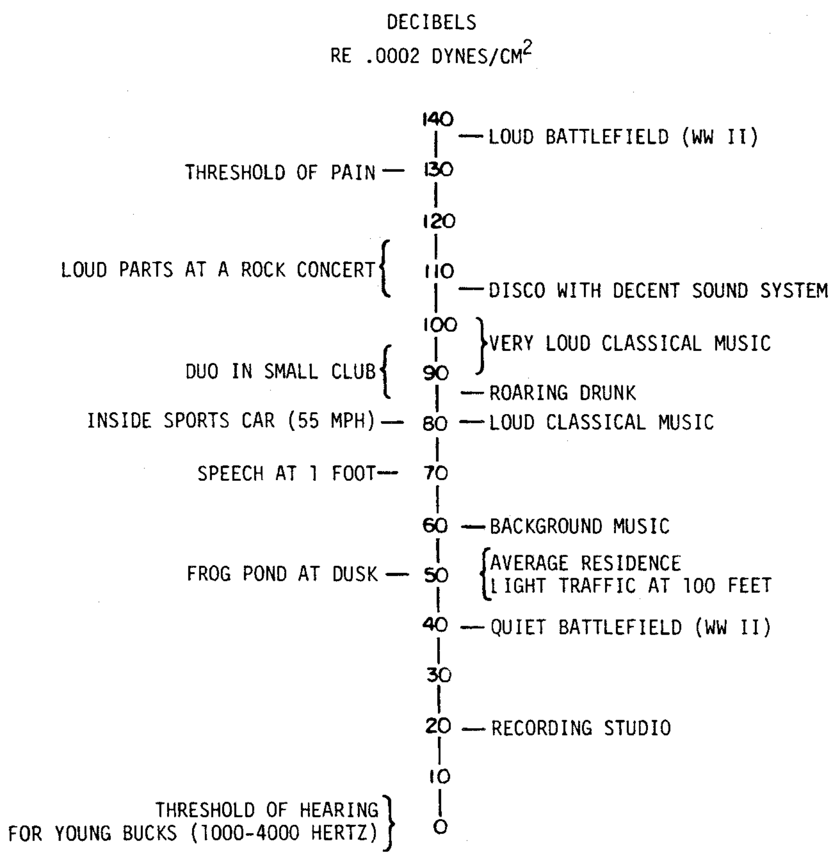


FIGURE 7 – Typical A-Weighted Average Sound Pressure Levels

be judged to be pretty loud except in the case of a full-tilt, high-energy rock band where sound pressure levels are likely to fall into the 105-115 dB range. (The SPL's we're talking about are "A-weighted," where the bass below about 500 Hz is rolled off. This makes the measurements correlate more closely to the loudness our ears perceive, since they are much more sensitive to midrange frequencies than to the bass.)

In most P.A. work a usable low-frequency capability of 50 to 75 Hz is quite satisfactory. All of the systems described in the following section fall well within this range. For extended low-frequency capability, on stage, a bass guitar speaker system can be employed (such as the E-V B115-M or B215-M) or for synthesizer reproduction down to 40 Hz, a wide-range system (such as the S18-3) can be employed. Examining the specification sheet for each E-V speaker system or component will give you some insight to the sound levels and frequency range it is capable of producing in the application you are interested in.

Small Size Room. If you need to supply sound to a fairly small room, say 31¼ feet x 32 feet x 10 feet high or a room volume of 10,000 cubic feet, the system design shown in Figures 8 and 9 shows a typical setup that will usually work well.

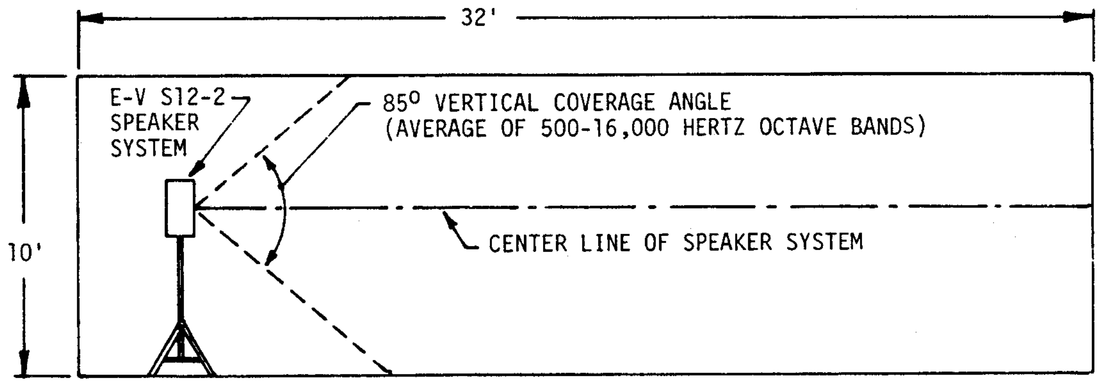


FIGURE 8 – Small Size Room Elevation View

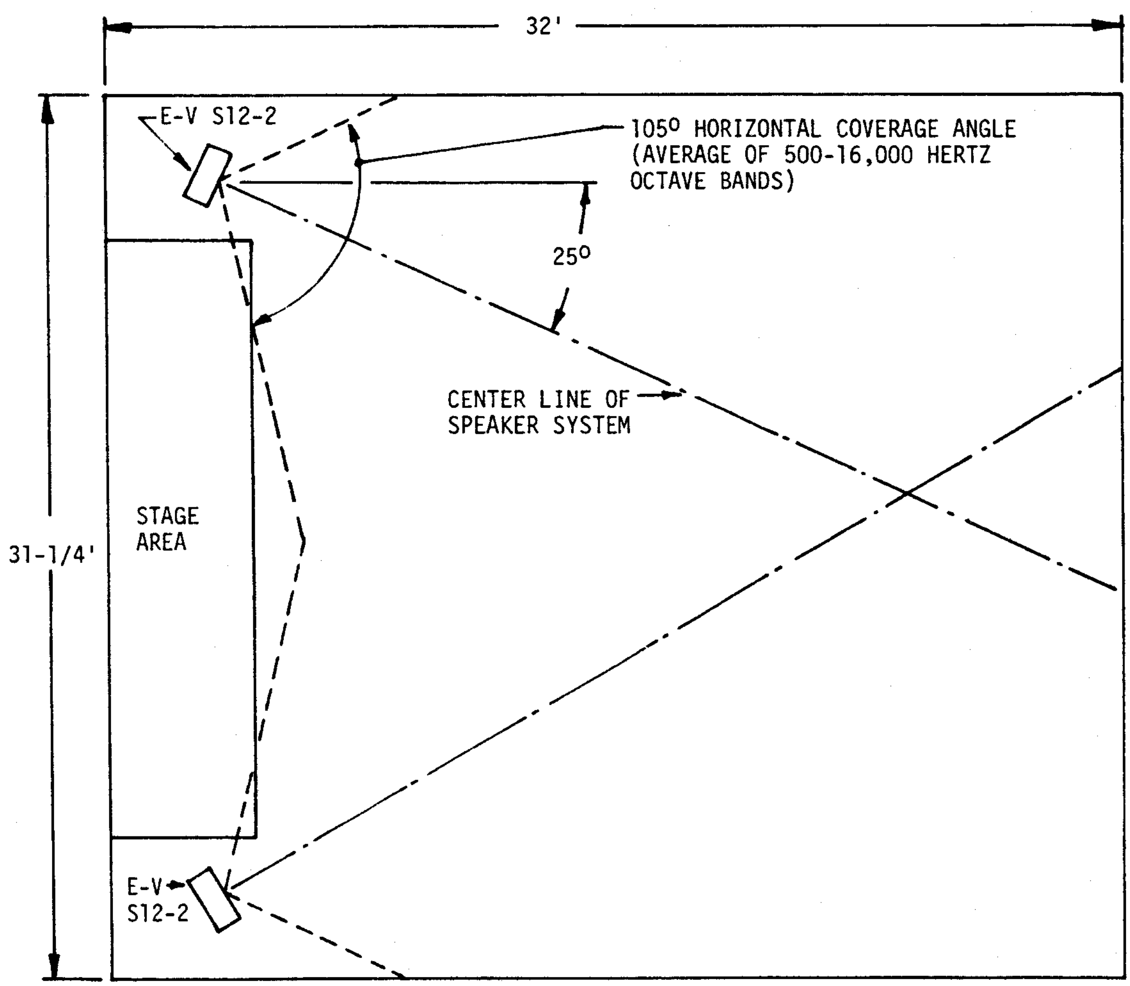


FIGURE 9 – Small Size Room Plan View

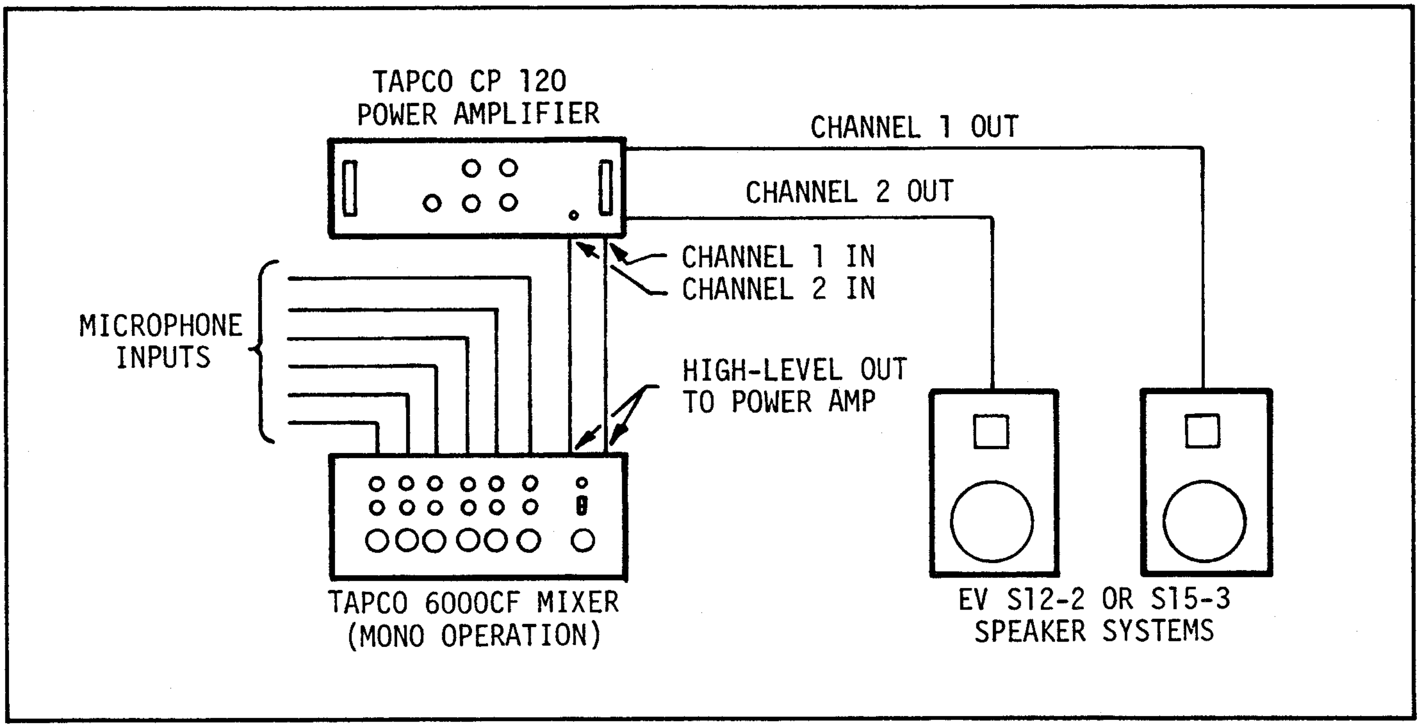


FIGURE 10 – Block Diagram for System Shown in Figures 8 and 9

The system shown is one that would be a typical portable setup using a pair of E-V S12-2 speaker systems. For increased dispersion and coverage in the midrange, a pair of S15-3's could be used in place of the S12-2's. In this particular room, either the S12-2's or the S15-3's, when hooked up to a two-channel power amplifier capable of producing 50 watts per channel with both channels driving an 8 ohm load (such as the TAPCO CP120), will be able to produce average midband sound pressure levels of 106 dB in the reverberant field of the room. Figure 6 in the "Room Reverberation Swamps Your Voice" section starting on page 4 will refresh your memory on the concept of reverberant field. System headroom before amplifier clipping is also great enough to reproduce 116 dB short-duration peaks (10 dB above the long-term average level), so your vocals and instruments will stay clean and undistorted. Remember, these SPL's are the most this system can do – chances are you will need only 85 to 100 dB depending on the type of material you are playing. You may want to refer back to the chart in Figure 7.

For increased dispersion *and* higher sound pressure level, a pair of Dominators could be used in place of the S12-2's. The Dominators, because their efficiency is about 15% compared to 5% for the S15-3, will be capable of producing 111 dB average sound pressure levels, with peak sound pressure level capability of 121 dB. Figure 10 shows the "block diagram" – which shows how things are connected together – for a complete system employing one channel of the power amplifier per speaker. This connection will produce a "monaural" system, which is more practical than "stereo" in such a setup. With a stereo arrangement, you would risk having part of the music on one side, and part on the other, with only the people near the middle of the room hearing it all.

For more headroom and higher sound pressure level capability a two-channel power amplifier capable of producing 150 watts per channel with both channels driving an 8 ohm load (such as the TAPCO CP500) could be used in place of the 50-watt-per-channel power amplifier previously described. With the 150-watt-per-channel power amplifier, the S12-2's or S15-3's could generate average sound pressure levels of 111 dB and peak sound pressure levels of 121 dB. The Dominators would be capable of producing 116 dB average sound pressure levels with peak level capability of 126 dB. If you wanted a mixer with reverb, you could use the TAPCO 6000R in place of the TAPCO 6000CF mixer shown. Other substitutions and additions could be made to suit your own personal needs. If you are designing a system to be permanently installed in the same room, you could mount a pair of PI12-2 or PI15-3 speaker systems on the wall close to the location shown in Figures 8 and 9.

Medium Size Room. To provide sound to a medium size room, approximately 40 feet x 50 feet x 15 feet high, or a room volume of 30,000 cubic feet, a system such as the one shown in Figures 11 and 12 could be used. It uses E-V Dominators and will produce 106 dB average midband levels and peaks of 116 dB. Note that the additional efficiency of the Dominator relative to the S12-2 or S15-3 has just made up for the SPL loss in going from the small to medium size room. If you don't need 106 dB average levels in the medium size room you could use the S12-2 or S15-3 for 101 dB levels. If you need more than 106 dB with the Dominators, you could go from the 50-watts-per-channel amp to 150-watts-per-channel and get 111 dB.

In the case of a medium size room, some of the earlier discussions about SPL drop with distance, the reverberant field, and dispersion come more into play. The job can be done with a high-efficiency, all-in-one system such as the Dominator. However, it is more appropriate to do it with a suitable component system such as the one shown in Figures 13 and 14.

The all-in-one system shown in Figures 11 and 12 is capable of producing 106 dB average sound pressure levels and peak sound pressure levels of 116 dB. The component system shown in Figures 13 and 14 can produce average levels of 108 dB, with peak levels of 118 dB. For higher levels, a larger amp could be used on the high-frequency horns, such as the TAPCO CP500 at 150 watts per channel. This would increase the sound pressure level to 113 dB, with peaks of 123 dB.

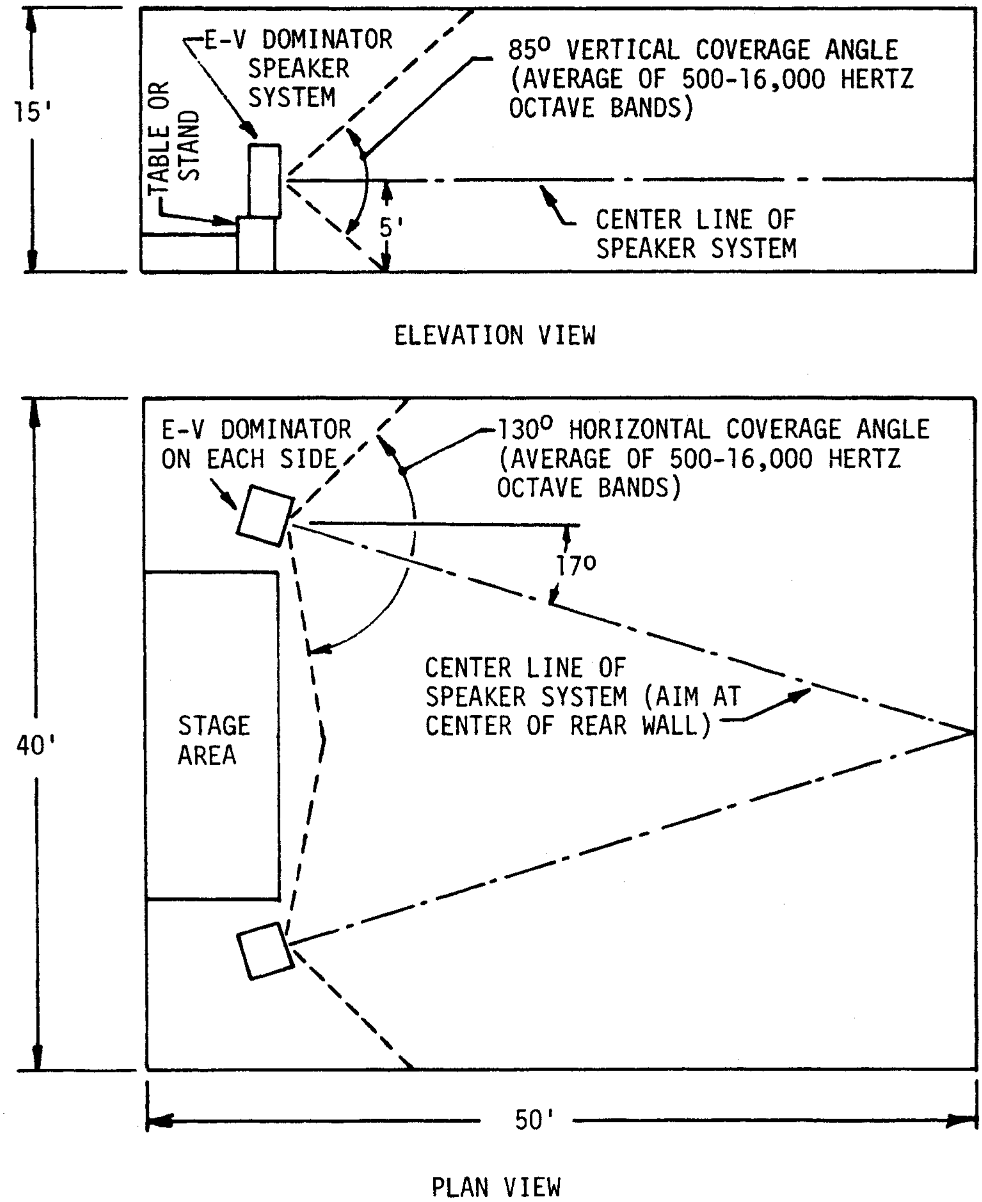


FIGURE 11 – Medium Size Room, Setup Number 1

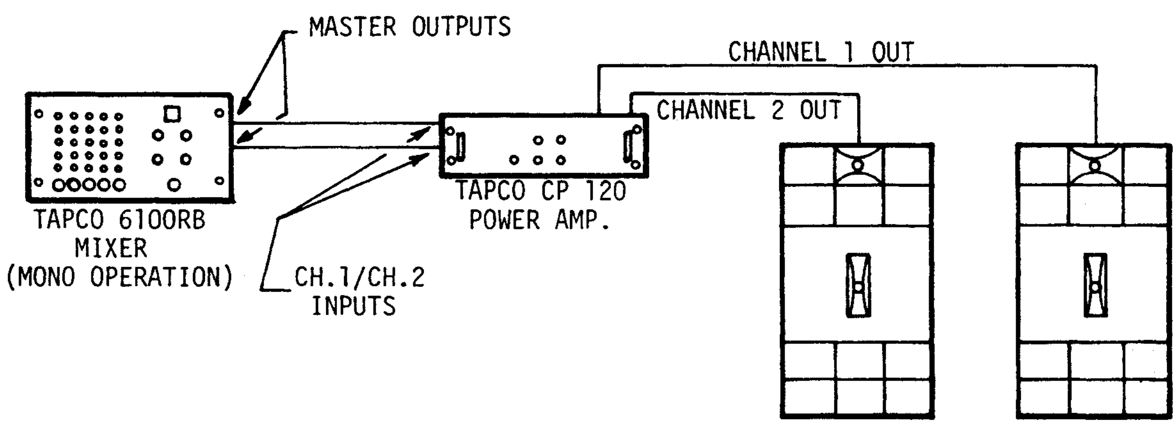
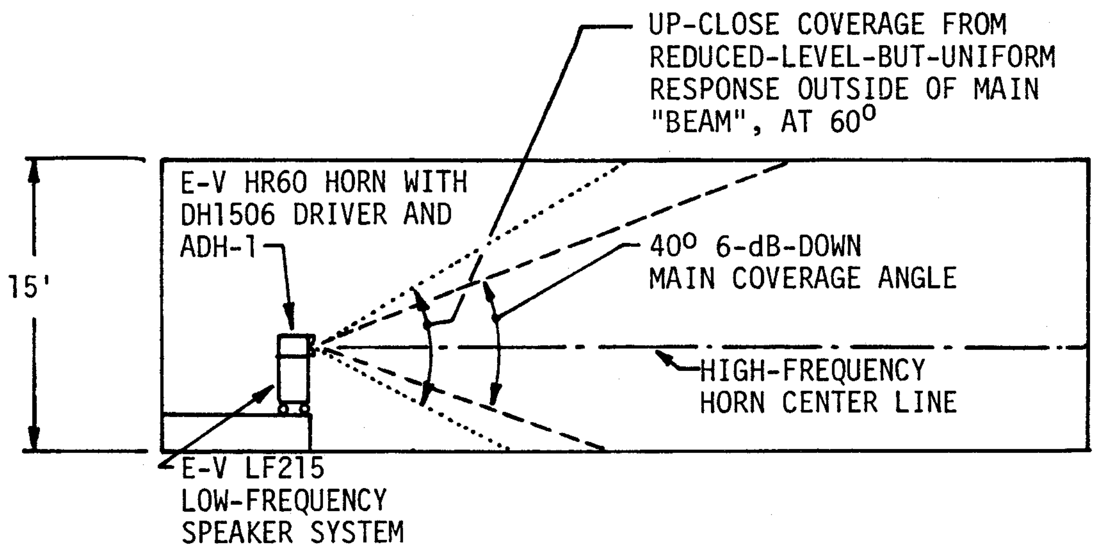
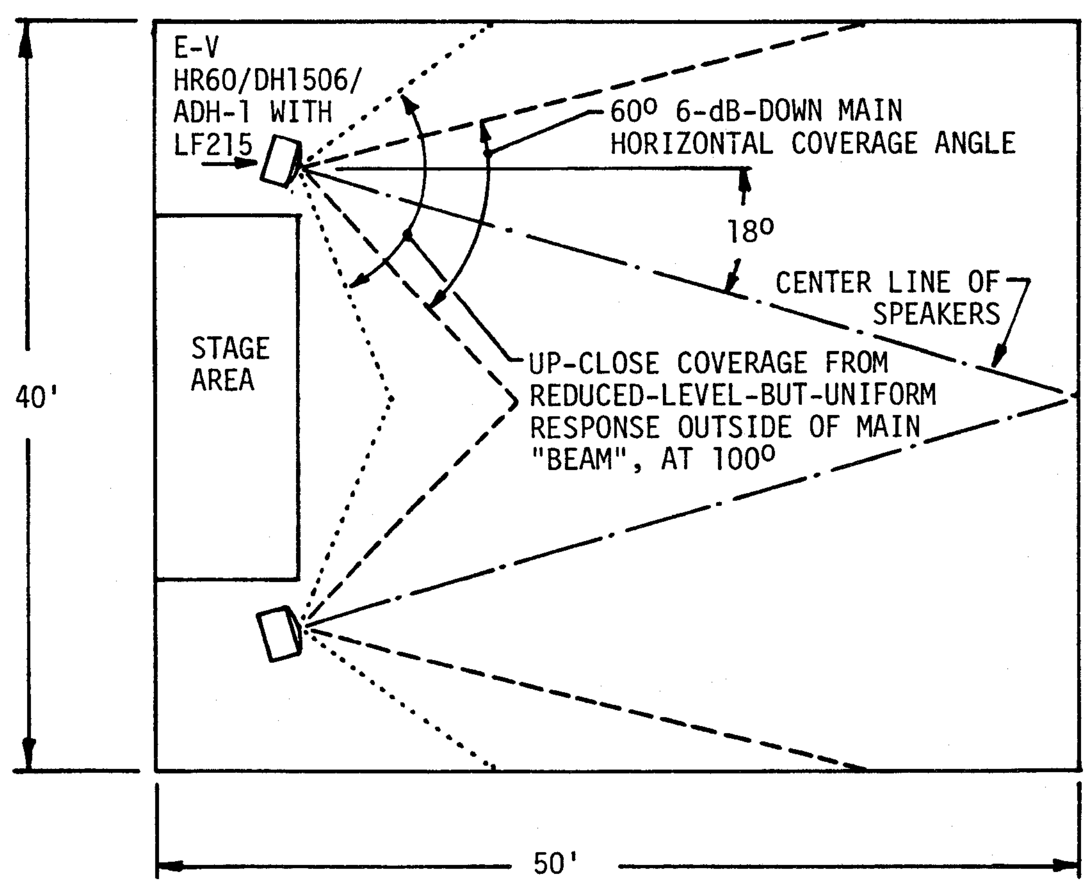


FIGURE 12 – Block Diagram for Medium Size Room, Setup Number 1



ELEVATION VIEW



PLAN VIEW

FIGURE 13 – Medium Size Room, Setup Number 2

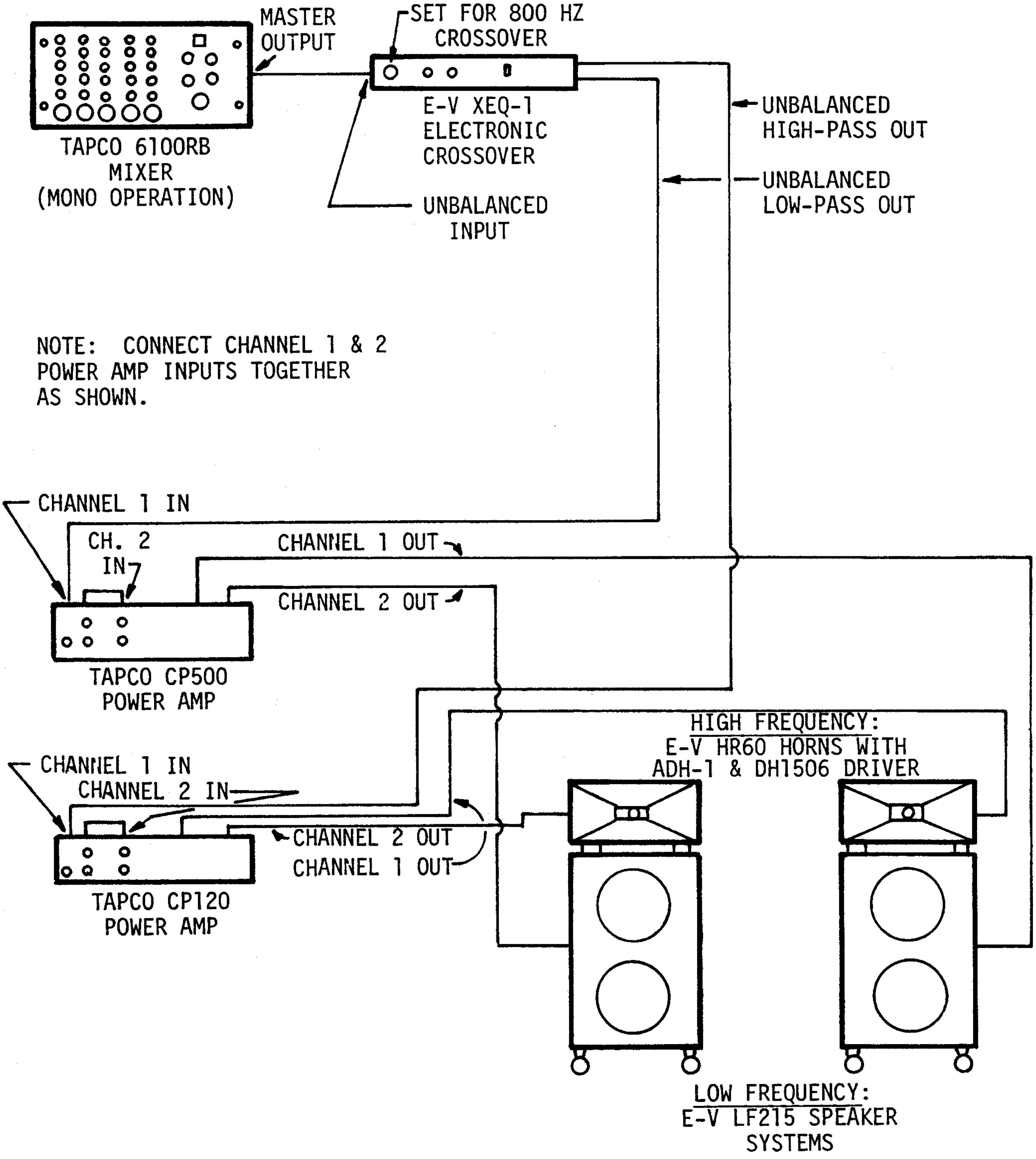


FIGURE 14 – Block Diagram for Medium Size Room (Setup Number 2)

The component system has important advantages over the all-in-one layout using the Dominators:

1. The narrower, controlled dispersion of the component system better fits the geometry of the room. The medium-throw 60° x 40° coverage of the HR60 horn directs more sound to the rear of the room. This gives:
   A. More uniform SPL throughout the room (helps get around the inverse square law.)
   B. More direct sound at the back of the room for clear, intelligible vocals there (helps keep room reverberation from swamping your voice.)
2. The component system is modular. It has the flexibility needed to expand to larger and more complex environments in the future.

Using the component system requires an electronic crossover (such as the E-V XEQ-1). This makes the system a "bi-amplified" system. Bi-amplification is a method by which crossover is placed *ahead* of the power amplifiers feeding the speakers. First, let's be certain that we know that a "crossover" is a device which separates full-range program material into the appropriate low-frequency and high-frequency portions for a two-way speaker system. The crossover directs the material to the speaker designed to reproduce it: the bass to the low-frequency speaker and the treble to the high-frequency speaker. Otherwise, the performance of the complete speaker system would be compromised, including blowing up the high-frequency driver with bass frequencies it can't handle. The frequency response of a typical, idealized two-way crossover is shown in Figure 15. The point where the curves "cross over" is called the crossover frequency. The slope rates of 6-, 12-, and 18-dB-per-octave designs are shown for reference.

Usually, the crossover is placed *after* the power amp, between the amp and the speaker components in question. Such a crossover is often called "high level" (meaning it comes after the relatively large voltages and power levels of the power amp) and "passive" (meaning it has no electronics or a need to be plugged into an AC outlet). In a bi-amplified system, the low and high division occurs *ahead* of the power amps, so that separate low- and high-frequency amps connected directly to the corresponding low- and high-frequency speakers are required. This is where the term "bi-amplification" or "bi-amp" comes from. Also, the crossovers used in bi-amped systems usually incorporate electronic components in them and must plug into an AC outlet, so they are called "active" crossovers. They are "low-level" active crossovers because they work on the few volts which come out of the mixer, instead of the high-level signals delivered by the power amps.

Bi-amplification is a *necessity* for a flexible component PA system. High-frequency horn/driver combinations are usually much more efficient than bass boxes – by a factor of 3-to-5 times. Only separately controlled high- and low-frequency power amps can efficiently compensate for this difference so that a smooth response in the room will result. Bi-amplification, with its separate high-frequency amps, also provides a means to balance short-, mid-, and long-throw horns in the more elaborate component setups (see the "Large Size Room" section beginning on page 13).

Also, bi-amplification reduces the audible effects of overdriven power amplifiers, a situation which occurs from time-to-time in even properly designed and operated systems. For example, in a conventional full-range system, if low frequencies clip the power amp the unpleasant high-frequency distortion products which are generated are reproduced all-too-cleanly by the speaker's high-frequency components. In a bi-amped system, these distortion products are fed only to the woofers, which do not reproduce them so well. Similarly, if high or mid frequencies clip the arm in a conventional full-range system, low-frequency distortion products may be generated which will be reproduced by the system woofers. In a bi-amped system, these distortion products would never get to the woofers for reproduction. (Some power amplifiers essentially eliminate these clipping-related distortions by preventing the amp from going into clipping in the first place. Examples are the TAPCO amps with PowerLock.™)

When using a bi-amplified system such as the one shown in Figures 13 and 14, the sound level of the high-frequency horns must be balanced to the sound level of the low-frequency speakers. One simple method of accomplishing this is to turn the gain controls of all power amplifiers all the way up and turn the high-frequency level control on the crossover all the way down. (If the crossover has no such control, use in its place the gain

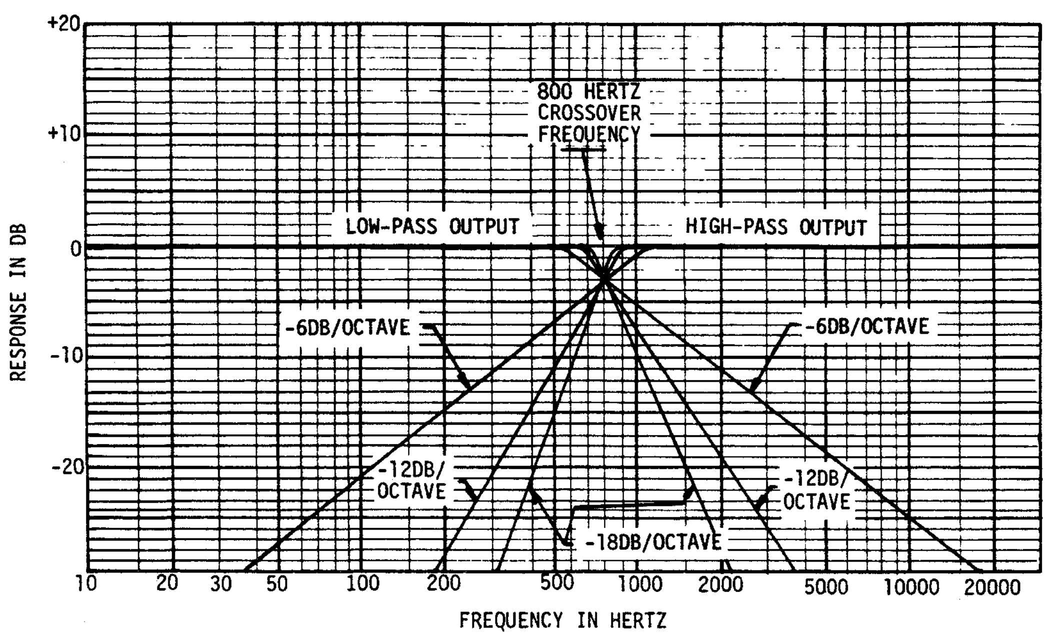


FIGURE 15 — Ideal Crossover Frequency Response

control on the high-frequency power amp.) Then, while speaking or singing into the microphone you usually use, bring the high-frequency level control on the crossover up until a natural, balanced sound is obtained. This balance may also be achieved in a more scientific way by employing acoustic instrumentation – such as a one-third-octave real-time spectrum analyzer – which will display the frequency response of the system in the audience area. Adjust the crossover high-frequency level control for the most uniform response in the crossover region (800 Hz).

Large Size Room. If you are designing a system to work in a large room, say 50 feet x 90 feet x 20 feet high, or a room volume of 90,000 cubic feet, the use of components is the only practical way to get proper coverage and sound quality. Please don't forget that stacking several self-contained, full-range systems (such as used in smaller rooms) *will not* produce good results in a large room!

The system shown in Figures 16 and 17 uses separate short- and long-throw high-frequency horns whose narrow coverage angles give good, intelligible coverage in the entire room. As illustrated in the elevation view of Figure 16, the highly directional HR40 40° x 20° horns are aimed straight back to fill the back of the room. The HR90 90° x 40° horns are aimed down slightly to evenly fill the front of the audience. You may want to refer to the "Double Distance Rule" and "Room Reverberation Swamps Your Voice" sections which start on page 4 to refresh your memory.

The system block diagram is shown in Figure 18. This system uses an electronic crossover (such as the E-V XEQ-1) as described in detail in the "Medium Size Room" section on page 12. Only one XEQ-1 is required since it can drive many amplifiers as long as only one crossover frequency is necessary. Once again, the TAPCO 6001-RB mixer was chosen for illustration. A TAPCO C-12 mixer with the additon of another XEQ-1 crossover could give you a stereo system with more channels and flexibility if you so desired. Remember, a PA system with stereo capability must be used with intelligence and care, since you run the risk of having part of your vocals on one side, and part on the other, with only the people near the middle of the room hearing it all. The stereo system can, however, be used for special directional effects which serve to enhance the basically monaural vocal mix.

Since the large room system shown in Figures 16, 17, and 18 is bi-amplified, you will have to balance the level of the high-frequency horns against the low-frequency speakers. A simple way of accomplishing this can be done by ear, while speaking or singing into the microphone you usually use. First – with the amp driving the short-throw high-frequency horns turned down or off – balance the long-throw horns against the low-frequency speakers. Start with the gain controls of the low-frequency and long-throw high-frequency amplifiers all the way up and the high-frequency level control on the crossover all the way down. (If the crossover has no such control, use in its place the gain control on the high-frequency power amp.) Then, speak or sing into the microphone and bring the high-frequency level control on the crossover up until a natural, balanced sound is heard in the last half of the room served by the long-throw horns. Finally, move into the front half of the room served by the short-throw horns and advance the gain on their amplifiers until a similarly balanced sound as achieved. The appropriate high-frequency/low-frequency balance may also be obtained in a more scientific manner by employing a one-third-octave real-time spectrum analyzer which will display the frequency response of the system in the audience area. Adjust the gain and level controls for smoothest response in the crossover frequency region (800 Hz).

Monitor Systems. Up to this point all that has been covered are main or house systems. A discussion of "monitor" or "fold-back" systems is in order. In its simplest form you can send a signal from the main mix to your monitor. One way to do this is shown in Figure 19.

A mixer (for example TAPCO 6000-CF) is used to deliver a signal to one half of a CP120 for the main system and one half of the CP120 for an FM12-2 floor monitor. To go farther than this, some mixers such as the TAPCO 6000-R or the 6001-RB provide a separate "monitor send" output. This enables you to make a special "monitor mix" such as vocals only, etc. With this more sophisticated monitor system, the FM12-2 and FM12-3 can be used in a configuration such as the system shown in Figures 16, 17, and 18. The E-V FM-type floor monitors can be physically placed in four configurations as necessary for your application on stage, as shown in Figure 20.

Some Thoughts on Permanent Installation Systems. In most cases where a temporary sound system is used, the system is stacked on either side of the stage such as the systems shown so far. This usually cannot be avoided by the one-time operation of most groups, but it does have some problems. It blocks the view of some listeners and it causes interference of sound waves in the room because sound is coming from two points separated by a large distance. One somewhat more refined solution is to make one central cluster such as the one shown in Figures 21 and 22.

This is another way to supply sound to the 90,000 cubic foot room previously described. This would be a desirable way of installing a permanent system, but it becomes fairly complex for the touring group. With the central cluster suspended, the horns can be aimed to obtain even coverage over the whole room. In a permanent installation, the large HR horns may be a desirable component to use. The large HR horns (HR9040, HR6040, and HR4020) offer reasonably tight vertical control starting at 800 Hz, whereas the small HR horns (HR120, HR90, HR60, and

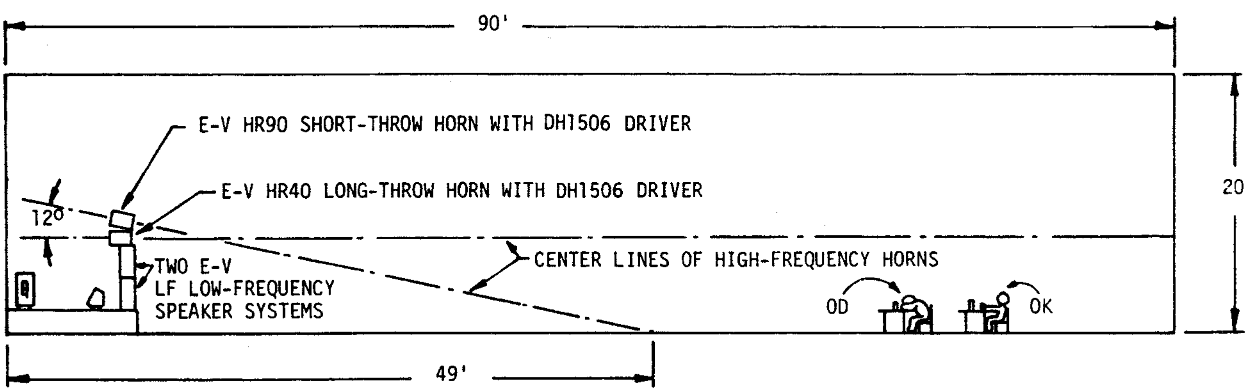


FIGURE 16 – Elevation View of Large Size Room

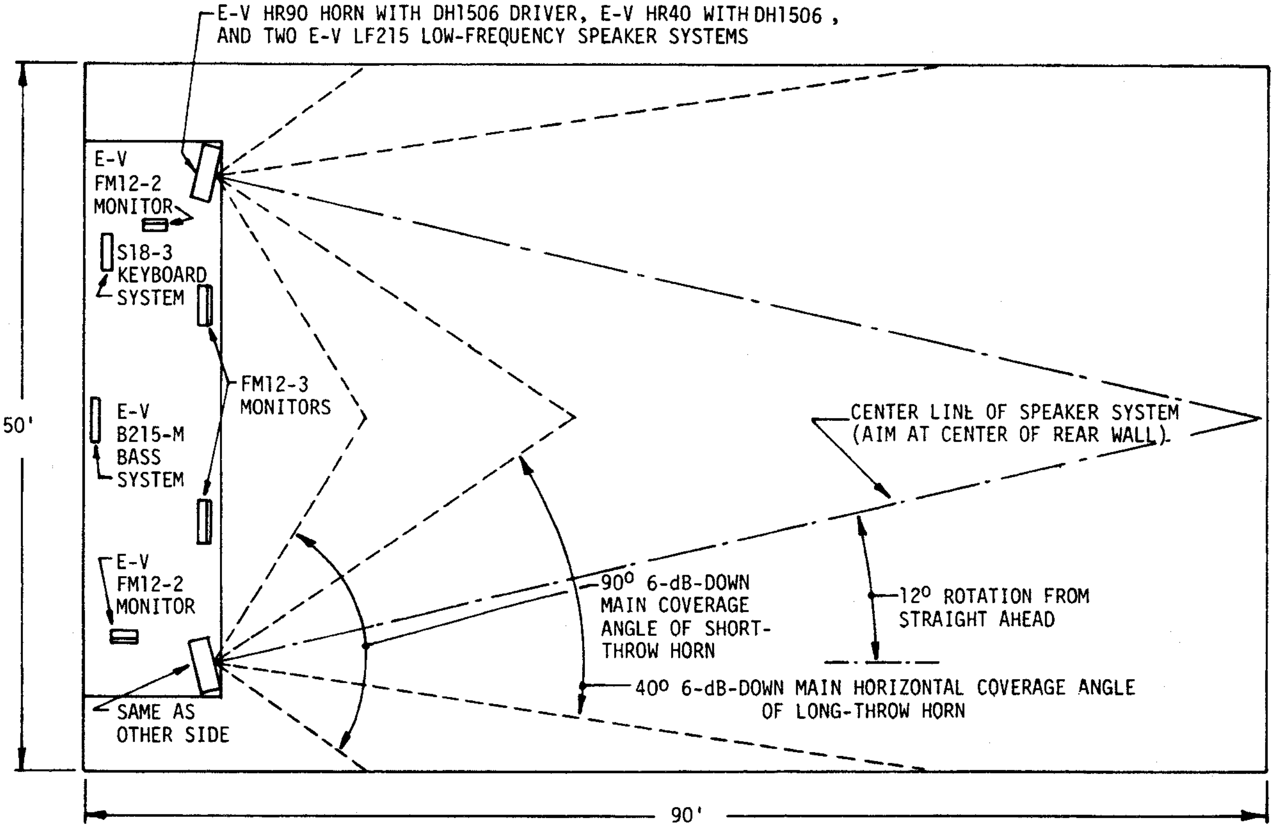


FIGURE 17 – Plan View of Large Size Room

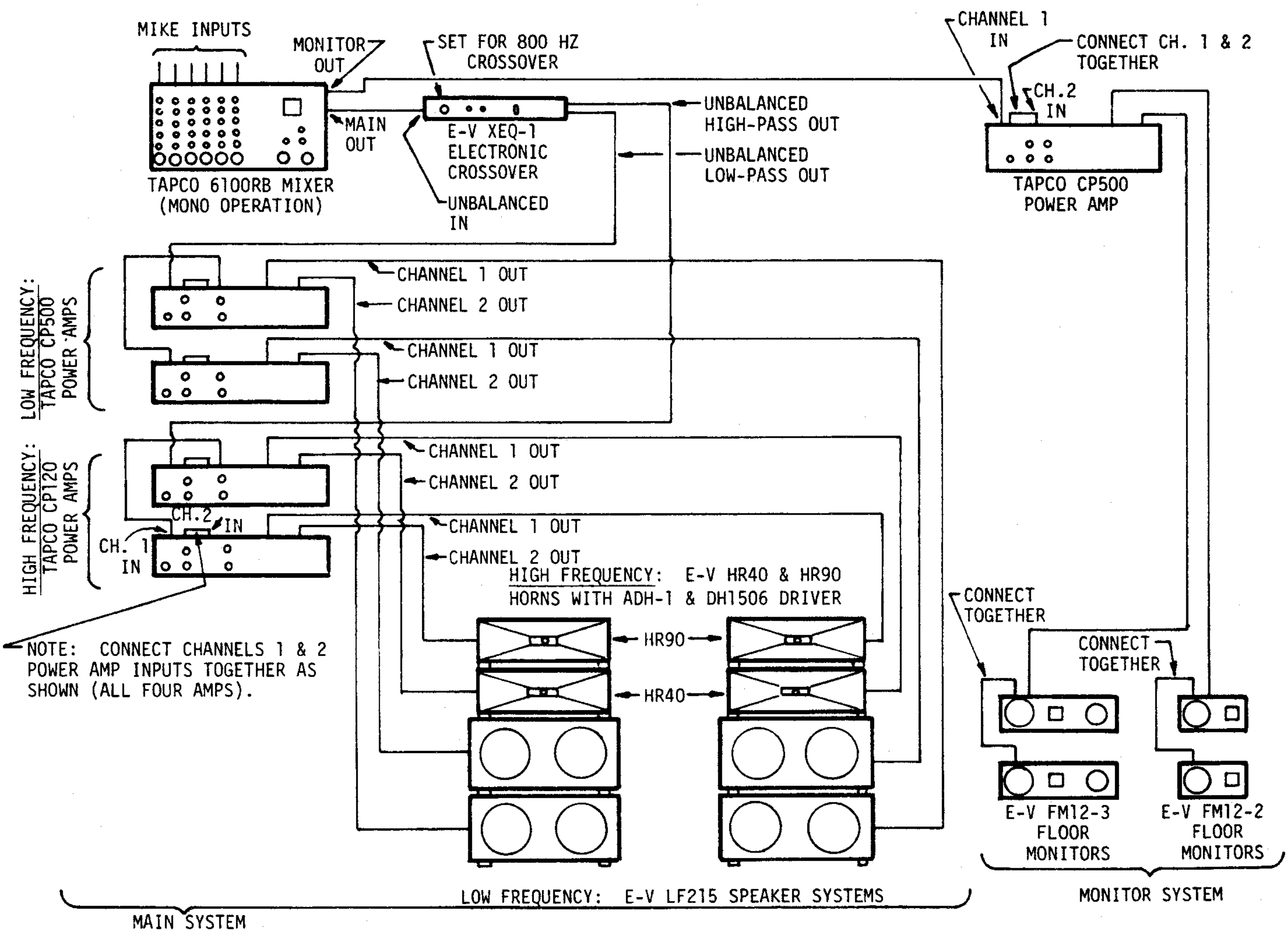


FIGURE 18 – Block Diagram for Figures 16 and 17

HR40) offer tight control starting at 1500 Hz. The large HR horns will, therefore, have better controlled coverage at lower frequencies and produce less reverberant energy than their smaller counterparts. They are the most desirable horn for a highly refined system when their increased size is acceptable. Bass systems such as those in the Electro-Voice TL series can be employed in permanent systems because the need for roadable enclosure construction can be eliminated. In this system two TL606D's are used to match the HR4020 and HR9040. This system will not have the same *maximum* acoustic output ability as the two-stack system shown earlier in Figures 16 and 17. However, it will only be 3 dB less overall.

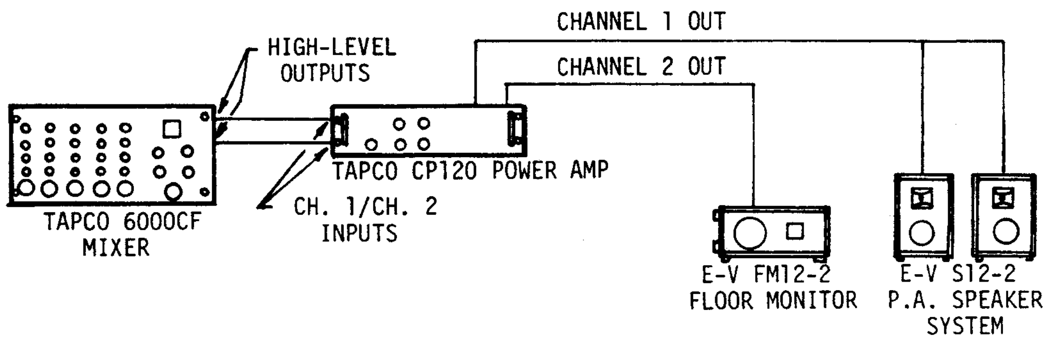


FIGURE 19 – Simple Monitor System

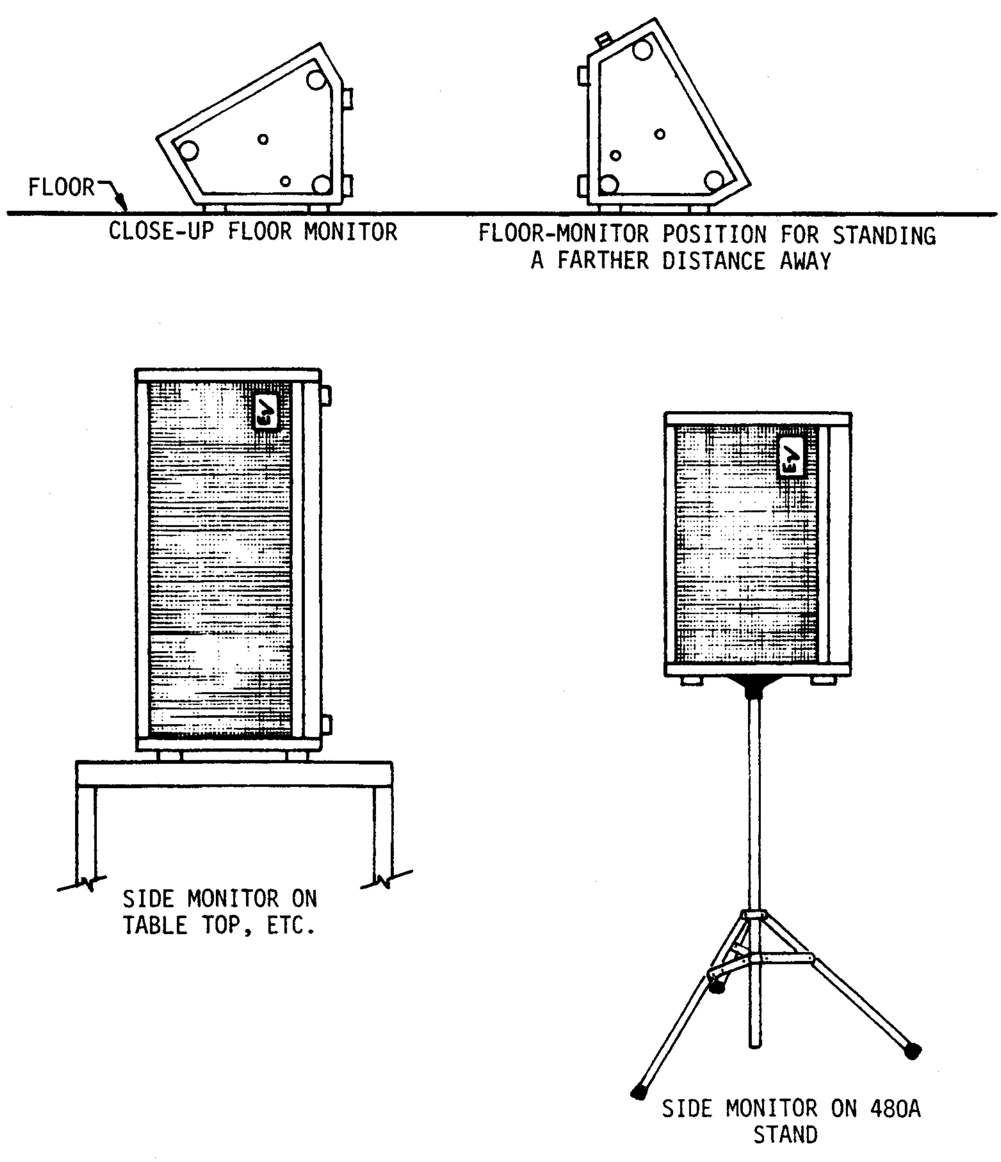


FIGURE 20 – Various Possible Placements of FM12-2 and FM12-3 Floor Monitors

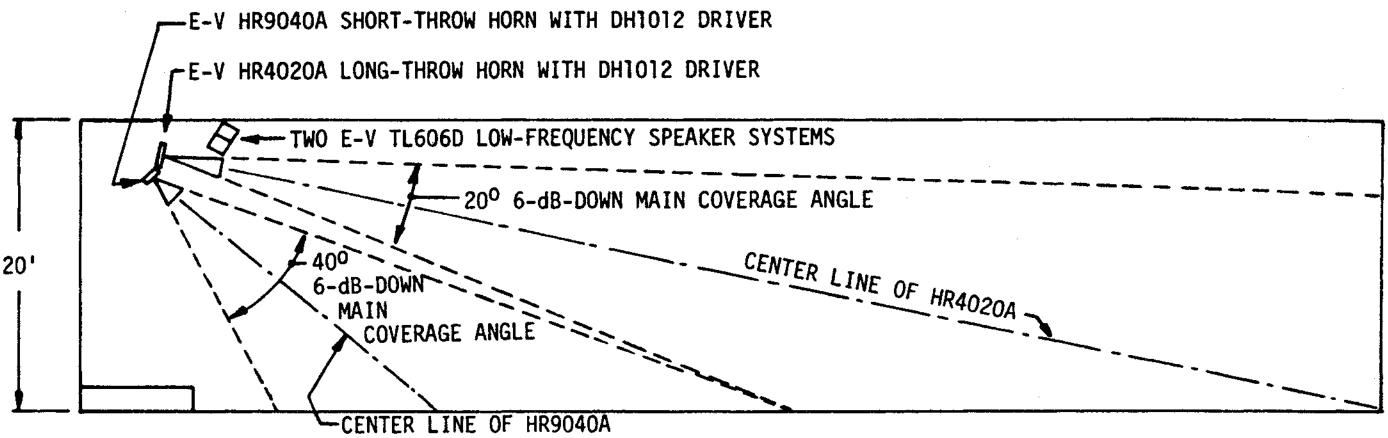


FIGURE 21 – Elevation View of Large Size Room, Permanent Installation

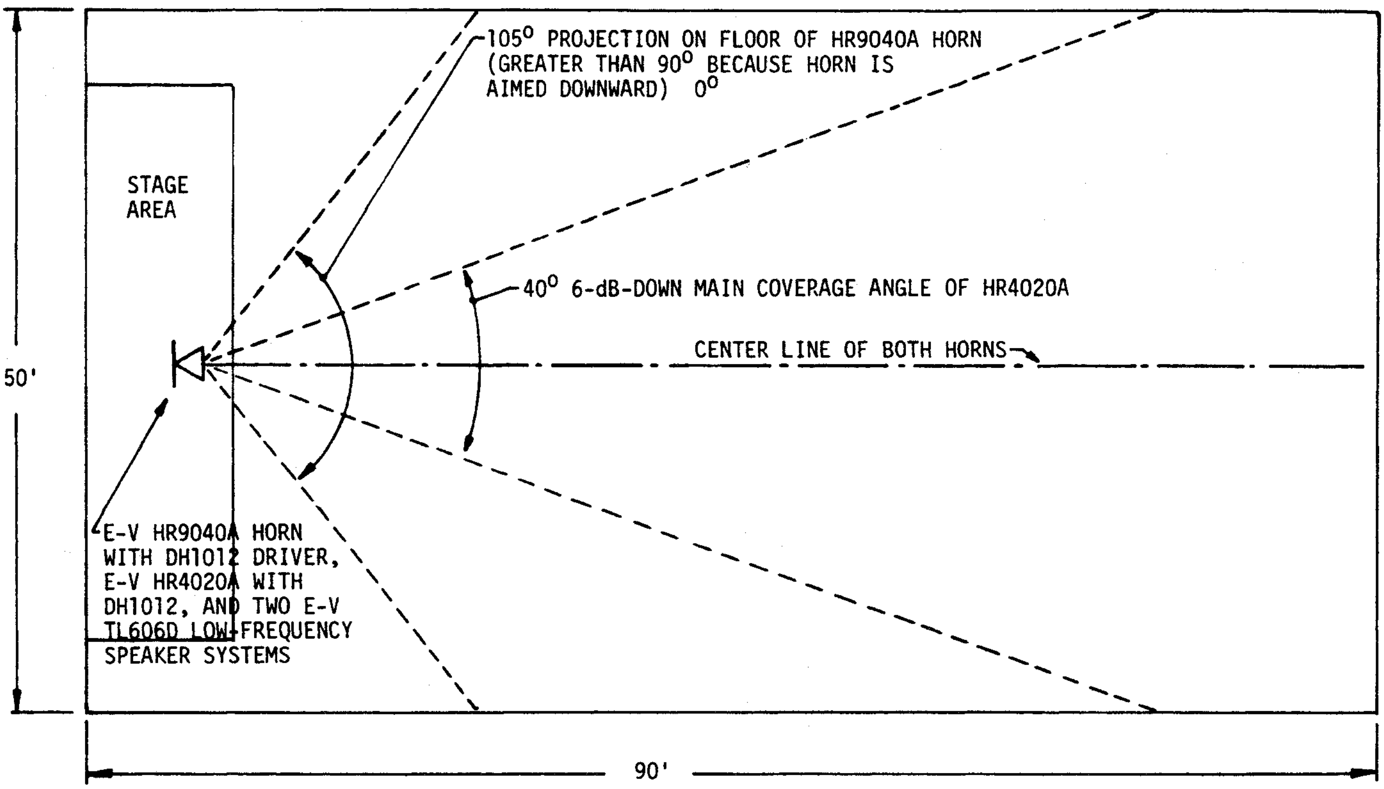


FIGURE 22 – Plan View of Large Size Room, Permanent Installation

SPECIAL NOTE TO THE READER

The E-V "PA Bible" has been prepared to help you solve your PA problems. In addition, we have begun a series of supplements which will expand upon and add to the basic "Bible". Let us know if you have any specific subjects in mind for us to tackle.

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THE PA BIBLE

ADDITION NUMBER EIGHTEEN
CONTROLLED SYSTEMS

What are Controlled Systems?

In a sense all loudspeaker systems are electronically controlled. They cannot function without an amplifier and this amplifier has control functions implicit to its operation. Such functions can be as simple as integral subsonic filters or even a single level adjustment knob. Control is a natural operating condition for all loudspeakers. The matter of significance is the degree and detail of control and the consequences associated with a given concept in controlled systems.

Historically, the earliest forms of explicit control may have been fuses and certain automatic devices intended to protect the more delicate components of a loudspeaker system. Such devices permit the system to be pushed closer to ultimate acoustic output capabilities with less damage risk than before.

During the last 10 or 15 years, the concept of controlled systems has been expanded and a number of manufacturers have offered their versions of this idea. These versions differ in various details of philosophy and execution. Figures 1 and 2 show a block diagram of a controlled system and a photograph of a typical system format.

What is the purpose? The intent of this concept is quite direct and simple. It is to permit maximum usable output to be achieved from a system with minimum risk of damage and minimal alteration of important sonic characteristics.

Note that once an electronic circuit becomes a part of a loudspeaker system, a number of possibilities beyond the core purpose become possible. As an example, it becomes relatively simple to incorporate circuitry to create systems such as Thiele's higher-order, assisted, vented box designs [Ref. 1] that require electronics for their realization. Electronic delays for loudspeaker acoustic alignment can also be included. However, it is best to separate out such possible additional functions from the basic concept to avoid clouding the main issue.

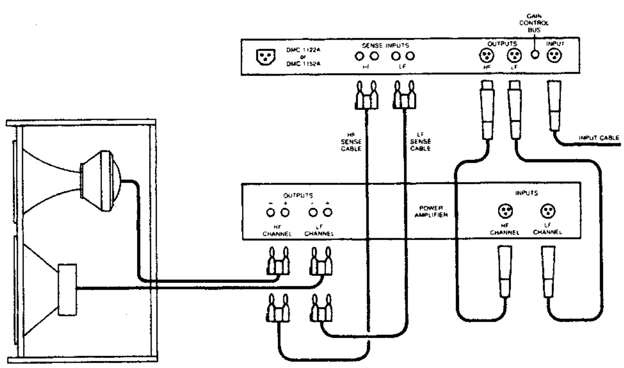

FIGURE 1 — A block diagram of a controlled speaker system.

Why Are Such Systems of Interest and Where Do They Apply?

Such systems are of interest because of the useful nature of maximum output from a given package with minimized risk of damage. This statement has been made previously, but it is important to keep in mind. This does not mean to suggest that all classes of loudspeaker products are appropriate for the full application of this concept.

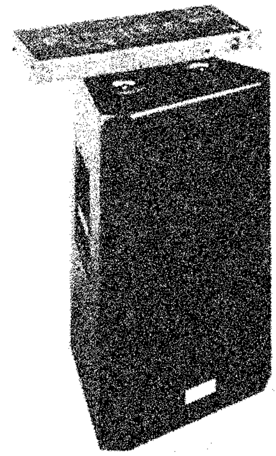

FIGURE 2 — DML-1122A speaker system with companion controller.

It should be stressed that this technique is not a substitute for good fundamental loudspeaker system design. The additional expense required to do a system of this type should not be secured by skimping on the acoustic components and then attempting to make the result more survivable. The intent should be to make pieces of fine machinery even more valuable when the circumstances warrant. The words "expense" and "value" should perhaps be expanded upon as their implications are not necessarily obvious.

A high-quality system incorporating a controller will cost more than a similar system not involving one. However, the value of such a system can be enhanced considerably. This is especially true when the system is pushed to near-ultimate output and cannot be overseen by human "controllers" who are intimately familiar with its characteristics under high-output conditions. The value of a system is substantially reduced when audible distortion intrudes on a performance. Its value is reduced to zero when failure occurs.

It might be argued that all systems should be moving in the direction of controlled protection. In fact this appears to be happening, although with various degrees of sophistication in the nature of the mechanics of control. Methods of control are being incorporated into many products, beginning with relatively simple devices such as fuses and relays that protect tweeters and ending with relatively elaborate multi-protection-mode devices with individual sections tailored to each loudspeaker in the system. As the degree of complexity increases, so does the degree to which the system can be pushed toward ultimate output with safety.

Are there arguments against controlled systems? Beyond the matter of additional expense associated with elaborate control methods (but recall earlier comments regarding the concept of value), arguments mainly fall into three categories. The first of these involves undesirable alterations in the sound of the system when the protection methods begin to kick in. To an appreciable degree this depends on the particulars of a given protection system. The next section will discuss this somewhat complicated topic and give some perspective to the situation. The second argument involves the loss of much human control since, to a substantial extent, the system is on "auto pilot" at high outputs. To a degree this is related to the first matter as the presumption is that skilled human control will result in audibly better results than those achievable under "auto pilot." The last argument involves the particular nature of controlled systems. This is that they are of a set, coordinated format that is unalterable. A given sophisticated controller cannot be switched to another loudspeaker system and function in an acceptable or even safe manner. Total systems are packages that are unalterable from their set format.

What is Being Controlled and What Are the Consequences?

A discussion of loudspeaker protection is typically divided into thermal and excursion considerations [Ref. 2 and 3]. The actual operating situation is made more complex by the interactions between these two conditions. A third category, not so commonly described, involves the consequences of exceeding strength of materials bounds. This matter also interacts with the prior ones to provide yet more complexity to the situation. One example will be given below to illustrate the nature of interactions.

This example involves a strong continuous music signal raising the temperature of a loudspeaker and reducing the strength of an adhesive joint near the voice coil. A sudden high-excursion producing explosive input signal (say a drum rim shot) accelerates the coil. The various conditions cited cause very high sudden stresses in a weakened joint resulting in complete or partial failure. This example is meant to illustrate the complexity in combined failure mechanisms which must be dealt with.

The matter of what is being controlled to ensure safe system performance differs from manufacturer to manufacturer [Ref. 4]. The main functions that are typically chosen for control are discussed below.

1. **Amplifier gain.** The gain of the amplifier can be reduced as maximum system output is approached. This can affect the various system channels (typically two) independently or together.

2. **Peak limiting.** The peaks of short-term transient output can be limited in various fashions when they exceed safe limits. This is usually a protection from over-excursion and avoids stress and damage to the moving system.

3. **Dynamic frequency response tailoring.** When used, this measure typically rolls off output at the frequency extremes to conserve available amplifier power for the middle registers. At low frequencies, over-excursion protection is again provided. At high frequencies thermal protection can be provided by reducing the normally provided amplifier boost used to equalize the upper two octaves of constant directivity horn designs.

4. **Crossover frequency shifting.** At high input levels the crossover frequency can be shifted upward, taking program away from the more delicate high-frequency driver and directing it toward the usually more robust woofer.

The control methods that cause the greatest concern are those which alter the spectral balance, or frequency response, of the system. At some point, enough of an alteration will be audible to a degree that most listeners would notice and judge undesirable. This can be imagined from the circumstance of going to a home music system and rolling off bass and high frequencies. It might also be imagined that a scheme that involved altering the general level of the woofer and high frequency section in a non-unison fashion could also prove unpleasant if done to any appreciable degree.

The condition of shifting a crossover frequency to a value more conducive to greater protection is a rather complex situation to definitively comment upon. In order to be effective, a shift of a half octave (a factor of 1.41) or more would be required. A number of loudspeaker behaviors would need to be well enough controlled in order to not be audibly objectionable. This would include:

a. smooth response over the alteration range;
b. sufficiently behaved phase characteristics to allow summing the net response well;
c. off-axis response (i.e. "polar" patterns) that don't change too abruptly over the desired alteration range.

Adequately good behavior of all these matters would normally be difficult to achieve over an appreciable frequency range and would make this protection method a chancy one at best.

The most workable protection methods are the ones that involve carefully done overall compression (uniformly applied to the whole system to avoid low-high spectral imbalance) and peak limiting done in the most audibly graceful manner achievable. It is felt that these "bread-and-butter" approaches, when done in a sophisticated and audibility-conscious manner, offer a legitimate means of providing a high degree of protection without undue compromise.

The Future of This Technique

In the future there is little doubt that controlled systems will become both more widespread and sophisticated. However, certain complicating and restrictive matters will be favorably modified or eliminated. Creative thinking and creeping Darwinism* are bound to exert their influence to elevate the general usefulness of this class of system.

In virtually all cases, current approaches presume a "model" for the behavior of components set into a specific system. Based on this model the controller exerts programmed input voltage alterations based upon determined ultimate output capabilities. This model is essentially built into the control electronics which then keeps track of the loudspeaker's input voltage and frequency. It then adjusts them in a programmed manner designed to keep the loudspeaker out of dangerous territory. This entire process is very

*Left alone, solutions to acoustic problems will eventually be resolved by a creeping Darwinian process: better solutions will occur in use until they eventually become the norm.

interconnected and is typically applied to a particular set of parts used in a well defined manner. Some of the specific matters that can determine the nature of the loudspeaker system model are as follows. (Note that these statements presume a two-way vented-box-format system. This is the most common type encountered.)

1. **Box characteristics**
   a. volume;
   b. tuning frequency;
   c. thermal insulating properties;

2. **Woofer characteristics**
   a. Thiele and Small (operational) parameters;
   b. excursion (cone travel) limitations;
   c. thermal input power capacity;
   d. changes in characteristics at elevated inputs;
   e. strength-of-materials limits;
   f. crossover frequency;

3. **High frequency section characteristics**
   a. excursion limitations;
   b. thermal input power capacity;
   c. changes in characteristics at elevated inputs;
   d. strength-of-materials limits;
   e. crossover frequency;
   f. acoustic impedance nature of the specific horn.

Two observations that come from this complex set of characteristics are that extensive and detailed testing is needed to arrive at the best controller dynamics and that the controller is intimately connected (dedicated) to the exact nature of the loudspeaker system. These observations should suggest some possible future directions. A third observation is that what is being dealt with is a combination of subsystems. The box and woofer, when treated together form one subsystem. The high-frequency section is a second subsystem usually contained within the low-frequency subsystem. Together these two sub systems, each requiring individualized but coordinated control, form the total system.

Although a starting point for determining appropriate controller operating dynamics are the "generic" power ratings of the components involved, a considerable amount of testing is required to reflect the influence of system details and the complex interactions that take place under actual use. "Actual use" is a complication within itself as the nature of an input signal can be very diverse and difficult to categorize because music and sonic events are by their very nature diverse and changing.

The generic power ratings used on components and systems (typically assessed using band-limited noise inputs) are useful for making comparisons but they are not sufficiently detailed and diverse enough for the highly sophisticated task of controlled system design. As a result, extensive testing using varied music signals is or should be used to set proper steady-state and transient controller characteristics. With time and perspective, it is suspected that improved ways of gathering endurance information will be developed. Perhaps data will be able to be gathered in a way which distills the number of variables to a more fundamental and accessible set.

Improved loudspeaker data gathering, as used to define the loudspeaker model, can only contribute to increased growth in the application of the technique. This should lead to greater diversity in the combinations of subsystems that can be chained together to form total systems.

Controlled systems are currently in a constantly evolving condition with new entries periodically appearing. This is a result driven by the basic merit behind the idea. In a sense, the concept began decades ago with simple mechanical relays and fuses. It has progressed to specialized systems with dedicated controllers whose control functions and general philosophy vary between manufacturers. The persistence of a good idea will cause continued refinement in the mechanisms of control. It is also expected that more sophisticated ways of assessing loudspeaker characteristics will result in a greater variety of systems suited to ever more general use.

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SPECIAL NOTE TO THE READER

The EV "PA Bible" and this addition have been prepared to help you solve your PA problems. Let us know if you have any other areas in mind for us to tackle. If you don't have a copy of the "Bible" and would like to,

You may call, write or E-mail any comments to:

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EV THE PA BIBLE

ADDITION NUMBER SEVENTEEN
WHAT IS MANIFOLD TECHNOLOGY™?

Manifold Technology™* is the blending of the output of several loudspeaker elements. This combined sound source acts as a larger, more powerful loudspeaker. It can project sound directly into an area or be used to power a horn.

The far reaching significance of this simply stated concept became the subject of numerous discussions and experiments at Electro-Voice. This gestation period began in 1983 and resulted in the introduction of the first completely manifolded product, the MT-4, in 1986. Electro-Voice is proud to be the first to recognize the potential inherent in Manifold Technology™ and to offer the first product featuring this concept. We would like to share our knowledge and enthusiasm with you through this PA Bible addition.

MANIFOLD TECHNOLOGY™ BENEFITS: MORE FROM LESS

Simply stated, Manifold Technology™ is a way of obtaining more acoustic output from a smaller, lighter, better performing and less expensive package. This is done by avoiding unnecessary duplication of certain parts of a loudspeaker system. As an example, the horn portion of a horn loudspeaker system can be fed by a number of horn drivers instead of stacking several horn-plus-driver assemblies together. The saving of space and weight is immediately obvious. What is not so obvious are the performance improvements that result as well. The useful characteristics of Manifold Technology™ can be summarized in five ways.

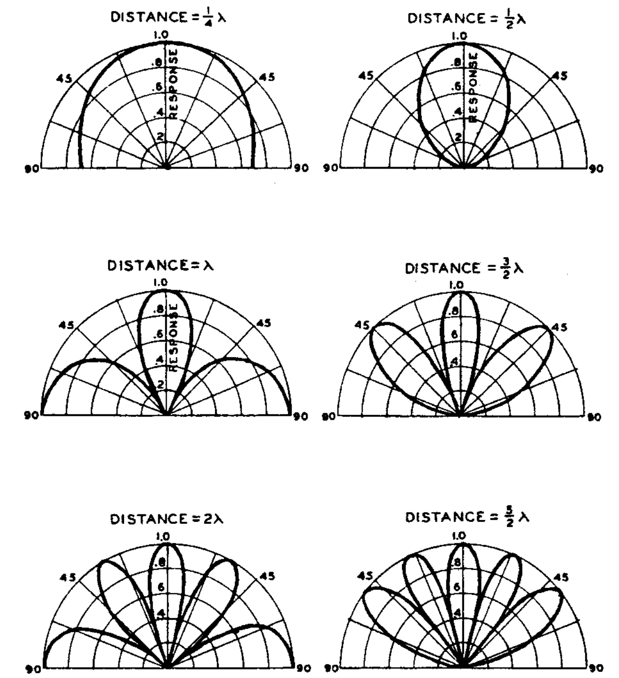
1. At low frequencies (below approximately 200 Hz) the outputs of a number of woofers can be effectively coupled to act as one very large loudspeaker. This maximizes acoustic output and minimizes the cone movement (excursion) needed at high output levels. Additionally, the variable air mass loading of the manifold chamber (see Ref. 1 for a detailed description) results in an enclosure size that is smaller than usual for a given level of performance. (Performance is meant to indicate the amount of usable acoustic output available over a frequency range.)
2. At higher frequencies the number of directivity-controlling structures (typically horns) in front of the loudspeakers or drivers can be minimized. This results in less bulk, lower weight, lower cost and the elimination of interference patterns caused by several sound sources operating at the same time.
3. Driver redundancy is provided. Sound will not be totally lost if some manifolded loudspeakers should fail.
4. The required total piston area needed for a given level of performance is subdivided into smaller sizes that don't "break up" (cease to act as a unified sound source) as readily.
5. Manifolding permits the fabrication of large voice coil circumferences for a given piston size. This has favorable thermal implications because of the increased coil surface area. More area results in reduced coil temperature. (Example: Four two-inch-diameter driver diaphragms with the expected two-inch-diameter voice coils, when taken together, act as a single four-inch diaphragm which has — as if by magic — an *eight*-inch coil diameter.)

**Electro-Voice has chosen this proprietary term to describe the physical process taking place.*

The increased output from a given volume of space that manifolding provides, leads to smaller loudspeaker system size for a desired level of loudness. It can also mean the ability to achieve greater output when system size or weight must be restricted.

INHERENT ENGINEERING PROBLEMS

The main problem lies in combining effectively the output from several loudspeakers or horn drivers. If the various outputs are not blended "in phase" there will be partial or complete cancellation of their outputs. The exact nature of these cancellations will depend upon the geometry of the various loudspeakers, the frequency involved and the position of the listener. A simple example of this phenomenon is shown in Figure 1, showing two sound sources displaced in space by a distance expressed in wavelengths. The resulting peaks and dips that occur at various listening positions are sometimes referred to as a comb filter effect.



Directional characteristics of two equal sound sources moving in phase as a function of their separation distance in wavelengths. The polar graph depicts the sound pressure level, at a fixed distance, as a function of angle. The direction corresponding to the angle 0° is perpendicular to the line joining the two sources (after Olsen, Ref. 4).

FIGURE 1

In the example of Figure 1, the cancellation process which causes the lobing takes place in the air in front of the two loudspeakers within the listening space. It would seem attractive to combine the two sources of Figure 1 into a single radiating device to avoid the anomalies shown. This is what Manifold Technology™ successfully achieves. However, the problems associated with combining a number of acoustic sources is difficult at high frequencies. This is primarily because:

1. It is hard to put acoustic sources close together in terms of the wavelengths involved. This is especially difficult at high frequencies where the wavelength can vary between a few inches to less than inch.
2. It is hard for sound waves to negotiate sharp bends without some cancellation occurring. This is particularly true when the dimensions of the bend are comparable to or larger than the wavelength of the sound.

The "yardstick" or critical unit of measurement that is used to help explain and categorize acoustic phenomena is the wavelength. Wavelength is related to frequency by the expression $\lambda = \frac{c}{f}$ where λ denotes wavelength, c is the velocity of sound (approximately 1130 feet per second) and f is frequency. Similar yardsticks occur in the evaluation of light and electromagnetic phenomena.

Sound wavelengths can vary from 69 feet (16.35 Hz) down to 0.68 inches (20,000 Hz). In order to have the outputs of several acoustic sources blend well, they must be spaced less than 0.5 wavelength apart at the highest frequency they are required to produce.

A crude but useful analogy can be seen by dropping two pebbles into water. (The wavelength or distance between the wave crests will usually be of the order of an inch in this case.) If the two pebbles are dropped close together they will act as one larger pebble. That is, the wavelets will go outwards from the drop point as well defined circles. However, if the pebbles are simultaneously dropped with a spacing of an inch or more between them, the situation will become more chaotic. There will be two distinct sources of wavelets which will tend to interfere with each other. So it is with sound waves.

At lower frequencies the wavelengths are relatively long. At 200 Hz the wavelength is 5.65 feet, at 100 Hz it is 11.3 feet and so on. At these frequencies it is possible to place a number of loudspeakers close together and have the centers between them be substantially less than a half wavelength apart. Their output act as one large speaker, without the spacing-induced interferences of Figure 1.

However, as frequencies increase, the combining problems increase as well. In many high frequency transducers there will be a directivity-controlling structure, such as a horn, in front of the loudspeaker (which in this case is often referred to as a horn driver or compression driver) so that the outputs of the combined units can be directed to the listener. Here the loudspeakers or drivers would be combined at the small end or "throat" or the horn.* At the lower frequencies mentioned above, directional control is difficult to accomplish due to size considerations. This is because the frontal dimensions of an *effective* directivity control structure needs to be roughly comparable to that of the wavelength being radiated. (At 50 Hz, for instance, these dimensions would approach 20 feet.)

At high frequencies the two conditions noted earlier make effective manifolding much more difficult. Being more specific, the size of the acoustic source can be as much as one to two wavelengths in diameter. An eight-inch source (representative of a 10-inch loudspeaker) is about 1.2 wavelengths across at 2000 Hz. In addition, especially in the case of compression drivers, the magnetic system that drives the diaphragm can be considerably larger than the diaphragm itself. A compression driver with a one-inch opening can have an outer diameter of four to six inches. These conditions make it difficult to crowd the acoustic sources close together (less than a half wavelength) so they will act together as a unified whole.

As a result, manifolding at high frequencies requires that the output of the various acoustic sources be channeled along passages that culminate effectively at the back of the horn. Invariably, this acoustic "plumbing" must be bent to permit the drivers to physically clear one another. This brings into play the second condition previously noted involving the difficulty of sound waves negotiating bends. Unfortunately, bent acoustic plumbing often alters the wavefront coming from the drivers so that effective blending does not take place. This situation is additionally compliciated as the number of manifolded drivers increases due to the increased complexity of the plumbing. Four drivers are more complicated than two, and eight drivers would be harder yet.

**A horn will be assumed in this discussion for sake of convenience However, it should be noted that it is possible to have directivity control structures that are not horns In a horn driver the acoustic source that would be manifolded is the hole in the driver casing This is often referred to as the throat size*

SOME SOLUTIONS

At low frequencies the usual system choices are either a horn or a vented direct radiator. Horns are sometimes selected because of their high-efficiency characteristics. It is not widely known that direct radiator systems can have efficiencies comparable to horns when multiple loudspeakers are used. Additionally, vented forms of direct radiators are more efficient users of space than horns [Ref. 1]. This translates into the vented format system having the smallest box size for a given level of performance. Since Manifold Technology™ in general offers maximum performance from minimum space, vented direct radiators fit nicely into the spirit of "more from less."

At higher frequencies the logical desire to control the directivity of acoustic output coupled with the availability of many fine horn drivers, makes horn transducers very attractive. Horns needed to cover the range above 200 to 300 Hz are relatively small compared to their low frequency counterparts, making them a desirable choice for reproducing all higher frequencies.

The problems inherent in creating high-frequency manifolds were mentioned earlier. They involve being able to pack the acoustic sources of the drivers close together and providing acoustic plumbing which does not permit destructive interference. Such interference can occur within the manifold or where the manifold exits into the throat of the horn.

The problems in manifold design usually lie at the top end of the range of frequencies they are required to reproduce. At these frequencies the wavelength is often becoming equal to or smaller than manifold dimensions. Here the technique of "geometric optics" (ray tracing) is starting to become a valid method of evaluating performance. Figure 2 illustrates a situation that can occur in a simple "Y" type of manifold. Here, rays that pass directly through the manifold passages can interfere with rays that reflect off sidewalls due to the differences in path length. A way of dealing with this kind of problem is to turn it into an advantage through a different design technique. Figure 3 shows a configuration in which most of the rays reflect off a suitably shaped wedge. In this case, all rays have equal path lengths and, therefore, do not interfere with each other when they assemble together at the throat of the horn.

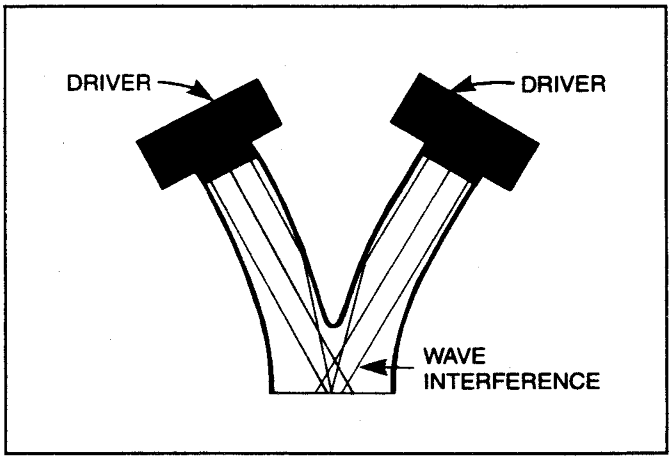


FIGURE 2
"Y" Manifold Showing Ray Interferences

It is possible to use this technique or a variation of it to combine the outputs of either cone loudspeakers used as horn drivers or actual compression drivers. In the former instance, it may be necessary to provide a suitable phasing plug in front of the cone to shape and phase-correct the wavefront that enters the manifold. The result is a compact manifold that can effectively blend together the outputs of several drivers. Although the illustrations show two driver situations, it is possible to extend the technique to greater numbers of drivers.

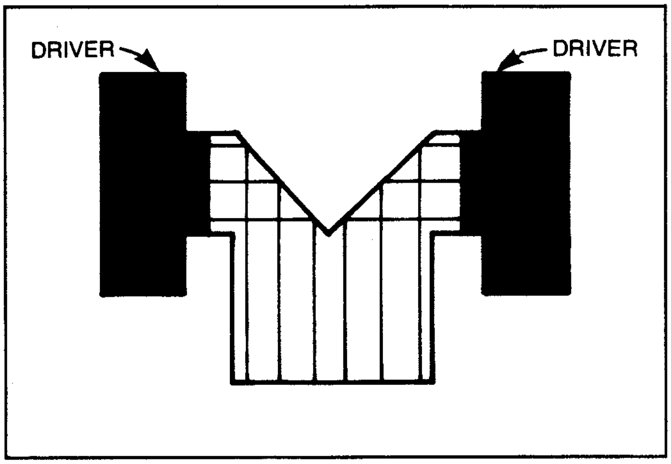


FIGURE 3
Wedge Type Reflective Manifold Showing Rays Coherently Reflected

SOME SPECIFIC REALIZATIONS

When the outputs of two drivers are combined effectively, the result is a doubling in output power. The subjective effect is quite audible and useful in many applications, but it is not dramatic. This is because hearing is not a linear process. A doubling of output is not heard as a doubling in loudness. If four drivers are combined, the result is very noticeable and begins to approach a doubling in perceived loudness. Four-driver manifolds were, therefore, chosen as a suitable upward step in the design of the first system to fully employ Manifold Technology™, the EV MT-4 [Ref. 2 & 3]. In this system the extreme bass is reproduced by transducers in their own separate enclosure and higher frequencies (160 to 20,000 Hz) are reproduced by three high-input-power horns. The critical decade range from 160 to 1600 Hz is handled by four ten-inch loudspeakers with specialized loading plugs feeding a manifold much like that shown in Figure 3. (The second set of drivers can be pictured as being below the first in this figure.) The throat of the wood horn connected to this manifold is a slot suitably proportioned to accept the geometry of the manifold. The upper part of this middle range (1600 to 8000 Hz) is reproduced by a small horn with mouth dimensions about the size of this sheet of paper fed through a circular two-inch opening. The manifold attached to this horn is a variation of that shown in Figure 3. It is shaped to blend the outputs of four horn drivers into a standard two-inch opening. The manifold is shown in Figure 4. The extreme top end (8000 to 20,000 Hz) is provided by four specialized tweeter drivers feeding an identical manifold/horn combination.

Low frequencies (40 to 160 Hz) are reproduced by a vented system with four 18-inch cone loudspeakers manifolded into a rectangular chamber. The loudspeakers are in reverse position to provide additional volume to the interior of the box and to put the heat producing part of the loudspeaker on the outside. This section (the MTL-4) takes advantage of variable air mass loading [Ref. 3] to provide an unusually compact box for the four 18-inch units. Figure 5 shows the entire system. The manifolding employed in each of the four sections is unique and the subject of several patents.

WHERE DOES THIS ALL LEAD?

Manifold Technology™ is such a fundamentally sound idea that there is little question about it finding increasing usage in the future. The concept is valuable whenever larger amounts of undistorted output are needed from relatively small packages. Recall that higher output does not necessarily mean louder. It more typically translates into achieving a desired loudness level at the listener's ears with the least amount of equipment. The concept is bound to eventually find its place alongside other acoustic milestones such as vented systems and constant directivity. It has the hallmark of a fundamentally correct concept waiting to be fully employed.

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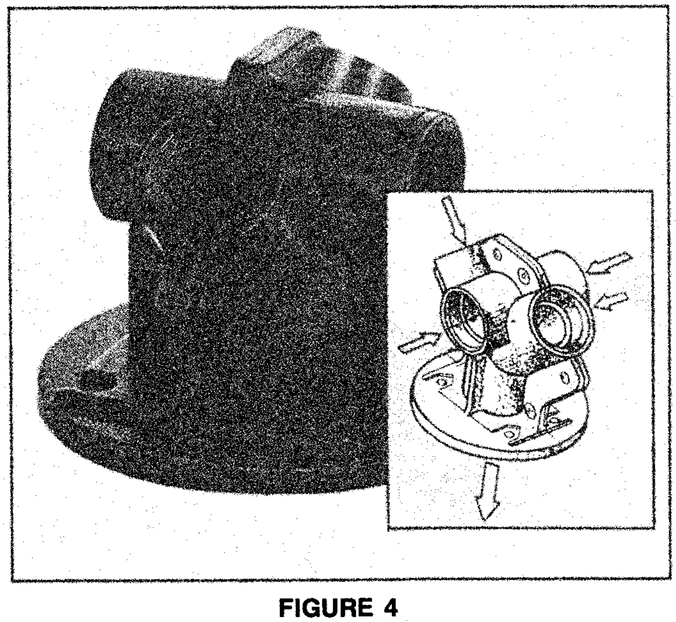

FIGURE 4
A Four-Driver Manifold for Use on Horns Having Two-Inch Throats (the accompanying sketch shows additional detail)

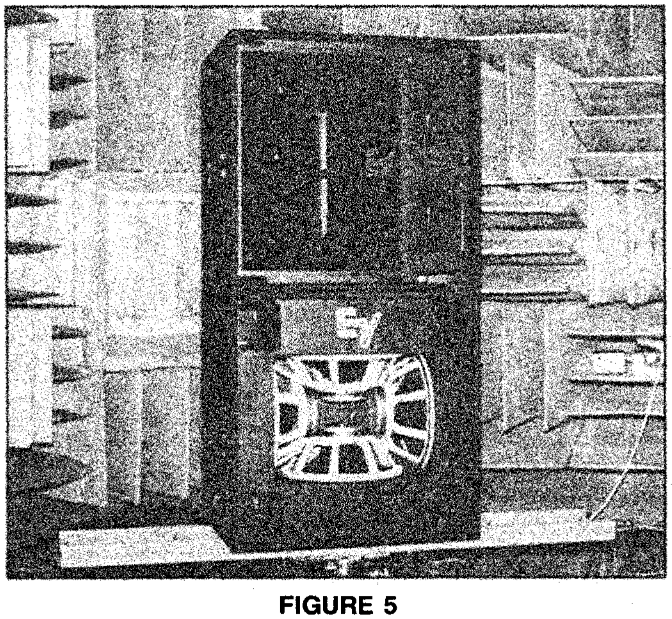

FIGURE 5
The MT-4 2-Box, 4-Way Concert Sound Reinforcement System. The MTH-4 High-Frequency Box is on Top with the MTL-4 Low-Frequency Box on the Bottom

EV THE PA BIBLE

ADDITION NUMBER SIXTEEN
MISMATCHING DRIVERS AND HORNS

INTRODUCTION

Professional audio components from various manufacturers are sometimes used together; essentially a "cross breeding" of different cultures or philosophies of sound equipment design or concept. This can be done successfully if certain practice is followed. It is the purpose of this article to show what will work and what won't.

Reasons for cross-breeding of drivers and horns vary. Here are a few common ones:

1) **In a Jam and/or The Show Must Go On.** This is a common occurrence. Eight drivers from manufacturer "A" blow up and a nearby dealer has drivers in stock, or for rent, from manufacturer "B." With two hours to show time, a decision to "make do" and bring the show up on time always is more heroic and favored than any other alternative.

2) **Availability and "Equivalent Units" on a Sound System "Spec."** This is identical to (1), only on a much longer time span. Cross-breeding on a fixed installation may occur by "picking" units from a shopping list of equivalents as a bid, depending on availability, contractor's stock, price, etc. Last-minute inability of a manufacturer to ship when promised may force a cross-bred system into existence.

3) **Listener Preference.** A customer, consultant or contractor may actually prefer the combination of a driver from manufacturer "A" and a horn from manufacturer "B" as a "best sound" combination. This is not unusual, but it does reflect a subjective opinion which may be singular. In a similar vein, certain combinations may exhibit a preferred performance, such as frequency response or power handling. Some or all of these may find their way into a specification.

4) **Some Existing Equipment on Hand.** A customer may want to upgrade his system with, say, new horns but doesn't want to buy new drivers when the old ones work just fine. He could buy the new horns, sell the old ones off and bolt on his old tried-and-proven drivers which he loves the sound of.

Sound familiar? It doesn't happen every day, but it does happen with some regularity. This article will attempt to clear up some mythology and misconceptions about the bizarre practice of cross-breeding high-frequency drivers and horns. It will also present recommendations for acceptable practice, based on available EV hardware.

LOUDSPEAKER MECHANISMS

Most "wide-range" professional compression drivers are similar in mechanism. Metallic, spherical diaphragms (aluminum, titanium, beryllium) are mounted in close proximity to a solid spherical-surface member, called the phase plug, which has slots or openings in it. These slots, on the spherical surface of the phase plug, join together inside the driver and "conduct" the sound generated by the moving diaphragm to the driver throat and its exit. The smaller the length of the sound paths on the surface of the phase plug, the better the high-frequency performance. All compression drivers have phase plug "throat" or entrance areas which are about 10% of the "effective piston area" of the diaphragms. This is the net sum of all the individual slot or slit areas. Figure (1) shows typical compression driver construction.

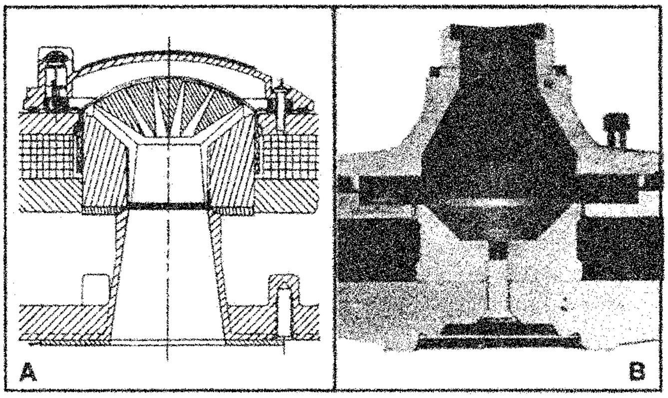

FIGURE 1
Compression Driver Cross Sections
A) Conventional Concave-Drive B) Convex-Drive (after DH2)

"SMALL-FORMAT" DRIVERS

Most "small-format" drivers are either one-inch bolt-on or one-inch screw-on, which is to say that the actual effective acoustic exit aperture is a one-inch diameter hole. These conventions are shown in Figure 2.

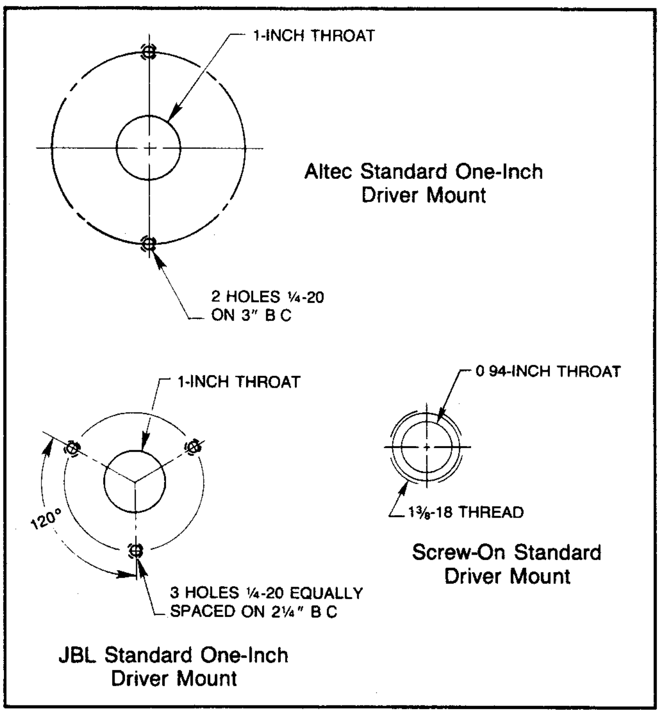

FIGURE 2
One-Inch Throat Configuration

Most "small" pro drivers have between 1.75 (JBL, Altec) and two-inch (EV) diameter effective pistons. There is some use of these drivers on small-format horns, where the horn merely bolts or screws onto the horn, and that's that. There's no real problem here. The EV ADH-1 will convert a one-inch screw-on driver, like the DH1506, to a one-inch flange mount configuration. However, there seems to be more of a proliferation of "adapters" for so-called "one-inch drivers" than for any other format. An example is shown in Figure 3. This is probably because:

a) there are more varieties of large-throat-format horns "out there"; and,
b) in low-SPL applications, such as churches, a small-format driver will fit the bill and is generally more cost-effective than a large-format unit.

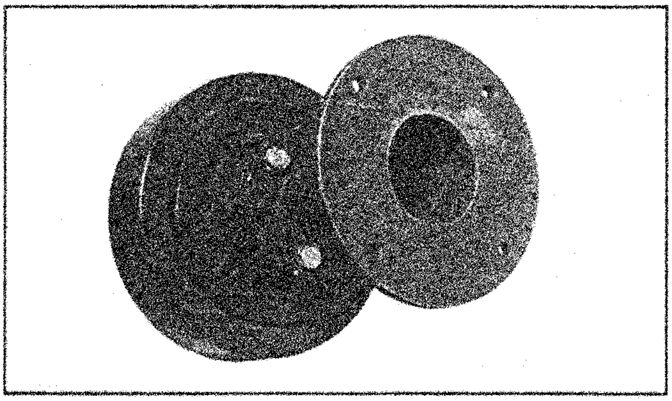

FIGURE 3
Driver with Adapter "Plumbing" for One-Inch to Two-Inch Conversion

A useful exception to this is a design philosophy developed at EV. We have lived for years with various "ADH" adapters, or acoustical plumbing, as a "necessary evil" of different throat formats. When we introduced the HP horn series with a standard two-inch throat (actually $1\frac{15}{16}$) we were again faced with the prospect of throat adapters, knowing full well the problems herein; the main one is expense of installation due to extra time needed to unpack an extra carton, sort out screws, gaskets and hardware, actually mount the adapter, etc. We concluded that the best solution was to just build the throat so it extended to a two-inch exit within the driver and *forget* adapters. This resulted in the unusual design of the DH2 (Figure 4). Not unexpectedly, this concept has been eagerly accepted by our existing and growing customer base.

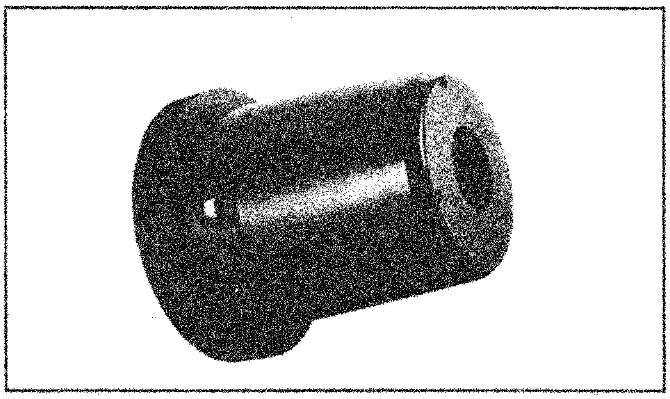

FIGURE 4
DH2: No Adapter

SOUND CHARACTERISTICS OF SMALL- AND LARGE-FORMAT DRIVERS

Small diaphragms have less low-frequency (400-800 Hz) output *capability* than larger ones according to the following equation:

$$\begin{aligned} \text{Peak } W_{out} &= 0.00472\ f^2\ X^2_{max}\ S_D\ T \text{ (watts)} \\ f &= \text{frequency (Hz)} \\ X_{max} &= \text{diaphragm-to-phase-plug spacing (in.)} \\ T &= \text{compression ratio } = S_D/S_T \\ \text{Where } S_T &= \text{throat area (in.}^2\text{)} \\ S_D &= \text{diaphragm area (in.}^2\text{)} \end{aligned}$$

Small-diaphragm drivers also tend to have stiffer suspensions or higher principle resonances, giving them less *relative* low-frequency response. Lastly, a smaller diaphragm begins to "break up" at higher frequency (in the 10-20,000 Hz region). Therefore, they sound smoother and tend to have smoother frequency-response curves. Actually, most larger drivers tend to have better high-frequency *level* or output around 10,000 Hz or so, but smaller diaphragms tend to be smoother. Larger diaphragms, on the other hand, tend to have a "fuller" vocal sound at high levels because of their greater low-frequency output and response capability, especially when used with a 500 Hz crossover. The point here is that you make your choice and accept the corresponding tradeoffs, whether you like it or not.

LARGE-FORMAT DRIVERS

High-frequency drivers whose coils are larger than about 2¾ inches fall into the "large-format" class. This more generally manifests itself as a certain size of exit aperture. A throat which is larger than one inch generally qualifies as "large format." Unlike one-inch throats, there are at least three conflicting large-format-bolt pattern/exit-aperture combinations: Altec/Yamaha, "old EV" and "standard two-inch" (new EV, TAD, JBL, Gauss, etc.). These are shown in Figure 5.

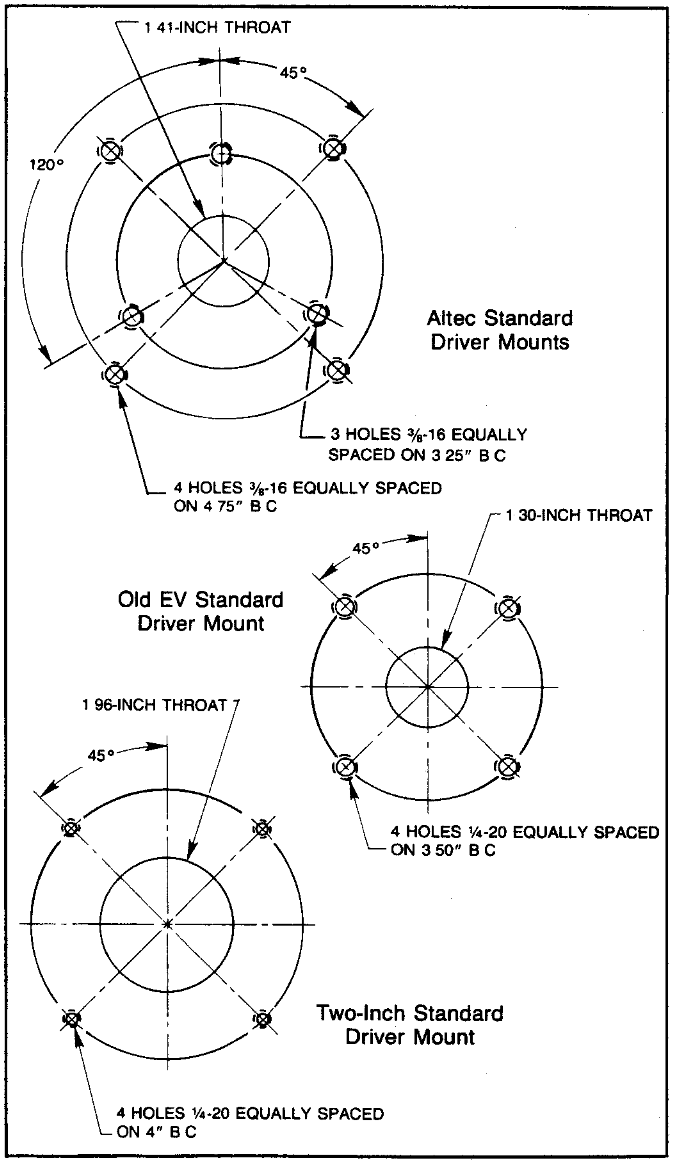

FIGURE 5
Large-Format Throat Configuration

Behind each of these throats is a relatively large diaphragm, representing a specific philosophy of driver design. Here are some differences:

Driver	Diaphragm Contruction	Compression Ratio	Coil Diameter	Throat Diameter
EV DH1	Aluminum w/Polyimide Suspension	10	3″	1.93″
EV DH2	Integral Titanium	10	2″	1.93″
Altec 288	Integral Aluminum	10	2.83″	1.414″
Altec 291	Aluminum w/Kapton Suspension	10	2.83″	1.414″
JBL 2445J	Integral Titanium	10	4″	1.93″
TAD 4001	Integral Beryllium	10	4″	1.93″
JBL 2482	Integral Phenolic	10	4″	1.93″

Here are what amount to a wide variety of different opinions on what music should sound like when it comes out of a compression driver. If you think in terms of musical instruments, they all should sound very different, being made of materials which have entirely different resonant character. However, of the "wide-range" devices here which are useful to above 15,000 Hz or so, differences tend to disappear dramatically when set into a balanced sound system in a reasonably sized auditorium. As an "acoustic appliance" all the popular items do a creditable job. However, when you try to interchange drivers and horns, several possibilities of "mismatch" present themselves. These are as follows:

a) Driver exit smaller than horn entrance.

b) Driver exit larger than horn entrance.

c) Driver "flare rate" different from horn with the same size throat.

All these combinations can lead to less-than-ideal behavior, due to the unplanned-for driver-horn interface. The first two can lead to high-frequency reflections which are harmonic or repeat themselves at regular musical intervals; "bumps" in response above a certain high frequency. Here are some examples of this. Figures 6 and 7 show the performance and impedance of a well-known driver/horn combination. Response and distortion are not unusual for a "constant-directivity" horn design, this combination being intended for 800-Hz-and-above use. The same driver (Altec 288-8K) is now fitted to an EV HP9040 2-inch throat horn, an area mismatch of about 2:1. The performance and impedance of this combination are shown in Figures 8 and 9. Note that a huge loss of output (a "hole") develops between 5,000 Hz and 12,000 Hz and that distortion levels (remember, percent distortion is *relative* to the fundamental) jump considerably in this region. The rest of the range is relatively normal. For a contrast, Figures 10 and 11 show the behavior of a correctly matched 2-inch driver/horn combination, the EV DH1 on the same HP9040. Compared to Figures 8 and 9, the gaping hole is gone and a smooth response is in its place. Distortion levels are generally lower here, depending on range, these being a function of the driver design.

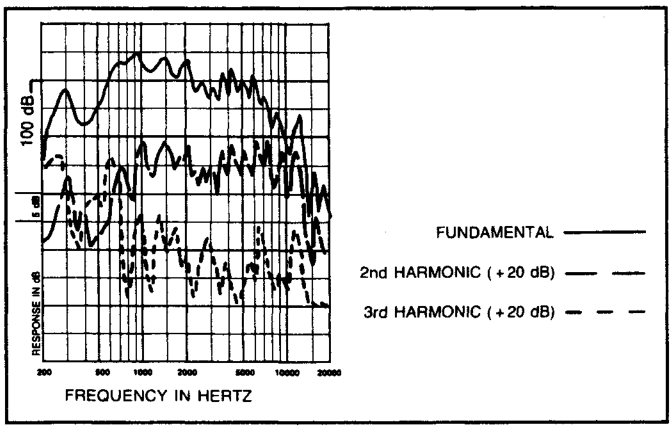

FIGURE 6 — Response & Distortion; Altec 288-8K Driver on Altec MR-94 Horn (1 watt, 10 feet)

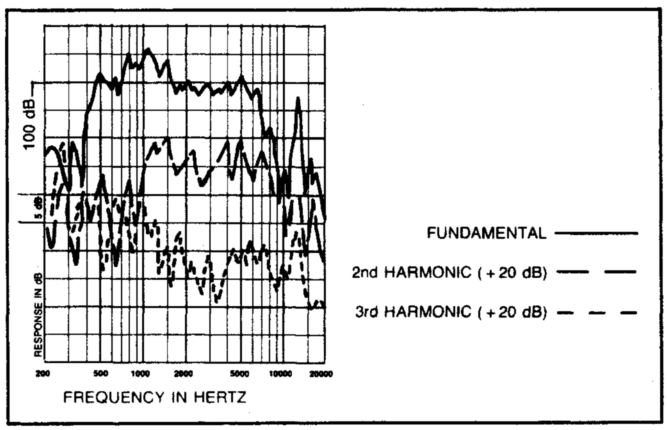

FIGURE 8 — Response & Distortion; Altec 288-8K Driver on EV HP9040 Horn (1 watt, 10 feet)

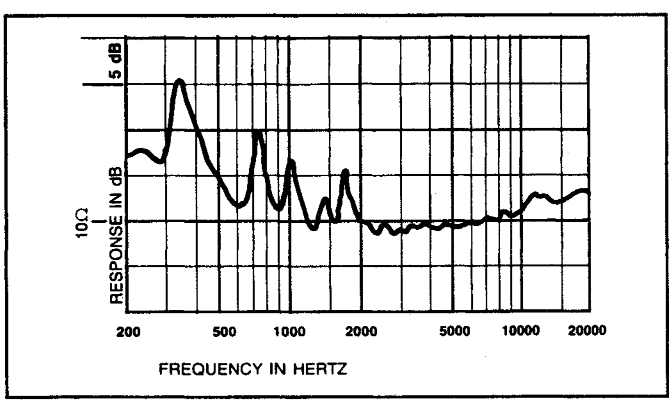

FIGURE 7 — Impedance Altec 288-8K Driver on Altec MR-94 Horn

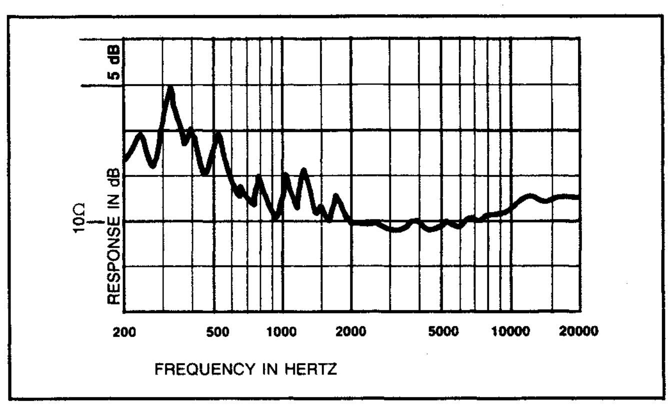

FIGURE 9 — Impedance Altec 288-8K Driver on EV HP9040 Horn

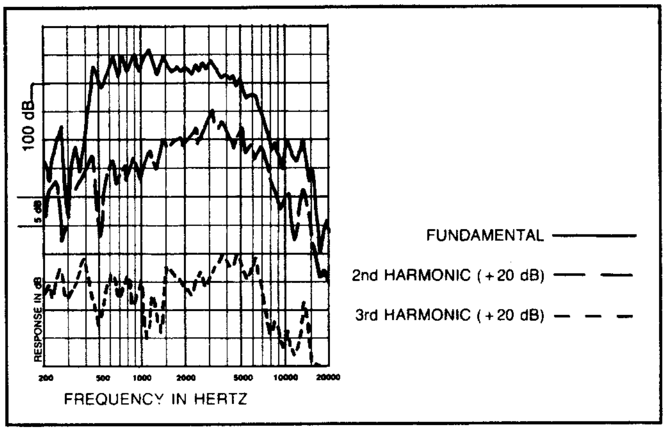

FIGURE 10 — Response & Distortion; EV DH1 Driver on EV HP9040 Horn (1 watt, 10 feet)

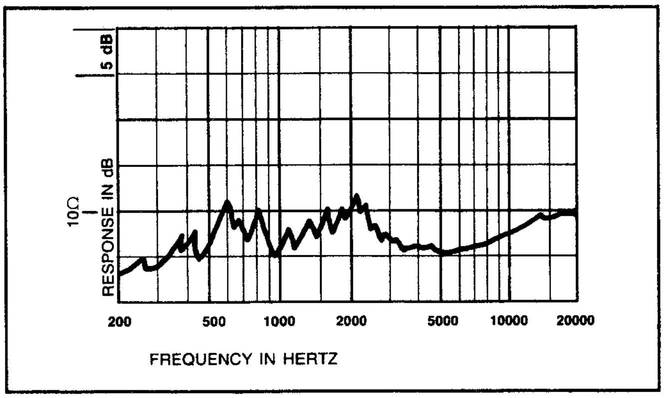

FIGURE 11 — Impedance EV DH1 Driver on EV HP9040 Horn

Try a horn whose throat is smaller than that of the driver. In this case, we used "the original CD horn," the EV HR9040 and, again, the DH1. This is another 2:1 area mismatch, but in the opposite way as in the Altec/HP horn example. Figures 12 and 13 show the performance of this combination, which is quite a bit worse than the other mismatch. Starting at 2,000 Hz, you can see a nasty, periodic "musical-horn"-type of behavior. Note the distortion. It also is very peaky and "high-Q" in nature, indicative of a relatively nasty sound. Also, the "step-down" mismatch gives an overall wide-band loss of level on the order of 3 dB. Interestingly, the overall harmonic distortion is lower here than the "matched" DH1/HP9040, this being due to overall output loss from the driver/horn combination. This combination will sound "nasty" but possibly tolerable.

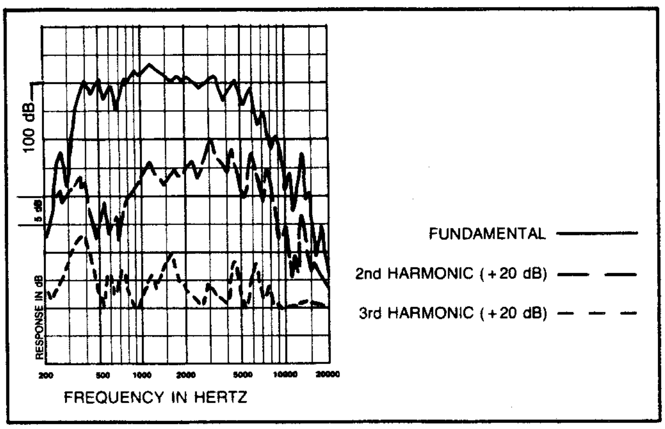

FIGURE 12 — Response & Distortion; EV DH1 Driver on EV HR9040 Horn (1 watt, 10 feet)

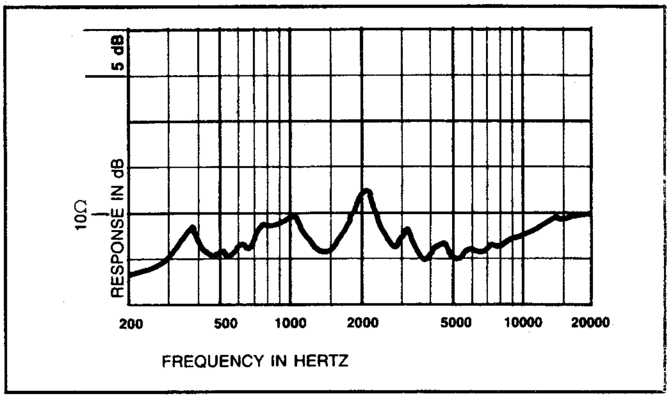

FIGURE 13 — Impedance EV DH1 Driver on EV HR9040 Horn

Lastly, a "mismatch" of horn flare and flare through the driver will mostly affect "ripple" at the low end of the horn. The high end will be largely unaffected by this, so long as the "interface" between driver and horn is matched, per the previous discussion.

THROAT ADAPTERS

Throat adapters are generally conical-expansion connectors which make a small-throat driver compatible with a large-throat horn. They are rarely used the other way. The "step-down" will cause an overall loss of level and will result in upstream and downstream reflections similar to those generated in Figure 12. For most uses, a conical throat adapter can be said to have a low "cutoff" frequency. If D_1 is the largest diameter, D_2 is the smallest and L is the length of adapter, then the cutoff frequency f_c is found as follows:

$$f_c = \frac{.255c}{L} \ln\left(\frac{D_1}{D_2}\right) \text{ Hz}$$

where "C" is the speed of sound in air, in the same linear units as D_1, D_2, L, and Sec^{-1}.

This "cutoff" will give about a one dB loss at that low frequency and more and more loss below that. Generally, throat adapters seem to receive more bad press than is warranted. If properly designed, they are of good use. Use the equation to check this out re: low end. If they are made accurately and are mounted and aligned properly, they will not affect response, even at high frequencies.

CONCLUSIONS

What have we learned here? First of all, we can see that severe driver mismatches, such as the ones shown, will still produce sound over a wide frequency range. In the case of a "step-up" in driver-to-horn area, we can get a large loss of high frequencies, as a single "dip" or "suckout." In the case of a "step-down" mismatch, harmonically-related peaks and dips result and an overall loss of efficiency or level will result. In both cases, audibly deteriorated performance will result.

So what do you do when you're out in the field and there is nothing but Altec 1.4-inch drivers and EV 2-inch horns (or vice-versa) and the promoter is threatening to sue? Obviously, the show must go on. Drill out new holes and bolt them together; they'll work! They will produce wideband sound, possibly correctable using equalization, and you can get by in a squeeze. It will be usable. If you have your choice and must match incompatible drivers and horns, purchase an (ADH) adapter and don't take a chance. The following are available EV horn-driver adapters which can be of great help:

ADH-1: 1 inch Bolt-on to 1.3 inch Bolt-on
ADH-2: 1 inch Screw-on to 1 inch Bolt-on
ADH-3: 1 inch Screw-on to 1.3 inch Bolt-on

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THE PA BIBLE

ADDITION NUMBER FIFTEEN
BARRIER MIKING

INTRODUCTION

The use of microphones mounted on barriers, baffles, acoustic boundaries, floors, walls, ceilings, and other surfaces — even the human body — has grown in practice in live performance and recording. With proper application, barrier miking can help eliminate acoustic interference and enhance the reproduction of voices and instruments by augmenting microphone response.

When an omnidirectional microphone is placed near a sonically reflective surface, the sound received by the microphone no longer has any dual path effects and, as such, resembles a simple pressure wave. The microphone is in a "pressure zone." The term Pressure Zone Microphone (PZM™) has been used by Crown International to describe their microphone made for this purpose.

The effect of this placement on microphone performance is the elimination of dual path interference, an increase in the apparent microphone sensitivity and a change in the microphone's high-frequency response that is attributable to the "baffle effect."

One should not depend on this technique to improve the basic performance and quality of the microphone. Therefore, your first choice should be a high-performance microphone from companies like Electro-Voice, Shure Brothers, Beyer, etc.

In this edition of the PA Bible we will review how this practice was first applied, some of the more recent developments in the field, and some of the most asked questions regarding the application of this technique.

EARLY DISCOVERIES

About 1970, Electro-Voice engineers discovered a way of solving a miking problem commonly experienced in sound reinforcement and recording of stage performances.

The typical miking technique was to mount mics on stands along the front edge of the stage. However, this practice suffers from a phenomenon known as phase cancellation or "comb filtering." This is a form of acoustic distortion caused as a result of the sound waves from the same source arriving at the mic at different times. Figure 1 illustrates how this can happen. Notice that the sound waves from the singer travel to the microphone in two different routes. (A simple

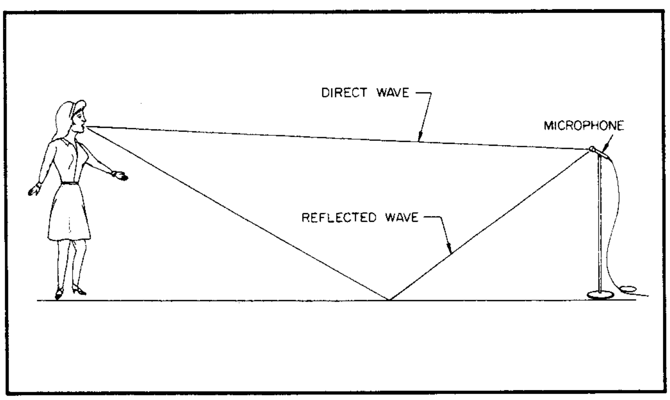


FIGURE 1
Typical Distance Miking

case, indeed!) The waves that travel directly to the mic are called "direct sound" waves. The ones that arrive after being reflected off the floor are called "reflected waves." In the sound system or on the tape, both sounds will be heard at different levels and at different times. Figure 2 illustrates the results of this graphically. Note the deep dips where the sound waves perfectly cancel each other; thus the term, "comb filtering." This was the problem that EV engineers set out to solve.

SOLVING THE PROBLEM

An understanding of the principles of barrier miking can be gained from studying some of the acoustic fundamentals that EV engineers employed.

In the case of microphone phase cancellation, the frequencies where cancellation occurs and the amount of cancellation is dependent upon the relative positions of the source, the microphone, and the reflective surface.

Look at Figure 1 again and notice that if the microphone is lowered closer to the floor (the reflective surface) the direct path and the reflected path become closer to the same length. As the two paths approach the same length, the distortion caused by the comb effect is audibly reduced.

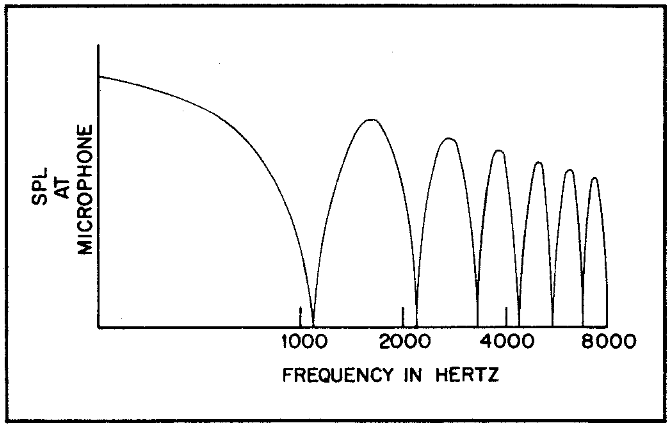


FIGURE 2
Comb Filter Effect

It was finally discovered that the closer the reflective surface was to the mic, the higher the frequency of the first dip (see Figure 3). Note that when the mic and the reflective surface are very close together, the dip will occur well above the audio range. The decision was then made to mount the mics on the stage floor, and the EV Mike Mouse was developed for convenient floor positioning (see Figure 5). This low-profile, Acoustifoam™ block not only uses reflected sound to increase gain before feedback, but also reduces interference resulting from floor reflections. Many types of microphones may be used with the EV Mike Mouse. A tripod microphone holder for floor positioning is available from Shure.

FURTHER DEVELOPMENTS

In recent years, this principle has been employed with very small microphones such as the EV CO94, the Crown PZM, and the Shure SM18. The fact that these types of mics are small not only allows their use inside instruments such as pianos, but they can be placed even closer to the reflective surface, thus assuring an in-phase arrival of the direct and reflected sounds at high frequencies.

COMMON QUESTIONS

Electro-Voice receives numerous questions regarding the use of microphones mounted on barriers, baffles, acoustic boundaries, floors, walls, and ceilings. Following are some of the most common questions and our answers.

WHAT IS BARRIER MIKING?

Barrier miking, also referred to as proximity mounting, is the technique of mounting a microphone on or very near an acoustically reflective surface. Laying an EV CO94 or a PL4 microphone in a model 370 Barrier Plate in the center of a conference table is an example of barrier miking.

A word of caution, barrier miking is not a "cure-all" and there is no "magic" in barrier miking. Each miking situation must be analyzed individually.

WHEN SHOULD BARRIER MIKING BE CONSIDERED?

Barrier miking should be considered: (1) when the situation requires relatively distant miking, and/or (2) when low visibility of the microphone is required, and (3) because mics used in barrier miking tend to be small, they should be considered for use inside instruments such as pianos.

The principal advantage of barrier miking is the elimination of a portion of the destructive interference that results from the presence of a reflected wave that is equal or nearing equal in strength to the direct wave but that has a slightly longer path length. This is, of course, the situation that results when distant miking is used. (See Figure 1)

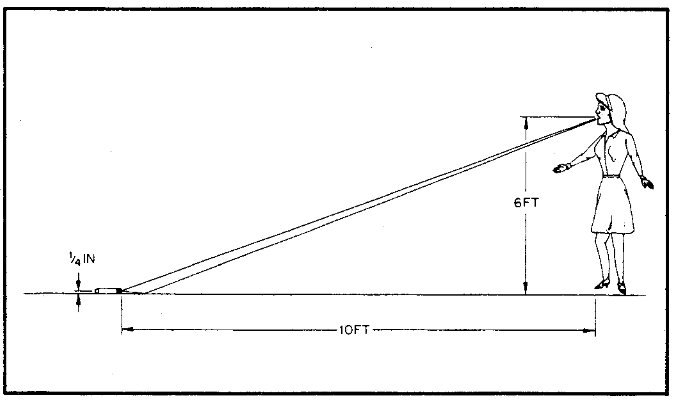


FIGURE 3
Time Delay

The visibility and size advantages are obvious. Barrier miking allows the microphone silhouette to blend into the podium, floor, wall, or instrument. In addition, the size advantage allows the body of an instrument to act as the barrier.

WHAT ARE SOME TYPICAL INSTRUMENT MIKING APPLICATIONS?

Barrier miking offers the user a means to express artistic freedom by using the influence a barrier and position can have on the resulting sonic character of the microphone's output. In instrument applications, a barrier microphone can be placed within the instrument and, through positioning, the sonic character adjusted to the user's preference.

Piano Miking

The piano is one of the most difficult instruments to mic successfully. Its large size and large number of discrete desirable sonic sources make it very hard to achieve the proper mix of these sounds. Many users have found they can obtain a very clean and well-balanced sound from a carefully positioned microphone mounted on the piano's sound board.

In situations requiring a well-balanced piano and good gain-before-feedback, some users have been able to locate a barrier microphone within a piano with a closed sound board. The resulting attenuation of room noise combined with the increased piano loudness can produce very good gain-before-feedback relative to one or more conventional microphones placed some distance from the piano. Since the "best" locations vary and depend to a great extent on the desired character, trial placement should be done carefully. Moving the unit a few inches on the piano's sound board can have a profound effect on the resulting sonic character and balance.

Drum Miking

Placing a small microphone against the inside surface of the drum's side wall can provide a very clean sound and, being within the drum, the resulting intensity, relative to the room, gives very good gain-before-feedack. As was mentioned in the piano section, exact placement will vary due to differences in drum design and user sonic preferences.

Miking Stringed Instruments

A barrier microphone can be mounted either on or inside a stringed instrument. By positioning the microphone carefully, the desired balance can be obtained.

Normally, the user must employ electronic equalizers to obtain frequency response other than that provided by the microphone's native response. By mounting a barrier microphone on relatively small surfaces, the user can create a sonic diffraction surface capable of changing the microphone's manifest frequency response.

Through experimentation the user may be able to create the desired sonic effect. Since small baffles also impart directional effects that are frequency dependent, this technique can produce sonic characteristics that are virtually impossible to obtain by any other means, including the use of expensive equalizers.

I'VE HEARD THAT A MICROPHONE MOUNTED ON A BARRIER IS MORE SENSITIVE THAN THE SAME MICROPHONE MOUNTED ON A STAND. HOW IS THAT POSSIBLE?

When the sound wave reflected from the barrier and the direct sound wave arrive at the microphone in the same relation, the two waves are added together, resulting in a microphone output that can be twice that of the direct wave, a gain of 6 dB.

WHAT EFFECT DOES CARPETING HAVE ON THE PERFORMANCE OF A BARRIER-MOUNTED MICROPHONE?

Since the strength of the reflected wave determines the additional sensitivity, anything that reduces the reflected wave energy will reduce the sensitivity of the microphone. Carpeting or acoustic tiles tend to reduce the intensity of the higher frequencies; therefore, the output from a barrier microphone on a carpeted floor will have a marked reduction of high-frequency sensitivity.

I'VE READ ABOUT USING PLEXIGLASS AND PLYWOOD PANELS TO IMPROVE MIC PERFORMANCE

The basic idea is to use deliberately placed reflecting surfaces to block unwanted sounds and/or reinforce, via the resulting reflections, the desired sounds. If the baffles used are big enough to reflect sufficiently low frequencies (to reflect sound, a surface must be big in relation to the wavelength of the sound) and are carefully arranged, then a very effective miking situation can result. This technique has often been used in miking singing groups such as choirs. When several panels are used to form a corner or pyramid shaped assembly, a crude "horn" is created and the resulting directional and frequency characteristics are very complex. The gain can greatly exceed the coupling associated with a simple single-surface reflector, and the resulting response variations are also exaggerated.

WHAT TYPE OF MICROPHONES CAN BE USED IN BARRIER MIKING?

The majority of the microphones in use as barrier microphones are small, omnidirectional electret condenser microphones such as EV's PL4. These units can be placed close enough to the barrier to assure an in-phase arrival of the reflected sound at high frequencies. The omnidirectional pattern can also be of use when complex reflectors are used, since omnidirectional microphones possess no directionality of their own and are predictable.

Recently, a number of manufacturers have developed barrier-mounted microphones based on directional transducers. These offer the additional advantage of directionality which can significantly improve gain-before-feedback.

Electro-Voice manufactures two accessories that are applicable to barrier miking: the model 370 Barrier Plate (Figure 4), suitable for mounting a lavalier microphone to a table top, wall, ceiling or instrument and the model 411 "Mike Mouse" (Figure 5), especially suitable for stage floor use.

We hope this addition to the P.A. Bible has contributed to your understanding of barrier miking. If you have further questions about appropriate applications or proper techniques for barrier miking, Joe Katowich, EV's Technical Communicator, will be happy to answer them.

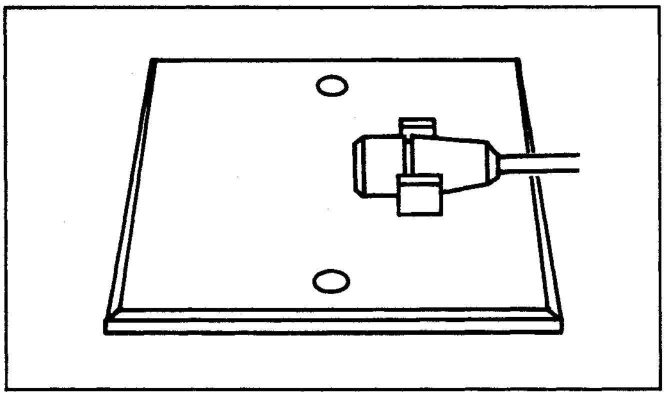


FIGURE 4
Small Omni Attached to a Barrier Adapter Plate

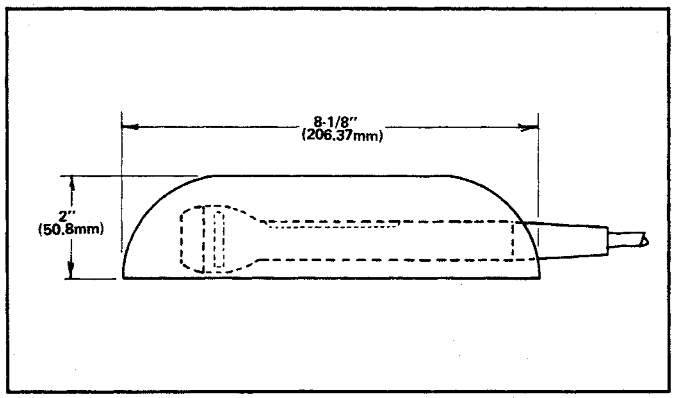


FIGURE 5
Cardioid Microphone in a Mike Mouse

EV THE PA BIBLE

ADDITION NUMBER FOURTEEN
LOUDSPEAKER SYSTEM TYPES

INTRODUCTION

In the third P.A. Bible addition the subject of microphone types was discussed. Like microphones, loudspeaker systems come in a variety of configurations. Previous P.A. Bible additions have briefly touched on loudspeaker system types. Addition 12, for instance, discusses vented type boxes for Electro-Voice FORCE loudspeakers and Addition 13 has a small section on cabinets. In this addition we wish to discuss the subject of loudspeaker system types in more detail in order to give some insight into this rather fundamental topic. We intend to concentrate mainly on the portion of a system that reproduces low-to-mid frequencies, as the geometry of this part usually determines the name given to the system type. This is probably natural, as in most cases the low-to-mid frequency section accounts for most of the systems bulk and weight.

This addition will also discuss the system design theories of Neville Thiele and Dr. Richard Small. You may possibly have encountered "Thiele and Small" in loudspeaker manufacturers' specification sheets, magazine articles on acoustics or the P.A. Bible itself. We will use some of this theory to help explain and sort out the performance characteristics of various loudspeaker system types. It is hoped that this material will help remove some of the mystery from loudspeaker system types and be of help in understanding the way they work, how they came about, and where they are most effectively used.

WHAT IS A SYSTEM?

A system of any sort is a collection or combination of things which form a coordinated and unitary whole. In the case of a loudspeaker system, the combination is the actual loudspeaker plus the enclosure in which it is put. The combination of these two elements should act together in the best possible way to achieve a desired performance goal in order to be classified as a good system. Putting an arbitrary loudspeaker into an arbitrary enclosure is not the way to achieve an optimized system. The knowledge of how to properly coordinate loudspeaker and enclosure in the best possible way is what the science of loudspeaker design is all about.

It is possible to achieve a good system by the familiar method of "cut and try". Such a method involves changing the variable elements of a loudspeaker and enclosure until a desired level of performance is obtained. However, if the system has any degree of complexity to it at all, it will be difficult to know when the desired result has been achieved in the best possible manner, thereby obtaining the desired performance in the least volume of space. In a loudspeaker system, basic performance goals usually translate into getting a specific low frequency limit (i.e., how low does it need to go) and efficiency level (the ratio of acoustic output to electrical input) from the least volume of occupied space. Effective space (and consequent weight minimization) is an important concept to keep in mind.

COMMON TYPES OF SYSTEMS

Over the years a variety of systems have been offered under various names. After stripping away the minor variations and multiple names, there are about six basic types that account for the great majority of systems on the market. These are:

1. dipoles (open-back baffles),
2. sealed boxes (acoustic suspension),
3. vented boxes (bass reflex, phase inverter, passive radiator),
4. horns,
5. combination boxes with horn midrange and vented bass sections, and
6. combination boxes with horn-bass and sealed-box midrange sections.

The names in parenthesis are some additional ones given to these types.

These system types are illustrated in Figure 1. Historically, horns are probably the oldest type of system. Early forms were megaphones and phonograph "morning glory" horns used to amplify stylus sounds in the days before electronic amplifiers. After the introduction of cone loudspeakers in the 1920's, simple open-back baffles, sealed and vented boxes and combination boxes evolved as various ways of allowing cone loudspeakers to reproduce sound effectively.

A few comments are in order to help distinguish the main characteristics possessed by these six types of systems.

Dipoles

This is the simplest type of system and, in its most rudimentary form, consists of a naked loudspeaker with the loudspeaker cone itself forming its own baffle! Due to its simplicity, it is a quite common type of system. It is found whenever a loudspeaker is mounted on a surface which has a substantial size opening on its backside. Examples can be found in open-back guitar amplifiers and dashboard-mounted car speakers. This opening allows the sound coming from the front and rear of the loudspeaker to interact and partially cancel each other at lower frequencies. This cancellation process begins to occur when the average baffle size is about half the wavelength of the sound being reproduced. A loudspeaker mounted on a 3-foot square baffle would allow cancellation to begin occurring at wavelengths longer than about 6 feet — a wavelength equivalent to a 190 Hz frequency. In general, it is hard to generate powerful low frequencies with dipoles because of the cancellation effect, although some forms of this system with rear cavities (see Figure 2) can produce mid-bass peaks. The rolled-off bass characteristic of this system type tends to make it "project" well. It is widely used in guitar amps.

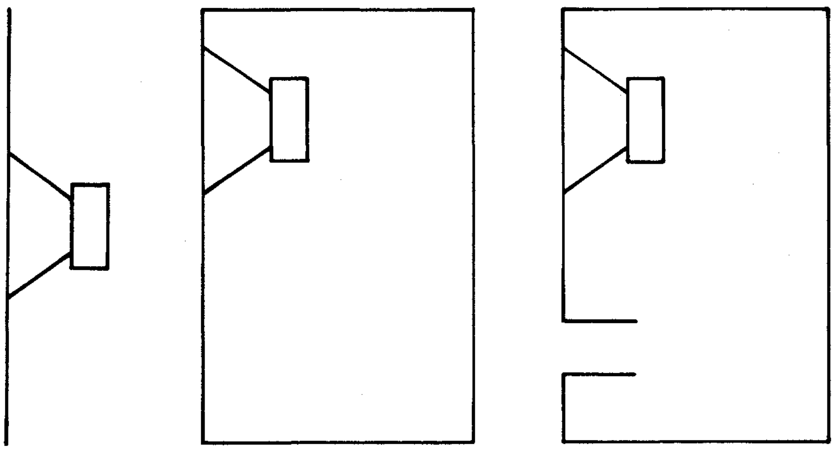
Sealed Boxes

This system type probably developed when someone tried to suppress the low-frequency cancellation inherent in a dipole system by closing off the system's rear. Sealing the box solved some problems and created others. It took until the 1950's for this type system to be sufficiently researched to allow the design of quality "bookshelf" systems for home use. These home systems offered smooth and extended low-frequency response in small-size boxes with rather low-efficiency levels of less than 1%. This type of system has tended to be used more for music reproduction in homes and background/foreground applications, than for musical instrument or fixed installation commercial applications.

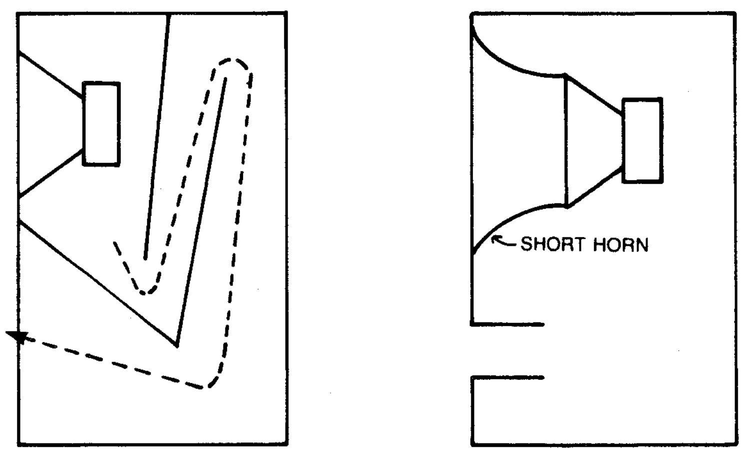
Vented Boxes

This system type appears to have been thought up in the mid 1930's. The basic idea was to put to work some of the acoustic energy that would otherwise be bottled up in a sealed box. To do this, the acoustic output from the rear of the cone needs to be phase inverted to compliment the output from the front of the cone over some appropriate range of frequencies. As it turned out, the proper coordination of all the variables of the box and the loudspeaker inside was a rather complex juggling act. Many of the early forms of this system type were uncoordinated to a degree that made listening to them unpleasant. One of the more common approaches was to attempt putting a vent on a system that worked reasonably well as a sealed system and getting either no noticeable improvement or (perhaps worse yet) a thumpy or boomy form of false bass. Figure 3 illustrates the nature of what has to happen in order to achieve usable response from this type of system. The variables of the box and the loudspeaker have to be coordinated so as to achieve a smooth net response from the combination of vent output and loudspeaker output. In essence, the vent acts as a second loudspeaker that takes over and reproduces the lowest frequencies — sort of a little sub woofer that automatically cuts in when needed. Thiele and Small were to describe this system type in great detail in the 1970's and show how to coordinate the loudspeaker and box to obtain smooth, usable response.

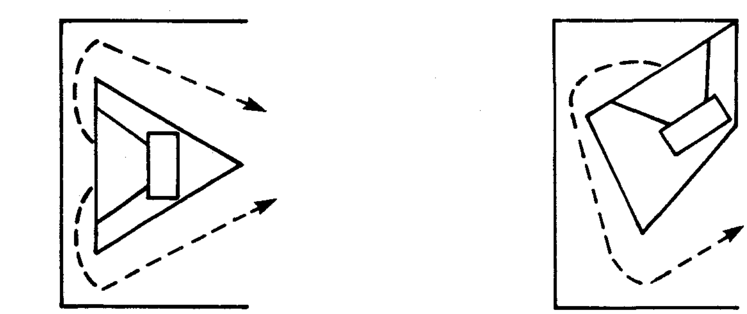
The vented system has become a widely used system type in musical



DIPOLE (Loudspeaker Mounted on a Finite Flat Baffle) — **SEALED BOX** — **VENTED BOX**



COMBINATION BOX (Horn Bass and Sealed Box Midrange) — **COMBINATION BOX (Horn Midrange and Vented Bass)**



HORN (folded - 2 methods shown)

FIGURE 1 — BASIC SYSTEM TYPES

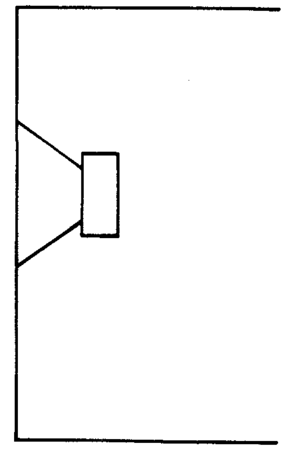


FIGURE 2 — Dipole with a Rear Cavity formed by Extended Sides.

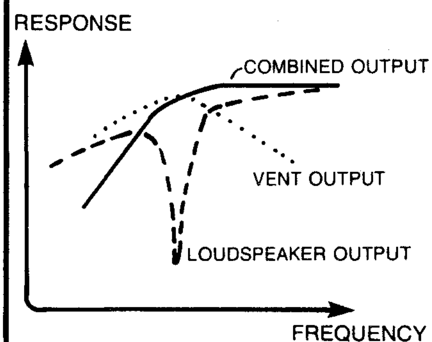


FIGURE 3 — How Loudspeaker and Vent Output Combine in a Vented System

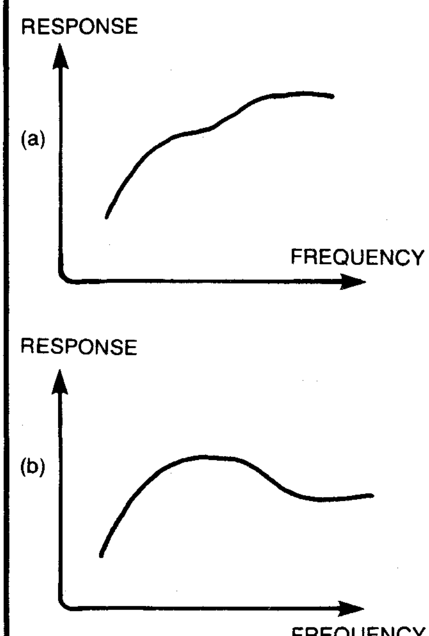


FIGURE 4 — Nature of Response of: (a) Combination Box with Horn Midrange (b) Combination Box with Horn Bass

instrument applications. It is capable of producing smooth, flat, "hi-fi"-like sound quality with good solid bass.

It is possible to install a properly weighted cone as a substitute for the vent in this system type. This vent substitute, or "passive radiator", duplicates the diameter of the vent it is replacing and has a weight equivalent to that of the air contained inside this vent. This type of system is, in reality, a minor variation of a vented system.

Horns

As noted before, this type of system is quite ancient. It was once used to amplify the feeble vibrations of the stylus on early phonographs, making these sounds audible to the listener. Horns usually possess the property of having very high conversion efficiencies — often in the 15 to 30% range. (When not horn-loaded, the highest efficiency obtainable from a single loudspeaker is in the 5 to 10% range.) Because of this, they were of special interest when the "talkie" motion picture appeared. Spacious motion picture theaters required relatively large amounts of acoustic output. Since early amplifiers had limited output, it was only natural that the inherently efficient horn system received considerable attention in the 1930's and 40's.

Horns that must reproduce low frequencies are rather large and lengthy devices, and it is common practice to fold them up in order to make a more compact package. Many possible folding geometries exist. Because of their long, expanding air columns, horns are relatively complex structures. The rate at which the horn's air column expands in going from the small (throat) end to the large (mouth) end is determined by the lowest frequency that the horn is expected to reproduce effectively. This frequency is referred to as the horn's "cutoff frequency". The horn ceases to operate well below this frequency and response drops off rapidly.

The main points to keep in mind when thinking of horn type systems are:

1. if properly designed, they can offer very high efficiency from a single cone loudspeaker used to drive them,
2. they tend to be large, and
3. they are of relatively complex construction.

Horns can produce tight, punchy, high-intensity sound.

Combination Boxes (Horn Midrange and Vented Bass)

In this form of system the box is subdivided into two sections which are not usually of equal size. One of the two sections is in the form of a short, horn-like structure with the loudspeaker at the rear or small end of the horn. As in the case of the vented box system, the very lowest frequencies come from the vents. Higher frequencies pass through the short horn but are relatively unaffected by it until the cutoff frequency of the horn is exceeded — typically somewhere above 100 Hz. Above this frequency, a combination of increased efficiency and greater directivity causes a response rise of three or four dB to occur. Figure 4a indicates the nature of this type of response. The change in both directivity and efficiency caused by switching from one type of system to another as frequency increases results in this form of shelved or stepped response characteristic. This rise in response can give a projected quality to vocals that many performers and engineers find effective.

Combination Boxes (Horn Bass and Sealed Box Midrange)

In this form of combination box, the tables are turned relative to the previous type. Here, bass is reproduced by the horn part. Almost all of the volume of the box is occupied by the horn in most cases. A small chamber houses the loudspeaker with the throat, or small end of the horn, beginning from an opening placed in this chamber. With this arrangement, frequencies between 100 and 300 Hz are radiated chiefly from the front of the cone, much in the manner of a sealed box type of system. The air chamber containing the loudspeaker acts as a shutoff valve for the horn above these frequencies. Lower frequencies are radiated from both the horn and also from the front of the loudspeaker. However, the higher efficiency of the horn tends to dominate the net low-frequency response. As might be expected, the response shape of this type of system favors bass over midrange in the manner illustrated in Figure 4b.

As in the case of the pure horn system discussed earlier, there are many possible schemes for folding up the large and lengthy horn. Several of these geometries result in the "scoop" type of appearance.

THE THIELE/SMALL CONNECTION

A selective discussion of the design concepts developed by Thiele and Small will help put the basic system types mentioned in better perspective. These concepts are basically a carefully reasoned way of coordinating loudspeaker, enclosure and, in some cases, auxiliary electronics so as to create pre-determined and desired low and midrange response curve shapes. These response curve shapes are given the names of the mathematical equations that describe them, hence Butterworth and Chebyshev responses of various kinds often appear in the description of systems which are based on Thiele and Small's work.

Even though Thiele's original work was titled "Loudspeakers in Vented Boxes"[1], sealed boxes are discussed as well, and Small's works are even more detailed in this respect. In the process of a thorough description of system possibilities, Thiele creates a table of 28 methods of achieving various selected responses from vented systems. This method of coordinating loudspeakers, boxes and (sometimes) auxiliary electronics are known as "alignments". In a sense, when all the significant speaker and box variables are aligned properly, the system produces the desired response. Small's work expands on Thiele's and, among other things, describes something that is best thought of as a system interrelationship equation which illustrates fundamentally important relationships between box volume, low frequency capabilities, conversion efficiency and system type.

This interrelationship equation is worth further discussion as it illustrates several very important matters about system behavior that no amount of wishful thinking will alter. It will help a user understand more thoroughly the implications of large and small systems, the problems of achieving higher levels of efficiency and more extended bass response and, finally, the ramifications of system type. Although the equation was meant to be applied to enclosed direct radiator systems, it can yield interesting information about horn systems as well. (Direct radiator is a term often given to systems where the loudspeaker directly addresses the air in the listening space.)

In its simplest form the equation is:

$$E = C\ V\ F^3\ K$$

where E is efficiency (the ratio of acoustic power out to electrical power put into the system),

C is a constant number dependent on the system of measurement used and the local environment the system is operated in,

V is the internal volume of the loudspeaker enclosure,

F is the frequency below which bass output begins to rapidly drop away, and

K is another constant number, but this time it is associated with the nature of the system response shape and, therefore, the system type. It is a very important number sometimes referred to as the "figure of merit".

Let us fix the system type for a moment so that K remains at a fixed value (think of it as having the value 1 if you wish) and look at a few revealing situations:

a) Extending the bass (making F smaller) makes severe demands on system design since F is multiplied by itself three times (or cubed). As an example, extending bass by one octave (making F one-half as large) would imply that efficiency would be reduced to one-eighth its former value ($\frac{1}{2} \times \frac{1}{2} \times \frac{1}{2} = \frac{1}{8}$) if loudspeaker volume (V) were to remain unchanged. Another implication would be that if it would be desirable to keep efficiency (E) the same but extend bass one octave, then loudspeaker volume would need to be increased by eight.

b) Reducing the size of a larger system can certainly be done, but either efficiency would have to be reduced or low frequency capabilities would have to suffer or some combination of both things would have to happen. The equation must, in all cases, remain satisfied.

It should be understood that making system changes of the type being discussed would, in general, imply that the basic makeup of the loudspeaker (i.e., magnet size, moving system weight, etc.) used in the system would need to change in a manner prescribed by Thiele.

COMPARING SYSTEM TYPES

Now let us examine the situation involving a change in system type which implies that the K, or "figure or merit", in the system interrelationship equation is altered.

In one system of measurement, the value of the figure of merit (K) is between 1.5 and 2 for almost all sealed box systems that would have response curves that would be of interest. *Dipole systems* are somewhat more difficult to characterize due to their lack of a well-defined enclosure volume, but some recent work [2] suggests that K values for this system type appear to be less than 1. *Vented boxes* would usually have a figure of merit value between 3 and 4 although some unique forms of vented systems described by Thiele that use auxiliary electronics can be higher yet. These high values for the figure of merit are the reason for an upsurge of interest in vented box systems. As an example, consider a change in the figure of merit (K) from 2 to 4 obtained by designing a vented system instead of a sealed one. In this case, the efficiency (E) could be doubled, thus making it possible to obtain 3 dB more of sound pressure level from a system without having to increase enclosure size, sacrifice bass or get a larger amplifier. If a smaller box were to be desired, this same figure of merit doubling could be used to halve box size without other sacrifices. Electro-Voice has been interested in the potential of vented box systems since the early 1970's and currently offers systems such as the S-1202, S-1503, S-1803, FM-1202, FM-1502, and TL bass boxes that embody the values inherent in this system type.

Although the interrelationship equation is meant to be used with direct radiator systems, it is possible to derive a figure of merit for *horn systems* by doing measurements and deducing K. Some work in this area [3] suggests figure of merit values for horns are at best similar to those for sealed boxes. Because of this, most of the comments made in comparing vented to sealed boxes would apply to horns as well. What about the matter of high efficiency? It is known that horns can have efficiency levels in excess of 10% but that single direct radiators seldom exceed 5 to 7%. The interrelationship equation suggests that putting several of the same systems close together is, in effect, increasing the box volume (V) and thereby increasing efficiency. This method can be used to push the net efficiency of direct radiator systems to values above 10%. The point being made here is that very high efficiencies are not the exclusive domain of horn-type systems. They can be achieved with direct radiator systems by using several closely spaced loudspeakers.

The *combination box with horn midrange and vented bass* is sometimes constructed so that the vented box portion of the system is most of the box. In this case, the comments made about vented boxes are applicable to this type of system. However, when the short horn takes over (usually somewhere above 100 Hz), the response will show a rise. This rise is due to a combination of directivity increase and some efficiency increase. The sonic balance caused by this type of response is sometimes favored. The Electro-Voice SH-1502 is a system of this type. If, however, this system type is constructed so that the horn portion of the box is a dominant part of the total system volume, then the system will behave more like a pure horn with a resultant reduction in the figure of merit.

The *combination box having horn bass and sealed midrange* is basically a horn system with a small section of the box apportioned to the chamber containing the loudspeaker. Because of this, the comments made about horns as compared to direct radiators apply to this system type, with the only appreciable difference being that the loudspeaker, which faces outward, provides some subdued midrange output.

SUMMARY AND APPLICATION COMMENTS

Six basic types of loudspeaker systems have been described in this P.A. Bible addition that range from simple, open-back baffles (dipoles) to quite complex horn-type systems. Within each system type there can be a variety of ways to balance or "voice" the sound quality of the system. However, there are some general sonic characteristics or "signatures" that can be assigned to each type that make them more suited to certain applications. The following table should be of help in summarizing the situation.

Electro-Voice has highly qualified technical personnel who can assist with any field problems which may arise, and are able to answer questions concerning any aspect of the application and performance of our products. Our technical correspondent is Mr. Joe Katowich, and our telephone number is (616) 695-6831.

TABLE OF CHARACTERISTICS OF LOUDSPEAKER SYSTEM TYPES

System Type	Figure of Merit (K)	Complexity of Construction	Typical Response Shape	Characteristic Sound Quality	Typical Use
Dipole (Open-Back Baffle)	Less than 1	Very Simple	Strongly rolled off bass due to cancellation	"Projects" well	Guitar Amplifier
Sealed Box	1 5 - 2	Simple	Can be flat or have bass somewhat rolled off	Tight sound punchy midrange	Guitar Amplifiers, some vocal
Vented Box	3 - 4	Simple	Usually Flat	Smooth "Hi Fi" character with solid bass	Floor monitors, vocal, bass
Horn	1 - 2	Very Complex	Can be flat or have bass somewhat rolled off	Tight, punchy midbass	Low Frequency building block for large systems
Combination Box (Horn Midrange)	3 - 4 (If horn is a small part of the box)	Complex	Response rise when horn takes over	Good projection	General use, Especially Vocal
Combination Box (Horn Bass)	1 - 2	Very Complex	Elevated Bass	Strong Bass Emphasis	Bass Instruments (Drums, Organ)

REFERENCES

[1] A.N. Thiele, *Loudspeaker in Vented Boxes: Part I,* Journal of the Audio Engineering Society, Vol. 19, pp. 382-392 (1971 May); Loudspeakers in Vented Boxes: Part II, ibid., pp. 471-483 (1971 June).

[2] R.J. Newman, *Dipole Radiator Systems,* Journal of the Audio Engineering Society, Vol. 28, pp. 35-39 (1980 Jan./Feb.).

[3] D.B. Keele, *An Efficiency Constant Comparison Between Low-Frequency Horns and Direct-Radiators,* presented at the 54th Convention of the Audio Engineering Society (May 4-7, 1976).

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THE PA BIBLE

ADDITION NUMBER THIRTEEN
THE ELECTRIC GUITAR LOUDSPEAKER, A UNIQUE DESIGN

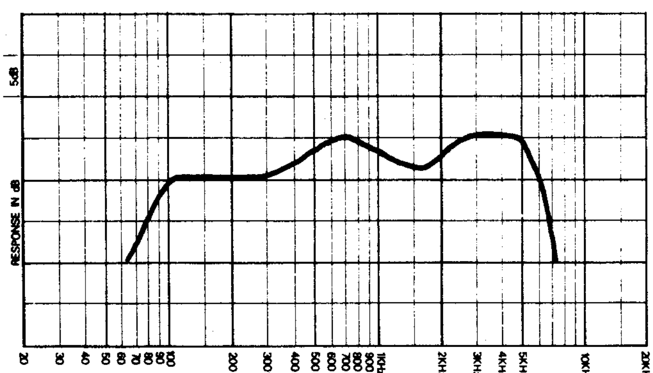
In this addition of the PA Bible, we will discuss the development of loudspeakers for electric guitars to illustrate the special character of these unique speakers.

Electric guitar amplifier designs of the early 60's depended upon hi-fi and general purpose loudspeakers that were then available. These loudspeakers were found to be deficient in two characteristics: they failed when driven with high power, and the tonal characteristics did not enhance the guitar sound.

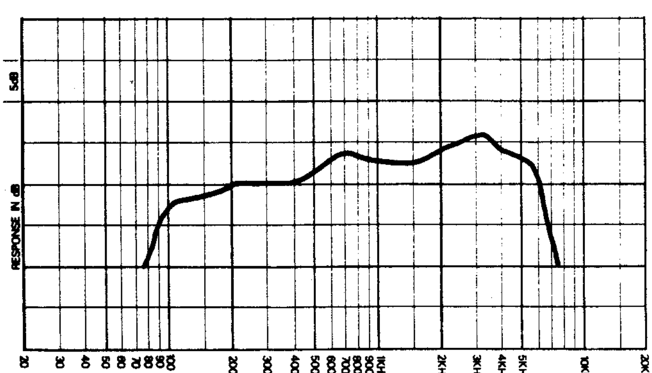
Engineering efforts to overcome these weaknesses, including extensive power testing and new materials searches, have resulted in unique designs, with special tonal characteristics and power handling capabilities unheard of prior to their development.

The Guitar Sound Approach

A significant part of the design specialization has been the recognition that electric guitar speakers are part of the instrument; that these speakers must be developed expressly for the electric guitarist; and that these speakers are designed with the guitar note in mind, not for playing records.



EVM-12L



EVM-12S

FIGURE 1 – Frequency Response

Individual preference is very much a part of this process, as it is with musical instruments.

The Electro-Voice EVM®-12L and EVM®-12S have found wide acceptance as quality electric guitar speakers and their characteristics will be referenced to illustrate the special character of the guitar speaker design.

The Special Tone Quality

The special tonal characteristics of electric guitar loudspeakers result from the selection of cone materials and shape, voice coil materials and size, cone suspension details, magnetic gap geometry and venting.

Because electric guitars are normally played at very high sound levels, their characteristic sound can only be fully described by listening to the instrument. Lower level frequency response measurements, however, are indicative of the overall character, and are useful in controlling the consistency of manufacture.

The tonal characteristics of the Electro-Voice EVM-12L and EVM-12S are thus indicated in the response curves shown in Figure 1, with the curves showing a 2000 Hz to 4000 Hz frequency rise which adds a brilliance or presence to the sound, with the EVM-12S having a brighter sound as indicated by the greater output in 1000 Hz to 3000 Hz.

To achieve this special response, the guitar loudspeaker designer must carefully select the size, shape and composition of the materials used in the cone, coil, dome and gap structure of the loudspeaker.

Cone

The selection of a single-piece felted paper curved-sided cone constructed of a proprietary pulp mixture, when driven by the correct coil size, results in the sound of the EVM-12L and EVM-12S, the sound preferred by most electric guitarists. By contrast, an inexpensive seamed, straight-sided cone tends towards the uneven tonal characteristic illustrated in Figure 2.

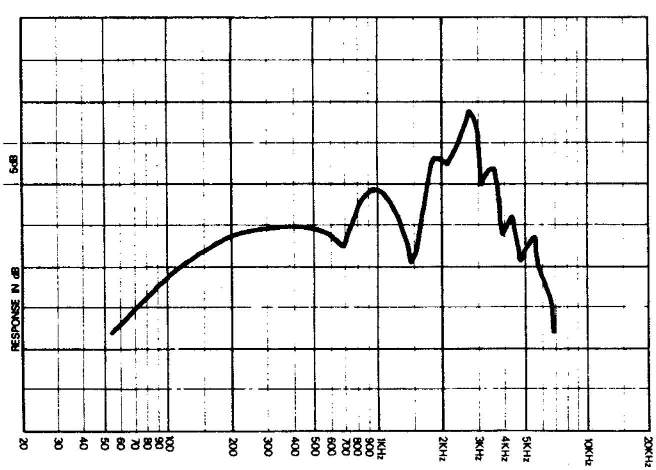


FIGURE 2
Inexpensive Small Coil 12 in. Guitar Speaker with Flat, Folded Paper Cone

Coil

The selection of a 2.5 inch diameter coil for the EVM-12L and EVM-12S is supported by the tendency of smaller diameter coils to dip in the region of 1500 Hz, as illustrated in Figure 3; and, conversely, by the lack of a high-frequency rise when larger coils are used (see Figure 4)

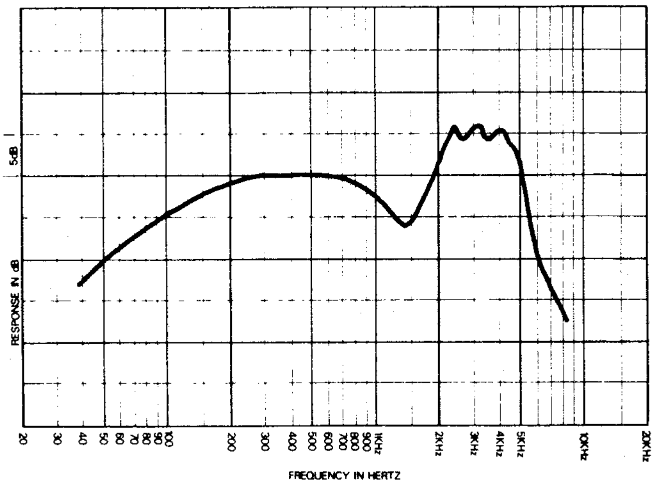


FIGURE 3 – Small Coil (1.75 in.) 12 in. Guitar Speaker

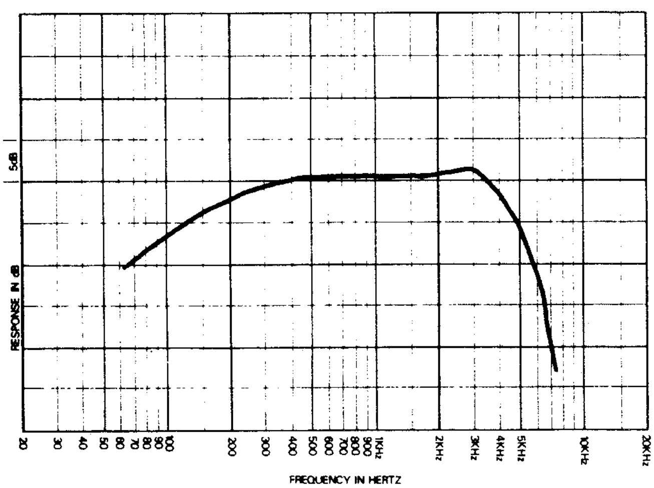


FIGURE 4 – Large Coil (4 in.) 12 in. Guitar Speaker

Dome

While the smooth, rising frequency response of the EVM-12L and EVM-12S are indicative of their tonal quality, other features add to the character of the sound. For example, a paper dome is used in the EVM's to avoid the harshness caused by the high-frequency breakup of aluminum domes.

Gap Structure

The use of an asymmetrical gap structure, and a coil height equal to the height of the pole piece, (see Figure 5) particularly illustrate the special character of the EVM electric guitar loudspeaker.

Symmetrical magnetic gap structures have been promoted as desirable in a guitar speaker. We have found this to be a fallacy. When driven at very high power, the coil, which is made to fill the gap height to obtain high efficiency, will be driven out of the gap. When so driven,

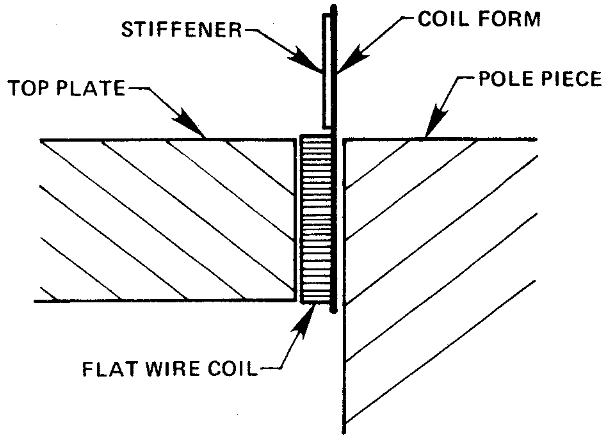


FIGURE 5 – Gap Structure

the coil motion in a symmetrical magnetic gap structure will produce only odd harmonics due to the non-linear magnetic field. A coil moving in an asymmetrical magnetic gap, conversely, will generate a mixture of odd and even harmonics resulting in a more complex, richer sound, with the additional benefit of lower coil temperature resulting from the longer time that the coil spends adjacent to the pole piece.

Thus, the special tonal characteristics of the EVM-12L and EVM-12S are the result of careful choices during the design process, choices specifically tied to the guitar sound.

Power Handling Capacity

Electric guitars are normally played at very high sound pressures, requiring a loudspeaker that is both efficient and capable of handling very high electrical power input. The EV EVM-12L and EVM-12S, for example, are designed to withstand 200 watts long term and 800 watts short term peak power.

To design loudspeakers that can withstand these power levels, two failure mechanisms must be eliminated. These are (1) material failure due to high temperatures, and (2) mechanical failure due to large displacements of the coil.

The heat rise of the voice coil at high power levels is substantial (see Figure 6) and requires special materials and construction to survive and control these temperatures. The wire insulation and coil support should be made from a material capable of withstanding temperatures in excess of 400 degrees fahrenheit. For example, the EVM-12s use a specially treated polyimide material for both the wire insulation and coil support. Special high temperature adhesives must be used on all adhesive joints. In the EVM-12s a special high temperature epoxy is used to secure the coil to the form, and this assembly then to the cone. The coil should be spaced very close to the pole piece and top plate to optimize the heat transfer from the coil assembly. The EVM-12's coil structure is spaced within thousands of an inch of the pole piece and top plate; and, further, the EVM structure uses a straight pole piece which tests show lowers the coil temperature 17 degrees F for 200 watts input. In addition, the aluminum frame used in the EVM-12s provides heat conduction away from the magnetic structure that is superior to stamped steel frames.

To prevent destruction during the high coil excursion, special materials are required to withstand the resultant stresses. The EVM-12s use special high strength rolls and spiders, and a special Kapton® stiffener to strengthen the coil form.

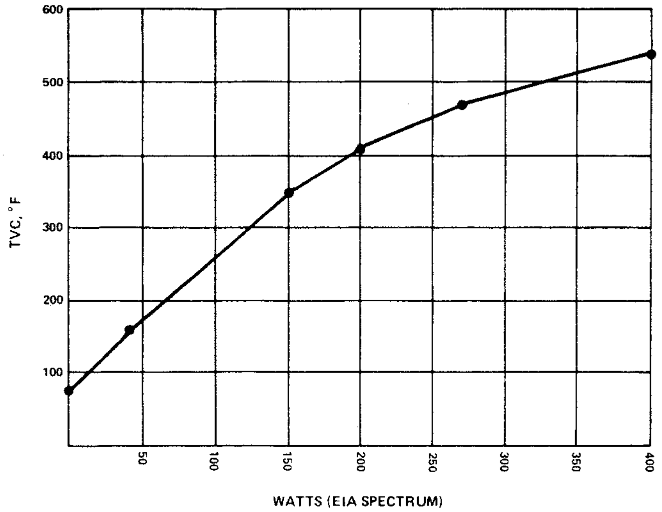


FIGURE 6 – Voice Coil Heat Rise

Power Test

The test used to determine the power handling capacity of an electric guitar loudspeaker is very important, and can mean the difference between long life and catastrophic failure. The test specified in EIA Standard 426A, which includes instantaneous peak powers that are four times the continuous (rated) power, more closely resembles the character of the guitar signal than does a sine wave; and, consequently, is a superior test for guitar speakers. EVM speakers are required to pass the eight hour test specified in EIA Standard 426A.

The reliability provided in these high power designs is enhanced by the use of an aluminum frame. These die cast frames provide a stable, stiff support for the cone and heavy magnetic structure; and, in addition, provide a major heat loss mechanism due to the excellent heat conductivity of aluminum. Aluminum is also light weight, non-magnetic and non-corrosive; and, is further enhanced by a reliable, chemical-resistant baked epoxy finish.

Venting

Unless special precautions are taken, air trapped between spider and top plate, and between the dome and pole piece, is forced through the air gap. Because of the physical restriction in the gap, high velocity turbulent air causes noise and an increase in the system stiffness. To eliminate this problem the EVM speakers use vent holes in the coil support, and, also, use porous spiders to vent the heated air entrapped beneath the spider.

Cabinets

The EVM-12L and EVM-12S will work well in open back, sealed or vented cabinets. We recommend the use of vented enclosures for the optimum size and bass response combination. The bass responses (see Figure 1) were measured with the speakers mounted in a 1.3 cu. ft. vented box, the TL806. Details on the TL806 design and construction may be obtained from Electro-Voice by requesting "TL806 Builder Plans," Form 1544–523.

Conclusion

In conclusion, it is apparent that a special, premium grade loudspeaker is required for the best electric guitar performance, including the details highlighted in Figure 7. This premium design, which includes special materials and special production processes, results in a product that sells at a premium price.

Recently, other alternatives have become available. For example, the EV FORCE® loudspeakers, which use cast aluminum frames, and the same cone and magnetic gap design as the EVM speakers, are priced only slightly higher than hi fi or general purpose speakers, but with performance characteristics only slightly lower than the EVM speakers.

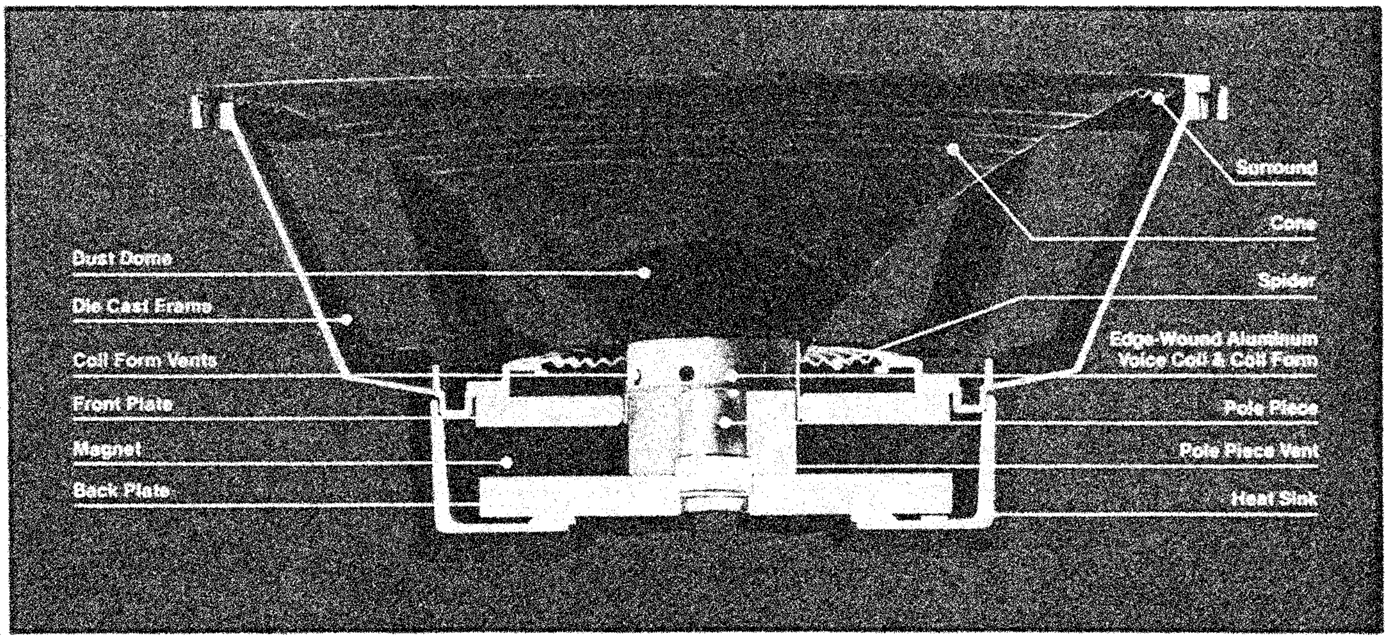


FIGURE 7 – EVM Cutaway

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THE PA BIBLE

ADDITION NUMBER TWELVE
"FORCE®" BOXES
FOR MUSIC SYSTEMS

INTRODUCTION

In response to many requests for information on enclosures for our FORCE® music loudspeakers, we are publishing this twelfth addition of the PA Bible. In doing so, we will furnish curves showing the expected performance, plans for enclosures for the FORCE 10, FORCE 12 and FORCE 15, along with suggestions for constructing the cabinets.

The phrase "Boxes for Music Systems" in the title of this PA Bible addition refers to enclosures used for the bass section of sound reinforcement systems for live performances. These boxes are not intended to be home hi-fi or stereo systems.

The sound reinforcement system must have a higher average power handling capability in order to maintain the sound pressure levels in a club or auditorium environment, and must be able to handle the higher peak levels (dynamic range) of the live performance.

The home stereo system, on the other hand, is designed to reproduce material for the intimacy of the average home environment, where the power requirements for sufficient sound pressures levels and dynamic range are modest when compared to a live performance in a club or auditorium.

There are many different types of enclosures that can be used with music systems such as the acoustic suspension, folded horn, infinite baffle and the vented direct radiator (bass reflex), to name a few. Each has its advantage and disadvantage, depending on the application.

For this addition, we will consider vented enclosures based on Thiele parameters.

It wasn't until A. N. Thiele[1] developed his "alignments" that it became practical to mathematically predict or tailor the performance of a vented system to a specific application. Since then, much has been written on the subject of speaker system design and the mathematics involved in the actual enclosure/speaker relationship.

Our subject here will only deal with the vented direct radiator (or bass reflex) enclosure for FORCE loudspeakers and the effects on the response curves resulting from different enclosure tuning and enclosure volume.

You, as the builder, have control of the two most significant factors in determining the performance of a vented system; the enclosure tuning and enclosure volume. By varying either of these parameters, the bass response characteristics can be tailored to realize a personally desirable sound.

ENCLOSURE DESIGNS

Figure 1 shows the computer generated curves depicting an Electro-Voice FORCE 10 speaker in a 1.4 ft^3 enclosure. Curve A represents the predicted bass response with the enclosure tuned to the free air resonance of the speaker (65 Hz) and would be considered a classic B^4 (Thiele's Alignment No. 5)[1] enclosure.

This tuning produces the flattest curve with the –3 dB level at 65 Hz and a –10 dB level at 52 Hz.

By simply retuning the enclosure to 1/2 octave lower (49 Hz) the response curve B starts rolling off about one octave higher than in curve A and the –3 dB level has shifted to 95 Hz with the –10 dB at 45 Hz. This could be considered an unequalized B^6 (Thiele's Alignment No. 15) enclosure and the bass would sound a little "thin".

Our third curve (C) in Figure 1 shows the effect of tuning the enclosure 1/2 octave above the free air resonance of 65 Hz or to 98 Hz.

This higher tuning has created a hump of +4.5 dB at 120 Hz as well as shifting the –3 dB level to 85 Hz and –10 dB level to 70 Hz. Aside from sounding like a "boom box", this enclosure would also accentuate any 120 Hz hum in the amplifying system.

The next set of curves (Figure 2) represents the effects on the system response by changing the enclosure volume while maintaining the enclosure tuning of 65 Hz as a reference.

Reducing the enclosure volume by one-half has the same effect on the response curve (Figure 2B) as lowering the enclosure tuning (shown in Figure 1B). The –3 dB level is also at 95 Hz but because of the faster rolloff, the –10 dB level has shifted to 60 Hz.

Doubling the enclosure volume to 2.8 ft^3 (Figure 2C), the hump is still apparent but has been reduced to about +1.5 dB. Also, the peak of frequency of the hump has shifted from 120 Hz to 70 Hz and a slight loss in level in the upper bass region has developed. The –3 dB level has shifted to 59 Hz and the –10 dB level to 50 Hz.

With the countless combinations of enclosure volume and tuning, it is obvious that almost any shape response curve can be achieved for a given driver (within the limits of the driver, of course).

Similar curves have been developed for the FORCE 12 and FORCE 15 speakers (Figures 3 through 6), and you will notice that the characteristics of these curves are the same except for frequency shifts and magnitude.

CONSTRUCTION SUGGESTIONS

In the actual construction of the enclosures it is assumed that one has the basic knowledge of woodworking methods. The plans, included, show uncomplicated butt joints for simplicity sake, but the joints can be modified according to the knowledge and expertise of the individual's woodworking skills.

It is most important that the enclosure be as mechanically well built as possible. There should be no rattles or buzzes, and all joints must be airtight. This includes air leaks caused by connector openings. Speakers can't tell the difference between an air leak at a joint and a designed port. The result could be a cabinet that is tuned higher than intended.

Material selection is important from the standpoint of sound quality, the application of the system, the beauty of its finish and, of course, the cost.

For sound quality, the material should have a high density and, in the case of plywood, should be free of any voids that could resonate and color the sound. The ruggedness of the enclosure would depend on the application. For a permanent installation, such as a club or auditorium, particle board or a good grade "chip core" veneer would be suitable. Also, for more appealing wood finish enclosures, select lumber core veneers or even solid wood planks could be used.

Road cabinets, on the other hand, which can be crushed, creased and punctured through abusive handling, require a stronger material such as marine plywood. Added protection for portable enclosures can be provided with edge extrusions, metal corner protectors and front cover panels over the grilles. Also, the final finish of the cabinet has to be considered. Some of the alternatives are painting, vinyl covering, Formica®, or even natural wood finishes, if desirable.

With the slightly recessed baffle, grilles can be easily fabricated for these enclosures.

If the fabric grilles are to be used, a frame slightly smaller than the baffle area should be assembled of 3/4 inch square material and the grille cloth stretched over and stapled to the frame. The grille cloth should be a very open weave fabric to avoid acoustic loading of the speaker or the vent opening.

The grille assembly can be secured to the baffle with velcro type fasteners for easy access to the speaker.

For portable or road type enclosures, perforated metal grilles can be cut to the baffle size and, using the 3/4 inch frame described above, can be assembled to the baffle with screws. There are, of course, many personal touches that could be considered in building your enclosures and each of them should be given serious thought.

As with any product, serviceability should be kept in mind, and driver accessibility is a must. The enclosure plans are all designed for front mounting of the speakers with our SMH-1 mounting hardware kit, which is available as an accessory.

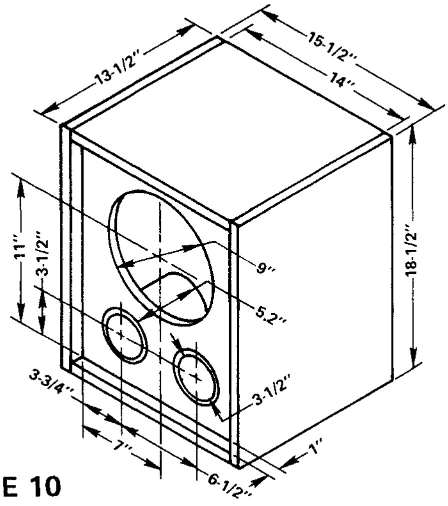
REFERENCE

1. A. N. Thiele, "Loudspeakers in Vented Boxes: Parts I and II." **Journal Audio Engineering Society**, Volume 19, PP. 382-392, (1971 May); PP. 471-483 (1971 June).

The chart below shows the venting required for each of the enclosures mentioned in this addition

	FORCE® 10					FORCE® 12					FORCE® 15				
V_B (Encl. Vol. in ft^3)	1.4*	1.4	1.4	.7	2.8	3	3*	3	1.5	6	5*	5	5	2.5	10
f_B (Encl. Tuned Freq. in Hz)	65	49	98	65	65	55	41	82	55	55	40	30	60	40	40
Sv (Vent Area in $Inches^2$)	19.2	9.8	31.8	9.6	19.2	19.2	19.2	66.4	15.9	47.5	19.2	19.2	66.4	15.9	66.4
Lv (Duct Length in inches)	5.2	5.3	1.7	6.2	.75	2.1	6.7	2.1	6.2	1.3	2.9	7.9	3.2	7.4	4.4
Vent Diameter (inches)	3½	2½	4½	3½	3½	3½	6½	6½	4½	5½	3½	3½	6½	4½	6½
No. of Vents	2	2	2	1	2	2	2	2	1	2	2	2	2	1	2

*The dimensions on the plans are for the optimum tuning and enclosure size.



FORCE 10

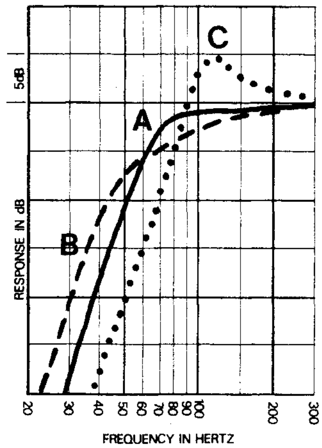


FIGURE 1

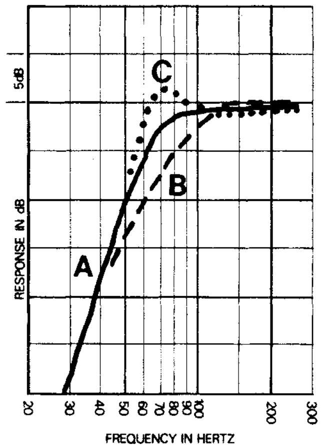
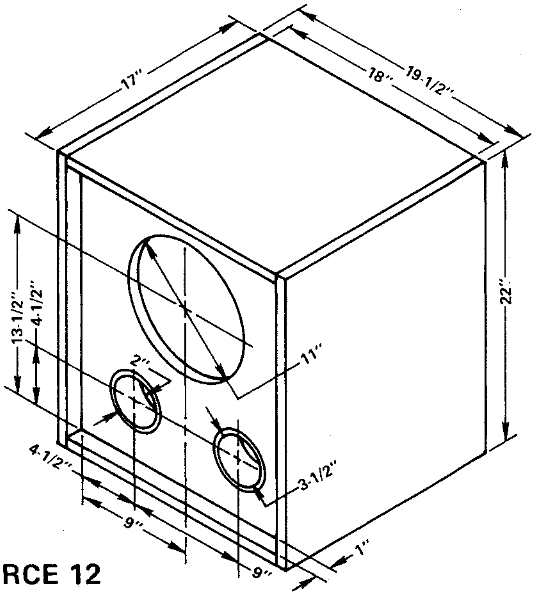


FIGURE 2



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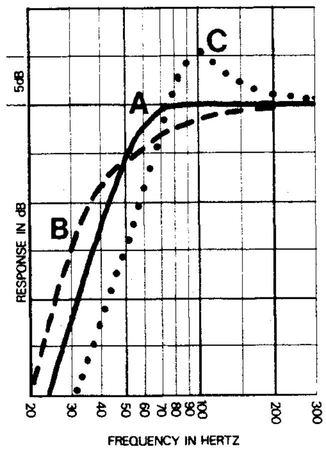


FIGURE 3

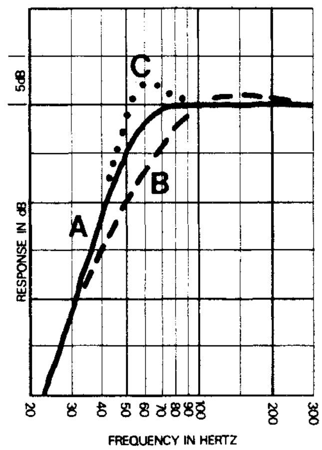
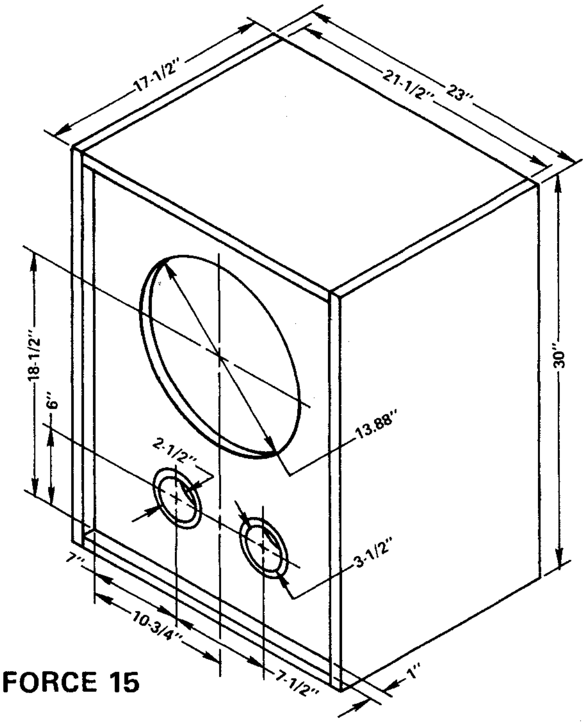


FIGURE 4



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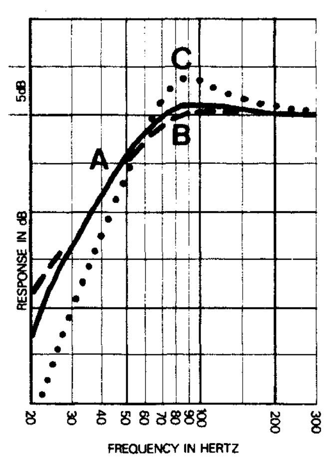

FIGURE 5

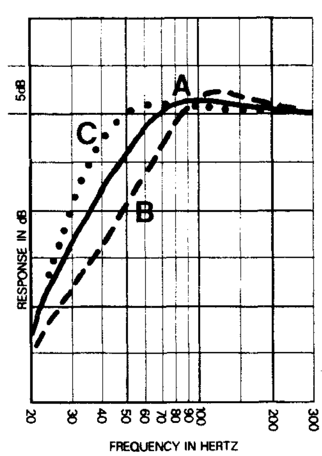

FIGURE 6

Electro-Voice®
600 Cecil Street, Buchanan, Michigan 49107
8234 Doe Avenue, Visalia, California 93277
Electro-Voice Div , 345 Herbert Street,
Gananoque, Ontario • Electro-Voice, S A ,
Romerstrasse 3, 2560 Nidau, Switzerland

THE PA BIBLE

ADDITION NUMBER ELEVEN
PORTABLE SOUND SYSTEMS FOR THE SMALL CLUB

SOME HISTORY

In the 50's, the club musician's portable music system was like the Knight system shown in Figure 1. It weighed 98 pounds, including a microphone with stand and cable. Both speakers could be carried in one hand with the amplifier-mixer in the other. It was small and light but sounded pretty bad by today's standards and wasn't very loud.

With the 60's came the era of rock and roll and the need for louder sound. Responding to this awareness was the Shure Vocal Master ® with its two columns and a mixer-amplifier, as shown in Figure 2. This system was louder but was no longer small and light, and still sounded bad by today's standards.

In further pursuit of louder, better sound, really large systems were developed for rock and roll concerts. One such system used twenty-six microphones, eight speaker systems (each using two 15-inch speakers, a 100-watt midrange, and four tweeters), four consoles, six 200-watt amplifiers, and 4000 feet of mike cable; all carried in a twelve-foot truck! These large systems were an important element in creating very loud sound and also a better sound quality. These developments, while affecting the requirements for club systems, offered little equipment that was practical for the club musician.

Various products came along for club performers but they generally suffered drawbacks: if small, they were very inefficient; if efficient and loud enough, they were too big to be easily portable; and, most used cut-and-try design approaches with honky sounding horns and highly colored bass response. The system in Figure 2, in its many guises, is typical of the heavy, bulky, and low-fidelity products most working performers had to accept.

THE NEEDED SYSTEM

To keep audiences coming back, and club managers hiring, today's performer needs a clean, intelligible sound. Every seat in the room should hear the entire range of frequencies without the distortion that makes full sound seem too loud. Equally important is portability, so the performer doesn't have a back-breaking setup. In other words, there is need for a **portable**, **distortion-free**, **accurate** system capable of delivering crisp, clear sound to every seat in the room.

Most sound equipment for the musician, advertised as portable, fails to meet the definition "easily carried."[1] Much of the equipment would more appropriately be described as capable of transport, requiring hand trucks and built-in wheels.

What are the criteria of portable equipment?
To be portable, the **Human Engineering Guide to Equipment Design**[2] suggests the following:

> "The best individual loads, if carried in either hand by means of handgrips, are about 60 pounds for short distances and 35 pounds for longer distances. The weight of bulky articles (around 30 inches to a side) should not exceed 20 pounds. In general, a weight is "heavy" when it reaches 35 percent or more of human body weight. . . "

The shape of the equipment should follow the long established practices of the luggage industry, with limited size, and handles located in line with the center gravity. A suitcase is a good model since it is carried by hand and is stackable, easy to slide, rugged, scuff resistant, and easily loaded into a car. The equipment should also be durable and able to withstand repeated transport.

FIGURE 1

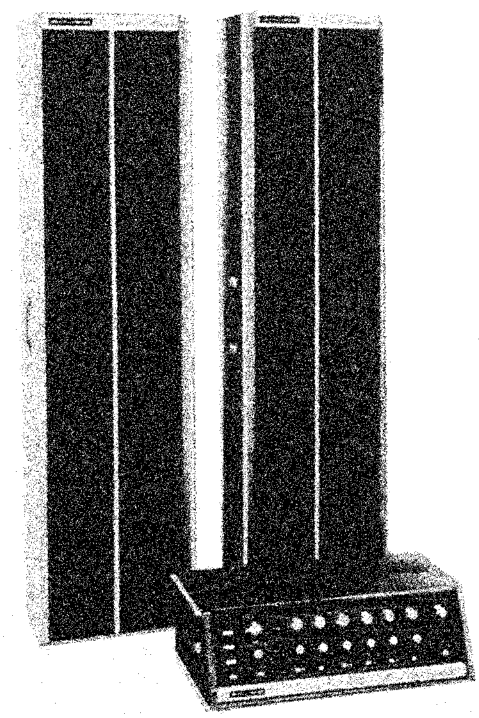

FIGURE 2 – Shure Vocal Master®

How much sound is enough?

Sound loudness is measured by the sound pressure that the system can generate in a club environment. To relate this sound pressure requirement to some recognizable sources, the chart shown in Figure 3 is reprinted from the original PA Bible publication. A further aid to the evaluation of these sound pressure levels is the speech interference criteria. A sound pressure level of 93 dB in the speech spectrum represents a serious impediment to normal conversation, requiring a person to shout at a distance of one foot.[3] As the sound levels in Figure 3 and speech interference criteria indicate, a sound pressure level of 93 dB would appear to be loud enough for the club environment. Actually, sound pressures higher than 93 dB are occasionally needed in a typical program, so a sound system needs to produce high peaks, up to 103 dB maximum output measured about 3 feet from the speakers.

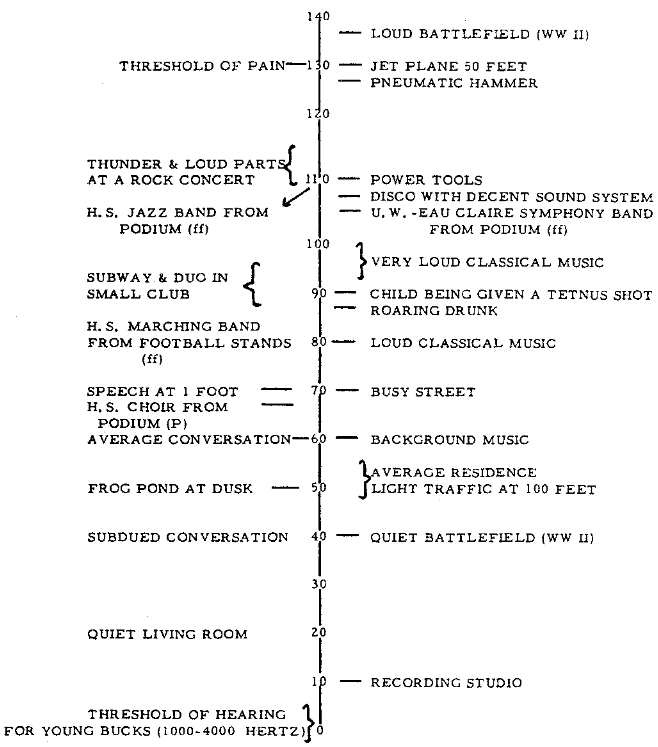

FIGURE 3 – Typical A-Weighted Average Sound Pressure Levels in Decibels (dB) RE .0002 Dynes/cm²

What frequencies should be covered?

Designs encompassing the range of 70 Hz to 15,000 Hz will cover all the needs of the typical club environment. Fig. 4 shows clearly that these frequencies go beyond the range of nearly all instruments and voices. It is important to note, however, that all frequencies should be reproduced about equally. Otherwise, "honky", and other forms of colored sound, will result. This is a common problem with available products, making them sound too loud before they are really loud enough, causing feedback problems, and delivering "muddy" sound ("I can't understand the words"). A system that reproduces sound over all of the needed frequencies, without coloration or distortion, is said to be "accurate".

How can we get this good sound to all seats?

Recent developments in technology have dealt with this problem. In systems using design techniques of several years ago (most available today) the sound heard off the center line of the speakers lack most frequencies above about 2000 Hz. Using a tweeter does not resolve the problem because the horns used with the tweeters aim the highs in a narrow band to only a small part of the audience. With these systems the audience in front of the system hear a fairly well-balanced sound, while persons off-axis hear a muddy, unintelligible sound. This effect can not be corrected with equalization because the "beaming" is inherent in the physical shape of horn or speaker. To correct the problem the horn shape has to be changed, and a significant accomplishment in this regard has been the development of Constant Directivity horns. Systems using Constant Directivity horns spread the sound uniformly at all frequencies so that listeners sitting off-axis hear a balanced spectrum of highs, mids and low frequencies.

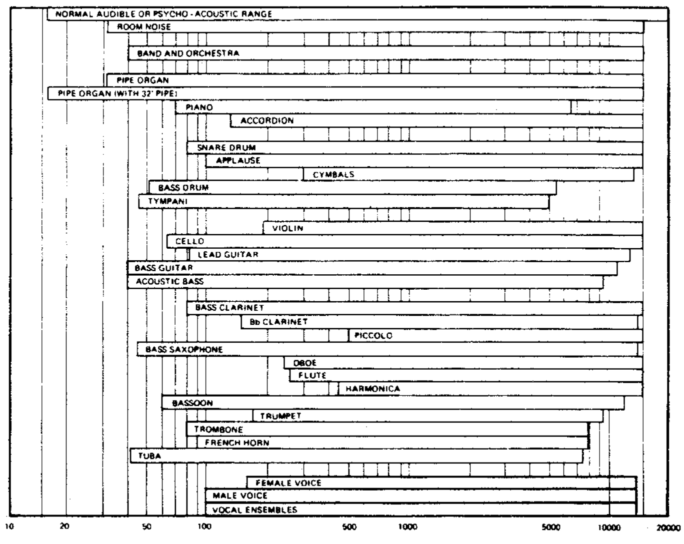

FIGURE 4 – Range of Sounds

A SYSTEM FOR TODAY

The systems available to the club musician today generally emphasize one or two of the necessary features, but fall short of being completely suitable. Some systems are loud enough but are neither accurate nor use constant directivity; some larger systems utilize constant directivity, and are loud enough, but are not portable.

A recent product from TAPCO demonstrates that new technology allows a solution.

The Entertainer™ system is a portable design that has the small size and light weight necessary to be portable, while providing the sound loudness and accuracy essential to the performer. The system provides two high efficiency loudspeakers systems based on Thiele concepts and a mixing board with two separate 100-watt amplifiers. It is conveniently packaged in three containers that do not exceed 30 inches in any dimension and weigh less than 36 pounds each. Thus, the system easily meets the criteria of portability (see Figure 5).

To give uniform coverage, a separate tweeter in each loudspeaker system utilizes a molded-in Constant Directivity horn to give a uniform sound dispersion (Figure 6) and the bandwidth shown in Figure 7.

Thus, it can be seen that the Entertainer fulfills requirements of the "needed system". At present there are no equivalent alternative products, but the Entertainer shows beyond doubt that today's technology permits performers, finally, to satisfy the vital need for a high performance sound reinforcement system that can be readily moved from job to job.

ADDENDUM

The combined capability of the Entertainer's two separate 100-watt amplifiers and two 100S speaker systems will produce a sound pressure level of 93 dB at 65 feet, out of doors. Used alone, each channel will produce 90 dB at the same distance. Figure 8 shows levels that may be expected at other distances.

Trying to figure out what sound pressure levels can be expected in a room is more complex, but it can generally be expected that a given system will be louder in a room than outside due to sound reflecting off of walls or other surfaces. As an example, in a reasonably "live" room 60 feet by 40 feet by 15 feet the same power as above to both speakers, the sound level would not be below 104 dB at any place in the room.

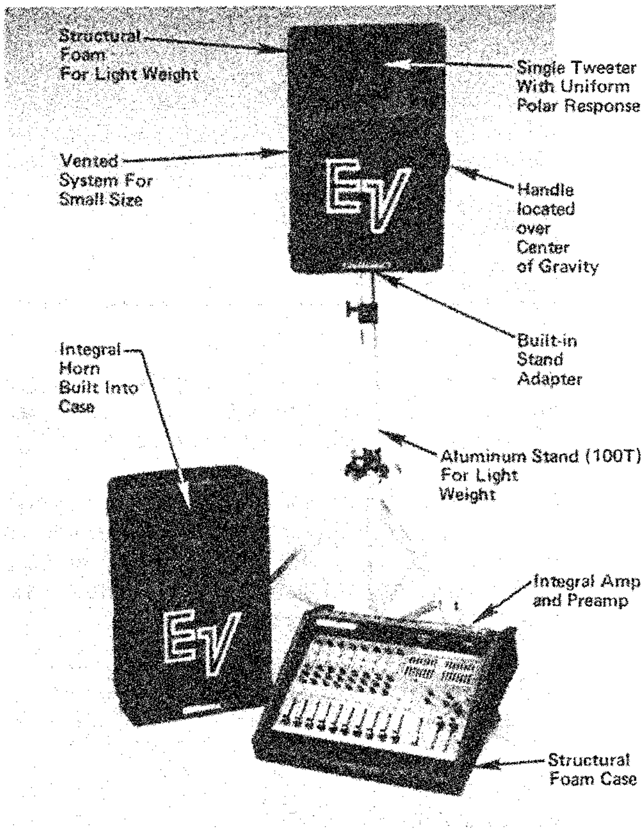

FIGURE 5 — The Entertainer™

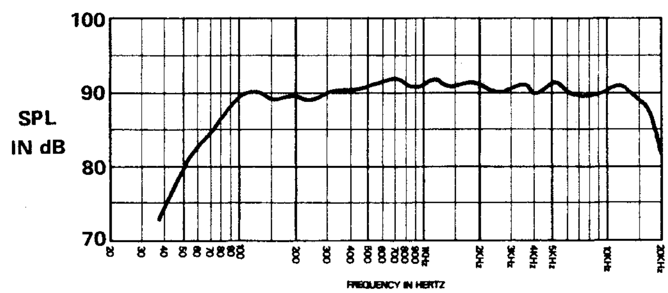

FIGURE 7 — Axial Frequency Response (Bandwidth)

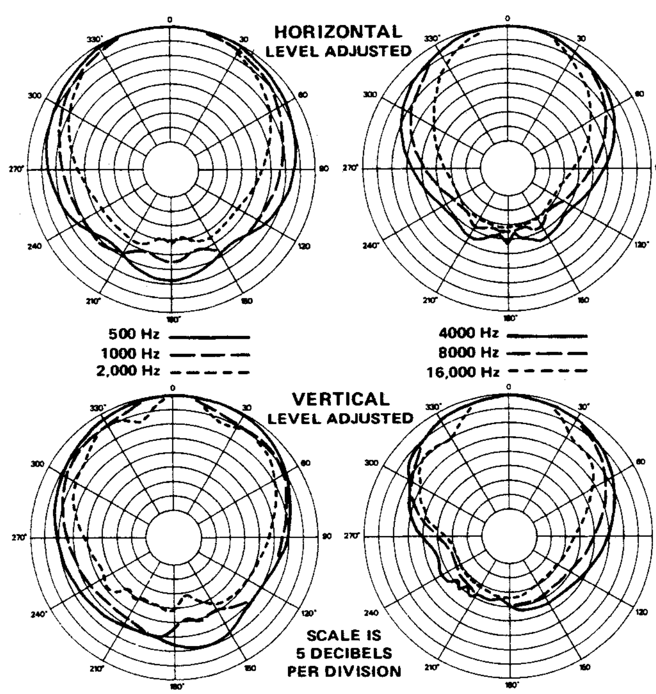

FIGURE 6 — 100S Polar Response (Dispersion)

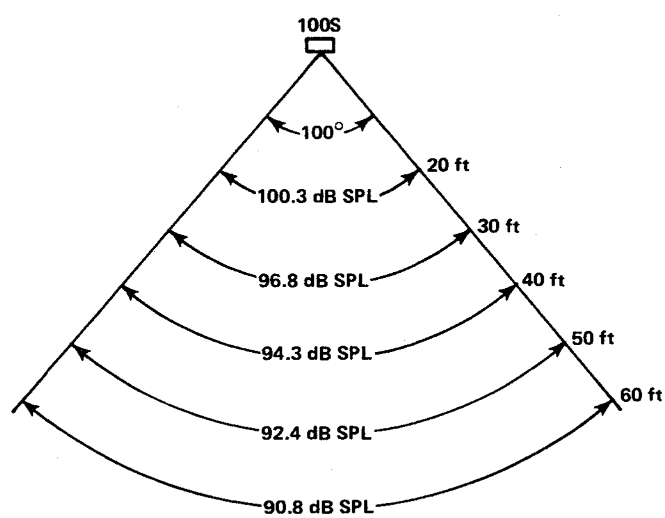

FIGURE 8 — Maximum Direct Sound Pressure Levels for 100S Loudspeaker System

REFERENCES

1. **Webster's New Collegiate Dictionary**
2. **Human Engineering Guide to Equipment Design**, ed. Clifford T. Morgan et al, (McGraw-Hill Book Company, Inc., 1963).
3. **Handbook of Noise Measurement,** Arnold Peterson and Ervin Gross, Jr., (General Radio, 1963), p. 38.

Editor's Comments:

In the last addition (No. 10) we described the sound systems installed in the Music Box, a nightclub in Mishawaka, Indiana. On the evening of April 10 a fire that started in an adjacent building damaged the interior of the Music Box, including the sound equipment. Portions of the building actually burned and the entire building filled with dense smoke. All of the equipment had to be cleaned, including the insides of the electronics. One driver failed due to a fire hose water jet entering one of the horns. The white horns turned yellow, but otherwise were O.K. One amplifier and the mixer were destroyed. The balance of the equipment was saved and placed back in working order, and the Music Box reopened on July 29th.

We had a couple of letters asking about the location of the mid-bass components of the Music Box system. For those that didn't write but also had the same question, the mid-bass cabinets were part of the central cluster and can be seen (dimly) just above and to each side of the horns in Fig. 5.

The next P. A. Bible addition will be about "Do It Yourself Boxes for Music Systems." If you would like to comment on this subject before the article is written, we would welcome your suggestions; and although we don't answer all letters we appreciate receiving your suggestions and criticisms.

Send your comments to:

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THE PA BIBLE

ADDITION NUMBER TEN
A CENTRAL CLUSTER SYSTEM FOR ROCK AND ROLL

The Music Box is an old movie theatre converted into a nightclub featuring a broad spectrum of high-energy live music (Figure 1). The club owner wanted a permanent sound system so good that word-of-mouth would squelch any rumors that groups were "being stuck with a house system", and he asked Electro-Voice to help design and install such a system. The conventional design for such an installation would be a split stack.

FIGURE 1 – The Music Box

Split Stacks

Traditionally, musicians pile speakers on both sides of the stage. Such split stacks can be seen everywhere, from large rock concerts to local musicians in small clubs.

Tradition aside, there are good reasons for using split stacks. Portability is a primary reason. Most road-worthy bass enclosures and high-frequency horns are stackable without special rigging; and, the position to the side of the stage area permits a lower height and simpler rigging. Using two systems allows for stereo operation and a better sound level distribution is produced than with one system on one side of the stage. If maximum sound power is limited, having systems close to the listening area produces at least one area of high sound pressure. Also, in some situations (such as L-shaped rooms) different sound pressure levels are desired in split systems.

There are problems with such a split system, however. With the speakers close to the listening area the sound pressure levels will be high close to the speakers but lower away from the speakers. Reverberation content at the back of the listening area will be high, while listeners close to the speakers will be in the near field. Listening positions at different distances from the two systems will suffer from interference effects due to phase cancellation between the two systems. In deep rooms the relatively low position of the stack makes separate coverage at the rear of the room difficult.

In contrast to the split stack, fixed installations, especially the deeper rooms, traditionally use a single "point source", or "central cluster" placed above the stage.

Single Cluster

The long narrow, reverberant room that resulted from the old theatre conversion suggested to us that such a system using a single cluster would function better than the conventional (for rock and roll) split system. The proscenium arch over the theatre stage would provide a good location for the central cluster, and the stage floor overhang would provide a location for the sub-woofers.

Two features of the Music Box, the dance floor in front of the stage and the balcony located at the back of the theatre, work against using a conventional split system. Such a system would create very high sound levels on the dance floor but much lower level, highly reverberant sound levels at the balcony. A single sound source from an elevated cluster permits separate coverage of theatre areas using constant directivity horns while eliminating the interference effects of multiple sound sources.

With the concept of a single cluster using constant directivity horns as a basis, we proceeded to design a three-way system using separate constant directivity horns for three separate theatre areas.

The Equipment

The block diagram, Figure 2, shows the system's components consisting of a sub- woofer section comprised of two folded

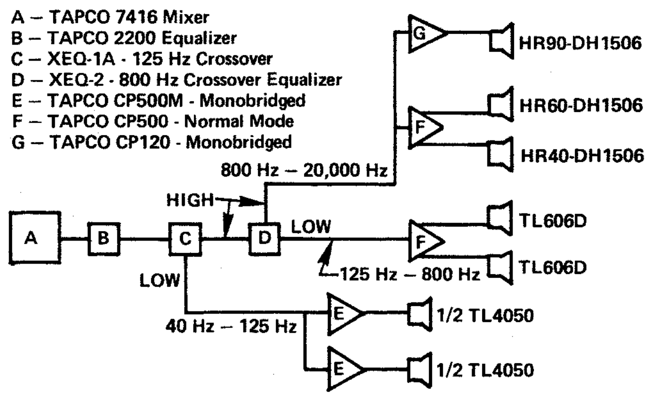

FIGURE 2 – Music Box Block Diagram

Qty	Model	Description
3	E-V DH1506	High-frequency driver
1	E-V HR90	90° x 40° short-throw high-frequency horn
1	E-V HR60	60° x 40° mid-throw high-frequency horn
1	E-V HR40	40° x 20° long-throw high-frequency horn
2	E-V TL606D	Mid-bass speaker system (optimally vented, including two 15-inch EVM15L Series II woofers)
2	E-V TL2025 (custom built)	Sub-woofer (horn loaded; equipped with one EVM15L Series II)
1	E-V XEQ-2	Electronic crossover, with X800 800-Hz crossover-frequency module and EQA horn-equalization module
1	E-V XEQ-1A	Electronic crossover with 125-B3 125-Hz crossover-frequency module
1	E-V/TAPCO 2200	Octave-band two-channel equalizer
4	E-V/TAPCO CP500	Stereo power amplifier (255 watts per channel into 4 ohms, average sine wave power)
1	E-V/TAPCO CP120	Stereo power amplifier (61 watts per channel into 4 ohms, average sine wave power)
1	E-V/TAPCO 7416	16 in, 4-out mixing console

FIGURE 3 – Equipment List

horn speakers using 15-inch drivers, a mid-bass section consisting of two vented systems, each with two 15-inch speakers; and three high-frequency constant directivity horns, one for each of three separate areas of the theatre.

A list of the equipment used in the setup is shown in Figure 3. All system components are stock E-V and E-V/TAPCO products, with the exception of the custom-built TL2025 sub-woofers. The TL2025 is half of a TL4050, a folded horn scaled up from the Sentry IVB low-frequency section to accommodate two 15-inch EVM15L Series II woofers instead of the two 12-inch woofers. The TL4050 can provide solid bass down to 40 Hz and a maximum acoustic output of 84 (!) watts. That is over 100 times that of a typical "high output studio monitor speaker system". On its side each TL2025 slips easily under the stage. One of the TL2025 sub-woofers is shown in Figure 4. (Plans for the TL4050 are available by writing to Electro-Voice, Inc., 600 Cecil Street, P.O. Box 186, Buchanan, Michigan 49107.) The cluster which handles the range above 125 Hz is shown in Figure 5.

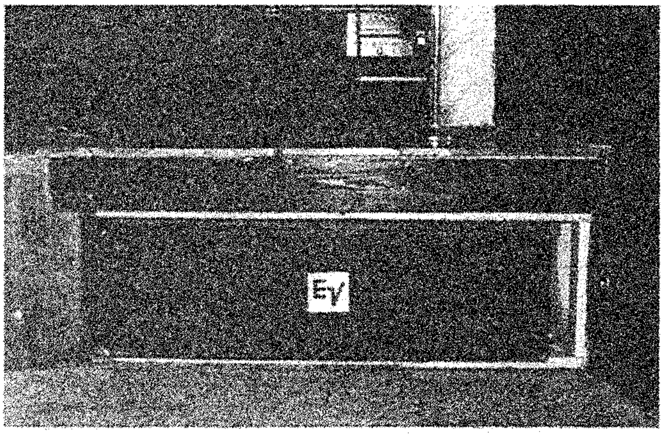

FIGURE 4 – TL2025

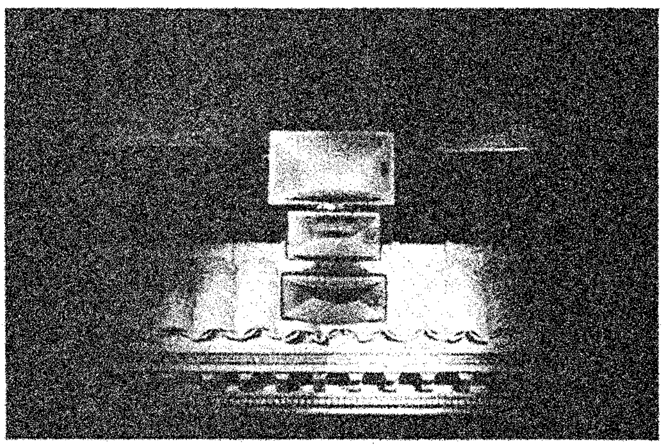

FIGURE 5 – Central Cluster

System Adjustment

The first step in adjusting the system was to correctly aim the high frequency horns to get uniform coverage over the floor and balcony areas. Horns were aimed as shown in Figures 6 and 7. Note that only one horn covered each audience area in order to virtually eliminate the interference effects. Such a design demands constant directivity horns, like the E-V HR series, whose coverage angles are essentially constant over the entire operating range.

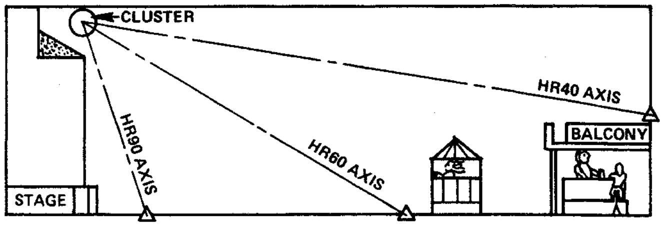

FIGURE 6 – Vertical Positioning

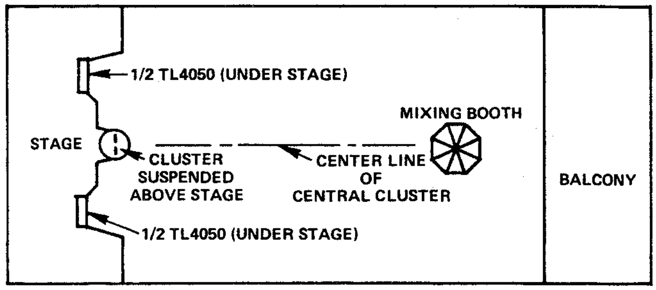

FIGURE 7 – Horizontal Positioning

From a practical viewpoint, the horns can be aimed visually, using the major conical (flat) portions of the horns as a guide. This is an important concept. The conical sections define the horn's 6-dB-down coverage angle. See Figure 8. For example, the angle between the HR90 and the HR60 was adjusted by viewing the cluster while walking backwards from a point within the HR90's coverage angle to a point within the HR60's coverage angle. The angle between the horns was set so the "top" conical section of the short-throw HR90 was leaving the viewer's line of sight as the "bottom" conical section of the mid-throw HR60 was coming into view. This resulted in a vertical angle of 40° between the horns with the horns' 6-dB-down vertical coverage angles approximately tangent. When all horns were aimed in this fashion, interference between any two horns was minimized and very uniform coverage achieved.

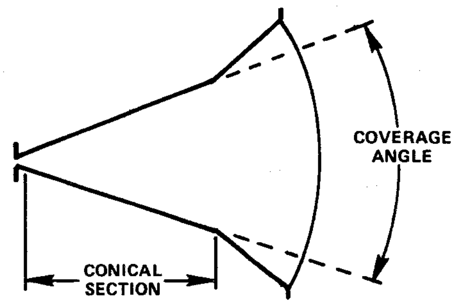

FIGURE 8 – Conical Section of Horn

As the block diagram shows, the system is tri-amped and the first step in adjusting the overall level and balance of the system was to adjust the relative levels of the horns, mid-bass, and sub-woofers.

While it was tempting to do this by ear using our favorite cassette featuring the Amazing Rhythm Aces (a popular way to EQ and adjust a rock-and-roll sound system) we decided to use the instrumentation employed by the professional sound reinforcement contractor. Specifically, we used an Ivie-30A real-time spectrum analyzer. The Ivie was driven with the output of three omnidirectional condenser microphones (custom manufactured at Electro-Voice). In this way, it was easy to assess the overall uniformity within, say, the rated beamwidth of a given horn, without "walking the house" to get a feel for the average. The output of the three mikes was summed by a White Instruments Micplexer. This device averages the amplitude of each mike's output, so that the average sound pressure level – without regard to the relative phase of each mike's output – can be measured. (Simply running the three mikes into a mixer and using that output would give misleading results. For example, if two microphones were located in the room in such a way that the sound pressure levels at their diaphragms were of equal amplitude but exactly out-of-phase, the output of the mixer would be zero. In contrast, there would be plenty of output indicated by the Micplexer, since it sums only the amplitude of the signals.)

To facilitate system evaluation and adjustment, the output of the IE-30A was interfaced to a Hewlett-Packard 7035B X-Y recorder via an Ivie IE-17A. This combination of equipment made it super-easy to run hard-copy frequency response curves with a resolution of .2 dB.

Since there are three high-frequency horns, three separate level adjustments were required. These adjustments were:

1. With the HR60 and HR40 horns turned off, and with the measuring microphones in the HR90 coverage area, the relative levels of the HR90, mid-bass and sub-woofer were set. The system was then equalized using the octave band equalizer.
2. The measuring microphones were moved to the HR60 coverage area; and, with system on and balanced as in step 1, the HR60 horn was added to achieve a balance for the HR60 coverage area. The HR40 remained off during steps 1 and 2.
3. The measuring microphones were moved to the HR40 coverage area and the HR40 level adjusted to achieve a balance in the third area.

Step 1

Starting with the HR90 horn, we placed the three microphones well within the coverage angle of the short-throw HR90 horn. First, we turned on the mid-bass system (125 to 800 Hz), taking care that ambient room noise was at least 10 dB below the measured signal so that our readings would not be affected. The resulting mid-bass-only curve is shown in Figure 9. Second, we turned off the mid-bass systems and turned up the level of the short-throw HR90 horn so that its output matched that of the mid-bass systems. See Figure 10. Then, the mid-bass speakers were turned on again. The composite curve is shown in Figure 11. The SEQ-2 crossover has a polarity-reversal switch as well as an unusual control which can delay its low-frequency output by up to about two milliseconds at crossover. We made certain that the settings of these controls were consistent with the smoothest response in the crossover region. Finally, the sub-woofers were added to the system. See Figure 12. The overall system response is quite smooth, typical in smoothness of responses found in high-quality fixed installations after the "house curve" has been equalized with a multi-band, one-third-octave equalizer. The curve closely follows the Boner preference curve to 10,000 Hz. There is some experience and evidence – beyond the scope of this article – which indicates that clusters made up of constant directivity horns "sound best" when the 3-dB-per-octave rolloff of the Boner preference curve is somewhat less. Thus, we opted to boost the response in the highest octave using the octave band equalizer. The resulting curve is shown in Figure 13.

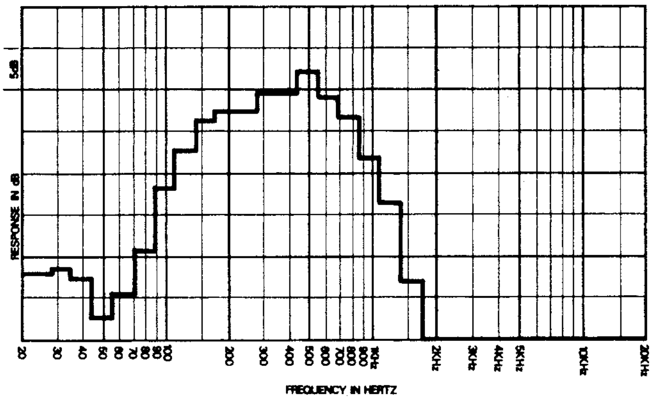

FIGURE 9 – Mid Bass Curve (HR90 Coverage Area)

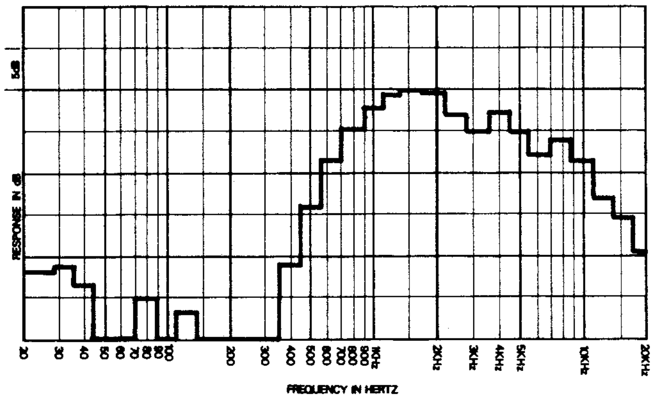

FIGURE 10 – HR90 Horn Curve

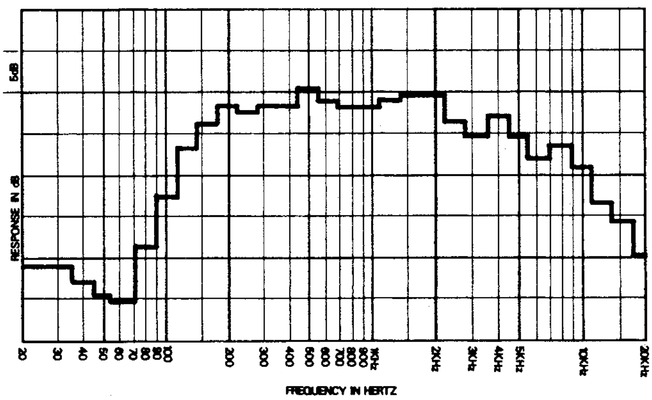

FIGURE 11 –
HR90 and Mid Bass Curve (HR90 Coverage Area)

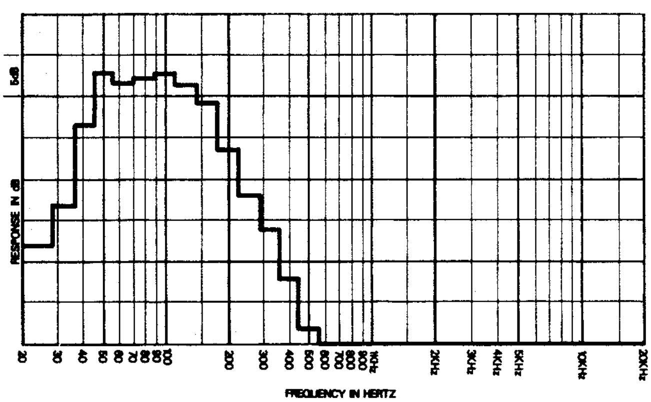

FIGURE 12 – Sub-woofer Curve (HR90 Coverage Area)

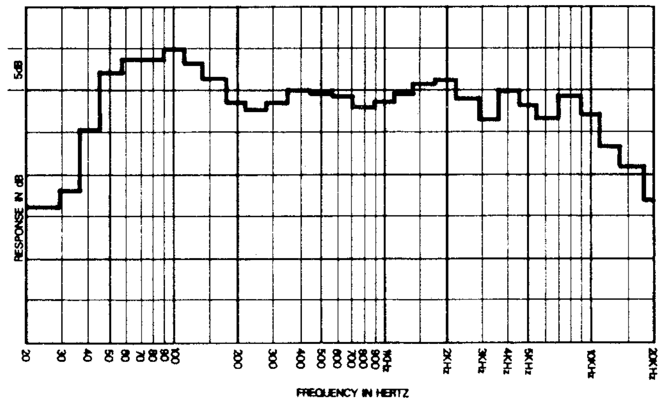

FIGURE 13 — HR90 Coverage Area (Final Adjustment)

Step 2
To set the level of the HR60 horn, the three microphones were moved to the HR60 coverage area. With the levels set as in Step 1, the level of the HR60 was increased until the frequency response curve matched the curve obtained in Step 1. Figure 14 shows the system response with the HR horn both on and off. No equalization settings were changed.

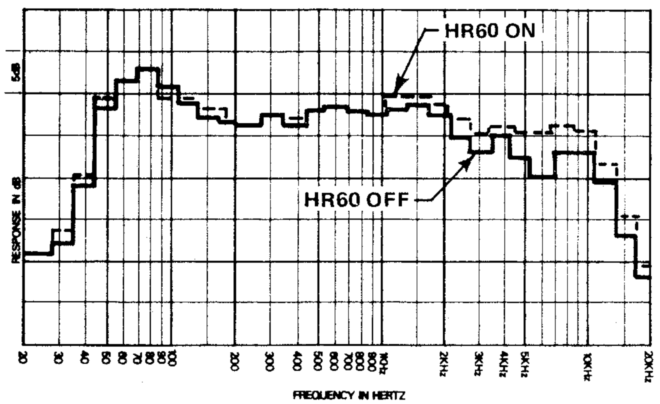

FIGURE 14 — HR60 Coverage Area Curves

Step 3
The final step was to incorporate the long-throw HR40 horn. This was done in the same fashion as the mid-throw horn. Figure 15 compares the response with the microphones in the beamwidth of the long-throw horn, with the level of the long-throw horn appropriately adjusted. As an experiment, the long-throw horn was turned on and off while spoken vocal material was played through the system. With the long-throw horn off, speech level was high, but the non-vowel speech components had a "muddy" quality.

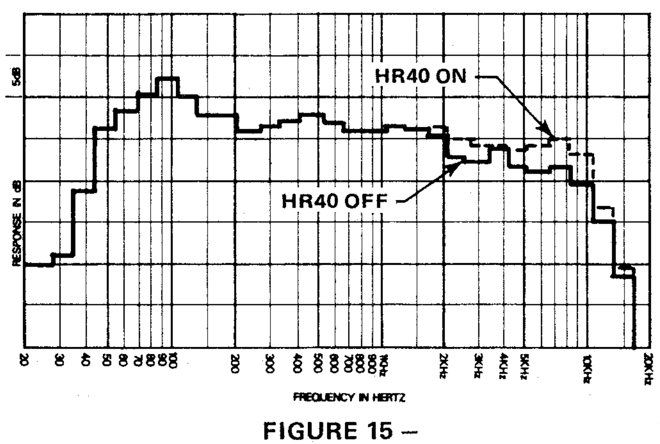

FIGURE 15 —
HR40 Coverage Area Curve (Horn On and Off)

Switching on the long-throw horn brought the sound clearly into focus, getting a quality strikingly similar to that observed from a position in the pattern of the short-throw horn.

Overall Coverage and Response Uniformity
Figure 16 shows an overlay of three frequency response measurements made with all system components operating but with the trio of measuring microphones in the three different locations: within the short-throw coverage pattern, within the mid-throw coverage pattern, and within the long-throw coverage pattern. Coverage is uniform within ± 2 dB over most of the frequency range.

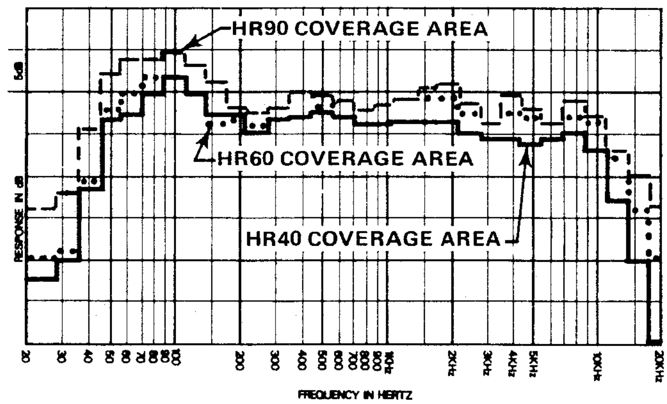

FIGURE 16 — Curves for Three Different Coverage Areas

Sound Pressure Levels
With wide range speech and music program, unweighted average levels of about 118 dB were obtained before the onset of significant amplifier clipping. This sound pressure level was judged sufficient for all Music Box activities, including rock and roll.

Any comments you wish to make about this addition or future additions of the PA Bible are welcome. Please send your response to:

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THE PA BIBLE

ADDITION NUMBER NINE
MIXING FOR THE LIVE PERFORMANCE

1. INTRODUCTION

Prior PA Bible additions have discussed speakers, microphones and their characteristics as applied to PA installations for the musician. The variety of subjects covered is shown in Figure 1, which lists all of the previous additions and their content.

While Addition No. 5 discusses the interconnection of equipment, no discussion of the control of the sound system elements was included. That, we felt, was the proper subject for a separate PA Bible addition; hence, the following discussion of "Mixing for the Live Performance".

Having discussed in some length the major elements that comprise the input and output of a PA system, it is appropriate that we now discuss the manner in which these signals are combined and directed. For the most part, this means discussing the procedure by which microphone and amplified instrument signals are combined, amplified and connected to loudspeakers.

The gear used to accomplish this task is called a "mixer". The operator is also called a "mixer". As the term is often used in PA sound work, the mixer is the operator but also has the overall responsibility for the PA system, including the microphones, amplifiers and loudspeakers.

While our discussion will center on the routing and control of the electrical signals comprising the main outputs of the mixing equipment, we will include those elements of the overall system that seem applicable. In effect, we are interpreting the phrase "Mixing for the Live Performance" in the broad sense, and will discuss the planning, set-up and operation of the PA system. While it is our intent to make this a broad discussion of mixing, we do not intend to cover the "art" of mixing, since we feel that the correct balance, equalization, loudness and overall "feel" of the sound varies from performance to performance depending upon the intent of the performance and the personal tastes of artists involved.

PA Bible Addition	Subject
Original Article	A Guide to setting Up a Speaker System for PA
No. 1	Drivers and Horns
No. 2	Speaker Power Handling Capacity
No. 3	Microphone Types
No. 4	Understanding Equalization and the Various Types of Equalizers
No. 5	System Interconnection
No. 6	The Constant Directivity White Horn White Paper
No. 7	Crossovers and Biamping
No. 8	Microphone Techniques

FIGURE 1 – PA Bible Additions

II. THE MIXER

The mixer is the equipment that combines, controls and directs the input signals to the required outputs, primarily program and monitor amplifiers. The inputs are usually from microphones with levels of –50 to 0 dBm or amplified instruments with levels from –20 to +30 dBm. A typical output from a PL80 dynamic microphone would be –40 dBm. By adjusting various input levels the mixer is used to provide a nominal line level program output of +4 dBm feeding a 600-ohm line. In P.A. applications this output is connected to one or more amplifiers, depending upon the particular set-up.

The mixer equipment provides a means for combining the various required inputs regardless of the differences in signal levels, and permits these combinations to be easily changed.

An important constraint on this equipment is that each audio signal must be preserved, and the mixer must not introduce distortion or modify the signal except by intention.

The primary element of the mixer is the channel. Each input signal is connected to a channel, and the number of channels in a mixer determines the number of signals that can be combined at any given time.

Figure 2 shows the basic elements of a single channel of a mixer.

A complete mixer consists of a number of these separate, isolated channels having the elements shown in Figures 2 and 3, combined into one main output (two for stereo) and a number of auxiliary outputs (such as monitor output). A commercial mixer such as the TAPCO C-12/Series Two, with expanders, may have as many as 44 of these separate channels.

The important features of a channel element include:

1. The phantom feed for electret and condenser microphones. This voltage is sometimes switchable.
2. Provisions for alternate line inputs. These optional inputs are usually higher level, unbalanced feeds, such as electronic organs and pianos, electronic guitars, synthesizers.
3. Attenuation prior to, or part of, the first stage of amplification. This attenuator (pad) is used to prevent overloading the first amplifier stage. Preventing overload and clipping of input signals is one of the most important elements of good mixing.
4. A gain stage prior to the fader and equalizers for an optimum signal-to-noise ratio.
5. Bridging connections which are electrically isolated from the main channel, providing monitoring, effects, and special device connections.
6. A fader to adjust the output of the channel to obtain the proper "mix" with mics and instruments from other channels.

These individual channels are combined in a variety of possible connections to provide, in addition to the main program output, a number of additional outputs. Figure 4 shows the block diagram of the TAPCO 7200 series which specifically illustrates these features. Only one channel is shown with the understanding that all channels are connected in parallel at each of the buses. In addition to P.A. speaker amplifier outputs, the inputs are bridged to provide solo, auxiliary, monitor, and effects signals that are electrically isolated from the main mixer output. These signals can be controlled separately from the main mixer output.

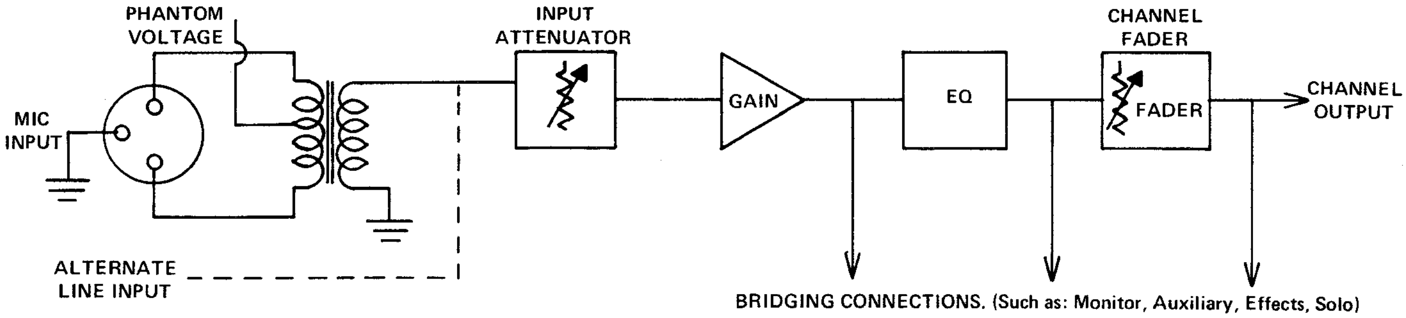

FIGURE 2 — Channel Schematic

The versatility of the modern mixer as exemplified by the 7200 series illustrates the complexity of the sound equipment that may be used for the music performance.

Separately controlled stage monitors, special effects and multiple microphone set-ups can add up to a very complex system and one that requires proper planning for success.

III. PLANNING

Planning obviously depends on the particular event and location and, of course, the "mixer's " relationship to that event; but, regardless of these constraints, certain basic information is required. Proper planning means having this information early, at least prior to the day of the performance. The required information includes:

1. The number of performers and their instruments.
2. A layout of the stage area showing the location of the performers.
3. Stage monitors required.
4. The size of the audience and theatre and location of audience and stage.
5. Sound level desired at the audience location.
6. Any requirements for special feeds such as recording and back stage monitoring.
7. Location of AC power outlets (or for large systems with many amplifiers, the circuit breaker box.)

Discussions with the performance organizer, performers and facility personnel prior to the performance are very important, if not essential. A visit to the location **is** essential.

From this information equipment requirements can be determined. Knowing the musicians, their location and the instruments they play, the number and type of microphones can be determined. Refer to PA Bible Additions 3 and 8 for information to assist in your selection. Microphone mounting requirements should be

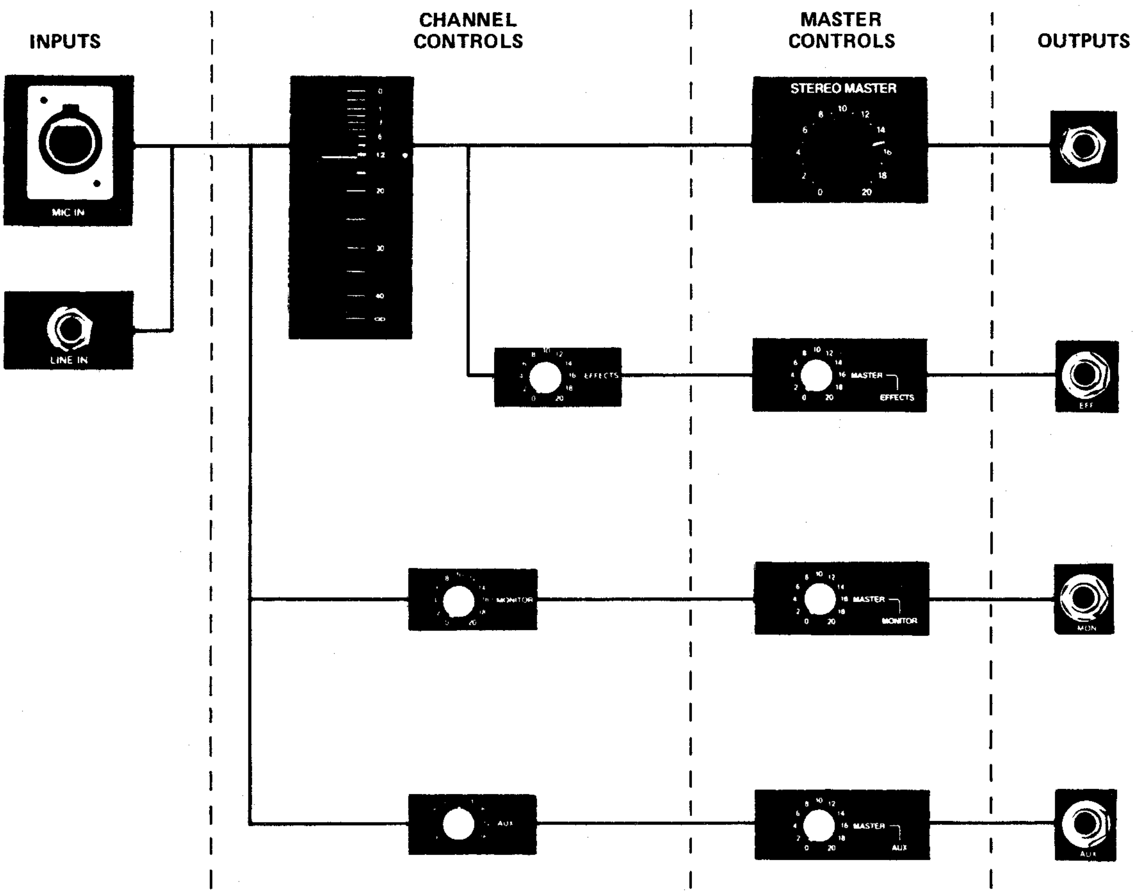

FIGURE 3 — Channel Pictoral

considered to determine the number of mic stands required, and any special mounting requirements such as boom mounts.

A layout of the stage area showing microphone positions combined with a knowledge of the physical plant permits a location for the mixer and amplifiers to be chosen.

Microphone and speaker cable runs (snakes) can now be planned.

With the number of mics determined and auxiliary equipment requirements known, the mixer requirements can be determined.

The size and physical layout of the "house" will determine the number and type of loudspeakers, and the size of the amplifiers that are needed. Consult the original PA Bible article and additions 1, 2, 6, and 7 for more information on power requirements and speaker placements.

Equipment requirements must be matched against the equipment available. Additional equipment may be needed. Limited equipment resources may require modification of the performance itself, with modifications in the number of instruments used or number of monitors requested.

Planning requires that these conflicts be worked out prior to the set-up. Needed additional gear must be obtained, and the prudent sound technician will test all new gear well in advance.

For large installations, check lists for equipment and tools are a good precaution. Without check lists, unusual items such as a flash light or paper labels are easily overlooked.

For maximum reliability redundant equipment should be provided whenever possible.

IV. THE SET-UP

The set-up can be considered in two steps:

1. Locating and interconnecting equipment.
2. Testing.

Location of equipment is fairly straightforward. Microphones are mounted on appropriate stands and placed at their approximate location. Loudspeakers are located in number and position by the size of audience and arena.

The amplifiers and any auxiliary equipment are located at one location whenever possible, and at a position where they can be viewed from the mixer position. The mixer should be in the audience area preferably, but, some performing facilities have specific areas designated that must be used.

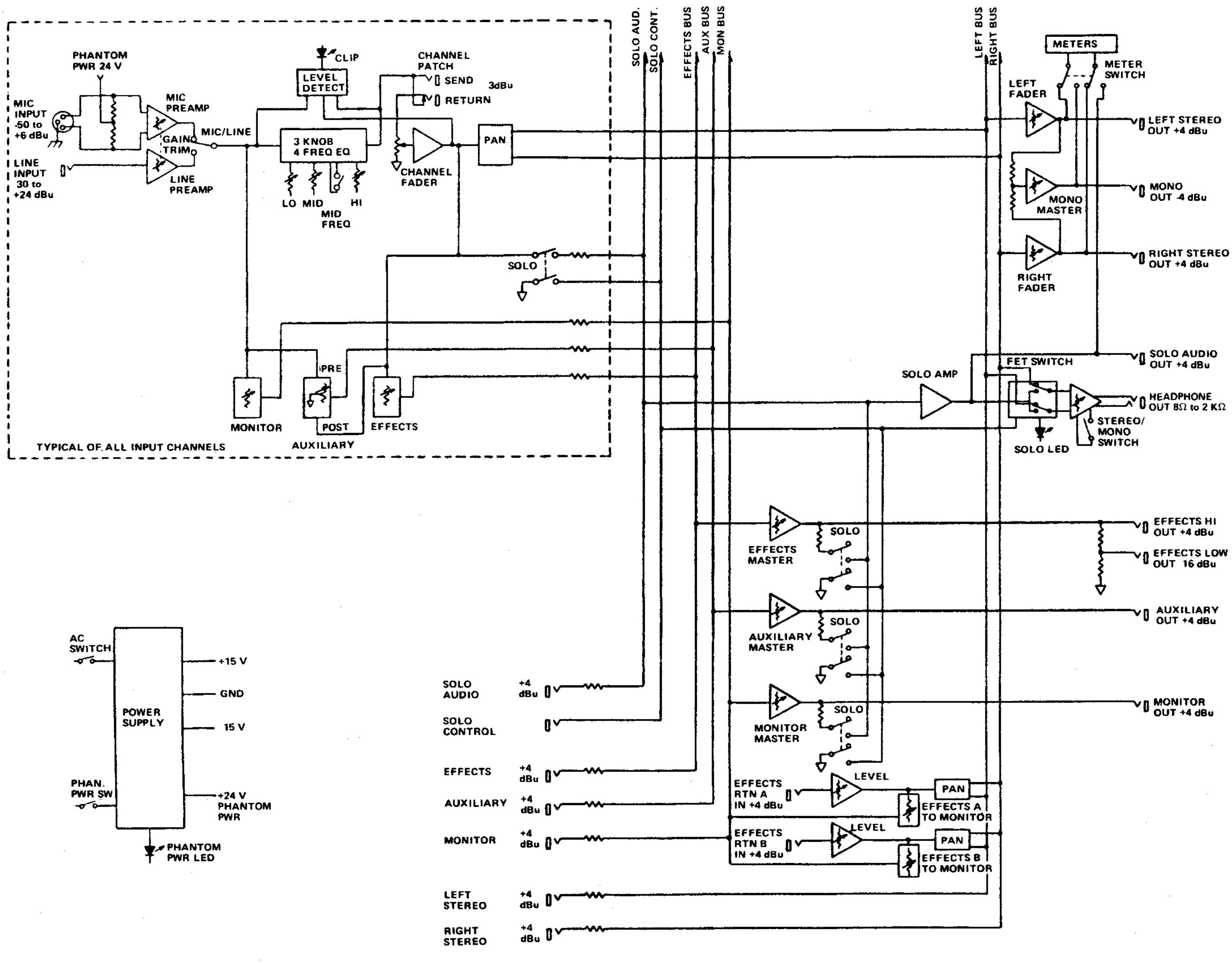

FIGURE 4 — Block Diagram Panjo 7200 Series

Prior planning will have provided the necessary tables and chairs.

It is essential that all equipment be tested as a part of the set-up and that vocal and musical inputs used for testing be near normal level. At this time, levels can be set and sound quality checked. There should be no indication of clipping, noise, distortion or feedback. Input attenuators may have to be adjusted.

A general procedure for testing and adjusting the system should include the following:

1. Equalize the main program outputs. This procedure may involve sampling microphones and sound level meters, or may simply use a known source and the mixer's ears.
2. Adjust input attenuators to prevent clipping, making certain that sufficient "headroom" is available.
3. Set the level on the vocal microphones.
4. Adjust instrument levels.
5. Set stage monitor levels and equalize to prevent feedback.
6. "Run through" each musical grouping for balance, sufficient level and lack of feedback.
7. Determine and adjust loudness of audience sound.

Overall loudness of the sound is a vital consideration. Loudness can relate to the quality of the performance and, also, to the audience being able to converse. Managers, owners, performers and the mixer can all be a part of determining the proper loudness.

Monitor and microphone positions should not be moved after the system has been adjusted.

During the set-up, consideration should be given to identifying the microphones on the stage and at the mixing equipment so that it is readily apparent as to which channel controls what microphone. With multiple microphones of the same model this can be tricky. Colored windscreens on microphones or colored wrapping on cables and connectors can be identified with a corresponding color patch at the matching channel on the mixer. Numbers or instrument and musician's names can be used to label each channel.

Proper identification of all inputs is absolutely essential.

A useful concept in mixing is to group inputs in adjacent channels, or to group inputs to feed sub group controls that are available on some mixers such as TAPCO 7400 Series. This grouping allows several microphones, such as those used on the drums, to be considered as a single unit and controlled as such. This reduces the number of elements in the set-up that require separate control. This is an especially useful concept for large installations.

It is also prudent to identify cables at a point near the connection to the mixer.

V. THE PERFORMANCE

The critical element of the music performance is that it is live and not subject to correction on a retake. Proper planning and rehearsal are the steps used to obtain a "smooth" performance.

For a performance with a number of acts and several band arrangements, a work sheet should be developed during rehearsal and used during the performance to guide the switching of inputs and control of levels.

Good practice will include switching off unused microphones.

Despite the best planning, unrehearsed, unplanned events often occur during a performance. Knowledge of your equipment and of the particular set-up are the basic tools that can deal with the unexpected, including equipment failure.

This knowledge will not only guide problem solving efforts, but will provide the confidence necessary to avoid panic.

Some specific actions that are useful in dealing with the unexpected include:

1. Using headphones and the "solo" feature available on some mixers to isolate problems, particularly distortion, in a specific channel.
2. Switching from an apparent dead microphone to a similar unused microphone in the same location.
3. Maintaining a supply of redundant mikes and cables.
4. Being able to quickly view all equipment to determine status, including clipping indicators.
5. Avoiding "cranking up" the gain when the signal level drops. Feedback is usually not far removed from normal settings. Knowing the feedback level for each control is helpful.

Despite the best intentions and preparations, "glitches" are likely to occur during live performances. The corrective actions available are limited. Knowledge of what is **not** possible or practical is as important as knowing the possible corrections. It is better to "live through" some problems than to thrash around making the problem worse – or at a minimum, calling attention to it. Experience is the best teacher in this regard.

Planning, preparation and experience are the necessities for the "live performance", that and a little luck.

We would appreciate receiving your comments about this addition or about topics for future additions. Please send your response to:

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THE PA BIBLE

ADDITION NUMBER EIGHT
MICROPHONE TECHNIQUES

Electro-Voice receives many inquiries regarding the selection and application of microphones. These questions can be very specific, such as "what model microphone should I use with a B-flat clarinet", or very general as, "recommend a good vocal microphone". It is very difficult to adequately answer these questions in a letter or telephone conversation. We hope that the following information will fill in the blanks in our responses.

This addition is divided into two sections, moving from the general to the specific.

In Section I microphone technique is analyzed to establish a basis for making specific judgments, with emphasis on the live performance of a contemporary (pop/rock) music group.

In Section II some specific microphone setups are described and the techniques involved considered.

Throughout this addition we assume that the microphones selected are operating properly, and that they are correctly matched for level and impedance. Our discussion will concentrate on the decisions of selection, placement and quantity, choices the user can make to affect the quality of sound.

SECTION I

In PA Bible Addition No. 3 different types of microphones are discussed, and some guidelines for selection are suggested. (The material in Addition No. 3 relates closely to the subject of microphone technique and the reader may wish to reread No. 3 as an introduction to the present addition.)

While the information in Addition No. 3 is involved in the selection of a microphone, it is not sufficiently specific to a given application. To achieve the "right" sound we need to choose a microphone with a specific **frequency response** and a specific **directional characteristic.**

In any microphone application we are concerned with sounds that we want to "hear" and sounds that we do not want to "hear". For example, we may accentuate bass frequencies of the voice that we want to "hear"; while, in a different situation, we may reduce bass frequencies to eliminate acoustic noise and reverberation that we do not want to "hear" In a similar way we may use a directional microphone to favor the pickup of sound we want to "hear" and eliminate unwanted sounds.

Directional Microphones. As suggested above, a directional microphone can be used to avoid the pickup of distracting, undesirable noises. By reducing the pickup of off-axis sounds, a directional microphone reduces the amplitude of undesirable sounds such as noise, reverberation, or music from an adjacent instrument or a vocalist.

The most commonly used directional microphones belong to the general class referred to as cardioid microphones. This class includes microphones with cardioid, hyper-cardioid, and super-cardioid polar patterns. Plots of these patterns are shown in Figure 1. The super-cardioid and hyper-cardioid microphones have a broader rejection in the back hemisphere and a greater overall rejection of random sounds, but there is the penalty of a back lobe. The hyper-cardioid microphone, because it is only down 6 dB at 180°, is less widely used than the cardioid and super-cardioid microphone. In some applications it is desirable to reduce sounds that arrive from a specific, known direction. As the polar plots show, these microphones can be "pointed" so that the null rejects sounds from a specific direction. Thus, pickup of direct sounds from adjacent instruments can be reduced by aiming the microphone so that the angle of lowest sensitivity points towards these instruments.

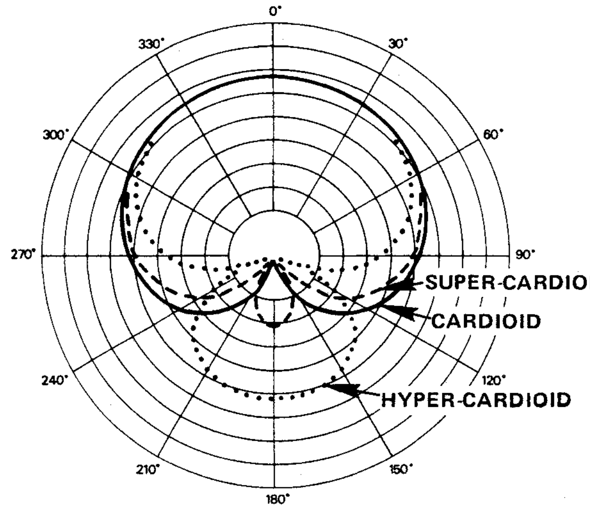

FIGURE 1 – Cardioid Polar Patterns

Directional microphones can also be used to reduce the pickup of sounds with a random directional character, such as noise and reverberation. This is especially true indoors where sounds are sustained, forming a reverberant field.

Thus, we can generalize that a directional microphone should be selected when it is necessary to isolate a sound source or to reduce room reverberation and ambient noise. It is also true that a directional microphone will allow a more distant placement. This is particularly true of a microphone with high directional characteristics, such as a line microphone.

In this regard, it might seem logical to always select the most directional microphone, such as a line microphone. However, line microphones are limited to long pickup distances because of their large size, and variation in directional characteristics with frequency.

A related question is "why not always use a directional microphone?" The following discussion will lead us to the answer "that there are characteristics of non-directional microphones which are advantageous in certain applications".

Non-Directional Microphones. The reasons for using directional microphones relate to their cancellation of unwanted sounds and their proximity effect (discussed later). Unfortunately, certain problems unrelated to the pickup pattern result from the construction of directional microphones. For this reason non-directional microphones (often called omnidirectional) are recommended because they can offer the following advantages over a directional microphone:

1. Less mechanical noise.
2. Less wind noise.
3. Less breath noise.
4. Wider bandwidth.
5. A more accurate pickup of reverberant sounds.

Weighing these advantages against those of a directional microphone is difficult. It is the old story of "apples vs. oranges". Often the decision is one of finding the least objectionable choice, or selecting an omni-microphone because of its simplicity and lower cost.

Frequency Response. In selecting a microphone to give the "right" sound, it is important to choose a microphone with the proper frequency response.

To record a pipe organ it is essential that the microphone selected respond to the very low frequencies of the organ, that the bandwidth of the microphone match that of the source. However, using the widest bandwidth is not always correct. Limiting the bandwidth of the microphone discriminates against noise sources, such as air conditioner sounds and reverberation sounds in large enclosures. These sounds can often be reduced by the attenuation of low frequencies, without affecting the higher frequencies associated with the human voice. This attenuation can be a part of the microphone design.

Finally, adjustment of frequency response can be used to enhance certain frequencies, usually to give a special "artistic" effect. This enhancement is often characterized by a boost of the bass frequencies called proximity effect, as discussed below, or by a broad peak in the 5 kHz region which projects the upper range of the human voice.

Proximity Effect. Proximity effect is the microphone characteristic that results in a boost in bass frequencies for close microphone spacing.

All cardioid type microphones exhibit a proximity effect, but by vastly different degrees. Single-D cardioid microphones exhibit substantial low frequency boost, while Variable-D ® cardioid microphones minimize this effect.

Two positive benefits of proximity effect are: (1) an enhanced artistic effect resulting from the tonal coloration of the voice and the control of this coloration by the performer and/or operator; and, (2) an increase in "gain-before-feedback" in sound reinforcement use. The enhanced low-frequency response of Single-D microphones has become something of a standard for live-performance vocals, and is sometimes a desired effect in recording studios as well. Single-D microphones are often designed specifically to provide a balance of enhanced bass response coupled with enhanced response in the 5 to 10 kHz frequency range. There are precautions, however, which should be noted when using Single-D microphones.

1. The increased gain-before-feedback should be viewed as an increase in loudness that accentuates low frequencies, and the microphone should be used close to the source for maximum increase.
2. The low frequency boost at 1/4 inch is not the same as that at 1/2 inch; and, in fact, diminishes from lips to about eighteen inches. Proximity effect makes these microphones sensitive to position, and therefore, control of position is essential to achieve the desired sound quality.
3. Single-D microphones can be notorious noise makers in the wind, including the wind from the mouth.
4. The spectral characteristics of Single-D microphones can vary from one model to another, and because artistic results are usually involved, the selection of a specific model can be both significant and meaningful to the result.

As these precautions indicate, the Single-D microphone is not for everyone or every microphone; and that is why Electro-Voice invented the Variable-D microphone, a different tool.

Variable-D microphones have considerably less proximity effect than Single-D microphones. Variable-D microphones are used where proximity effect is not desirable, but rejection of sounds is desired. In P.A. uses, where the microphone is not used close to the source, the Variable-D microphone provides improved gain-before-feedback compared to Single-D and omnidirectional microphones. In recording, isolation of sources and reduction in reverberation pickup also make the Variable-D microphone a correct choice.

PLACEMENT

Having selected an appropriate microphone type, good technique requires the correct placement of the microphone. Correct placement involves finding the "right" **angle** and **distance** of the microphone position relative to the source.

Angle. Choosing a particular angle from the microphone to the source requires a knowledge of the radiating characteristics of the source; knowledge of the effects of objects near the source and microphone; and knowledge of the microphone polar characteristics. Musical instruments are usually composites of several sources, each with different spectral and radiation characteristics. This complexity needs to be recognized and taken into account. More on this in Section II.

It is obvious that objects between the source and microphone should be avoided. Most users recognize that a "shadow" effect results from such objects. Not as obvious is the effect of an object adjacent to either the source or microphone. Reflections from such objects "color" the sound in a complex relationship involving size, wavelength and transit time, and can result in creating both peaks and dips in the output of the microphone.

Distance. The distance from the microphone to the sound source is often a critical factor in determining sound quality.

As sound radiates outward from a source, it decreases in intensity, and with increasing distance there is a corresponding reduction in microphone output. A degradation in the ratio of desired sounds to unwanted sounds results as microphone spacing is increased. Thus, problems with reverberation, wind noises, mechanical noises and particularly feedback become more difficult. To avoid these problems, placing the microphone close to the sound source is a good starting procedure. The consequence is a good signal-to-noise ratio, excellent isolation, and the individual control of multiple microphones. These effects relate to the prevalence of "close miking" in the recording and sound reinforcement of popular music. The tight, isolated sounds resulting from this spacing permits the control and manipulation that is desired.

Close spacing is often only a starting procedure, however. As indicated in the discussion of angle, musical instruments often have multiple sources. Microphone placement too close to one of these sources can alter the perceived character of the instrument. Moving the microphone away results in a more balanced pickup by reducing the differences in distances from the various source locations to the microphone, i.e., close spacing requires control of the microphone position. Being close, small changes in spacing can cause large changes in microphone output. For moving sources such as a clarinet or saxophone, maintaining a fixed, close, position may not be possible.

The above factors **tend** to move the microphone placement away from the source. A factor that often **mandates** moving the microphone location is visual distraction. Some visually oriented activities, such as television and motion pictures, will not tolerate a microphone in the field of vision. The result is a reliance of specialized products, such as line microphones. In the extreme, no microphone position is adequate and "dubbing" results.

A common result of moving the microphone away from the source is the increase in reverberant content when "miking" indoors. When some reverberant sound is desired, microphone spacing can be adjusted for the desired content.

Feedback in public address applications is a complex subject involving many factors, one of which is microphone placement. In this regard, a rule that applies to microphone placement is that it should be located in the direct sound field of the performer, and in the reverberant sound field of the loudspeakers; i.e. the microphone is placed near the performer, but far from the loudspeakers.

One specialized placement that has found favor lately is to position the microphone close to a large reflective surface, especially in locations where such a surface cannot be avoided. An excellent example of this technique is the mounting of the microphone near the stage floor in miking stage plays. Elimination of the reflections of sounds off the floor results in a substantial improvement in the sound pickup quality. Spacing to the floor should be in the size range of one-half inch or less.

NUMBER OF MICROPHONES

Proper placement and correct selection may not be a sufficient microphone technique. More than one microphone may be required. Just when and where more than one microphone should be used is the subject of this section.

Redundancy is occasionally a requirement, as when separate house feeds and broadcast feeds are used. Placement in this case is no problem, and there are no restrictions in using separate microphones for each feed.

By contrast, where two microphones are connected additively, the microphones should be mounted very close to each other or spaced sufficiently apart to avoid phase cancellation. If the microphones are placed so that the sound arrives at both microphones with comparable amplitudes, and if the microphones are spaced at different distances from the source, then phase delay occurs. This phase delay introduces a noticeable degradation in the signal. A useful rule to reduce this effect is the three-to-one rule. By spacing the microphones apart three times the distance to the source, phasing will not be a problem.

Isolation and control of sound sources is a primary reason for using multiple microphones. The use of a separate microphone to pick up crowd noise is a good example. Another is the pickup of individual sources within a group. Common in recording, this technique permits balance and the use of special effects, and permits experimentation in obtaining a desired sound.

The pickup of instruments, such as the piano, often requires more than one microphone. The instrument may be large and have multiple polar patterns associated with different parts of the instrument, requiring separate physical locations for pickup.

A mix of a "tight" close sound and the reflected, "open" sound can readily be adjusted using two microphones. Phasing in this instance is not a problem because the microphones "hear" different sounds.

A stereo signal is an obvious source for the requirement for two or more microphones. Several stereo recording systems use coincident microphone pairs, such as the MS and XY systems. The ORTF system uses a pair of cardioid microphones, separated by 17 cm and angled 110°. Other systems use microphone pairs, special microphones, and spaced microphones to produce the stereo signal. Each of these systems have well defined requirements for the number and type of microphones used in the basic system.

HINTS AND OTHER MISCELLANY

There are some additional subjects bearing on microphone technique that need mentioning.

Elimination of Distracting Signals. Air movement past a microphone, whether it be caused by motion of the microphone or by breath blasts or wind, can cause distracting, noise-like microphone outputs. Windscreens are provided for such problems and should be used as they are very effective.

Mechanical excitation is a frequent source of unwanted microphone signals. Tapping the case of a microphone can produce an electrical output, as, for example, the "click" of a performer's ring while holding a microphone. Shock mounts, both internal and external to the microphone, are available to the user. The user should be aware of the available accessory shock mounts and the sensitivity to mechanical shock of the microphones he owns. In general, omnidirectional microphones are less shock sensitive than directional types; and condenser microphones are less shock sensitive than moving coil (dynamic) microphones.

Special Microphones. Certain special microphones are available to the user to improve microphone technique.

Microphones that have special spectral characteristics for the human voice, such as the 635A, are available.

Lavalier microphones permit the fixing of microphone distance where movement is a factor, and these microphones have special spectral characteristics to match their position near the user's chest.

Line microphones, such as Electro-Voice Cardiline® microphones, have polar patterns narrower than cardioids, and are especially useful for pickup of sounds at greater than optimal distances, as in boom mounts in TV and motion pictures.

Some General Guidelines. There are some characteristics of microphone sources and environments that allow general guidelines to be stated for some applications.

For use out-of-doors, where the microphone is handheld, the non-directional microphone is an excellent choice. The non-directional microphone does not color the sound, and has low sensitivity to wind and mechanical shock.

In sound reinforcement applications, especially for handheld uses, the Single-D microphone can offer significant gain-before-feedback amplification. This result, however, includes the bass boost that is associated with Single-D microphones, and the specific microphone and user should be tested in the intended environment.

In extremely noisy environments, noise cancelling, close talking microphones can improve communications. These are special microphones, designed for voice communications. They are subject to breath noises and spacing control is essential; but they do work well in very noisy environments, as in a helicopter.

The microphone, as a tool to assist in the recording or reproduction of an artistic performance, poses a challenge to microphone technique. Appreciation of classical music in live performances is dependent upon receiving reflected sounds, and in recording classical music the inclusion of reflected sound is essential. For this reason symphonic orchestras are usually not miked close; but, rather, with microphone placements that achieve a balanced pickup, including reflected sound. Therefore, multiple microphones are often required to achieve the correct balance for the complete orchestra. It is essential that phase interference resulting from a source that is "heard" at two different microphone locations, not equally spaced, be avoided. Since the bandwidth of the microphones should match the spectrum of the instruments, very wide bandwidth microphones should be used.

The recording and reproduction of popular music tends to be creative, as opposed to the realism sought in classical recording. Balance is achieved in the "mix". Reverberation can be added artificially or with separate microphones. Close miking is used extensively to provide separation of instruments and allow control of the balance. Close miking allows the use of more than one microphone on an instrument to control the "sound" of the instrument. Microphone selection and placement under these conditions permits more selection and variety. The microphone can be a part of the performance in the sense that it can be selective in its pickup due to placement, directionality, bandwidth, and proximity effect.

A potential problem with close miking is the change in level with microphone position. Since the microphone is close, small changes in the distance to the source can cause large level changes.

A technique that is used to avoid this problem is to mount the microphone on the instrument. Two precautions should be observed when doing this: the microphone pickup will be selective in the sound that it picks up, and the microphone may need to be shock mounted to eliminate mechanical drive from the body of instrument to the case of the microphone.

For general guidance in selecting microphones, the guide shown in Figure 2 can be useful.

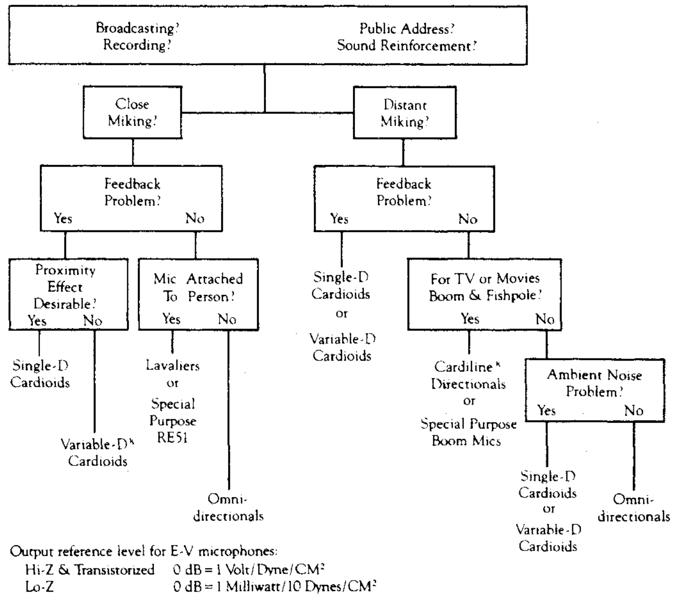

FIGURE 2
Microphone Application/Recommendation Chart

SECTION II

In Section 1 the elements of technique were analyzed without reference to a specific application. In this section a specific microphone installation will be discussed.

Figure 3 shows a hypothetical stage setup for a small music group consisting of a drum, electric/bass guitar, piano, vocalists, and acoustic guitar. The particular microphone installation shown consists of eight microphones: two for vocals, one for the electric/bass guitar, one on the piano and acoustic guitar, and four on the drums.

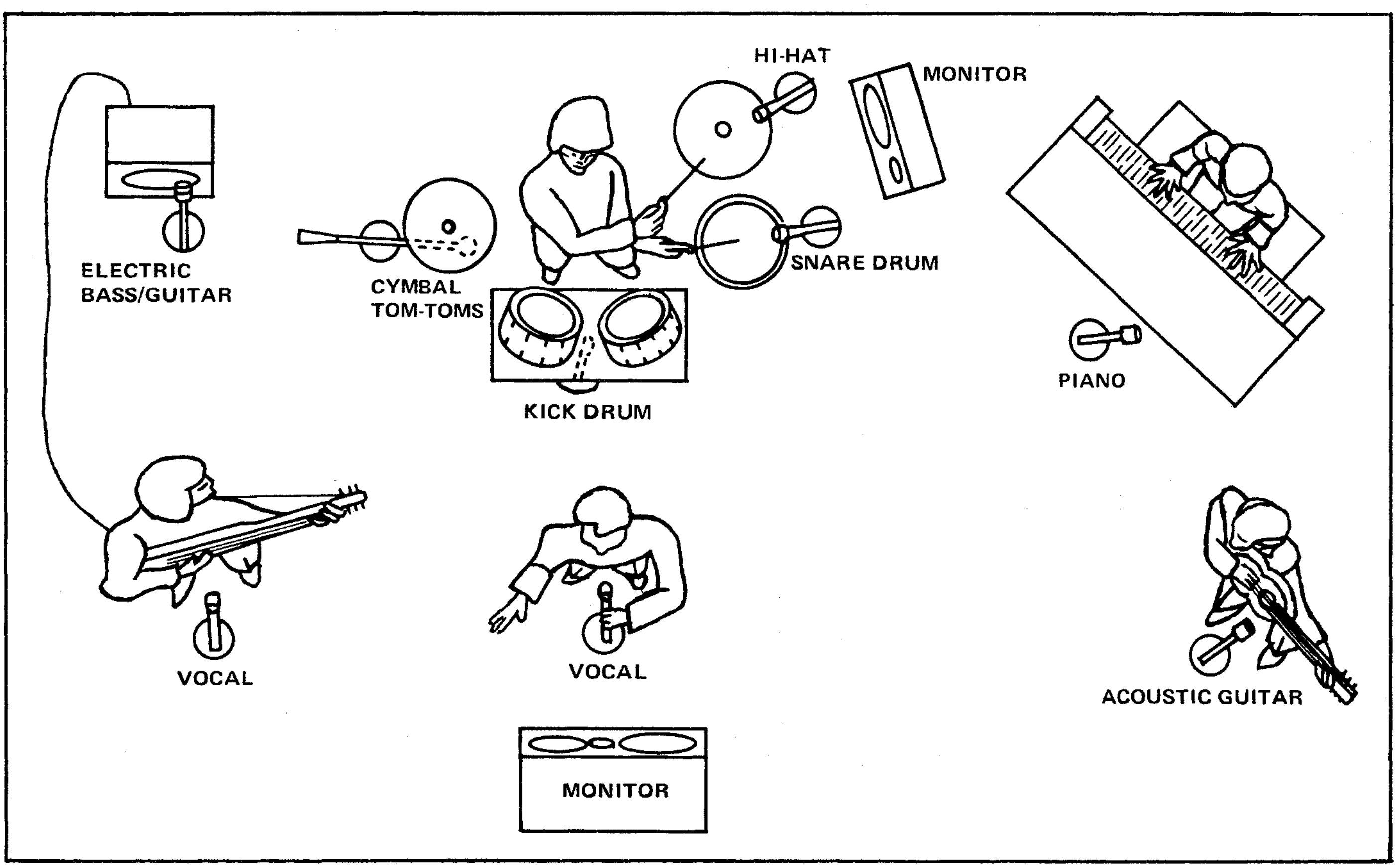

FIGURE 3 — Musical Group Layout

The grouping of the instruments follows standard practice, with the drummer at the rear center and main vocalist center front. The lower volume instruments have been placed together, away from the amplified instrument.

Because this is a P.A. setup with high sound levels involved, close microphone spacing is used throughout. Wherever possible, cardioid microphones have been placed so that the "dead" spot is pointed towards the nearest stage monitor, or other instrument. The P.A. speakers should be placed out front and to the sides of the group, taking care to avoid directing them toward close reflecting surfaces.

Vocalist Microphones. Single-D, cardioid microphones are recommended for vocal use. The microphone should be stand-mounted, as shown in Figure 4, or handheld, so that the cardioid dead spot is pointed towards the stage monitor, which would normally be placed in front of the vocalist. Recommended Electro-Voice microphones include the PL80 (preferred), PL91A, PL76A, and PL77A, with the PL95 being considered for maximum gain-before-feedback. Both microphones should be the same model to avoid the need for level and equalization compensation at the mixer.

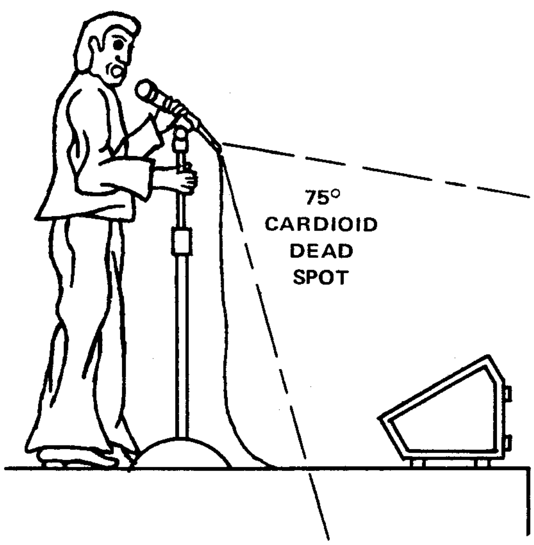

FIGURE 4 – Vocal Microphone Use

For the out-of-doors installations, the Model 376 windscreen should be available.

Electric/Bass Guitar Pickup. Place the microphone very close to the guitar loudspeaker at a point midway between the center and edge of the cone. (See Figure 5.) Select one speaker out of a cluster of identical speakers. A Variable-D cardioid microphone, such as the PL6, or PL11, has been selected for its excellent polar response and smooth, flat frequency response.

Piano Microphone. A single, Variable-D cardioid is shown mounted on a floor stand close to the soundboard (Figure 6). A second microphone, if available, could improve the balance by allowing separate microphones for bass and treble, Variable-D cardioids are recommended.

Acoustic Guitar. Because of the low amplitude of this instrument, pickup is difficult. The microphone, a cardioid Variable-D, is floor-mounted below the hand, pointed upwards toward the hole. (See Figure 7.) Close placement is essential, but realistically must allow for movement of the instrument. As previously suggested, point the rear of the microphones toward the nearest stage monitor (or house P.A., if closer).

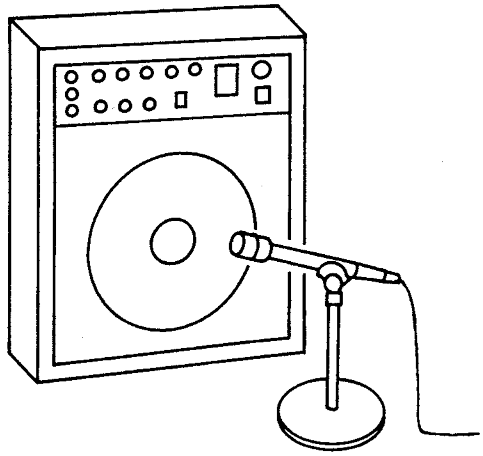

FIGURE 5 – Amplified Guitar Microphone Placement

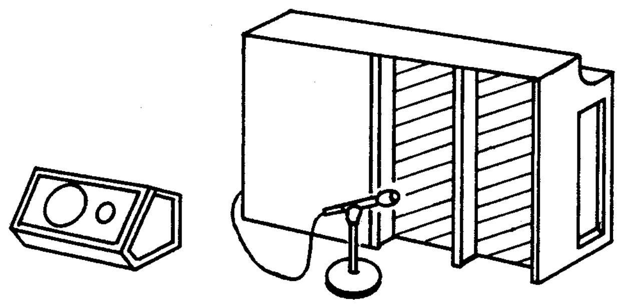

FIGURE 6 – Piano Microphone Placement

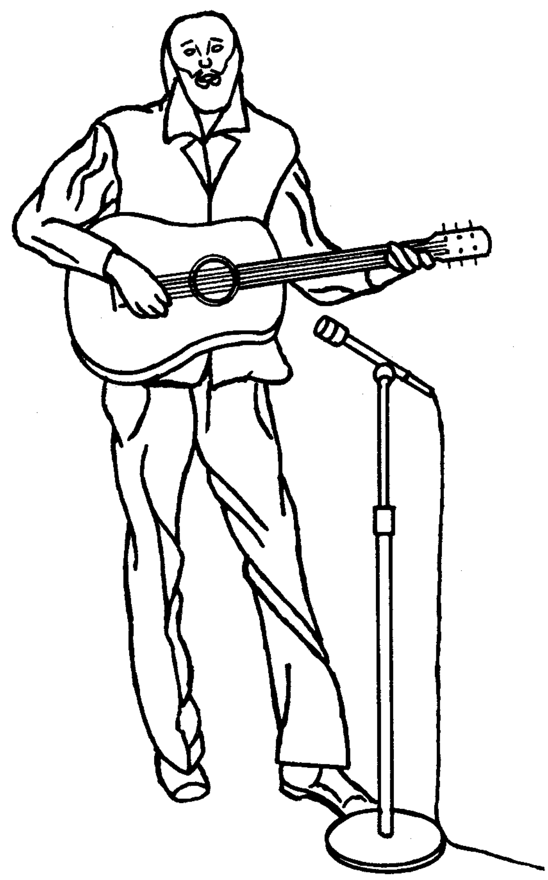

FIGURE 7 – Acoustic Guitar Microphone Placement

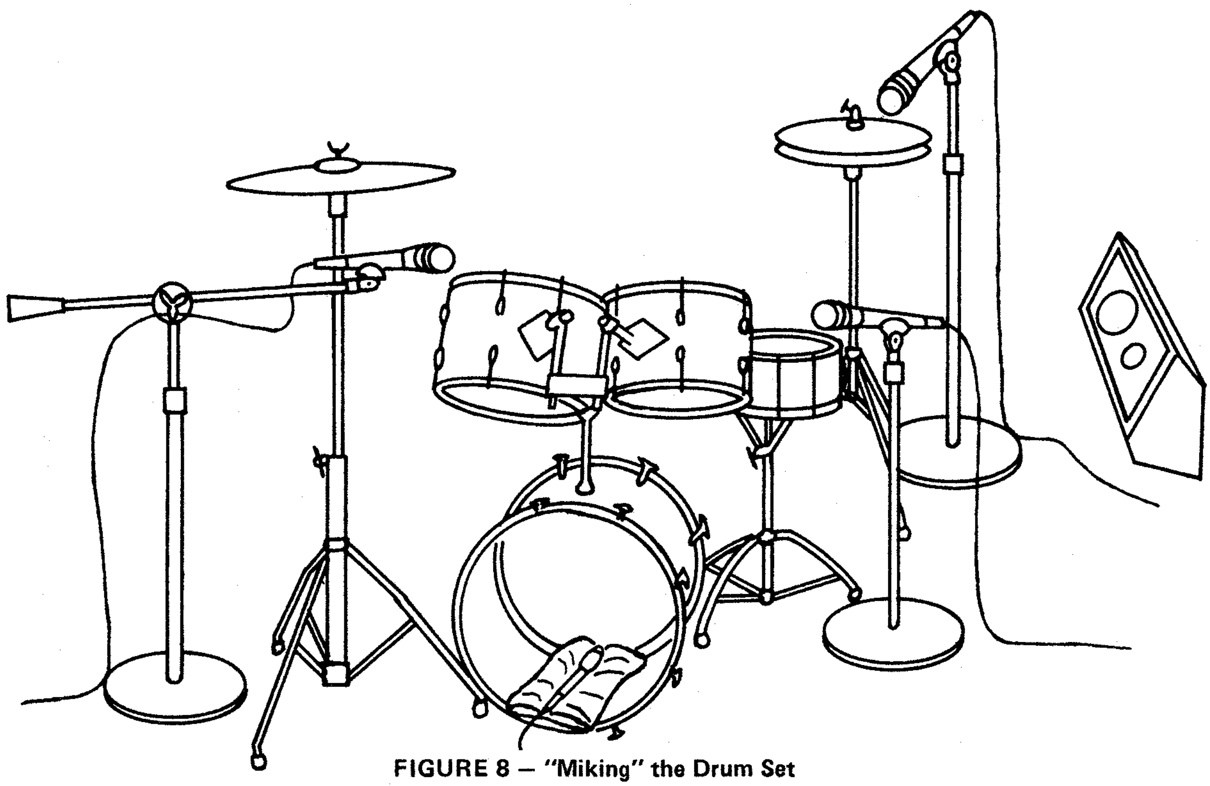

FIGURE 8 — "Miking" the Drum Set

Drum Microphones. Considerable variation in the number and placement is possible in miking drums. In the setup shown in Figure 8, four microphones are used; one for "kick" drum, one for hi-hat, one for snare drum, and one for tom-toms and cymbal.

The "kick" drum microphone, an omnidirectional dynamic such as a PL9 or PL5, is placed on a pillow inside the open end of the drum. A variation that is sometimes used is a PL20 (Variable-D cardioid) on a floor stand pointed into the open end.

Because of high frequencies generated by the hi-hat, cymbals and snare drum, a condenser cardioid microphone is recommended. The Electro-Voice PL77A, and CS15P, are good choices.

The snare drum microphone is mounted above the drum, opposite the strike area, with the front of the microphone protruding over the edge of the drum. A technique which is very effective, if an additional microphone is available, is to place a second mike below the snare drum just inside the rear plane.

The hi-hat microphone is mounted opposite the strike area, above and to the side of the top cymbal.

The microphone used with the cymbals and tom-toms is mounted below the cymbals, pointed towards the tom-toms, using a 90° pickup for the cymbals.

CONCLUSION

Not all readers will find the specific microphone placements in our example applicable to their own requirements. For those readers we recommend experimenting with different techniques, following the guidelines listed in Section I. One precaution for our technically oriented readers: involvement of someone responsible for an artistic result is essential.

Trying different microphones, altering placement and critically evaluating the resulting sound is the heart of microphone technique, and the development of a critical "ear" an essential element. Knowledge, experience, experimentation and a critical ear will result in judgments that comprise good microphone technique.

For readers wishing more information, the following publications are recommended:

1. Lou Burroughs, "Microphones: Design and Application", Sagamore Publishing Company, Plainview, New York, 1974.
2. Alec Nisbett, "The Use of Microphones", Focal/Hastings House, New York, 1974.
3. Bruce Bartlett, **db**, December 1980, p. 32.
4. John M. Eargle, **db**, December 1980, p. 38.
5. Electro-Voice, Inc., **PA Bible**, Addition No. 3.
6. Carson C. Taylor, **Journal of the Audio Engineering Society**, September 1979, Vol. 27, No. 9, p. 677.
7. John M. Eargle, "Sound Recording", Van Nostrand Reinhold Co., 1976, p. 129-137.
8. John Woram, "The Recording Studio Handbook", Sagamore Publishing Company, Plainview, New York, 1979, Ch. 4.
9. Bruce Bartlett, **Recording Engineer/Producer**, June 1980, p. 82.
10. Bruce Swedien, **Modern Recording**, August 1978, p. 50.
11. Wieslaw V.R. Woszczyk, **Recording Engineer/Producer**, October 1979, p. 78.

We would appreciate receiving your comments about this addition or about topics for future additions. Please send your response to:

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THE PA BIBLE

ADDITION NUMBER SEVEN
CROSSOVERS AND BIAMPING

In the original "PA Bible" we discussed the crossover and some aspects of biamplifying. In this supplement we will take a more in-depth look at crossovers and the whys, hows, and advantages of biamplifying.

A crossover is a device which separates full-range program material (electrical signal) into appropriate low- and high-frequency ranges for two-way systems, or into low, mid, and high frequencies for three-way systems. A crossover is necessary to separate and direct the electrical signal from the power amplifier to the various speaker components in a speaker system. Different size speakers are best able to reproduce different parts of the frequency spectrum. Large cone speakers (usually 12, 15, or 18 inches in diameter) are designed to reproduce the low bass frequencies. Their relatively large cone areas and cone excursion abilities are necessary to produce high-level bass. Smaller speakers, such as tweeters or compression drivers on horns, are better able to reproduce the high frequencies. They are also capable of providing wider dispersion of these high frequencies. Since different speakers are necessary to properly reproduce the total frequency spectrum, the crossover is necessary to direct the appropriate frequencies to each speaker.

If low-frequency energy is applied to a small fragile tweeter, the tweeter will try to make the large excursions needed for low-frequency production – and disintegrate. If high-frequency material is applied to a woofer, no damage to the speaker will occur but you won't get many highs. The large, heavy woofer cone simply cannot move fast enough to reproduce the high frequencies.

CROSSOVER FREQUENCY AND SLOPE RATE

Figure 1 shows the frequency responses associated with three typical crossover response characteristics. The "high-pass" responses are for the crossover section which lets the high frequencies through. The "low-pass" responses are for the low-frequency section. The "crossover frequency" is the frequency where the low-pass output and high-pass output cross over one another. In Figure 1 the crossover frequency is 800 Hz.

Figure 1 also shows three different "slope rates." The slope rate is the degree to which the low-pass (or high-pass) section rejects the unwanted frequencies above (or below) the crossover frequency. For instance, a "6-dB-per-octave" slope rate means that the crossover reduces the speaker's signal level by 6 dB every time the musical tone is one octave (or one more octave) removed from the crossover frequency. A one-octave change is two times or one-half the reference frequency, e.g., 800 to 1600 Hz or 800 to 400 Hz.

The other two slope rates shown are 12 dB and 18 dB per octave. These slopes cut off more rapidly, which provides better low-frequency-overdrive protection for the high-frequency speaker components and reduces the interaction between the low- and high-frequency components in the frequency range around the crossover point.

TYPES OF CROSSOVERS

"Passive" Crossovers. A "passive" (or "high-level") crossover is what you would probably find in most multi-way speaker systems (such as P.A. and hi-fi systems). These "high-level" crossovers are made up of capacitors, inductors and resistors. The circuit requires no auxiliary power to operate (such as 120 volts ac) which makes it "passive." Because these crossovers are usually inside the speaker system, they must handle the high power levels delivered by the power amplifier (thus the name "high level"). See Figure 2 for a depiction of a simple high-level crossover.

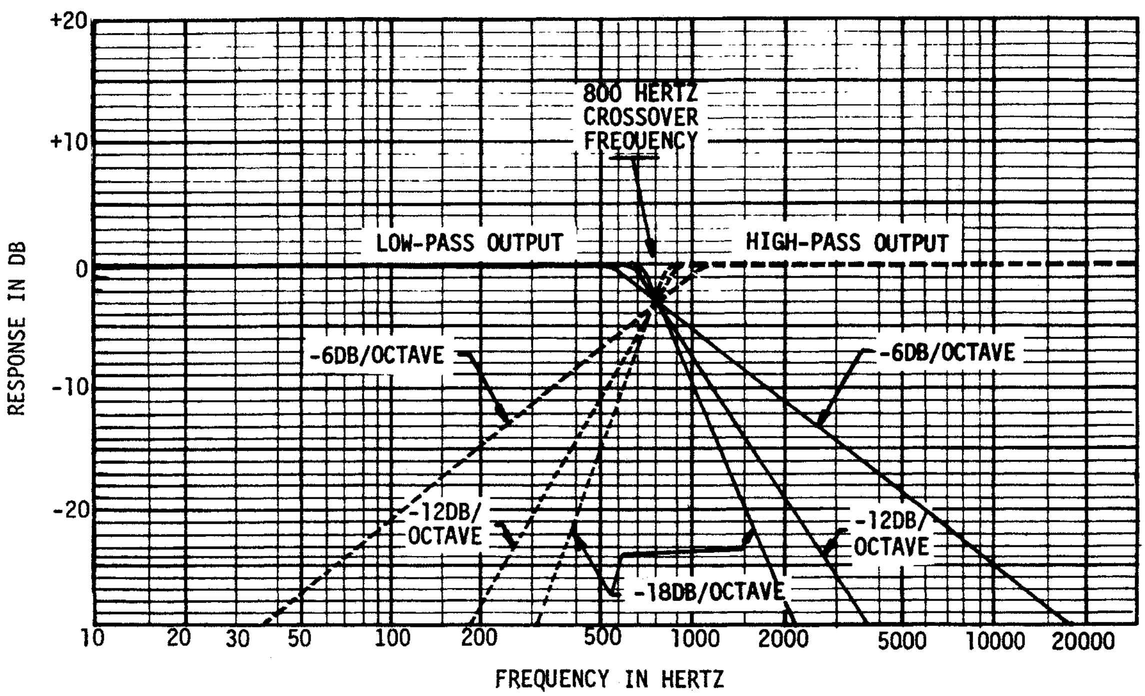

FIGURE 1 – Ideal Three Crossover Frequency Response Shapes

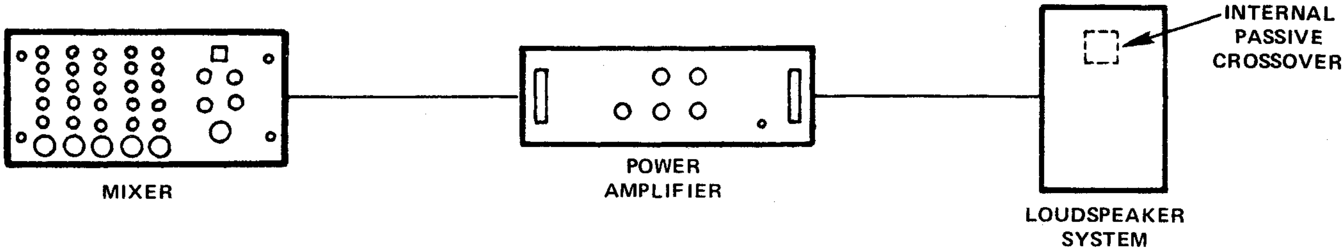

FIGURE 2 – Depiction of Passive Crossover

"Active" Crossovers. The "active" (or "low-level") crossover is placed ahead of, and feeding power amplifiers connected to, each loudspeaker channel (see Figure 3). The low-level crossover works on just a volt or so out of the mixer (thus the term "low-level"). The crossover still performs the frequency separating function, but each output of the crossover feeds its own amp, and the amp feeds its own speaker. (More on that later.) Since the "low-level" crossover drives two (or more) amps, the term bi- (two) amp is used. If an active three-way crossover were employed, the term would be tri-amp. The low-level crossover is usually composed of transistors, op-amps, and associated electronic parts that require an auxiliary voltage to operate (thus the term active crossover).

WHAT TO USE – ACTIVE OR PASSIVE?

When small sound systems are used and high sound pressure levels are not required, the economy of a passive crossover is attractive and the performance very acceptable. Speaker systems such as the E-V S15-3 have an integral passive crossover specially tailored to yield the best overall performance from the component speakers it utilizes. Passive crossovers are a useful and practical approach to many speaker systems. An active crossover, in many instances, would offer no advantage and be quite a bit more expensive.

When a larger number of components are used (such as high-frequency horns, drivers, bass bins, etc.), the active crossover may yield superior performance and better cost. One crossover can be used to drive many amplifiers and, therefore, many loudspeakers.

WHY BIAMP?

Biamping means driving your low-frequency speaker with one channel of a power amplifier and using the other channel to feed the high-frequency driver (i.e., using one amplifier for the low end and a second amplifier for high end). A biamp setup requires a low-level crossover connected between the mixer and the power amplifier. It doesn't have to handle large amounts of **power.** Since the amplifier's input is almost purely resistive, there is no interaction between the network and its load. (If interaction would occur, the crossover point or slope rates may change.)

Active crossovers are generally easier to use and adjust than passive units. Some recent units (such as the TAPCO EX-18) even have continuously variable crossover frequency controls, so you can adjust frequency coverage of different high-frequency (HF) and low-frequency (LF) drivers for different situations. LF and HF speakers can differ in efficiency by factors of two to ten (3 to 10 dB) or more. In a conventional system this difference in efficiency works against you. The usual way to get equal level from the drivers is to resistively pad (attenuate) the more efficient driver (usually the HF driver). This is really a waste of power, because the pad or attenuator merely converts the power, according to its setting, into heat. To get around this problem, the Electro-Voice XEQ808 and XEQ804 passive crossovers use capacitors to attenuate the high-frequency driver. This capacitor method also provides equalization of the driver on the horn and since it is **not** resistive it does not waste power like resistive padding methods do. Transformers can also be used to step the voltage down, but this is usually very expensive in comparison to resistors.

In a biamped system, the difference in driver efficiency can be balanced where it should be, at the level controls in the system in front of the power amps, so the signal delivered by the power amplifier is making music instead of heat.

To illustrate the difference between conventional and biamped operation, let's start with a conventional two-way speaker system with high-level crossover and one channel of a power amp. (You don't necessarily have to do the following, just read about how it works.) With the system connected as shown in Figure 2, send a low-frequency tone through the system. Note C below middle C on a piano will do nicely. It is about 126 Hz. Turn the level up until the amp clips. (It will sound just terrible!)

The harsh buzzsaw sound is amp clipping and all the loudspeaker system can do about it is to reproduce it the best it can: the fundamental (wanted) plus the harmonics which are produced by the clipping (unwanted). The clipped waveform is on its way to becoming a square wave, which is an infinite series of odd-order harmonics. If you have less clipping, you get less harmonics, which means the buzzsaw isn't as loud.

Something else happens when the amp clips. The average power delivered to the HF driver can be increased dramatically due to the addition of the harmonics. On a long-term basis, this can mean early diaphragm failure, and it costs money. Even though the tone is quite low, 126 Hz, you can get high-frequency driver damage because the tremendous number of harmonics produced can overpower the horn driver. When the amp clips, it doesn't care what is coming through, it clips everything.

Changing the input signal from a tone to music does little to change things, except the percentage of time the amp spends clipping. Because of the energy distribution in music, and the lower efficiency of the bass driver, the lower frequencies demand, and get, most of the amp's power. The high frequencies, which by their nature are faster transient spikes, don't need all that much continuous power. But added to the low frequency requirements, they can easily push the amp into clipping on peaks. For this you get harsh sound.

Recorded music usually has a peak-to-average ratio of at least 10 dB. Some program sources, like the unlimited signal from a mixing console, may have considerably more. The crash of a cymbal can easily generate transients 20 dB greater than the average signal level. For a 100-watt amp, a 10-dB peak-to-average ratio means that the average power delivered will be 20 watts, with peaks going to 200 watts. (You get 200 watts, instead of 100 watts, because the top of the sine wave signal used to rate a power amp is twice that of the average or "RMS" level.) The difference between the average level and peak clipping is called headroom. You want as much headroom as you can get.

Obviously, one way to get more headroom is to buy more watts – the most powerful amplifier you can get your hands on. But once you have it, and have reverently connected the monster to your speaker system with the power off (always off), you turn it on, sit back, and listen. You probably get a cleaner sound, and everything's fine until the amp clips because you turned the volume up too much, or worse yet, something happens in the line that sends a high-level transient (POP, SCREECH, HONK) through the system. What happens? Well, all of a sudden the poor HF driver, which may have only a 10-to-40-watt rating on peaks, sees a several-hundred-watt transient and does the only thing it possibly can do in this instance: BLOWS UP! The silence following is deafening. Later, when you can get around to it, the guy who recones your drivers takes your money and puts a little of it aside for his trip to Hawaii. And at $50 a throw, it won't take him long to get there.

We sort of got off the track, but there are really two points to all of this. Don't feed your drivers more than they can handle, and use some means of speaker protection. If you don't know the power rating of the speakers in your system, find out, and refer to "PA Bible" Addition No. 2 to refresh your understanding of power handling capacity.

So much for conventional operation; on to biamping. Connect a low-level crossover as shown in Figure 3. Hook your power amp into the crossover; channel 1 for the bass and channel 2 for the treble. Finally, connect the LF speaker or speakers to the bass channel and the HF speaker to the treble channel. Be certain the power is OFF while making the connections.

Okay, again put in the C below middle C on the piano and crank up the level the same as before. The sound stays relatively clean and just gets louder and louder. If the low-frequency amplifier clips, only the low-frequency speaker is fed harmonics instead of the high-frequency speaker, and it isn't very good at reproducing these harmonics. The high frequencies will be reproduced cleanly by the un-clipped high-frequency power amplifier, and the overall sound will be much cleaner. That's what you want, isn't it? Try **that** with a high-level crossover! No – don't do it. Save your money.

If you were now (in the same circumstances as above) to strike two notes on the piano, one low note and one high note, the combination would sound much better than the same signal through a high-level crossover system because even though the bass note may be clipped, the high note comes through clean and tends to mask it. In test setups where system components are operated near their design limits (that probably means **you**), biamp operation delivers a really transparent sound.

To sum up, there is minimal interaction between low-level crossover and amp, thus reducing distortion. The amp for the HF driver can be properly sized, thus minimizing the driver's exposure to high-level transients. The net result is a biamped system that can deliver somewhat higher sound pressure levels, lower distortion, and greater reliability. Yes, it may be somewhat more expensive in the short run, but compared to driver repair and downtime, it is a drop in the bucket. System expansion is easier – all you do is add more speakers and amps.

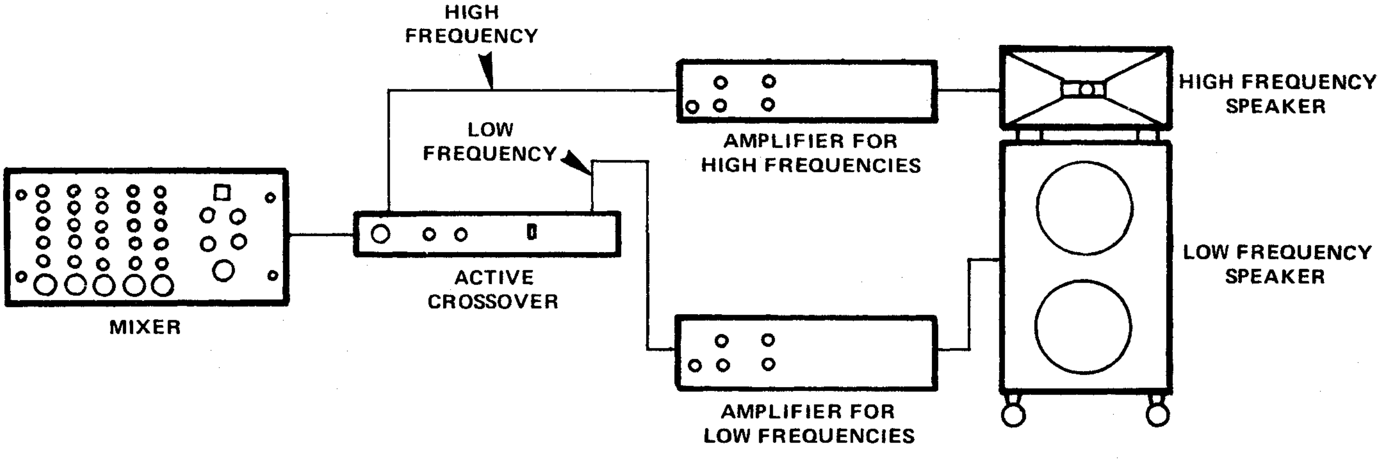

FIGURE 3 – Application of Active Crossover

POINTERS FOR SPEAKER SYSTEM BIAMPING

Many types of electronic components and speakers are available for constructing complete systems. Speakers should be picked to complement each other's characteristics. Amplifiers of the appropriate size for each speaker should be carefully chosen (see "PA Bible" Addition No. 2). The crossover should have crossover slopes suitable for the speakers being used, and should be electrically compatible with the other electronics in the system. Be certain the crossover will cover the crossover frequency you desire and beware of variable crossover dial markings so you don't accidentally set one for 100 Hz when you want 1,000 Hz.

The original "PA Bible" discussed initial balancing of high-frequency horn levels to low-frequency levels. Be certain to consult the HF driver manufacturer's specifications for the safe, permissible lowest crossover frequency or recommended crossover frequency. Higher than recommended crossover frequencies will not damage anything, but less than desirable results might be obtained.

Do not try to use an equalizer to make up for poor balance between highs and lows. This adjustment should be done at the crossover or power amplifier. Equalization is great for room tuning and performer enhancement, but over "EQ-ing" for a poor system will probably be so demanding that some part of the system will go to Hobart Tasmania for a long rest. This is particularly true of the very low and very high-frequency ranges (below and above system capability).

In a biamplified system high-frequency drivers end up being connected directly to the amplifier which drives them. This is great for most things, but if the amplifier has turn-on or turn-off transients, or if the amplifier ever fails, so might the drivers. Protective capacitors which will help minimize this problem are highly recommended. The manufacturer may be of help for type and value but if not, the following can be used. Non-polarized type capacitors (such as motor starting capacitors) with a voltage rating higher than the amplifier is capable of producing (70 volts to 120 volts) are recommended. The addition of the capacitor provides an additional 6 dB per octave roll-off and will also block any dc voltage that might appear. To calculate the appropriate value a frequency of one-half the crossover frequency should be used in the following formula so the protective roll-off will not affect normal operation. In the following formula "C" is the value of the capacitor in microfarads, "π" is 3.14, "f" is the frequency as discussed above and "Z" is the impedance of the driver $C = \dfrac{500000}{\pi \times f \times Z}$.

In passive, high-level crossovers this protection is an integral part of the crossover.

Be cautious when setting up so the low-frequency amplifiers aren't accidentally connected to the high-frequency transducers. The results are probably obvious. During initial sound check, turn the level up to a very **low** SPL, and listen to each speaker to be certain highs are coming from the place they should be and lows are coming from their associated speaker. (This is also a good thing to do, just to be certain all speakers are working.) If all is well, then proceed with your normal sound check.

CONNECTING THOUGHTS

By now you should be aware of the different types of crossovers and the role they play in making a speaker system function. Passive crossovers offer cost effectiveness while doing a good job whenever high-power inputs and multiple speakers are not being utilized. Biamping, on the other hand, with an active crossover, is desirable when multiple speakers and power amplifiers are used and is very flexible from a user standpoint. The choice is not always clear cut, and you must decide which will do the best job in your own application.

SPECIAL NOTE TO THE READER

The E-V "PA Bible" has been prepared to help you solve your PA problems. In addition, we have begun a series of supplements which will expand upon and add to the basic "Bible". Let us know if you have any specific subjects in mind for us to tackle. If you are reading a friend's copy of the "Bible" and would like to get your very own

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THE PA BIBLE

ADDITION NUMBER SIX
THE CON'·STANT DI·REC·TIV'I·TY
WHITE HORN WHITE PAPER

In prior "PA Bible" material we dealt with sound system design and various supplements have gone more thoroughly into specific related areas. In this addition, we will take a look at the unique class of horns known as "constant directivity" horns. This addition is complete in itself, but you may want to use Addition Number 1 – which discusses basic horn types and their function – as background.

"CONSTANT DIRECTIVITY" DEFINED

One way to get a handle on the concept of "constant directivity" is to consider the common garden hose and the nozzle at the end of it. Let's start with "directivity." Water pressure corresponds to the driver which produces the sound output. The nozzle corresponds to the horn which spreads – or "directs" – the sound about the listening area. At some settings, the nozzle spreads the water over a wide angle. A horn designed to do the same thing with sound would be said to have "low directivity," since it spreads the sound over a wide area. The nozzle can be readjusted to direct the water over a very narrow angle. The analogous horn would be "high directivity." (The water also goes farther – an important point that we will expand upon later.)

Other phrases are often used to describe how a horn directs sound. Since a low-directivity horn spreads sound over a wide area, it is said to have a "wide coverage angle," a "wide dispersion angle," or a "wide beamwidth." Such a horn is also said to be a "short throw" horn, since the driver's sound output is spread over a wide angle and, therefore, isn't "thrown" very far. A horn with *high* directivity would be a "*long* throw" horn, with a *narrow* coverage angle, dispersion angle, and beamwidth.

Well, directivity was easy, but what about the "constant" part? With the nozzle, there's no problem; you adjust the nozzle to direct the water where you want it. Ignoring the local wind conditions, that's pretty constant. Unfortunately, nearly every horn fails to behave like the nozzle. At low frequencies the coverage angle is one thing, like 80 degrees. At high frequencies it's another thing (usually narrower). At mid frequencies it can be something else again. In other words, most horns are *variable* directivity, nonwithstanding the claims of the advertising department to the contrary. A few designs are much more special. They behave like our nice nozzle! They have essentially *constant* directivity – or coverage angle, dispersion angle, or beamwidth – over a wide frequency range. The concept of constant direcitivity is new, powerful, and unique. Don't forget it!

The dramatic coverage differences between a constant directivity horn and a radial horn of conventional design are summarized in a visual way in Figures 1, 2, and 3. Figure 1 shows the basic comparison. Figure 2 shows in a more technical and detailed way how the coverage angle of a conventional radial horn changes with frequency. The angle changes a *lot!* The maufacturer calls his horn 60° x 40° but that's true only at a few frequencies – otherwise, it varies from here to Newport, Tennessee. In contrast, the constant directivity horn has an essentially uniform coverage angle over a wide frequency range. Take a look at Figure 3.

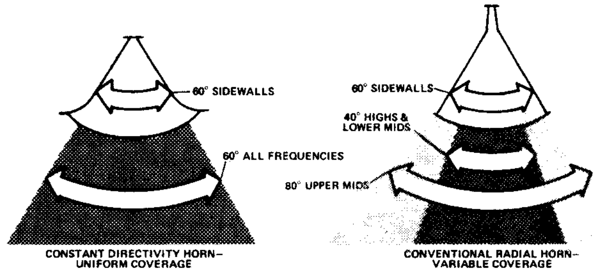

FIGURE 1 – Horizontal (Side-to-Side) Coverage Angle Comparison, Constant Directivity and Coventional Radial Horns

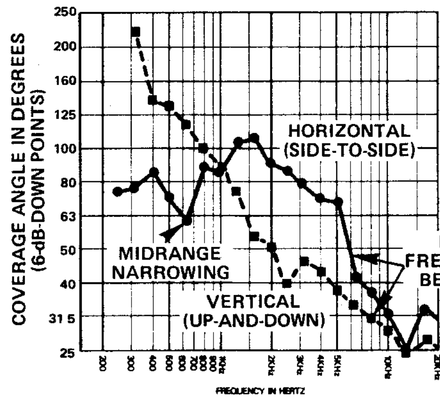

FIGURE 2 – Coverage Angle of a Conventional "60° x 40° Radial Horn

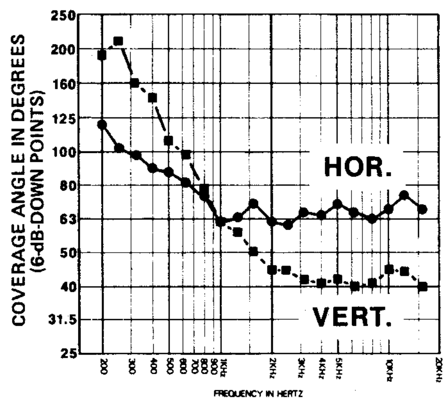

FIGURE 3 – Coverage Angle of 60° x 40° Constant Directivity Horn (E-V 6040A)

WHAT CONSTANT DIRECTIVITY HORNS DO FOR YOU

The advantages of constant directivity horns are fundamental:

1. **Constant directivity horns give a well defined zone of coverage that you can count on.** Every one in the audience within that zone will hear the full range and brilliance of the program. Dead spots and bright spots can be eliminated once and for all. This feature alone is very powerful.

2. **You can use fewer constant directivity horns.** No longer will you need to overlap conventional horns in an attempt to get uniform coverage at high frequencies. That saves money and van space. And you will have eliminated the "interference" which occurs when two or more horns are serving the same portion of the audience. The interference produces large dips and peaks in the direct field frequency response, which "dulls" the sound quality and tends to negate the very coverage you're aiming for.

WHAT MAKES A CONSTANT DIRECTIVITY HORN?

From twenty feet away all horns look alike. They are small where the driver gets attached and bigger at the other end. But if you put your glasses on you'll see that constant directivity horns look different. To make it easier to get specific, consider our HR6040A constant directivity horn which has a coverage angle of 60 degrees in the horizontal (left-to-right) direction and 40 degrees in the vertical (up-and-down) direction. Each special characteristic eliminates one of the problem areas of conventional variable directivity horn designs:

1. **A constant directivity horn is fed by a small opening.** This opening is usually, but not always, near the driver. The small opening assures that high-frequency sounds coming from the driver will be spread wide enough to fill, and be controlled by, the main section of the horn. Simply put, the laws of physics dictate that a *large* opening at the entrance to the main horn section *cannot* spread the highs to fill the main section. The problem is related to the principle of why the tweeters in you hi-fi speakers are much smaller than the midrange and woofer parts. For a look at a typical small feed section in a constant directivity horn, see Figure 4.

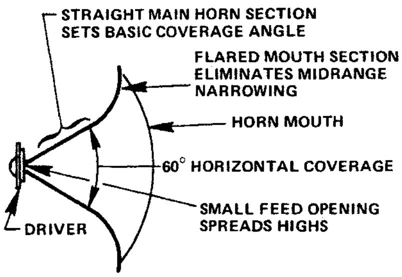

FIGURE 4 – Top Section View of Constant Directivity Horn (E-V HR6040A)

Most conventional horns have a long tapered "throat" that has a rather large opening by the time it reaches the main section of the horn. This means that the highest frequencies are *not* spread enough to be controlled by the main section and tend, instead, to beam straight ahead (the higher the frequency, the more straight ahead). See Figure 5 for a horn whose large throat opening cannot spread the highs.

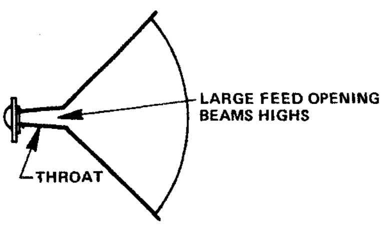

FIGURE 5 – Top Section of Conventional Radial Horn

2. **A constant directivity horn has straight sidewalls over a major portion of its length in both the horizontal and vertical planes.** These straight sections are basically responsible for establishing the horn's constant directivity over a wide frequency range. The particular coverage angle of the horn corresponds (nicely) to the angle between the two straight sides. Refer to Figure 4. The curved vertical section of a traditional exponential radial horn *cannot*, by definition, have constant directivity. However, there are many horns made this way and one is shown in Figure 6.

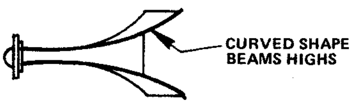

FIGURE 6 – Side Section of Conventional Radial Horn

3. **Constant directivity horns have an addtional wide-flare section near their mouth openings.** This flare is shown in Figure 4. The additional flare eliminates the substantial narrowing of midrange coverage angle that occurs in all horns of conventional design (refer back to Figure 2).

4. **Constant directivity horns are usually bigger than conventional horns.** This is for a good reason, not just because bigger looks impressive. To maintain constant directivity down to the typical crossover frequency (like 500 or 800 Hz), the mouth of the horn needs to be larger than has been customary in the past. A conventional 90° x 40° radial horn designed to load a driver down to 500 Hz might be, say, 8 inches high. On the other hand, the E-V HR9040A, which maintains its rated directivity over a wide frequency range, is about 17-½ inches high.

A DRIVER ON A CONSTANT DIRECTIVITY HORN NEEDS EQUALIZATION

The Newman Criteria for Drivers

We have seen that a constant directivity horn takes the output of the driver and spreads it around the room evenly. Great! But this stellar performance reveals an interesting and fundamental performance characteristic of *all* compression drivers that is not well known. If the output of the driver *itself* is isolated – unmodified by any particular horn to which it might be attached – it has anything but the "flat" response we've all been taught is good. An excellent real-world compression driver will convert about 30% of the amplifier's output into acoustic output up to about 3000 Hz. (In more technical terms, we would say that the driver's "midband efficiency" can be about 30%.) Above 3000 Hz, output falls at about 6 dB per octave. This curve shape is shown in Figure 7. Note that at 15,000 Hz, output is about 15 dB down. This corresponds to a mere 1% efficiency, similar to that of an inexpensive cone tweeter! We call the line shown in Figure 7 the "Newman Criteria," after our Chief Loudspeaker Engineer, Ray Newman, who postulated the criteria in the course of his driver development work. The Newman Criteria is an excellent standard against which to compare any compression driver or tweeter.

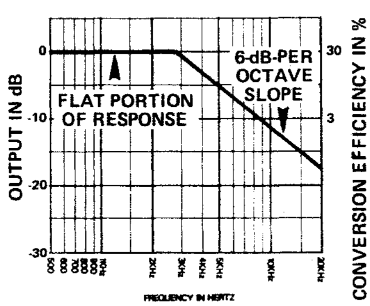

FIGURE 7 – "Newman Criteria" Total Output Reference Standard for Compression Drivers (No Horn)

The rolloff inherent in the Newman Criteria results because of the practical problems involved in making the mass of the moving parts (diaphragm and voice coil) any lower and/or the strength of the magnetic motor (magnet and related steel parts) any stronger. Really good drivers meet the Newman Criteria line over a substantial portion of the frequency range, occasionally exceeding it in certain parts of the range by a small margin. Unusual materials (such as beryllium diaphragms) can provide modest increases in high-frequency output but the resultant performance still basically follows the Newman Criteria. This situation is sometimes missed by enthusiastic advertising departments who find it convenient to confuse the output of the driver *itself* with the output as modified by a non-constant directivity horn. Other driver designs get "more highs" by a substantial (and typically unmentioned) sacrifice of midrange efficiency. Most vocal energy is in the midrange and – all other things being equal – a driver with low midband efficiency will have increased amplifier power requirements, a higher probability of blowout, and a reduced maximum sound pressure level capability. The output of a typical high-quality, wide-range driver with high midband efficiency is shown in Figure 8. The Newman Criteria is displayed for comparison.

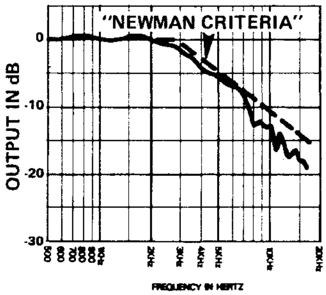

FIGURE 8 – Total Output of Typical Wide-Range Compression Driver (No Horn)

Horns Affect Driver Output

Now, let's get back to the horn. Figure 9 shows the response of the driver in Figure 8 mounted on a constant directivity horn (E-V HR9040A). Notice how closely the response matches the driver output shown in Figure 8. This makes sense because the horn, being constant directivity, spreads driver output over the same basic coverage zone over a wide frequency range. But the response curve certainly doesn't look too nice! No highs! You would probably say, "my brand X driver gives a nice flat frequency response curve a lot better than that." And you could be right. But you would have it on a horn (*not* constant directivity) like the one in Figure 2; and maybe the driver would be one of the low midband efficiency types to give the effect of more highs. The horn's collapsing high-frequency coverage angle makes the frequency response of the driver and horn *combination* look flat on the axis (in front) of the horn.

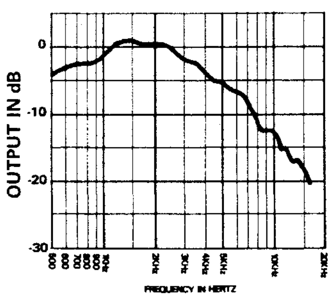

FIGURE 9 – Response Typical Wide-Range Driver on Constant Directivity Horn (E-V DH1506/ HR9040A)

Pick up that garden hose again. Of course, water flow doesn't have a frequency (high and low) like sound does, but it's fun to stretch the analogy a bit. Let's say a big high-pressure pump corresponds to the lower frequencies. We'll set the nozzle for a wide angle. Because we have lots of pressure we can adjust the nozzle for a wide angle of spray and still get the water as far as we want it to go. Now, let higher frequencies correspond to a low-pressure pump. If we leave the nozzle at the same spray angle, our water won't reach the flower bed any more. What to do? The only way to get the water to reach the flowers is to readjust the water for a narrow angle. Unfortunately, that gets water to the flowers but only to *some* of them! That's the plight of the old-fashioned variable directivity horn. It gets its nice flat response on axis only by depriving the listeners at the side of all those nice highs.

Utopia by Equalization!

The solution is to get a bigger pump for those high frequencies and put it on a constant directivity nozzle. Back in the real world of horns and drivers, that means putting the driver on a constant directivity horn and applying the appropriate *equalization* to get the flat response you want and need. "Equalization," you say. "That's a dirty word." What you need to understand is that the variable *coverage* of a conventional horn (which can be quite irregular) cannot be changed. It's fixed by the (hard) geometry of the horn. But the *response* of a driver on a constant directivity horn can be changed by appropriate equalization, easily. Read the last three sentences again and grab the concept before you go on!

By boosting or equalizing the high frequencies, a flat response can be obtained *and* at all angles of the stated coverage of the constant directivity horn. Then you'll have the open, transparent, super-clear sound you maybe thought you couldn't get from a horn. The effect of equalization on frequency response is shown in Figure 10.

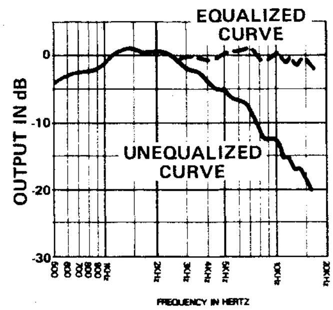

FIGURE 10 – Equalized and Unequalized Response of Wide-Range Driver on Constant Directivity Horn (E-V DH1506/ HR9040A)

Since the demand placed on a driver at high frequencies is not great compared to midrange and low frequencies (see "PA Bible" Addition Number 2, "Power Handling"), the boost required will not damage the driver or require excessive power from the amplifier in any practical situation. The Electro-Voice XEQ series of crossover/ equalizers has been specially designed to provide the proper boost to complement each of the seven E-V constant directivity horns. This leaves equalizers free to be used for enhancement or room tuning as desired (see "PA Bible" Addition Number 4, "Understanding Equalization and the Various Types of Equalizers").

DO CONSTANT DIRECTIVITY HORNS REQUIRE ANY OTHER SPECIAL EQUIPMENT?

What you will need besides the appropriate equalization depends entirely on your existing setup. If you are currently using a component system consisting of low-frequency (bass) enclosures, high-frequency horns, and a graphic equalizer (octave, half octave, or one-third octave), you probably have all the equipment you will need, including that required for horn-driver equalization. If you currently have an integrated speaker system (all speakers in a single enclosure), you will need some additional equipment. If you are designing a system for the first time, the recommendations in the "Basic Approach to System Design" section of the "PA Bible" will be of help.

The basic requirements for a two-way component sound system are shown in Figure 11. The low-level active crossover can be one of the E-V XEQ series, which also provides the high-frequency boost required, or a conventional unit which only provides the basic crossover function. If a conventional crossover is used, a graphic equalizer (such as the E-V/TAPCO C-201, 2202, and 2200) will be necessary to provide the proper high-frequency boost. (The engineering data sheet covering the E-V RC series of encased horns – which includes the DH1506 driver – shows recommended settings.) Even if the E-V XEQ crossover/ equalizer is used, a graphic equalizer is still useful for enhancement and room tuning as discussed in Addition Number 4 of the "PA Bible."

Driver-to-horn mounting arrangements vary. All Electro-Voice HR series constant directivity horns will accept most 1.3" diameter throat bolt-on drivers (including the E-V

DH1012). Using the ADH-1 adaptor allows most bolt-on drivers with 1" throats and all screw-on drivers with 1-3/8"-18 mounting threads (including the E-V DH1506) to be mated to the HR horns without any performance degradation. Some drivers have 2" throats and can only be mounted on horns or adaptors specifically designed for this largest throat size.

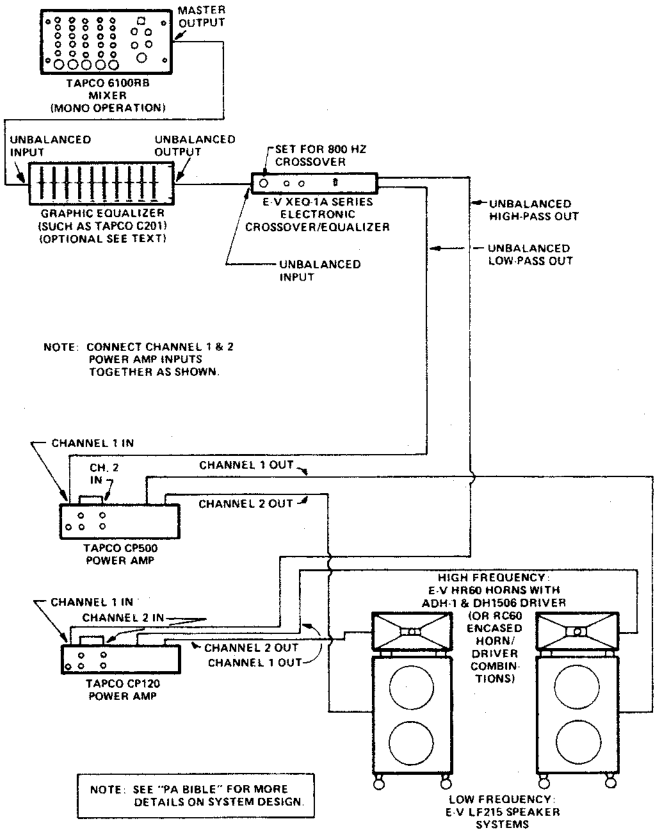

FIGURE 11 – Block Diagram of Typical Two-Way Component System

WHY ARE THERE A NUMBER OF DIFFERENT CONSTANT-DIRECTIVITY HORNS?

Coverage Angle

All-in-one PA speakers have what ever coverage angle they have, in the hopes that the specific performance is adequate for a relatively broad range of applications. When you step up to separate PA components, you can deal more accurately with the different physical situations you are likely to encounter. In the "PA Bible" we presented some typical examples and decided which horns would properly cover the listening area. Because listening areas vary in size and configuration, so should the speaker coverage angles. Constant directivity horns are usually available in three standard coverage patterns: 90° x 40°, 60° x 40°, and 40° x 20°. At least one other available constant directivity horn has 120° x 40° coverage. These coverage patterns provide a useful range to select from for most applications. Determining the proper coverage is the first step in deciding which constant directivity horn will do the job.

The second step is to know if long-throw horns will be required. Reviewing the section on "Room Reverberation Swamps Your Voice" from the "PA Bible" will be helpful here. In highly reverberant rooms, low-directivity, wide-angle horns (like 90° x 40° or 120° x 40°) will not provide enough "direct" sound (that which comes right from the horn and has not been reflected by one or more of the room surfaces) for listening positions past the approximate mid-point of the room. Sound past this point is likely to be mostly "reverberant" (reflected by one or more of the room surfaces) and, therefore, unintelligible. The solution is to get more direct sound in the back of the room (without blasting the high-paying seats in the front of the room) by aiming a high-directivity, narrow-angle horn (such as the E-V HR40 or HR4020A 40° x 20°) at the back.

In most clubs of small-to-medium size, under normal conditions, one type of constant directivity horn will satisfy your requirements. A 90° x 40° or 60° x 40° horn can usually do an effective job for these environments.

Minimum Crossover Frequency

The different constant directivity horns available "load" their associated drivers down to various "mimimum crossover frequencies," typically 500 or 800 Hz. For example, the "large" E-V HR4020A 40° x 20° loads down to 300 Hz. The other "large" HR's (HR6040A and HR9040A) load down to 400 Hz. The "small" HR's (HR40, HR60, HR90, and HR120) load down to 500 Hz. Each HR may be used down to its stated limit, as long as the driver itself can sustain full rated power down to that frequency. (For example, the E-V DH1012 can sustain full rated power down to 400 Hz; the DH1506 sustains full power down to 800 Hz.)

Minimum Frequency for Coverage Angle Control

At some low frequency, chosen by the horn designer, all constant directivity horns lose their coverage control and the coverage angle begins to widen. (Figure 3 shows this phenomenon.) Horns with larger mouth sizes maintain their rated coverage angle to lower frequencies. Horns which maintain coverage to low crossover frequencies (like 500 Hz) are very large, say 30" x 30" at the mouth. Often, compromises in maintaining the rated coverage angles are desirable, due to practical size and cost considerations. For example, compare the "large" E-V HR9040A (approximately 18" x 39" x 22" hwd) to the "small" E-V HR90 (11" x 24" x 14" hwd), both nominal 90° x 40° horns. Both horns are wide enough to maintain horizontal (side-to-side) coverage down to 500 Hz and below. The smaller vertical dimension of the HR90, however, limits its strict vertical (up-and-down) control to about 2000 Hz, versus the larger HR9040A's 1000 Hz. It is also good to keep in mind that complex multi-way speaker arrays which use high crossover frequencies (say, 1500 Hz and above) can use a relatively small constant directivity horn. For example, the large HR9040A provides no performance advantage over the small HR90 if crossed over in the 1000/2000-Hz range.

SPECIAL NOTE TO THE READER

The E-V "PA Bible" and this addition have been prepared to help you solve your PA problems. Let us know if you have any other areas in mind for us to tackle.

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THE PA BIBLE

ADDITION NUMBER FIVE
SYSTEM INTERCONNECTION

INTERCONNECTING THE SYSTEM

The importance of proper interconnection of components in a system can hardly be over-emphasized. Not only must the wire or cable be an appropriate type, but the compatibility of component input and output parameters must be considered. In this supplement we will discuss such parameters as signal level, impedance, and balanced versus unbalanced terminations. Our approach is to begin with the lowest electrical signal level and follow it through the line level stages to the power amplifier and speaker circuits. A short glossary of definitions appears at the end of this supplement to assist in reading this material.

LOW LEVEL CONNECTIONS

Microphone level circuits are potentially the most prone to hum pick up and noise interference because of the low signal voltages involved. Therefore, microphone cables and connectors must be shielded to prevent electrostatic pickup. Also, dynamic microphones, because they use a coil of wire, are sensitive to external magnetic fields, such as those produced by transformers, motors, fluorescent lamp ballasts, and high current wiring. If operation in the vicinity of such devices is anticipated, a model with a hum bucking coil or special shielding should be used. Magnetic phonograph cartridges, guitar and piano pickups also use coils of wire and may require physical separation or shielding to prevent hum pickup.

Attenuators

Although the voltage levels in low impedance microphone circuits often run in a low millivolt or microvolt region, it is possible to encounter substantially higher levels, even exceeding one volt, when a high output microphone is used in extremely loud sound fields. This might occur, for example, when a condenser microphone is used for close pickup of a rock vocal or trumpet soloist. The high level microphone output may overload the input of the microphone mixer or preamplifier. To eliminate this problem, a built-in or external attenuator can be used. These attenuators usually allow a selection of 10, 20, 30 or even 40 dB signal reduction. It should be noted that some external attenuators cannot be used with a phantom powered microphone. See Figure 1.

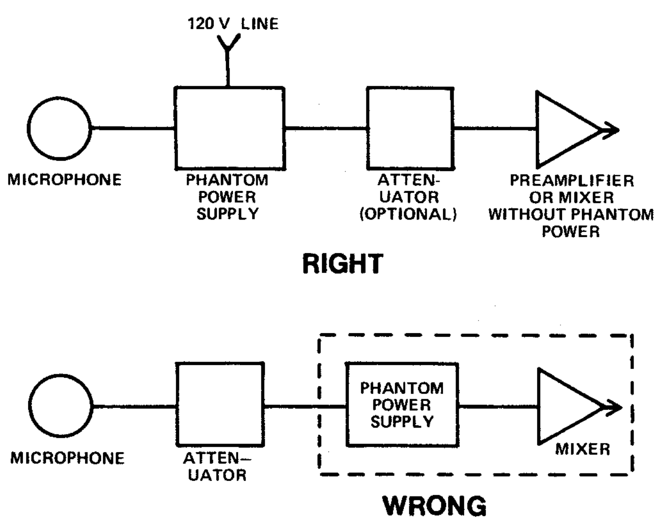


FIGURE 1 — Phantom Powering

Input overload creates raspy distortion when severe, but it is more difficult to detect if marginal, especially under live P.A. conditions. To assist in determining overload, some equipment is provided with LED input overload indicators. When using such equipment, simply insert enough attenuation so that the LED seldom flashes under the loudest signal conditions.

Impedance

Modern microphones can be segregated into two general classifications: high and low impedance types. High-Z microphone impedances lie in the range of 10,000 to 40,000 ohms, and are typically around 20,000 ohms. Low-Z values range from 50 to 500 ohms, and are frequently a nominal 150 ohms. Because transformers with very low DC resistance are often an integral part of the microphone, it is not meaningful to attempt to measure microphone internal impedance with an ohmmeter.

An exact match between microphone impedance and its associated equipment is not necessary. However, high-Z and low-Z microphones must be connected to separate inputs; and, where required, matching transformers are available to convert from high to low, or low to high. Such transformers may be plug-in options provided on the electronics, or be add-on, in-cable types. To minimize hum pickup, these transformers must be designed with adequate magnetic shielding, and be positioned well away from motors, power transformers, and other equipment generating large AC magnetic fields. High-Z microphones have become less popular in recent years because cable lengths greater than ten or fifteen feet cause a reduction of high frequency response; and, in addition, high-Z connections have somewhat greater susceptibility to hum and noise pickup, even though a good grade of low capacitance, single conductor, shielded cable is used. To avoid ground-loop noise and hum, the housing and stand of a high-Z microphone should not be allowed to come into contact with other grounded items (conduit, metal flooring, other equipment, etc.).

Low impedance microphones are recommended for all permanent installations and wherever cable lengths must exceed fifteen feet (see Chart I). Two conductor shielded cable is required for the hookup. The shield, which totally surrounds the signal conductors, may consist of a thin, metallized foil and "drain" wire running the length of the cable, or alternately, a braided mesh woven of fine wires may be used. The braided construction has a longer flex life, so is preferred in applications involving frequent cable movement. Avoid the use of a widely spaced mesh or a spirally wrapped shield, since they are more prone to interference pickup, particularly SCR hash from lighting circuits. Stretchable coil cords, such as those used on guitar pickups, are frequent offenders in this respect.

Microphone Impedance (ohms)	Cable Length (feet) Causing 1 db Loss at 10 KHz
50	920
150	310
250	190

CHART 1 — Cable Capacity Loading

Phantom Connection

Some types of microphones require power to operate internal electronic circuitry. Such power may be supplied by integral or external batteries, or it may be provided from the following mixer or preamp by feeding the required DC back over the same cable conductors which carry the signal (see Figure 2). This is called a "phantom" connection. If required by the microphone, and no provision is made in the following equipment, a separate phantom power supply module may be placed in series with the mic cable. Although the balanced nature of the phantom supply minimizes interaction (and noise) between the DC and signal, a severe transient can be generated during insertion and removal of connectors in the circuit, so this should not be attempted on a live channel.

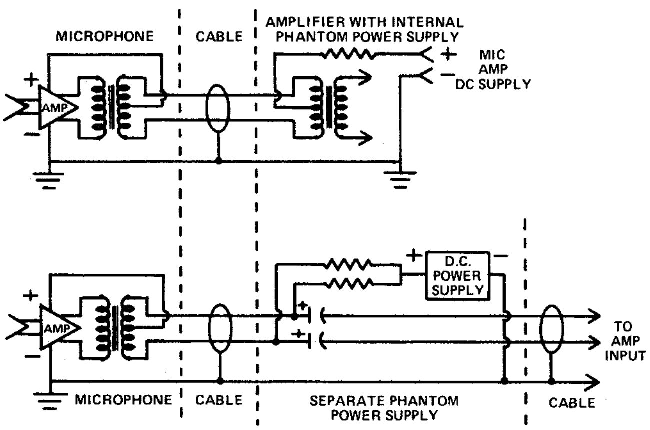


FIGURE 2 — Phantom Connections

Connectors

Usually, 1/4 inch phone plugs are used for high-Z microphone connections, and 3-pin XL style connectors are used for low impedance balanced lines. By convention, the cable ground braid, or shield, is always connected to pin 1 or to the phone plug shell. On XL connections, pin 2 is the in-phase terminal (positive pressure produces positive voltage on pin 2), in accordance with EIA standard RS-221-A. These same connectors are also popular for line level use, and the same comments on polarity apply. Figure 3 illustrates some of the most popular connectors suitable for low level and line level connections.

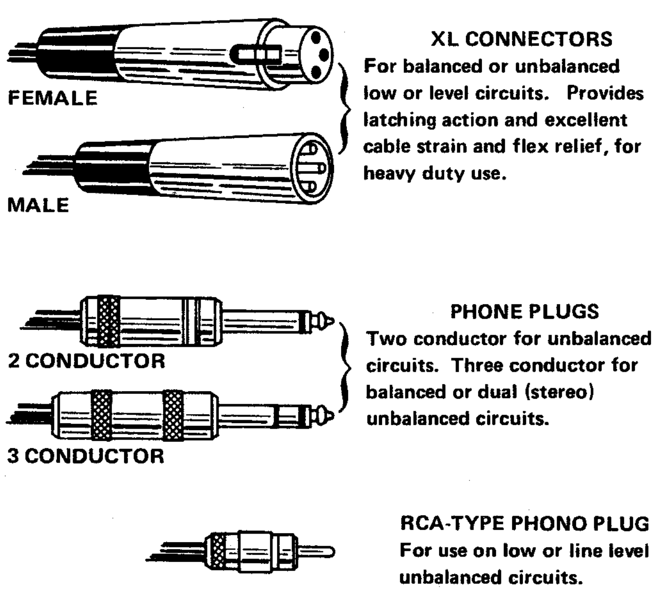


FIGURE 3 — Industry Standard Connectors

LINE LEVEL CONNECTIONS

The output of a typical microphone mixer usually provides a signal level on the order of one volt, which is considered to be "line level." If such equipment has VU meters, they are often calibrated so that zero VU reading corresponds to an output level of +4 dBm (decibels relative to one milliwatt).

Frequently, several pieces of line level equipment may be interconnected to yield the desired system performance. For example, a microphone mixer, equalizer and low level crossover may all be connected in series. Often these units are designed with a gain of unity between their input and output and have compatiable level requirements. This is not always true and it is essential that the equipment specifications be checked to assure compatible input and output levels.

Another consideration regarding compatibility of units relates to input and output impedances. It is particularly important to connect each piece of equipment to its recommended *load impedance.* Recommended load impedance, as its name implies, is that value which the device should "see" when looking forward into the circuit it is driving. Usually a *minimum* recommended load will be specified, meaning that the unit will operate properly as long as it sees a load of any impedance above this limit.

The internal impedance of an active device is not necessarily related to its load. It is that value which we would "see" looking back into the output terminals of the device. Internal impedance tells us, among other things, how much the gain will vary with changes in the external load. As stated above, active line level devices do not operate properly below a minimum *load impedance,* a value often above the device's internal impedance.

Balanced Versus Unbalanced

Another consideration when interconnecting electronic components is whether their inputs and outputs are balanced

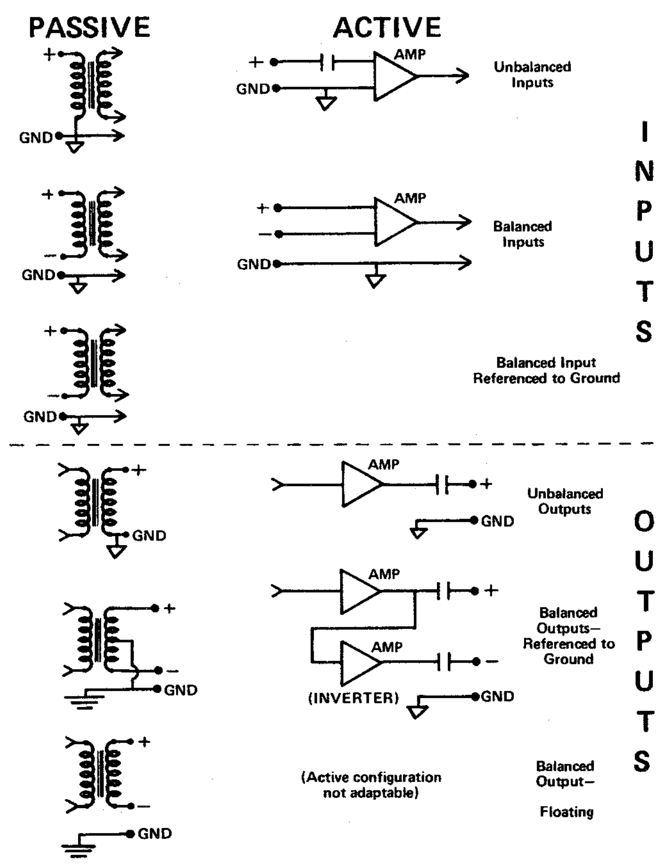


FIGURE 4 — Balanced Versus Unbalanced Connections

or unbalanced with respect to ground. A balanced system is usually preferred, particularly in more complex and sophisticated systems, due primarily to its superior freedom from interference and avoidance of ground loops. This is achieved by separating the signal and its return from the ground and shielding both paths. Figure 4 shows typical input and ouput variations.

Unbalanced systems utilize the ground for a signal return and require only two terminals, variously referred to as "hot" and "ground", "high" and "low", or "+" and "–". Balanced connections require three terminals, with ground being separate from the high and low. In preparing balanced cables, it is wise to make an ohmmeter check to insure that neither signal lead has been inadvertently shorted to the cable shield or the shell of either connector. The positive and negative terminology often used on balanced units refers to the polarity of a signal at any instant, allowing the user to tell if a signal is being reversed in phase in the equipment. This becomes important in multiple channel systems where the same phasing between reproducers must be maintained.

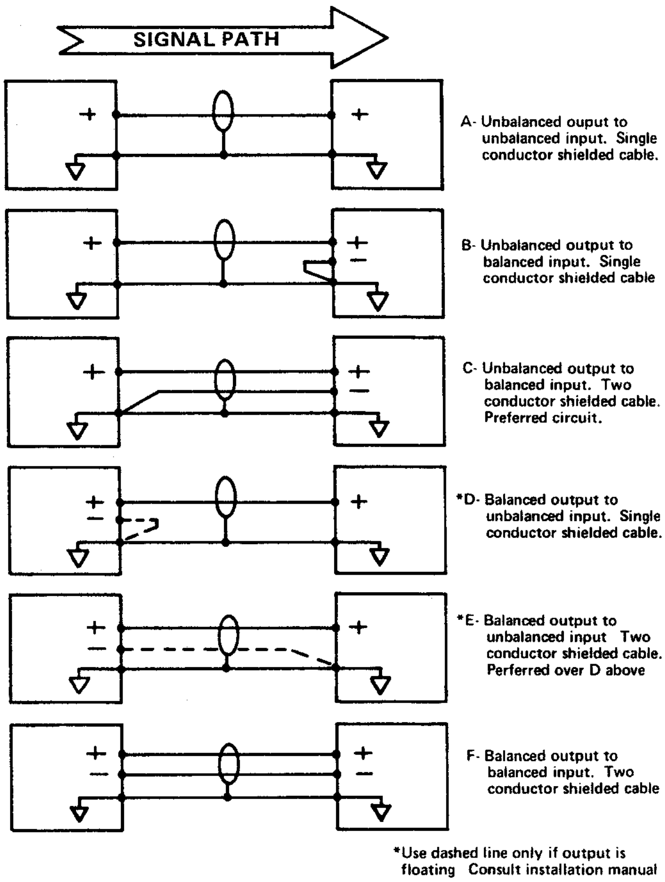


FIGURE 5 – Interconnecting Circuits

Some equipment is designed to allow either balanced or unbalanced operation. In other instances, transformers may be added as an outboard module or as a plug-in option. Figure 5 illustrates some typical connections. If hum is introduced using circuit B (as when the two pieces of equipment are separated by a long distance), circuit C can be used to avoid noise from ground currents in the shield. Similarly, circuit E may be used to advantage, but only if the output is not referenced to ground (for example, an ungrounded transformer secondary). Note that in circuits C and E, the "low" signal wire and the shield braid must be tied together only at one end, as shown. Otherwise, ground and signal currents will be combined in the signal path return, defeating the advantage of these circuits.

Circuit F, showing a completely balanced input and output provides the best interconnection and should always be selected, if compatible with the equipment involved.

It is desirable to place all line level equipment at one physical location, such as in one or more adjacent equipment racks, thus minimizing connecting cable lengths and providing good grounding. This is helpful in minimizing interference such as SCR hash and RFI (see "Definitions").

One or two-conductor (as required) shielded cable should normally be used for all line level interconnections. However, because the voltage level is typically 40 dB greater than in a low-Z microphone circuit, noise pickup is somewhat less troublesome. In fact, unshielded terminal strips can be employed as connectors, particularly when fed from a low internal impedance device. The presence of large grounded areas of metal in the vicinity, such as the chassis, adjacent equipment, equipment racks, etc., also help to minimize any electrostatic interference pickup by exposed signal terminals.

Interference Pickup

Interference from the outside enviroment (as opposed to hum and hiss generated within the equipment) may include hum (usually picked up from nearby power wiring) and radio frequency interference such as AM or FM broadcasting, citizens band transmitters, television sync buzz, or X-ray diathermy equipment. Also, it may include the sharp buzz generated by nearby lighting control dimmers, i.e. SCR hash.

A system may pick up interference in three ways: 1) electrostatic coupling, 2) electromagnetic induction, and 3) ground loop conduction.

Two conductors in space exhibit an electrical capacitance between them, which increases as their area becomes greater or as they move closer together. Electrostatic pickup occurs by means of this capacitance (capacity coupling, if you will). It can be largely eliminated by placing a grounded metallic shield between them, or around either one.

When two insulated wires are run next to each other for a distance, and an AC current is caused to flow through one of them, an AC voltage will appear across the ends of the other wire. This effect is called electromagnetic induction. The amount of energy transferred is a function of the mutual inductance between the wires, which increases with closer spacing between wires or with greater length. When the wires are wound into coils, the mutual inductance is further increased. This is why signal transformers are so susceptable to external magnetic fields. To reduce the coupling into (or from) a wire or coil, a magnetic shield must be employed. Such a shield is made of a magnetic material such as soft iron, mu-metal, or a number of special alloys manufactured for the purpose. The shield should enclose the part, forming a closed cylinder or box around it. If this magnetic shield is grounded, it may also serve as an electrostatic shield at the same time. Magnetic shielding is ordinarily required around all microphone and line level signal transformers. Even so, it may be necessary to separate them from equipment producing high magnetic hum fields. If such a problem exists in an equipment rack, try increasing the spacing between low level and high power equipment. Even five inches can often make a substantial improvement. For the same reason, avoid running signal and power cables in the same bundle or conduit.

System Grounding

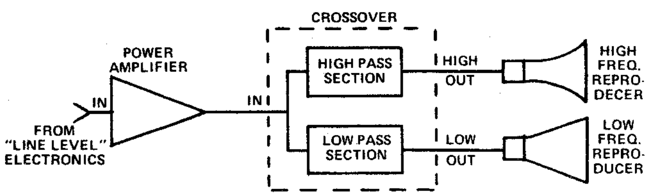
The term "ground" refers to conductors with a common potential. Usually, it refers to any connection to the chassis or external metal cabinet of a piece of equipment, with the ground terminals on each piece of equipment tied together forming a common "system ground." An "earth" ground is the zero or reference potential of moist earth as measured in a system of conductors buried in the earth. The metal pipes of a water supply system usually provide an excellent earth ground.

The ground, or third, wire of a power line also provides an approximation of an earth ground. However, due to ground currents flowing through the finite resistance of such wires, no two points are usually at exactly the same potential. For this reason, ideally, the system ground should be tied to an earth ground only at one point. In an enviroment where strong RF signals are present, additional care in grounding may prove beneficial. The sequence in which these connections are made can be important in preventing the formation of a ground loop, which may introduce hum and noise into the system.

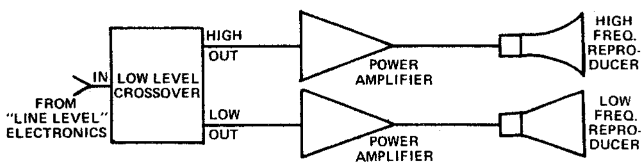
Various safety requirements mandate that most professional equipment be supplied with a three wire power line cord. This connects each of the various chassis to the power line ground, and largely eliminates the possibility of incurring a potentially lethal shock when touching a defective piece of equipment and a true ground at the same time. Unfortunately, it can also complicate the elimination of ground loops. This subject becomes quite involved, and will be discussed in a later supplement.

Crossovers

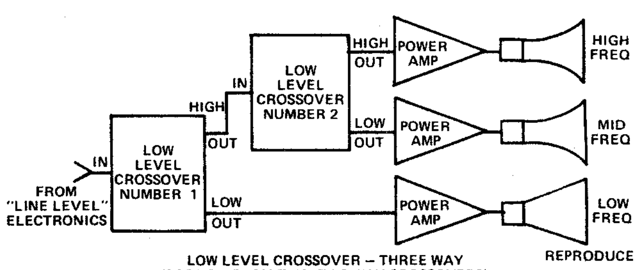
It is sometimes advantageous to divide an electrical signal into two (or more) frequency bands and connect each band to a transducer optimized for a limited frequency range. The device which divides the signal into these appropriate



HIGH LEVEL, TWO-WAY, PASSIVE CROSSOVER



LOW LEVEL CROSSOVER – TWO WAY



LOW LEVEL CROSSOVER – THREE WAY
(FORMED FROM TWO, TWO-WAY CROSSOVERS)

FIGURE 6 — Crossovers

frequency bands is called a crossover. A crossover may be connected between the power amplifier and reproducers as shown in Figure 6. Passive circuitry, consisting chiefly of capacitors and inductors, is employed in a high level crossover, so named because of its circuit location following the power amplifier. In some instances, it may be built into the transducer or into the system enclosure.

Low level crossovers, on the other hand, may be either active or passive, and are wired as shown in Figure 6. The low level crossover system offers a number of advantages over high level crossovers, but requires additional power amplifiers. The term "low level" indicates a crossover operating at line level in front of the power amplifiers.

POWER AMPLIFIER CONNECTIONS

The signal input to most present day power amplifiers is considered to be line level, requiring in the region of 1/2 to 2 volts to drive the unit to its full output. Power amplifier output levels are much greater than those at line level; and, in addition, the impedance is considerably lower. These factors combine to eliminate the need for shielding of the output circuit wiring. To avoid excessive power losses, however, a much heavier gauge of wiring must be employed.

Two distribution methods are in use to deliver the audio power from the amplifier to the speakers. Where it is necessary to adjust the comparative loudness level of a number of speakers, a 70 volt line distribution system is popular. Let us first consider the other method, however, in which a direct connection from amplifier to voice coil is used. Since the impedance of a horn driver or speaker is typically only eight ohms, any small resistance (impedance) in the wiring, which is effectively in series with the load, will result in a voltage drop and hence a power loss. Unlike most transducers, the impedance of this wiring is almost identical with its DC resistance, so these terms may be used some-what interchangeably. Also, readings from an accurate low-range ohmmeter are valid for checking losses. To measure wiring resistance, remove system power, place a heavy jumper wire across the load, and after disconnecting at least one lead from the amplifier output, measure the resistance at the amplifier end of the wiring. As an extreme example, wiring having a 3.3 ohm resistance, driving an 8 ohm load, will rob half (3 dB) of the amplifier power output. This power loss shows up as a slight heating of the wires. A more reasonable design would allow a power loss of 1/2 dB (approximately 12%). Total resistance with an 8 ohm load would then be held to less than 1/2 ohm.

Wiring resistance should vary with load impedance, with a 4 ohm circuit requiring heavier conductors, and with proportionately relaxed requirements for 16 ohms. A wire table (Chart 2) tells us that for a distance of forty-eight feet between amplifier and a 4 ohm load, 14 gauge wire is required. This wire may be either solid or stranded copper, depending upon flexibility requirements. Of course, halving the distance will also cut resistance in half, just as doubling it will increase resistance by a factor of two. In calculating resistance using Chart 2, do not forget that since there are two wires in the circuit, the total wire length is twice the distance between the components. For permanent installations, 12-2 or 14-2 house wiring is popular, while the flexibility of 14 or 16 gauge "zip cord" is favored for portable setups.

Obviously, the terminals used in these low impedance circuits must not contribute appreciable resistance. For permanent installations, solder lugs, binding posts, and heavy screw terminals are ideal. For portable equipment, banana plugs and phone plugs can also be used.

AWG Size	Resistance (Ohms/1000 Feet)	Low-Impedance			High-Impedance Systems				
		4 Ω	8 Ω	16 Ω	100 W/70.7V 12½ W/25V (50 Ω)	50 W/70.7V 6¼ W/25V (100 Ω)	25 W/70.7V 3 1/8 W/25V (200 Ω)	5 W/70.7V 5/8 W/25V (1000 Ω)	1 W/70.7V 1/8 W/25V (5000 Ω)
10	1.00	120	240	480	1,500	3,000	6,000	30,000	150,000
12	1.59	75	150	300	940	1,800	3,800	18,000	94,000
14	2.50	48	96	190	600	1,200	2,400	12,000	60,000
16	4.02	30	60	90	370	740	1,500	7,400	37,000
18	6.39	19	38	76	230	460	920	4,600	23,000
20	10.1	12	24	48	150	300	600	3,000	15,000
22	16.2	7	14	28	93	190	380	1,900	9,300

CHART 2 – 2-Wire Copper Cable Lengths For 0.5 dB Loss in SPL

Load Impedance

Unlike microphone circuits, a power amplifier should be matched to its rated load impedance by a factor of two to one, or closer. Too high a load impedance will simply reduce the amount of power which the amplifier can supply. Too low an impedance may not only reduce available power, but may cause protective devices (fuses, thermal cut-offs, etc.) to open, interrupting system operation. Increased distortion can also result. Some amplifiers provide multiple taps to match a variety of load impedances, while others are designed for only one, or a narrow range of values. The impedance of a typical transducer varies widely with respect to frequency, but the nominal rating is usually close to the minimum value obtained within its normal operating range. When more than one transducer is to be driven from a single amplifier, the effective load impedance may be altered, as shown in Figure 7. Parallel connection of two eight ohm transducers would match a four ohm amplifier output, for example. The series connection should be used only with identical units, which are to be driven at equal power levels.

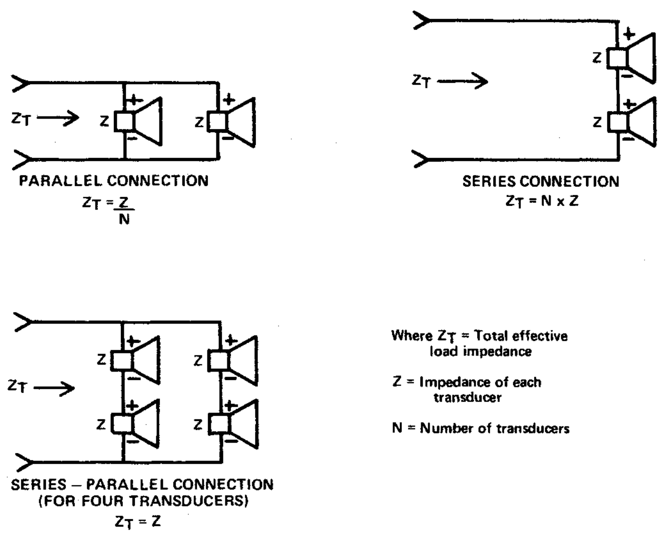


FIGURE 7 – Loudspeaker Connections

Polarity

It is important that the loudspeaker systems in a multiple array be connected in phase with each other. If one or more units is reversed in polarity from the others, frequency response, polar pattern, and/or sound levels may be adversely affected.

Output Impedance Matching Transformers

Some installations require a very large number of speakers to be operated at a relatively low level. In such applications, the 70 volt speaker line can be utilized to advantage, and is connected as shown in Figure 8. The amplifier is designed so that it will deliver 70 volts (RMS) of signal when adjusted for full power output. A transformer, with multiple taps to adjust level, is associated with each speaker. Often, these taps are labeled directly in watts corresponding to maximum amplifier output (70 V RMS). In this way, the level of each speaker may be independently adjusted. Each transformer need handle only the power drawn by its associated speaker. There is no limit to the number of speakers which can be so connected, as long as the sum of their power settings does not exceed the amplifier's rating. Not all power amplifiers operate satisfactorily into a transformer load. Therefore, unless a 70 volt output is specifically provided, the manufacturer should be consulted as to its suitability for this application.

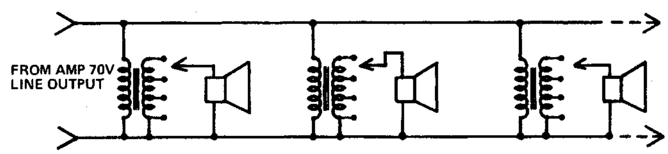


FIGURE 8 – Typical 70 Volt Line Connection

Decibel (dB)
Fundamentally, a unit of loudness. However, it is convenient to express *ratios* of electrical voltage, current, or power in terms of dB. For example, a 6 dB voltage increase to a speaker voice coil would produce a 6 dB increase in sound level, regardless of the starting point. A 3 dB increase represents the doubling of power, and a 6 dB increase, a doubling of voltage. Voltage gain of equipment is often expressed in dB.

dBv
The electrical voltage level compared to a one volt reference level. Thus, –6 dBv corresponds to 1/2 volt, +6 dBv to 2 volts, or +12 dBv to 4 volts.

dBm
The electrical power level compared to a reference level of one milliwatt. If dBm is used to indicate a voltage, the circuit impedance must be stated or understood. For example, in a 600 ohm circuit, 1 mw = 0 dB = .775 volts. Thus, at 600 ohms, a level expressed in dBm is always 2.2 dB greater than if expressed in dBv.

Active Devices
Devices requiring operating power (battery or other) in addition to the signal. Usually contain transistors, tubes, or IC's. Includes amplifiers, mixers, equalizers, etc.

Passive Devices
Devices requiring only signal power, and containing only resistors, capacitors, transformers, etc. Includes dynamic microphones, high level crossovers, speakers, etc.

Impedance
In an alternating current circuit, the ratio of the voltage to the resulting current which it causes to flow. The counterpart of resistance in a DC circuit.

Ground Loop
A condition existing when components in a system are tied to each other or to ground with more than the minimum number of wires to accomplish the connection. The duplicate paths can form a loop, which may allow circulating interference currents to flow, resulting in possible hum and noise.

RFI
Radio Frequency Interference. High frequency signals may enter at various points in the PA system. They include AM and FM broadcasts, television (continuous buzz), citizens band, nearby radar, and diathermy.

SCR Hash
Silicon controlled rectifiers, triacs, and various other semiconductors are finding increased application in light dimming and motor speed control. They generate extremely sharp wavefronts which, like RFI, can result in audible background buzz in sound systems if proper precautions are not exercised.

Signal
An AC voltage or current, representing the desired input to the system. The signal is amplified and conditioned by various system components, and appears at the output.

NOTE TO THE READER
We hope this addition of the "PA Bible" has shed some light on the diverse subject of interconnections. We appreciate receiving your comments and suggestions. Send them to

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EV THE PA BIBLE

ADDITION NUMBER FOUR UNDERSTANDING EQUALIZATION AND THE VARIOUS TYPES OF EQUALIZERS

Material for this addition was supplied by Larry Blakely, a well known audio and recording industry consultant and writer.

Larry recently completed a book on high-fidelity equipment, and has written articles for Modern Recording, Stereo Buyers Guide, Hi-Fi Buyers Guide, Tape Deck, Recording Engineer/ Producer, Modern Recording Buyers Guide, Studio Sound and Billboard. Larry is the co-founder and current president of the Creative Audio and Music Electronics Organization, and is on the Board of Industry Presidents of AMC and NAMM, and a member of AES. He is exceptionally well qualified to discuss the subject of equalization, having participated in the design, marketing, and application of equalizers for recording studios and for the live entertainment industry.

In prior P.A. Bible material we have discussed the general topic of sound systems with emphasis on certain topics relating to loudspeakers and microphones. In this addition we will cover the various types of equalizers, their operational characteristics, equalizing for enhancement, and system equalizing.

It is not our intention to discuss applications in great detail, especially the fine art of room equalization. We do want to acquaint you with certain basic concepts and guidelines that will enable you to jump in and learn by experience.

INTRODUCTION

There are many different types of equalizers that are used for a wide variety of applications. Equalizers will range from simple "tone controls" (bass and treble) which you will find on almost every record player or hi-fi system to the more sophisticated types of equalizers, such as the graphic or parametric. Equalization can place a great deal of flexibility at your fingertips. When used properly equalizers are excellent tools, but they are often misused and provide less than desirable results.

To help one determine if he needs an equalizer, as well as which type he should obtain, it is very helpful to have an overview of equalizers: how they work, the various types available, which types are best suited for various applications, and some simple guidelines for their use. For those who already understand this basic information, this will serve as a review, while giving the un-informed reader new knowledge that will greatly aid him in intelligently choosing and using equalization.

If you are using audio equipment and desire to hear something louder, you turn up a level (volume) control. When such a control is adjusted for a louder level, the level of all audible frequencies (entire audible frequency spectrum) is increased by the same amount. This simple level control is no respecter of frequencies; turn it up and the result is an audible increase in overall loudness (over the entire audible frequency range). However, when you turn up the "bass control" on a record player, only the low frequencies are increased in level, which will usually make the sound richer and fuller. When you turn down the "bass control", low frequencies are decreased, and the sound is usually thinner or even tinny. When you turn up the "treble control", high frequencies are increased, and the sound becomes bright or brilliant; turn it down and the sound will be dull and sometimes even muddy. These common bass and treble controls are "equalizers". In simple terms, an equalizer is a volume control (to increase or decrease level) for specific frequency ranges. To further expand our understanding, let us take a close look at frequencies and the audible frequency spectrum.

FREQUENCIES

Sound is made up of vibrations. These vibrations are referred to as "cycles". The number of these vibrations (cycles) that take place in one second are referred to as "cycles per second". One vibration (cycle) in one second would be called one cycle per second. 15,000 vibrations in one second would be 15,000 cycles per second. The number of cycles per second determine the actual frequency (pitch) of any audible sound. For those who are musicians, concert "A" is 440 cps (cycles per second).

For many years the term "cps" (cycles per second) was commonly used in the music and audio industries. In recent years the term "cycle" was substituted for another term called "Hertz", abbreviated "Hz". Hertz (Hz) is the unit of measure of frequency (cycles per second). One Hertz is one cycle per second. The range of human hearing is usually considered to be from 20 to 20,000 Hz for a person with very good hearing. When frequencies are in the thousands of Hertz, they are often indicated with a "k" (designating thousand) prior to the symbol Hz. Therefore, 20,000 Hz would be indicated as 20 kHz.

OCTAVES

As mentioned earlier, concert "A" is 440 Hz. If a musical note one octave higher were played, it would be "double" that frequency or 880 Hz. If a note were played one octave below 440 Hz, it would be "half" of that frequency or 220 Hz. Therefore, double the number of any frequency and you have one octave above; halve the number of any frequency (divide by 2) and you have one octave below. Much of music is generally referred to in octaves or portions of an octave.

FUNDAMENTALS AND HARMONICS

Musical sounds are made up of "fundamentals and harmonics". The fundamental frequency will sometimes (but not always) be the most powerful member of the harmonic series of a given instrument. The harmonics are multiples of the fundamental frequency. The 1st harmonic is the fundamental frequency, the 2nd harmonic is two times the fundamental frequency, 3rd harmonic is three times, etc. The harmonic structure of a musical sound has a great deal to do with its perceived qualities and/or texture (timbre). A typical relationship of fundamental to harmonic energy is shown in Figure #1.

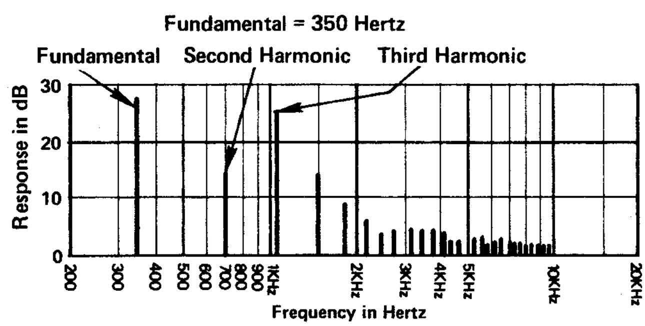

FIGURE 1 — Fundamental to Harmonic Energy Relationship

AUDIO FREQUENCY RANGE

Let us take a look at the audible frequency range to determine the location of commonly known sounds with respect to frequency. Refer to Figure #2 as you read the following

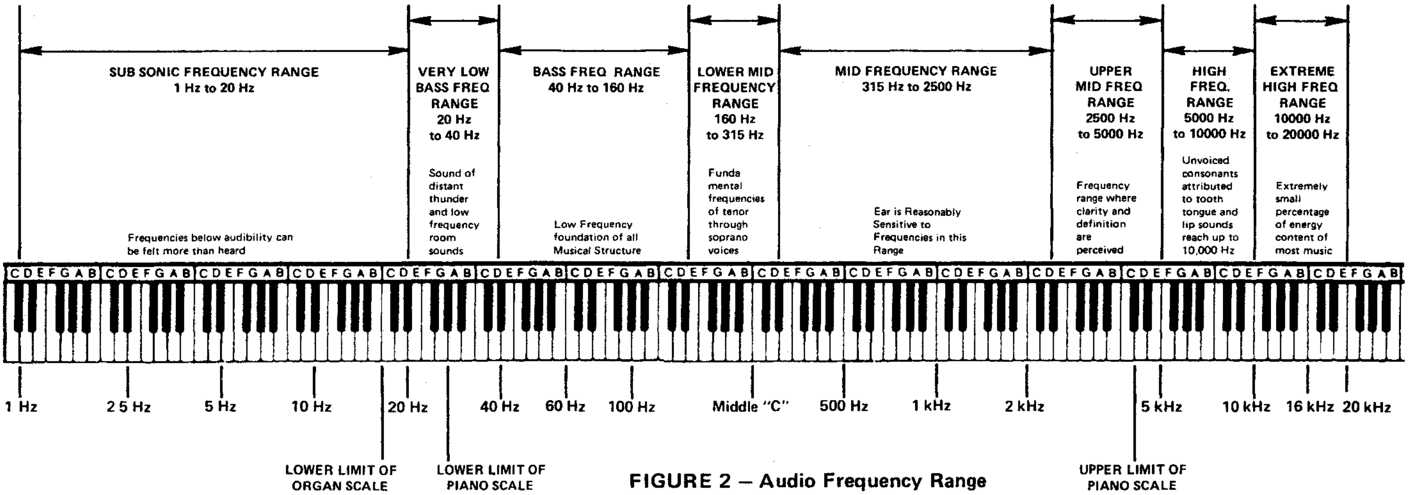

FIGURE 2 — Audio Frequency Range

to provide a visual aid. It was mentioned earlier that the range of human hearing is 20 Hz to 20 kHz. This would be the frequency range for a person with very good hearing. Most types of professional audio equipment will have a specified frequency response of 20 Hz to 20 kHz. For our purposes we will now investigate the frequency range from 1 Hz to 20 kHz.

Sub-Sonic Frequency Range – 1 Hz to 20 Hz (Approximately 4 Octaves)
In this portion of the frequency spectrum are frequencies that are usually not audible. One can *feel* their effect when they are at sufficiently high level as pressure sensations rather than as distinct musical tones. Some very large pipe organs and earthquakes have frequency components in this range.

Very Low Bass Frequency Range – 20 Hz to 40 Hz (1 Octave)
Examples of sounds with frequencies in this range are wind, room rumble, low frequency sounds from air conditioners, the sound of distant thunder, etc. Frequencies near 30 Hz are illustrated by the lowest fundamental frequencies of medium size pipe organs, the piano, and the harp.

Bass Frequency Range – 40 Hz to 160 Hz (2 Octaves)
Most of the low frequencies of drums, the piano, the organ, and string or electric bass fall within this frequency range. In this particular range is the low frequency foundation of all musical structure.

Lower Mid Frequency Range – 160 Hz to 315 Hz (1 Octave)
This octave is a transitional range *between* frequencies that would be perceived as bass and those that would be considered to be mid-range tones. Many of the fundamental frequencies of tenor through soprano voices and the lower frequencies of several instruments, such as the trumpet, clarinet, oboe and flute, are contained in this range.

Mid Frequency Range – 315 Hz to 2,500 Hz (3 Octaves)
The ear is reasonably sensitive to frequencies in this range. In fact, this range of frequencies, heard alone, would have a "telephone-like" quality. As most instruments are rich in lower harmonics, the majority of sound energy is found in the frequency range *up to* 2,500 Hz.

Upper Mid Range or Presence Frequency Range – 2,500 Hz to 5,000 Hz (1 Octave)
The ear is very sensitive to frequencies in this octave. A sense of clarity and definition is imparted to complex sounds by frequency components in this range. In fact, many horn type public address speakers are designed to deliver maximum energy in the 3,000 Hz region. The *apparent* loudness of a sound can be appreciably influenced by the frequencies in this region. Singing voices often have harmonics in this frequency range.

High Frequency or Brilliance Frequency Range – 5,000 Hz to 10,000 Hz (1 Octave)
This range makes things sound brighter, but there is normally little acoustical energy in this portion of the frequency spectrum. In fact, the highest fundamental frequency of musical instruments is approximately 4 kHz. These frequencies account for some clarity and brilliance; however, they are usually a small percentage of the total musical or speech energy. Often unvoiced consonants which are attributed to tooth, tongue, and lip sounds are high in frequency and reach up to the 10,000 Hz range.

Extreme High Frequency Range – 10,000 Hz to 20,000 Hz (1 Octave)
This is the top octave of the audible frequency range. It contains an extremely small percentage of the energy content of most music. Hearing loss associated with advancing age can remove an appreciable portion of the upper part of this octave. Only very high harmonics of some tones contribute to this region. The loss of most of this frequency range may not be noticed on much musical material – this is especially true of the upper *half octave* of this range. However, on very wide range frequency sources that are rich in harmonic content, the reproduction of this range can add a small but definite touch of realism.

EQUALIZER CHARACTERISTICS

An "equalizer" was defined earlier as a volume control (to increase or decrease level) of specific frequency ranges. When an equalizer is used to *increase* the level of a specific frequency range, it is called "boost". Likewise, when a certain frequency range is *decreased* in level it is called "cut". The amount a given frequency range is boosted or cut is typically indicated in decibels (dB). The decibel (dB) is the unit of measure for sound level (amplitude).

There are two basic types of equalization:

Shelving
Shelving is an equalization curve that changes in level until it reaches the "indicated frequency" at which point it levels off and changes no more. When illustrated, this curve looks like a shelf, hence the name "shelving". Once the indicated frequency is reached, all frequencies beyond will remain at the same level. A shelving equalizer is shown in both boost and cut modes in Figure # 3. The common bass and treble controls found on most types of hi-fi and inexpensive mixing equipment typically utilize shelving type equalization. Perhaps the most common indicated frequencies for these typical bass and treble controls are somewhere between 50 to 100 Hz for bass and 5,000 to 10,000 Hz for treble.

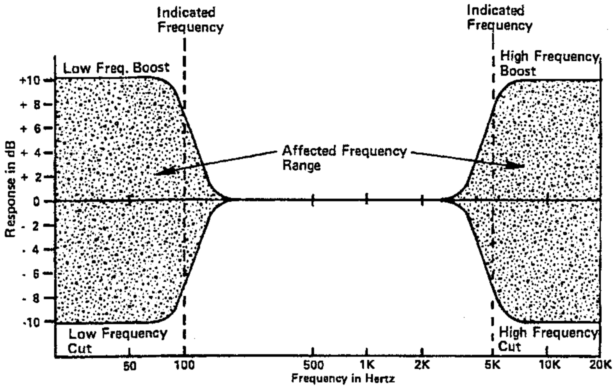

FIGURE 3 – Shelving Equalization Curve

Peak/Dip

Peak/Dip is a curve that starts at "0" (no change in level) and changes in level until it reaches its "center frequency" (maximum change in level) and then again changes in level until it returns to "0". When illustrated, this curve looks like a mountain or "peak" in the boost mode and a valley or "dip" in the cut mode. This type of equalization curve will provide maximum boost or cut at the center frequency as well as other frequencies close by, and will have less effect on frequencies further away from the center frequency. A Peak/Dip equalizer is shown in both boost and cut modes in Figure #4. Audio equipment with equalizers that have a "presence" or "mid-range" control (section) almost always utilize the peak/dip type of equalization curve.

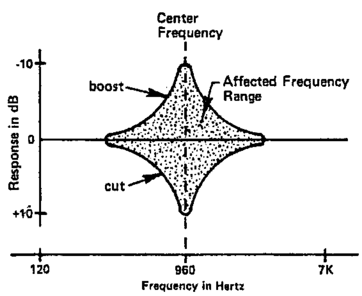

FIGURE 4 – Peak/Dip Equalization Curve

"Q" is another basic characteristic of equalization curves and pertains to both the shelving and peak/dip types. The technical definition of "Q" is beyond the scope of our purposes. Let us just say that "Q" is the property of an equalizer that determines the range of frequencies that are affected on either side of the center frequency in peak/dip type equalizers, and prior to the indicated frequency for shelving type equalizers. In Figure # 5 peak/dip type equalization curves are shown with both high and low "Q".

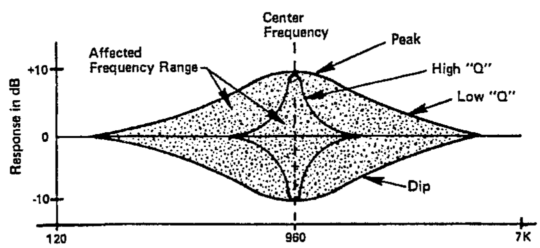

FIGURE 5 – "Q" of a Peak/Dip Equalizer

The "Q" on most types of equalizers is fixed and cannot be changed by the operator. However, some of the more elaborate types of equalizers (usually the peak/dip) have an additional control which will allow the operator to change the "Q". Shelving type equalizers are also affected by "Q" in that it determines the range of frequencies that are affected prior to its indicated frequency which is sometimes referred to as "slope". Figure # 6 shows shelving type equalization curves with both high and low "Q". The "Q" or slope of almost all shelving equalizers is fixed and not adjustable by the operator.

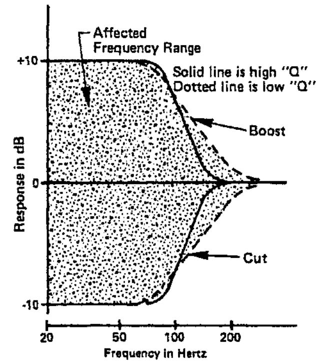

FIGURE 6 – "Q" or Slope of a Shelving Equalizer

BASIC TYPES OF EQUALIZERS AND THEIR APPLICATIONS

There are many types of equalizers available on the market to suit a wide range of applications. In many cases the addition of a specific type of equalizer to an audio system would provide a dramatic increase in quality, but often the user does not know that an equalizer would help. Again, there are those who desire to add equalizers to their systems, but do not know what type would best suit their needs. A real problem exists because many people purchase equalizers without any knowledge and are often sold equalizers that are far more elaborate and expensive than necessary. On the other hand, equalizers are frequently purchased which are not suitable or capable of performing well in a given application.

Equalizers are usually packaged in one of two ways; "inboard", which means that it is a part of another piece of equipment, such as a mixing console. "Outboard", sometimes referred to as "program" equalizers, are self contained equalizers that are usually self powered and can be patched (plugged) into almost any audio system.

Equalizers which utilize active components such as transistors or operational amplifiers are referred to as "active equalizers". When an active equalizer is connected to an audio system, there will typically be no loss in signal level when the equalizer is inserted into the signal path. Another less common type of equalizer, called the "passive equalizer", has no active electronic components and will cause a fixed signal (level) loss of several dB when inserted into the signal path. This is referred to as "insertion loss". Passive equalizers were the most popular type during the 1940's, 50's, and 60's, and some of these units are still available – usually second hand. There are also some sophisticated models of passive equalizers available for acoustic control and laboratory purposes. Passive equalizers must usually be terminated with specific impedances and have a fixed insertion loss. Passive type equalizers would not be an ideal choice for the unknowledgeable or new user of equalizers.

Tone Controls

Tone Controls are usually of the two knob type (one for bass and one for treble). Shelving equalization curves are most commonly utilized for this type of equalizer. The indicated frequencies are fixed and are typically somewhere between 50 and 100 Hz for bass and between 5 kHz and 10 kHz for treble. The "Q" or slope of these equalizers is fixed as well. The two knobs (controls) on this type of equalizer are for the amount of boost and cut that is available. The equalizer will have no effect at the "0" dB (no boost or cut) position which is usually at the center of rotation (rotary control) or center of travel (straight line control). These tone controls are actually a two knob (fixed frequency) equalizer. See Figure #7.

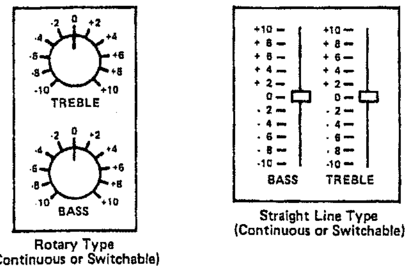

FIGURE 7 – Various Types of Tone Controls

Move the control in one direction for boost or in the opposite direction for cut. The available amount of boost and cut will vary depending upon the equalizer; however, it will usually be in the area of 10 dB of boost and cut (± 10 dB). Tone controls are found on almost every type of hi-fi equipment. Many types of mixers may have two knob equalizers on each input and/or on the main outputs. This is the most basic type of equalizer and can be easily used to enhance sound for a wide range of applications without the chance of getting into much trouble, and will usually provide very good results.

Two Knob Equalizers (Selectable Frequency)

This is a more sophisticated two knob (sometimes referred to as two band) equalizer which still has only two knobs and which will boost or cut two different frequency range simultaneously, just like the simple two knob "tone control". However, one or two additional controls are added to enable the operator to select various frequency ranges. A two knob shelving type equalizer that will select three bass as well as three treble frequency ranges is shown in Figure #8.

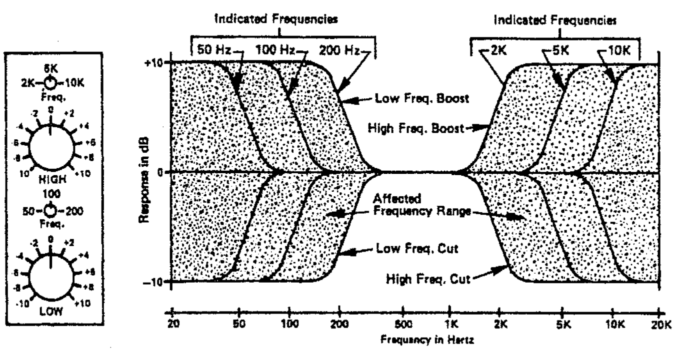

FIGURE 8 – Two Knob (six frequency) Selectable Frequency Equalizer

The "Q" of this equalizer is fixed. This equalizer will allow the operator to work closer to certain desired frequency ranges without affecting as much additional unwanted frequency range. For example, a low frequency room rumble could be reduced by "cutting" at the 30 Hz frequency range. The common tone control bass knob (typically 100 Hz) would serve the same function, but it would probably cut desired low frequencies from the program. The particular frequencies that are provided along with the available amount of boost and cut are determined by the manufacturer. Any switchable frequency equalizer can be a fine tool in the hands of a user who understands where the various instruments, vocals, and other sounds are located in the frequency spectrum.

Three Knob Equalizers (Fixed and Selectable Frequency)

These equalizers have three knobs (bands) which will boost and cut three frequency ranges simultaneously (sometimes referred to as a three band equalizer). One knob is for low frequencies, one for high frequencies, and one for mid-range frequencies. Three knob equalizers typically utilize shelving curves for the high (treble) and low (bass) frequency ranges while using a peak/dip curve for the mid-range. A "fixed frequency" three knob equalizer is shown in Figure #9.

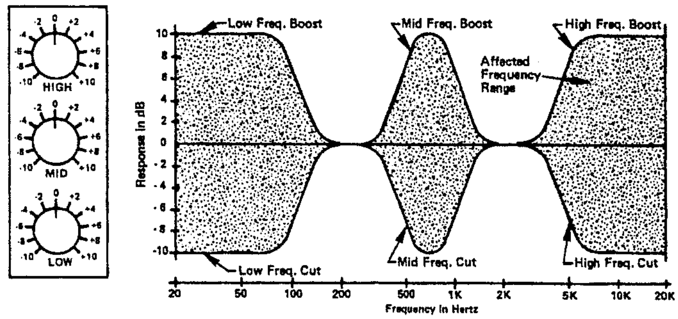

FIGURE 9 – Three Knob (three frequency) Fixed Frequency Equalizer

Another very common three knob equalizer also has fixed low and high frequency shelving curves but with a switchable peak/dip mid-range section as shown in Figure # 10.

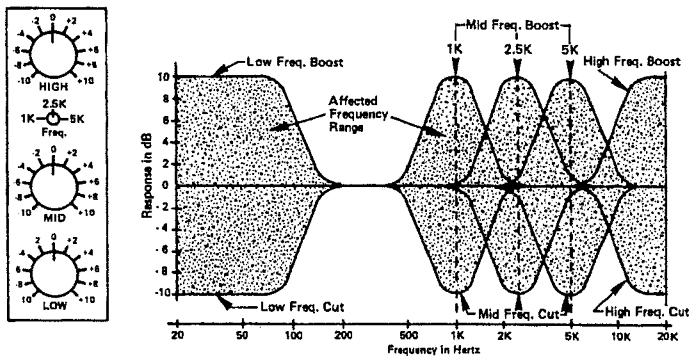

FIGURE 10 – Three Knob (five frequency) Equalizer with Switchable Mid-Range

Both of the equalizers in Figure # 9 and # 10 have fixed "Q". Again, the amount of boost and cut (in dB) as well as the available frequency ranges depend upon the manufacturer. Three knob equalizers add the flexibility of mid-range equalization to the low and high frequency ranges. This type of equalizer is typically found on higher priced mixing consoles and offers the user a great deal of flexibility.

Four Knob Equalizers (Fixed and Selectable Frequency)

Now we have four frequency ranges that can be boosted or cut simultaneously; low frequencies, lower mid-range, upper mid-range, and high frequencies. Most equalizers of this type utilize shelving for the low and high frequency sections, and peak/dip for both midrange sections. Almost all models of this type have switchable mid-range frequency ranges and sometimes even switchable low and high frequency ranges as well. In Figure # 11 there is a full blown, four knob selectable frequency equalizer.

This equalizer has ten frequency ranges with fixed "Q" that will allow the operator to boost or cut at frequency ranges that are somewhat close together. The available frequencies and amount of boost and cut depend, as always, upon the manufacturer. These sophisticated equalizers are typically found in the very expensive and more elaborate type of mixing consoles and in some models of outboard equalizers. Some equalizers of this type even have adjustable "Q" in the mid-range sections.

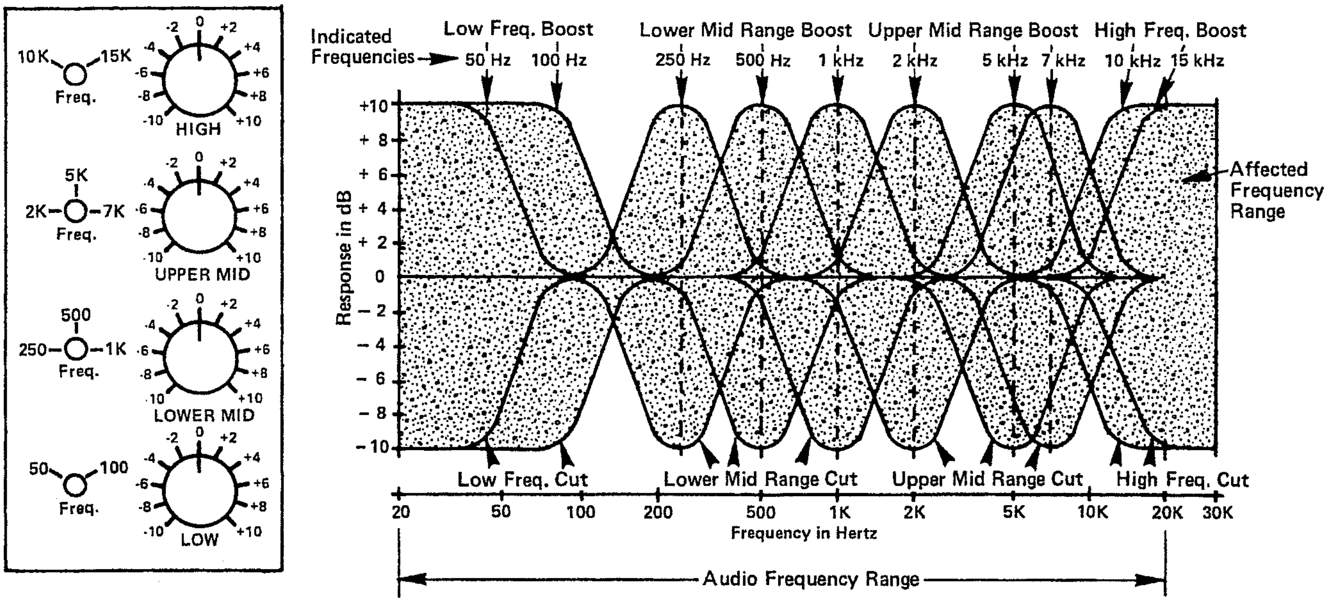

FIGURE 11 – Four Knob (ten frequency) Selectable Frequency Equalizer

Graphic Equalizers

All of the equalizers that we have investigated to this point have had the capacity to equalize in two, three, or four frequency ranges at the same time, depending upon the type. It is sometimes an advantage to have the ability to equalize in a larger number of frequency ranges at the same time. It can be easily seen that the two, three, or four knob equalizers do not have this capability. A "graphic equalizer" does have the capability of equalizing a large number of frequency ranges (usually 10 or more) at the same time. Graphic equalizers typically utilize the peak/dip type of equalization curve and the "Q" is usually fixed. Graphic equalizers have the ability to contour the sounds of different voices, effects, or music by boosting or cutting at a number of frequency ranges simultaneously. For example, equalizing a voice in the mid frequency range will add more presence and intelligibility while boosting the lower mid-range and bass frequency ranges will add warmth. If there are some low frequency, unwanted sounds like room rumble at very low frequencies, the 30 and 50 Hz regions can be cut to reduce the audible effect of these unwanted sounds. Now, all of these things can be done with the same equalizer that had the ability to boost or cut in a large number of frequency ranges (bands).

ISO Center Frequencies. The ISO (International Standards Organization) has established standard center frequencies for graphic equalizers. These specified frequencies are utilized by most manufacturers of graphic equalizers. It should be pointed out that an equalizer not using the ISO center frequencies is necessarily less desirable than one that does. Standardization of center frequencies enables the operator to use almost any graphic equalizer in the same way because it has the same exact frequency ranges available to equalize. This makes it possible to duplicate a certain sound or equalization setting at a different location when utilizing unfamiliar equipment.

One Octave Graphic Equalizer. Perhaps the most common graphic equalizer is the "octave" graphic equalizer. This is a graphic with usually ten frequency ranges that can be boosted or cut, with center frequencies placed at one octave intervals. Most equalizers of this type have fixed "Q". The ISO center frequencies for such a one octave equalizer are: 16, 31.5, 63, 125, 250, 500, 1000, 2000, 4000, 8000, and 16000 Hz. You may notice that there are 11 one octave ISO center frequencies. Most one octave equalizers do not have the 16 Hz frequency band because there is little or no musical energy at this range. A typical one octave equalizer along with its equalization curves is shown in Figure #12.

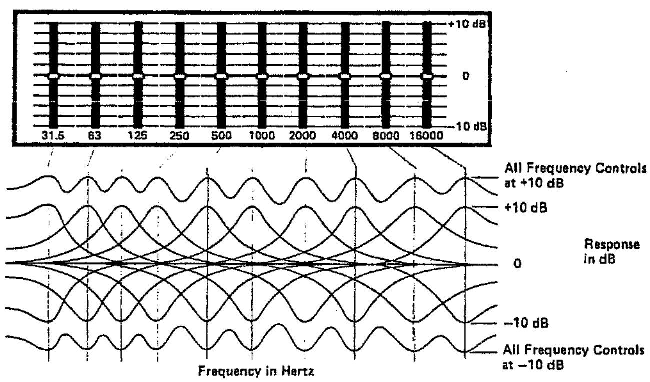

FIGURE 12 – One Octave Graphic Equalizer

1/3 Octave Graphic Equalizer. This graphic has three times as many frequency bands as the one octave type and will provide greater control over the frequency spectrum by breaking it up into smaller segments. This equalizer will typically have 30 frequency ranges that can be boosted or cut at the same time. Most equalizers of this type have fixed "Q". 1/3 octave graphic equalizers are widely used for speaker or room voicing, acoustic feedback suppression, as well as laboratory applications. The ISO center frequencies of the 1/3 octave equalizer are: 16, 20, 25, 31.5, 40, 50, 63, 80, 100, 125, 160, 200, 250, 315, 400, 500, 630, 800, 1000, 1250, 1600, 2000, 2500, 3150, 4000, 5000, 6300, 8000, 10000, 12500, and 16000 Hz.

1/2 Octave Graphic Equalizer. Often, for some applications, a one octave graphic equalizer will not equalize at enough frequencies, and the 1/3 octave equalizer is too elaborate and expensive.

This means that there is room for another equalizer in the middle. So for those who need additional frequency ranges over the one octave graphic, but who neither need the full flexibility nor can afford the 1/3 octave graphic, there is the 1/2 octave graphic equalizer. This type of equalizer is of the boost or cut type and usually has a fixed "Q". The ISO center frequencies for the 1/2 octave graphic are: 16, 22.4, 31.5, 45, 63, 90, 125, 180, 250, 355, 500, 710, 1000, 1400, 2000, 2800, 4000, 5600, 8000, 11200, and 16000 Hz.

It is common for one who realizes the flexibility of a graphic equalizer to say, "Boy, what I could do with that!". Graphic equalizers can be very valuable tools because they have the ability to equalize in a large number of frequency ranges (bands). This tool can also be a detriment because it can also do a great deal of sonic damage to a signal. We do not mean to discourage the use of these sophisticated equalizers; however, it is important for the user to make sure the equalizer is suited to his application and to operate it properly if he is to achieve the desired audible results.

Sweepable Frequency Equalizers

All of the equalizers mentioned previously have had either a small number or a large number of *fixed* frequency ranges (bands) which could be equalized. Utilizing these (fixed frequency) types of equalizers, one would first listen to the signal, then equalize at the nearest available frequency range to provide the desired effect. If one wanted to equalize a vocal in the mid-range and the equalizer would only boost and cut a 2 kHz and 5 kHz, it would be necessary to use one of these available frequency ranges (bands). In most cases the use of one of these two available ranges would probably improve the sound of the vocal, but the amount of flexibility would be restricted by the available frequencies on the equalizer. Suppose a mixing console had a four knob six frequency equalizer and the available mid-range frequencies were 800 Hz, 1.2 kHz, 2.5 kHz, and 5 kHz. We want to equalize for a particular type of mid-range sound on a vocal solo, and we find that it cannot be obtained with any of the available mid-range frequencies on the equalizer. One option is to keep trying outboard equalizers (perhaps a graphic) until you find one that equalizes at the right frequency to provide the desired sound.

Wouldn't it be nice if you could have an equalizer that would tune over a wide range of frequencies like a radio? Now it would be possible to adjust the center frequency of the equalizer (with a knob) until you had obtained the desired sound or effect. This would allow the selection of any center frequency within the equalizer's available frequency range. This is called a sweepable frequency equalizer. There are usually two knobs per equalizer section, one to tune (adjust) the center frequency over the available frequency range and another to adjust the amount of boost or cut (specified in dB). The "Q" for this type of equalizer is fixed. A simple one knob sweepable frequency equalizer would have equalization characteristics similar to the ones shown in Figure # 4, except that the center frequency would be sweepable (tunable).

It is a common feature to find three knob equalizers with fixed frequency high and low bands and a sweepable mid-range section. This type of equalizer will provide the operator with a great deal of flexibility. They are a very popular feature on many of today's mixing consoles. There are also two knob versions of sweepable frequency equalizers.

Parametric Equalizers

The sweepable frequency equalizers have fixed "Q". However, there are applications where it is desirable to be able to adjust the center frequency and the "Q" as well. Such an equalizer that has both tunable frequency and adjustable "Q" is called a parametric equalizer. The adjustable "Q" control operates in a tuning fashion similar to the adjustable frequency control. One can now tune the center frequency for the desired basic sound or effect, then have the additional flexibility of adjusting the "Q". By adjusting the "Q", the frequency range on either side of the center frequency can be restricted to that which provides the most desirable results. This equalizer will provide a great deal of flexibility for the knowledgeable user. A single knob version of the parametric equalizer would have equalization characteristics similar to the ones shown in Figure # 5, except that the center frequency would be sweepable (tunable).

There are a number of versions of the parametric equalizer available. The three and four knob (band) versions are the most popular of the outboard types. Parametric mid-range sections are sometimes used in three and four knob inboard equalizers on mixing consoles.

Paragraphic Equalizers

These equalizers are a hybrid made up of the properties of both the parametric and the graphic equalizers, hence the word paragraphic. One may ask why one would desire something like this. There are some applications where many frequency bands must be equalized and where it is also desirable to be able to adjust the center frequency and the "Q" of the bands. The most common type of paragraphic is similar to a one octave graphic made up of ten frequency bands placed on one octave centers. Each of the ten frequency bands are tunable (adjustable) over its one octave range to allow the center frequency to be adjusted to anywhere within that one octave band. Once the center frequency is set the "Q" can be adjusted for that band to provide the desired effect. A paragraphic is shown in Figure #13.

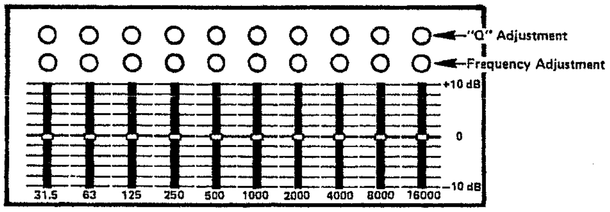

FIGURE 13 – Paragraphic Equalizer

Paragraphic equalizers are very popular for "room voicing" where there are often certain portions of the frequency spectrum with a peak or dip that may not coincide with the ISO center frequencies. Having the ability of adjustable frequency and "Q" as well will enable the equalizer to provide the right amount of complimentary boost or cut in the exact frequency range to correct the problem. This method will allow the corrective equalization to be specifically tailored to the problem or the room.

Sweepable Graphic Equalizer

An equalizer similar to the paragraphic is the sweepable graphic equalizer. This equalizer is identical to the paragraphic in every respect except it does not have adjustable "Q". Obviously, this equalizer is less complex and less expensive.

BASICS FOR THE USE OF EQUALIZERS

At this point you should have a good idea of the various types of equalizers available and their basic features as well as some of their ideal applications. Once you have reviewed this information and have selected an equalizer for your purposes, you have only just begun. Now you must know how to use it properly to realize its full potential. It is virtually possible to write a book on the operation and use of each type of equalizer that has been described. However, certain basic information will be of great benefit to you in your use of almost any type of equalizer.

KNOW THE FUNDAMENTAL FREQUENCIES OF VOICES AND MUSICAL INSTRUMENTS

It is possible to equalize the sound of an instrument or voice and not obtain the desired effect – if any effect at all. This is usually due to the fact that the boosting or cutting occurred at frequencies above or below the dominant energy of that voice or instrument. If you are to utilize equalizers effectively, you must know the frequency content of instruments and voices so you will know what frequency ranges can be equalized. For this purpose the chart shown in Figure # 14 may be used.

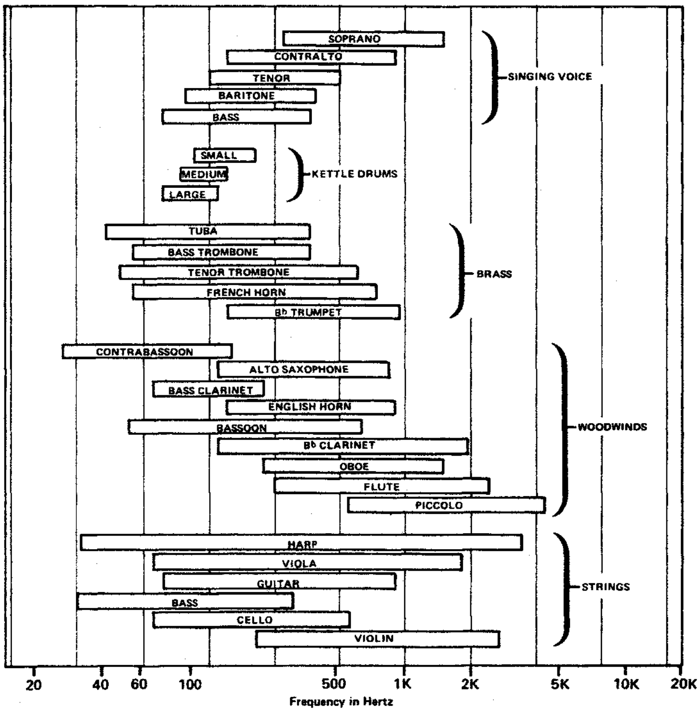

FIGURE 14 – Frequency Ranges for Fundamental Frequencies of Voices and Musical Instruments

It is important to point out that the information on this chart only applies to the fundamental frequencies. All musical sounds have harmonics which will go beyond the frequency ranges shown in Figure # 14. These harmonics affect the sound of voices and instruments; however, they are usually much lower in level than the fundamental frequency. At times good effects can be obtained by boosting or cutting in the frequency range of these harmonics rather than in the frequency range of the fundamental frequency. In any case, it is preferrable to always start your equalization in the fundamental frequency region. If you do not like the sound, try equalizing at surrounding frequencies until you get the desired effect.

Let us look at some of the things that happen as well as some things that must be avoided when applying equalization to some basic musical instruments and vocals.

Vocals

When using a mixer with simple bass and treble controls, only the low and high frequencies of the voice can be equalized. Often the low frequency sections of such equalizers will have little if any effect on female vocals because they are above the equalizer's effective frequency range. However, this low frequency control will usually prove quite effective on male vocals. One must be careful with the amount of treble correction that is used because the high frequency section of basic two knob equalizers are most effective in the 5 kHz region or above. Excessive equalization of vocals (male or female) in this frequency range will increase the amount of sibilance in the voice and will usually cause a "spitting" or "hissing" sound when the singer pronounces words with the letter "s". If a vocalist has an excessive amount of sibilance even when not using equalization, an equalizer can also cure this problem. Try cutting in the 4 kHz to 7 kHz region to reduce the undesired effect. However, you must not use too much cut in as it can sometimes cause a dramatic loss in the presence of the voice. A great deal of presence can be added to vocals with the use of mid-range equalization (boost) usually in the 1 kHz to 3 kHz region (depending upon the particular voice). Again, one must be careful because too much boost in this frequency range can cause excessive amounts of sibilance. The boosting of lower frequencies in the 500 Hz to 800 Hz frequency range will often add a warmth to the sound of a vocal. The boosting of frequency ranges that are either above or below the frequency range of the voice will rarely have any audible effect and will often cause the *unwanted* sounds of other instruments (which are in that frequency range) to be picked up by the vocal microphone.

Electric Bass

It is the normal tendency to boost the sound of the electric bass at very low frequencies (in the 30 Hz to 60 Hz range) to give it more "balls". When making recordings it is important to point out that during the disc cutting process, frequencies below 30 or 50 Hz are usually filtered out in an effort to cut the disc at a louder level and place more music (time) on the disc. For live stage sound, most loudspeaker systems will not produce a great deal of acoustic energy in the frequency region below 50 Hz. Usually a far greater effect can be obtained by boosting in the 100 Hz frequency range. Boosting low frequencies will usually provide a good effect; however, excessive amounts of boost in the low frequency region will often cause the electric bass to sound "muddy" or "boomy". Definition can also be added by boosting in the 800 Hz to 2 kHz region. This will tend to emphasize the percussive attacks of the strings. A little of this is often desirable.

Drums

Let us look at the drums in four separate parts: bass drum, snare drum, tom toms, and cymbals.

Bass Drum. For most recording or stage sound applications the microphone is usually placed a few inches in front of the drum. When equalizing the bass drum, many of the techniques as well as problem areas of the bass guitar apply. Do not boost at extreme low frequencies (30 Hz to 50 Hz). Usually better effects can be realized when boosting in the 100 Hz range. Additional definition can often be added when boosting in the 800 Hz to 2 kHz region. Still more definition can often be added by cutting at the extreme low frequencies (30 Hz to 50 Hz).

Snare Drum. The dominant energy from the snare drum is in the 1 kHz to 2 kHz region. This is the frequency region where the snare should be boosted. Boosting at frequencies in the 500 Hz to 800 Hz range may make the snare sound a little fatter, but beware of picking up leakage from the bass drum or tom toms in the snare drum microphone. A more crisp quality can be obtained by boosting at higher frequencies in the 2 kHz to 4 kHz region.

Tom Toms. Depending upon the toms and how they are tuned, one will usually find the most effective range for equalization in the 250 Hz to 2 kHz region. Some nice low frequency sounds can be obtained from the tom toms; however, it is important to maintain definition. Definition can usually be added by boosting in the 800 Hz to 2 kHz region.

Cymbals. The dominant energy for the cymbals lies in the 2 kHz to 5 kHz region. There is often a tendency to boost cymbals at frequencies in the 10 kHz region which make them sound paper thin. Far better results can usually be obtained by boosting in the frequency where the dominant energy lies.

There are basic rules and guidelines for all the other types of musical instruments. However, you can see from the instrument and vocal information already given that the guidelines are simple and basically revolve around working in the area of the fundamental frequencies. It does one no good to equalize at frequencies which a sound system will not adequately reproduce or which cannot be placed on a record.

ROOM EQUALIZATION (Room Voicing)

As we stated previously, it is not our intention to discuss room equalization in detail. This subject could be a book in itself. The material that follows will, however, offer some simple guidelines to help familiarize you with some important aspects of this topic.

Equalizers can be used between mixers and power amplifiers in environments such as recording studios, auditoriums, listening rooms, and theaters, to contour the overall frequency response of a speaker system within a room. In professional installations where a real-time analyzer (a piece of test equipment used to show the total energy present at all frequencies on an instantaneous basis) is available, the frequency response can be measured to determine what equalization will be needed to obtain a desirable frequency response. Even though the speaker system may produce an acceptable response alone, the room usually modifies the response and creates peaks and dips in various frequency ranges. A graphic equalizer can remove many of these peaks and dips by boosting and cutting in the appropriate frequency ranges. The 1/3 octave graphic equalizer is a popular choice for this application because of its large number of available frequencies.

In most cases where a real-time analyzer is not available, another analyzer can be used – the ear. One approach is to play recorded music you are familiar with through the system and adjust the equalizer for the desired sound. When the "EQ" is adjusted for the best sound, just be sure that no portion of the audio spectrum is given an extreme amount of boost or cut. Note as a special case that attempting to boost the output below the speaker system's low frequency capability will not produce much improvement and may cause damage to the woofer due to excessive excursion.

Another popular use of graphic, parametric, or paragraphic equalizers is to reduce feedback in sound reinforcement or stage monitoring systems. One method of achieving this is to make a sound check with all the "EQ" controls set flat. After a good mix is obtained, slowly increase the system gain until feedback or ringing begins to develop. When feedback occurs, it will often initially start in one portion of the frequency spectrum. The frequency range at which feedback will initially occur varies, depending upon the equipment used and the particular acoustical environment. The frequency at which the system is ringing, or beginning to feedback, may be reduced by lowering or cutting the appropriate frequency band of the equalizer. Which frequency band? It will be necessary to search for the correct one. The ringing will diminish when you adjust its corresponding frequency band – you will know when you find it. Now the overall sound level of the sound reinforcement system can be increased until feedback occurs again, when the same procedure is repeated. When this process is done several times, there will be a number of frequency ranges that have been "cut" (usually only by a few dB). This will now give the sound reinforcement or monitor system the ability to generate a higher sound level before reaching feedback. The operator must be careful because cutting the frequency response at these various frequencies places audible holes in the frequency spectrum. Sometimes after this procedure is done, things just don't sound right. It is often necessary to utilize smaller amounts of cut to enable the sound system to sound natural. This method of acoustic feedback suppression is a process of trial and error, and one of compromise (amount of additional gain before feedback versus the overall audible quality of the system).

DO NOT USE EQUALIZATION TO EXCESS

Equalizers give you the ability to change the frequency characteristics of instruments and vocals, but excessive amounts of equalization can make things sound very strange. This can be useful only when you desire an unnatural or freaky effect. When equalization is used in the boost mode, you are making certain frequency ranges louder, and this will reduce the amount of available headroom in your mixer or audio system, which can lead to distortion. This does not mean that using an equalizer will usually cause distortion, but that excessive equalization can in some instances cause problems. Equalizers, when used properly, are valuable tools providing that they are used with care and moderation.

Take the time to know the frequency content of the various musical instruments and voices. Spend time working and experimenting with your equalizer to become more familiar with it and the effects that can be obtained. Learn by trial and error. You are bound to make many mistakes – we all have! Review the information in this PA Bible addition carefully and become familiar with it. Blend into this the information on sound systems and their components contained in the original PA Bible and its previous additions for a more thorough understanding of the total system. Couple that knowledge with experience and a few goofs on the way and you will soon be mastering the art of equalization. Happy Equalizing!

SPECIAL NOTE TO THE READER

The E-V "PA Bible" and this addition have been prepared to help you solve your PA problems. Let us know if you have any other areas in mind for us to tackle.

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THE PA BIBLE

ADDITION NUMBER THREE MICROPHONE TYPES

In our prior P. A. Bible material we have dealt with acoustic transducers that convert electrical energy into sound (loudspeakers). Another type of acoustic transducer is the microphone. The microphone is similar to the loudspeaker, but its function is to convert sound into electrical energy.

The following discussion will describe the various types of microphones in terms of four factors that help define them. In addition, some information will be given that should help in selecting a microphone type and a few important operating tips will be mentioned. It is not our intent to go deeply into applications in this supplement. We intend to reserve that complex topic for a later issue.

MICROPHONE TYPES AND OPERATION

All microphones have two basic components: the diaphragm, and the generating element. The diaphragm is a membrane which vibrates in accordance with the pressure variations of sound. The generating element converts the diaphragm vibrations into electrical voltage. This generating element is one of four factors which determine the type of microphone. The kinds of generating elements vary greatly in expense, fidelity, complexity, ruggedness, and longevity.

Ceramic and Crystal Generating Elements. The diaphragm of a crystal or ceramic microphone is attached to a special material which produces an electric output voltage when it is moved. Such materials are termed "piezoelectric." A typical ceramic microphone is diagrammed in Figure 1. Such microphones generally provide insufficient fidelity and ruggedness, even for the most modest requirements of the professional and serious amateur.

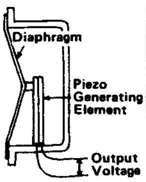


FIGURE 1—Ceramic/Crystal Generating Element

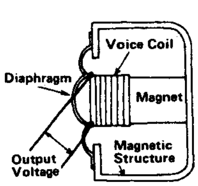


FIGURE 2—Dynamic Generating Element

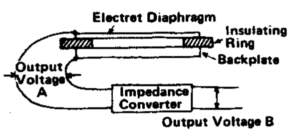


FIGURE 2A—Condenser Generating Element

Ribbon (or "Velocity") Generating Elements. Ribbon microphones are similar to dynamics, except that a very thin metal-foil ribbon serves as both diaphragm and voice coil. In order to obtain adequate frequency response and output level, the thin ribbon must be exceedingly light. Older ribbon microphones could easily be destroyed by mechanical shock or a suddent blast of air which would stretch and destroy the fragile ribbon. However, the best current designs have been improved for satisfactory durability.

Dynamic Generating Elements. The diaphragm of a dynamic microphone is attached directly to a coil of wire (voice coil) located close to a magnet. When the voice coil vibrates, a voltage is produced. A dynamic microphone is shown in Figure 2.

The dynamic microphone is a proven tool for the public address and instrumental miking requirements of the professional performer. It provides excellent fidelity, extremely stable performance characteristics, and a high degree of ruggedness – all at a reasonable price. These same characteristics are ideal for conventional sound reinforcement and recording, as well. In addition, the diaphragm of a well-designed dynamic microphone is able to withstand the close miking and high sound levels often employed by musicians; all without damaging the microphone or distorting its output. The many desirable features inherent in the dynamic microphone make it a good choice for most applications.

Condenser Generating Elements. The diaphragm of a condenser microphone is a movable plate of a condenser (capacitor), a common component in electrical circuits (Figure 2A). When polarized by applying a direct current voltage, motion of a diaphragm in relation to a fixed backplate produces an output voltage. The extremely high impedance of the condenser generating element is matched to typical inputs by an impedance converter in the microphone. Condenser microphones, many of which are capable of very wide frequency response, have been widely used in recording studios for years. For the performer, due to their relatively high output level, condensers may produce input overload distortion (distortion caused by too great an input signal to a mixer) unless appropriate precautions are taken.

Modern day electret type condenser microphones can offer ruggedness comparable to dynamic microphones. The electret microphone can often yield superior performance at the frequency extremes (high and low) when compared to dynamic types. Because electrets utilize an impedance converter to match the diaphragm signal to the mixer input, they require either a battery or phantom power for operation. Phantom power is a means by which power is supplied to the microphone from either a mixer or power supply by way of the microphone cable. Phantom power eliminates the need for batteries and the problem of replacing dead batteries. Even though electret microphones are more complex in construction, their performance advantages are making them an increasingly attractive choice for exacting applications.

MICROPHONE PICKUP PATTERNS

A microphone's pickup pattern is three dimensional in character and shows how the microphone responds to sound from different directions. Omnidirectional microphones pick up sound from all directions. Unidirectional microphones reject or reduce sound from their sides or rear. The pickup pattern is the second of four factors which determine the type of microphone.

Omnidirectional Pickup Pattern. The pickup pattern of an omnidirectional microphone may be represented as an inflated balloon with the microphone at the center, as shown in Figure 3. Usually a polar pattern is used to represent the pickup pattern, illustrated in Figure 4. The polar pattern shows the loss in output (in dB) experienced as a constant-output sound source moves 360° around a fixed microphone at a fixed distance from the microphone.

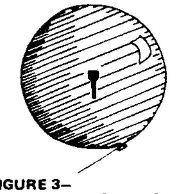


FIGURE 3—Omnidirectional Pickup Pattern

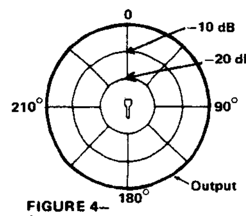


FIGURE 4—Omnidirectional Polar Pattern

How does an Omnidirectional Microphone Work? The case of the microphone shown in Figure 5 is totally sealed, so that sound pressure can strike only the front of the diaphragm.

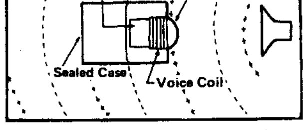


FIGURE 5—Omnidirectional Microphone

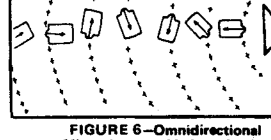


FIGURE 6—Omnidirectional Microphone at Various Angles

Pressure variations passing over the diaphragm move it no matter how the unit is oriented with respect to the sound source! This phenomenon is shown in Figure 6. Thus, microphone output is constant regardless of orientation.

Why an Omnidirectional Microphone? In systems where extremely close working distances are employed, say touching the lips to six inches, the omnidirectional microphone, if it can be used, has several advantages in its favor:

1. For a given price, an omnidirectional microphone generally has a smoother frequency response than its cardioid counterpart. Such smoothness of response is important because any roughness invites feedback.
2. An omnidirectional microphone is significantly less susceptible to breath pops than its cardioid counterpart.
3. An omnidirectional microphone is significantly less sensitive to mechanical shock than its cardioid counterpart.
4. An omnidirectional microphone is often more rugged than its cardioid counterpart.

Unidirectional Pickup Pattern. The most common unidirectional microphone is called a cardioid. Cardioid is a mathematically descriptive term that denotes the geometric form of the pickup pattern. The pattern happens to be crudely heartshaped (hence the term "cardioid"). Side pickup is moderately reduced in a cardioid microphone and rear pickup is dramatically reduced. The polar pattern of a cardioid microphone is shown in Figure 7. The apple shown in Figure 8 would be a good three-dimensional model of the cardioid pattern with the stem representing the microphone.

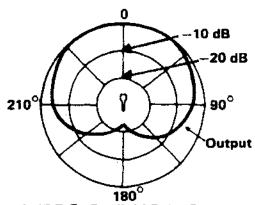


FIGURE 7–Cardioid Polar Pattern

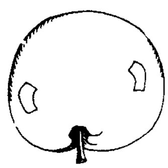


FIGURE 8–Cardioid Pickup Pattern

How does a Unidirectional Microphone Work? In a cardioid unidirectional microphone, the case is not sealed. The sound pressure is permitted to contact the diaphragm from the rear as well as the front. The rear contact occurs through a port which is precisely located in the microphone case.

A simple cardioid microphone is shown in Figure 9 with the sound source at its rear. With sound originating from the rear of the microphone, diaphragm motion is neutralized by equal, in phase sound pressures arriving at each side of the diaphragm, resulting in zero net force acting on the diaphragm. (Note the "plus" sound pressures on both sides of the diaphragm.) However, with sound originating from the front as shown in Figure 10, a delay in the sound pressure reaching the rear of the diaphragm due to the increased distance (that is, distance to the rear opening plus the distance back to the diaphragm) permits motion of the diaphragm. Microphone output results. (Note the "plus" sound pressure on the diaphragm's front, and the "minus" pressure at the rear.)

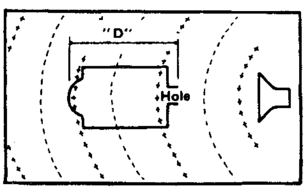


FIGURE 9–"Single-D" Cardioid Microphone with source at rear

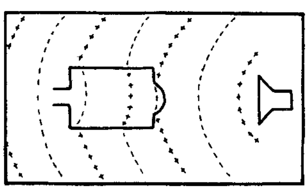


FIGURE 10–"Single-D" Cardioid Microphone with source at front

Why a Cardioid Microphone? The pickup pattern of a cardioid microphone – relatively dead at the sides and rear – tends to increase the working distance (the distance between the sound source and the microphone). The limiting factor is when the distance becomes so great that amplifier gain must be increased until:

1. The sound becomes over-reverberant due to room reflections.
2. The pickup of random background noise becomes excessive.
3. Sound system feedback results from P.A. or monitor speakers.

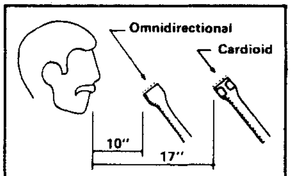


FIGURE 11– Working Distance Comparison

This increase in working distance is theoretically 1.7 to 1, as shown in Figure 11. For instance, if the maximum effective working distance of an omnidirectional microphone is ten inches, then theoretically a cardioid mic can be used at seventeen inches with the same effectiveness!

The feedback-reducing characteristics of cardioid microphones would seem to make a clear-cut case for the use of a cardioid microphone by professional performers. In marginal feedback situations, the cardioid will produce a higher level in the room before feedback occurs. This situation is often encountered in portable P.A. systems and other systems employing high-level stage monitors, where high levels of direct speaker sound reaches the microphone from the sides or rear. Usually in such instances, the loudspeakers are closer to the microphones than would be desirable from a sound-system design standpoint, and care must be taken to maintain proper gain without feedback.

Two vastly Different Types of Cardioid Microphones. A 'Single-D" cardioid gives big bass. The simple cardioid microphone described previously (the one with a single port located in the case) has a frequency response which varies strongly with working distance! As shown in Figure 12, at one-quarter inch, the bass response is boosted fifteen dB over the response at 24 inches and beyond! In engineering terms, this type of cardioid is called a "Single-D", named for the single distance between the rear sound entrance and the diaphragm.

The close-up emphasis of bass tones of the Single-D cardioid, called "proximity effect", provides a big, no-mistake-about-it bass sound – and for certain vocal applications, this is a popular sound. The Single-D sound, however, may not provide the super-clarity often desired by today's performer.

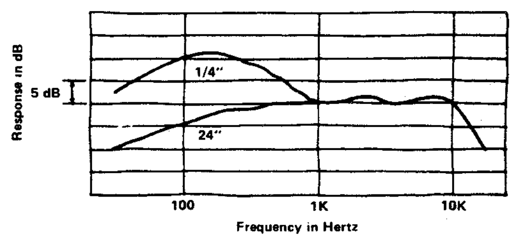


FIGURE 12–"Single-D" Proximity Effect

A "Variable-D®" cardioid emphasizes clarity. In order to reduce bass-boosting proximity effect, Electro-Voice developed and patented the "Variable-D" microphone. In a Variable-D, multiple ports are used with high frequencies entering the port closest to the diaphragm, mid frequencies entering midway along the length of the microphone case, and low frequencies entering the port farthest from the diaphragm. A Variable-D microphone is shown in Figure 13.

The virtual elimination of proximity effect of a Variable-D microphone is shown in Figure 14 in comparison to the strong bass boost of the Single-D cardioid microphone.

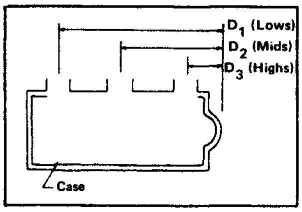


FIGURE 13— "Variable-D" Cardioid Microphone

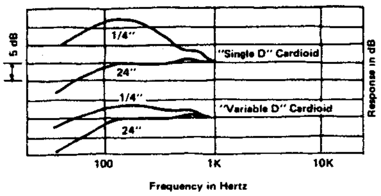


FIGURE 14— Proximity Effect Comparison

Latest Electro-Voice designs employ a variation called "Continuously Variable-D" where the mid- and low-frequency ports are replaced by a long, slotted entrance which has a continuously varying frequency acceptance along its length, with the lowest frequencies entering at the farthest point from the diaphragm. The frequency discrimination of Variable-D or Continuously Variable-D microphone ports can be effectively demonstrated by speaking, with lips touching, into the front, then mid, then rear openings. The change in vocal character will be readily apparent, with the sound very "bassy" at the rear port of the microphone and with much more "treble" evident toward the front port. A Continuously Variable-D microphone is shown in Figure 15.

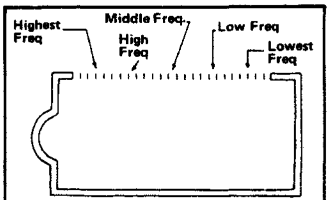


FIGURE 15—"Continuously Variable-D" Microphone

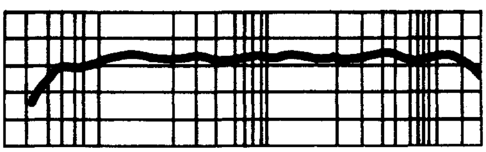
In addition to reduction of the Single-D's proximity effect, Variable-D and Continuously Variable-D cardioids have reduced breath popping and shock sensitivity. Thus, the popularity of the Variable-D microphone is due to its combining of the omnidirectional's clarity and inherent pop and shock resistance with the cardioid's feedback reduction and working distance advantages.

Variable-D and Continuously Variable-D (CV-D) are registered trademarks of Electro-Voice, Inc.

MICROPHONE FREQUENCY RESPONSE

The third factor which determines microphone type is frequency response. Response information for each microphone will help you select for special results. For instance, a microphone with "rising" response will emphasize the brightness of a trumpet or other brass instrument; one with proximity effect (single "D" cardioid) will add bass boost to a close working "thin voiced" singer. Communications microphones almost always have rising response or a "presence peak" to add intelligibility to voice transmission. A flat response, for most accurate sound reproduction as shown in Figure 16, would be typical for studio recording under ideal room and low noise conditions.

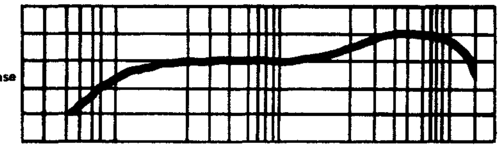
FIGURE 16 – Nominally Flat Response



Choose an appropriately shaped response (an example is shown in Figure 17) for special requirements: rolling-off frequencies of unwanted background noise (such as in a vehicle or plant machinery) and room reverberation; boosting bass; brightening high frequencies or increasing intelligibility.

When choosing a microphone, be certain its frequency response is wide enough to reproduce the sounds you need to reproduce with no perceptible change in quality. It should be pointed out that the response curves shown in Figures 16 and 17 (as well as curves shown on data sheets) were produced with the microphone at a sufficient distance so as to be in the "free field." When microphones are measured at very close distances, changes in the response curve can occur from microphone characteristics, such as proximity effect, which we described earlier. The chart shown in Figure 18 shows the frequency ranges of various instruments as well as the human voice, and it should be of some help when selecting an appropriate microphone with respect to frequency response.

FIGURE 17 – Typical Shaped Response



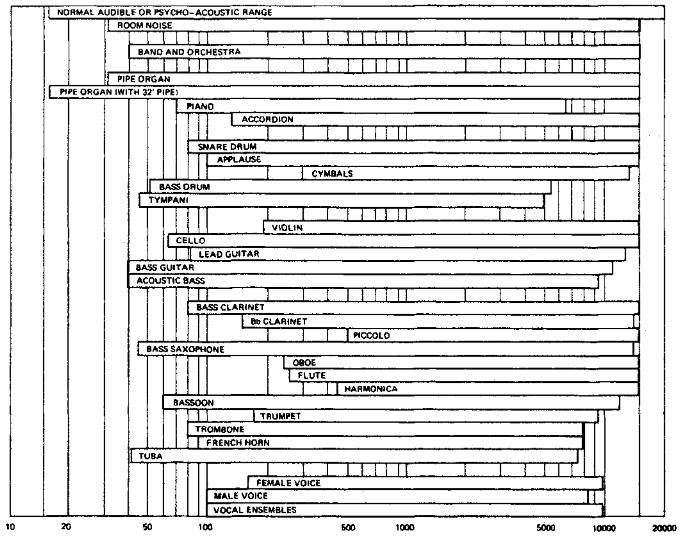


FIGURE 18 – Range of Sounds

MICROPHONE IMPEDANCE

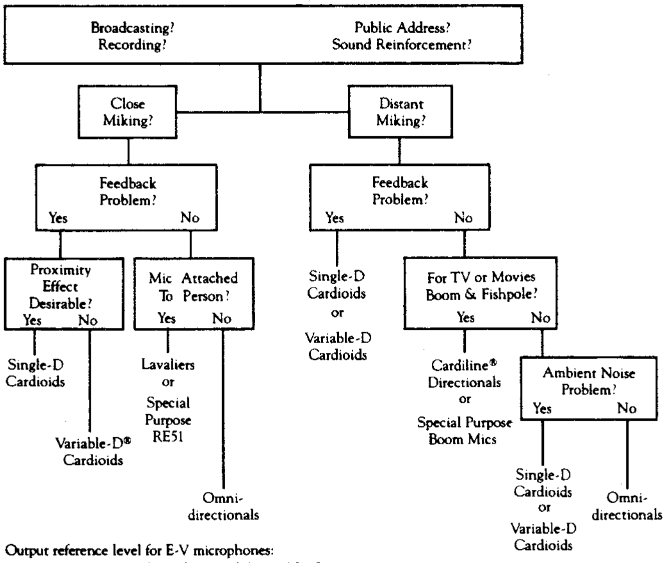
Choosing Between Low-Z and High-Z Microphones. Microphone impedance is the fourth factor that determines microphone type. High impedance microphones have higher output than low impedance types (about 20 dB). However, low-Z microphones permit the use of longer cables without high-frequency rolloff. Therefore, if microphone cables will be longer than fifteen or twenty feet, only low-Z microphones should be used if the maximum clarity of extended high-frequency response is desired! Low impedance microphones have become the industry standard due to their versatility and the availability of equipment which accept low impedance inputs.

HOW TO CHOOSE THE RIGHT MICROPHONE

Knowing how microphones operate and taking into consideration frequencies of sound, pickup patterns, impedance, and proximity effect, you should now be able to choose an appropriate microphone. To accomplish this use the chart shown in Figure 19. It will allow you to pick the microphone that fits the application. Start at the top of the chart and work your way down. Answer the questions in each box and the chart will indicate the type of microphone recommended for your application.

OPERATING TIPS

Since we are dealing with the subject of microphones, a few tips on applications should be of interest. As mentioned earlier, we intend to devote a future supplement totally to applications, but we would like to include a few of the more important topics right now.



Output reference level for E-V microphones:
Hi-Z & Transistorized 0 dB = 1 Volt/Dyne/CM²
Lo-Z 0 dB = 1 Milliwatt/10 Dynes/CM²

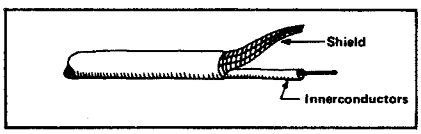
FIGURE 19 – Microphone Application/Recommendation Chart

Impedance Matching for Dynamic Microphones. In usual practice, high-Z microphones operate properly when connected to high-Z mixer inputs only. Connection to a low-Z input results in drastic low-frequency attenuation. Low-Z microphones are designed to operate through low-Z inputs. However, they will usually operate in high-Z inputs when the sound system has sufficient gain, and the microphone output level is large. This technique, incidently, is often used to control input overload since a voltage drop of approximately 20 dB usually results when a low-Z microphone is moved from the mixer's low-Z input to its high-Z input.

Connecting the Microphone to the Mixer Input.

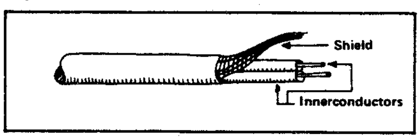
Hi-Z Cable. High impedance microphone cables are single conductor shielded, as shown in Figure 20. The output of a dynamic microphone voice coil is carried by the inner conductor and the shield, which also acts as ground to prevent hum. High impedance mixer inputs have two connections, with the shield going to the mixer's ground. Because one of the microphone voice-coil leads is connected to ground in such hook ups, the inputs are called unbalanced.

FIGURE 20 – Single Conductor Shielded Cable



Lo-Z Cable and Inputs. Low impedance microphone cables have two inner conductors and a shield, as shown in Figure 21. In such low impedance cables, the voice coil signal is carried on the two inner conductors, while the shield acts only as a hum and noise protector. This configuration is termed balanced line since neither voice coil wire is connected to ground. The balanced line arrangement provides hum and noise protection superior to the unbalanced lines used with high-impedance microphones. Low impedance microphone mixer inputs generally have three connections, with the shield going to ground.

FIGURE 21 – 2- Conductor Shielded Cable



Occasionally, mixer inputs for low-impedance microphones will have two connections like typical high-impedance inputs. Such low-impedance inputs are unbalanced similar to their high-impedance counterparts. In order to use an unbalanced input with a standard low-impedance microphone cable, one inner conductor must be connected to ground before the system will work. This must be done by connecting either one of the inner conductors to the shield at the mixer input.

Avoiding Multiple-Microphone Interference. Voids which can occur in frequency response when the outputs of two microphones are combined can be avoided if the microphones are at least three times as far apart as either is from the user. The 3-to-1 ration is illustrated in Figure 22. The ratio is frequently violated in stage microphone placement and is often the culprit for strangely inadequate sound-system performance. Some illustration of "good" and "bad" multiple-microphone placements are shown in Figures 24, 25, and 26. The "bad" placements can ruin the performance of an otherwise excellent sound system!

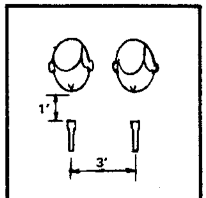


FIGURE 22

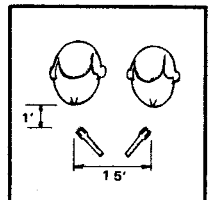


FIGURE 23

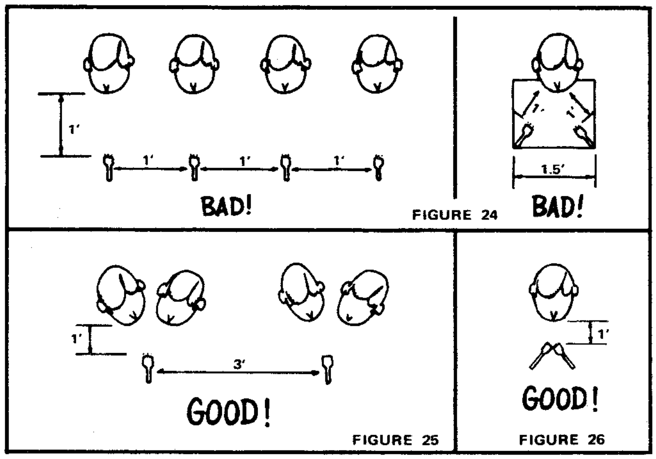


FIGURE 24

FIGURE 25

FIGURE 26

When two microphones must be used close together, multiple interference can be avoided by placing the heads directly together. (See Figure 26)

When cardioid microphones are used, the 3-to-1 ratio can be reduced somewhat by angling the microphones away from each other as shown in Figure 23.

The 3-to-1 ratio studies were first performed by Lou Burroughs, and are described in Lou's book, MICROPHONES: DESIGN AND APPLICATION. This is an excellent text for the microphone user. The book is published by Sagamore Publishing Co., Inc. (1120 Old Country Road, Plainview, N. Y. 11803).

We intend to expand on microphone uses in later issues of the P. A. Bible, as well as covering other subjects of interest to P.A. practitioners. We would appreciate receiving your comments about the first issues of the P. A. Bible and would welcome suggestions regarding future issues. Send your comments to P. A. Bible, Electro-Voice, Inc., 600 Cecil Street, Buchanan, Michigan 49107.

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THE PA BIBLE

ADDITION NUMBER TWO
POWER HANDLING CAPACITY

Excessive power is one of the worst enemies of a loudspeaker. Overpowering is probably the number one killer of loudspeakers, with rock bands jamming their guitar necks through the cones being a close second. To understand how speakers operate and what happens when they fail to do so will be covered in the following discussion.

LOUDSPEAKER PARTS AND OPERATION

Loudspeakers convert electrical energy into acoustic energy, which you hear, and another form of energy, heat, which you don't hear (more on heat later). To understand what happens in a normal operating cone loudspeaker (woofer), see Figure 1. When alternating electrical energy (power) is applied to the leads of the voice coil, it creates forces which interact with the magnetic field in the gap of the magnetic assembly. This interaction results in cone motion (excursion). Since the electrical energy is alternating in nature, the speaker voice coil will move in and out of the gap. Because the voice coil is rigidly connected to the cone, the cone, spider and surround move with the same motion as the voice coil. This motion moves air and this produces sound. Everything works great until some part is stressed beyond its design limits.

HOW POWER DESTROYS LOUDSPEAKERS

There are two different ways of overpowering a loudspeaker. First, excessive power applied over a long period of time will cause "thermal failure" of the loudspeaker by heating the voice coil to the point where some part of its structure literally melts, breaks, or actually burns up. Temperatures in the voice coil may exceed 300 degrees fahrenheit under normal operating conditions. Second, "mechanical failure" occurs when excessive power input moves the cone so far that the cone becomes separated from the voice coil form, the voice coil separates from the coil form, the surround rips, or the spider rips. Any one of these will cause the loudspeaker to fail. When the surround or spider rips, this will eventually lead to voice coil rubbing because the cone assembly is not suspended properly. Small rips will usually go unnoticed, but after some time they become large ones, and the failure is soon to follow.

Failure can be a composite of the two above failures. For example, when a very large transient is delivered by the power amplifier (such as dropping a microphone on the stage), the loudspeaker will try to reproduce the waveform. This **may** end up with the voice coil assembly traveling outward so far it leaves the magnetic gap and, when it tries to return, it may be stressed off center and miss. This causes the whole operation to become locked up with the cone protruding forward from its normal rest position. Even though the loudspeaker is motionless, power is still being applied to the voice coil. Since the coil is out of the gap (which, under normal conditions, acts as a nice heat sink to keep temperatures of the voice coil in operating range) it will quickly overheat and burn.

A properly designed woofer should be able to sustain the necessary excursion at the manufacturer's rated power and frequency range. High power woofers can usually handle maximum mechanical excursions of about one inch peak-to-peak if distortion is ignored. If distortion is considered, the linear, not mechanical, excursion limits might be 1/8 to 1/3 this value, depending on the particular design.

So far this has dealt only with cone loudspeakers (such as woofers). The same failures common to cone speakers can occur in midrange loudspeakers or tweeters. A mechanical failure which may occur in some high-performance compression drivers is dome shattering. Shattering of metal domes is caused by overstressing the metal by constant flexing. What happens when the dome is over-flexed is that it

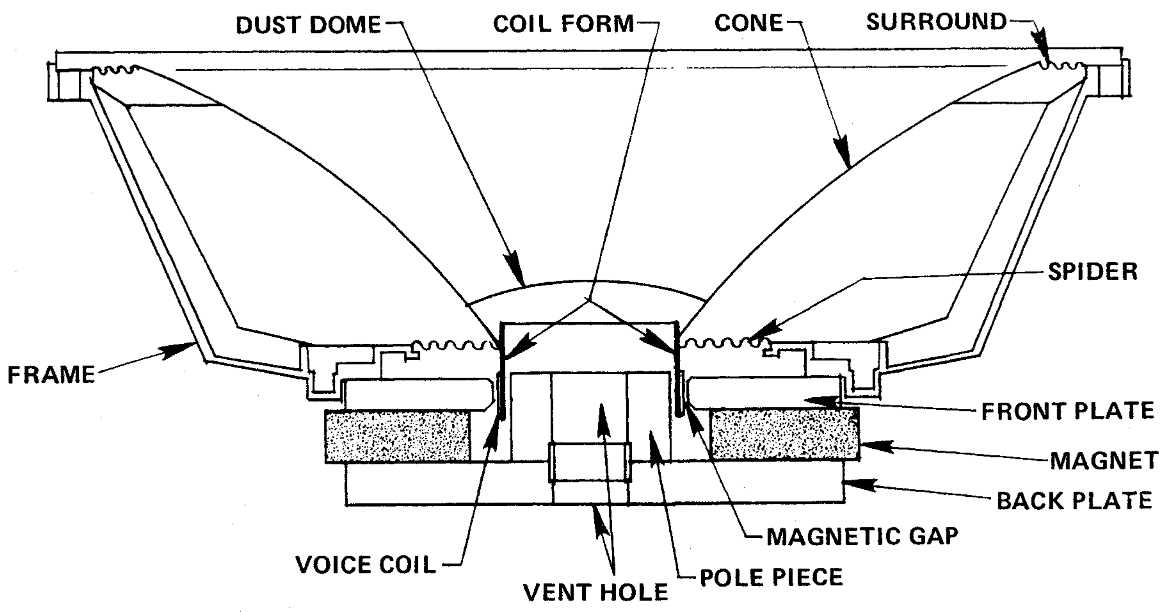

FIGURE 1 — Parts of a Cone Loudspeaker

breaks up into many small pieces similar to shattered glass and ceases to function. (See Addition Number 1 of our "PA Bible" for details on high-performance drivers.)

THE RELATIONSHIP BETWEEN THERMAL AND MECHANICAL FAILURE

Real-world vocal or instrumental program material has, first, a "long-term average level" (anything from a second on up) which pretty much determines how "loud" the sound is to our ears. Second, this material has short duration peak power levels (small fractions of a second) which can be on the order of ten times the long-term average levels. Although these peaks contribute little to perceived loudness, they are necessary for clean, accurate reproduction. It is the long-term average power input which heats up the voice coil (thermal failure). The short-duration peaks, even if they are many times greater in maximum power than the long-term average, will not damage the voice coil thermally because they are not applied long enough to increase its temperature. Also, the peaks found in typical real-world program material – say 4-to-10 times the long-term average (6-to-10 dB) – will not damage the usual speaker mechanically, either. However, large peaks from various "accidents" (like dropping a microphone or electronics turn-on transients) can cause enough mechanical stress to damage the speaker.

THE RATING GAME

Since every manufacturer would like to claim their loudspeaker can handle more power than anything else on the market, there are almost as many power ratings as there are manufacturers. The Electronic Industries Association (EIA) and the Audio Engineering Society (AES) have attempted to define standard testing procedures for the industry, but most manufacturers have not adopted the outlined methods for one reason or another. Therefore, you will find ratings with absolutely no definition at all as well as ratings that are well defined, but do not effectively accomplish a power test that means much in the real world. Some of the words are "program," "continuous," "peak," and "music power."

TEST SIGNALS

In order to understand how rigorously a speaker or speaker system is being power tested, you need to know two things about the input signal. First you need to know what frequencies are being applied. For instance, if you use a 1,000 Hertz single tone in a system with multiple drivers and an appropriate crossover network, only the speaker which reproduces the 1,000 Hertz tone is being tested. Such a test would tell you nothing about how rugged the woofer and tweeter were.

Some sort of real-program material would provide a more realistic test, in that the various frequencies that make up the real material would be appropriately fed to the test speaker. The problem with using program material is that it is not repetitive enough. In other words, it is pretty hard to get real "program material" out of a lab instrument in a repeatable way. Another possibility is to make use of some sort of "random noise." Random noise, over a period of time, contains all frequencies. It sounds quite similar to inter-station hiss on an FM tuner without inter-station muting. "White noise" is a type of random noise which has, over time, equal power at every frequency (for example, the same power in the region from 40 to 80 Hertz as in the region from 4000 to 4080 Hertz). "Pink noise" is another kind of random noise arranged to produce constant power in each musical octave (like 100-200 Hz, 200-400 Hz, 400-800 Hz, etc.) (The power content in each octave of white noise increases as you go up in frequency since there are many more frequencies in the 10,000-20,000 Hertz octave than, say, the 100-200 Hertz octave.) Both pink and white noise, although they contain all frequencies in a manner similar to general program material, have far more power at the frequency extremes than typical vocal or instrumental program material. Thus, they do not represent a particularly realistic test, as they will subject woofers and tweeters to a **much** more rigorous test than they would generally be subjected to in a real life.

The second thing you need to know about the input signal is how long it is applied. This is especially important in view of the above discussion about how most speakers take much greater power for a very short time than they can withstand for a long period of time. A 1,000 Hertz RMS sine wave test is not very good in this respect. If it is applied for a long time, it tests the thermal limit of the speaker at that frequency. However, the instantaneous peak power of a sine wave is only two times greater than the long-term RMS value. This means that the mechanical portions of the speaker are hardly put to a test at all, since real-program has instantaneous peaks that can be ten or more times the long-term average. Random noise can offer a more realistic test. Random noise can be arranged to have instantaneous peaks many times the average value in the manner of real program material. In addition, tests involving random noise are repeatable.

A MEANINGFUL TEST

Since random noise is closely related to real-program material, it becomes the heart of a meaningful power or "life" test. To just use pink noise would be too demanding on a speaker or speaker system in the extremes of the spectrum (low frequencies and high frequencies). Therefore, shaped or filtered pink noise can be used for a more realistic test. The shaping usually conforms to the bandwidth of the speaker or speaker system being tested or to a standard specification. Even though shaping is applied, the signal can be arranged to contain more energy at extremely high and low frequencies than the typical actual program which the speaker or speaker system would normally be required to reproduce, resulting in a product with an extra measure of reliability.

The noise test signal generates not only the overall "long-term average" or "continuous" level which our ears interpret as loudness but also the short-duration peaks which are many times higher than the average, just like actual program material. The long-term average level stresses the speaker thermally and the instantaneous peaks test mechanical reliability.

So far, noise appears to be a test which duplicates the real-life situation, but there is one more thing – how long will the speaker withstand this punishment? The length of time a speaker can sustain (without failure) a given input is the other factor important to a meaningful test. In actual use, long-term average levels exist from perhaps a second on up, but a practical test involves a number of hours.

Electro-Voice became interested in noise testing in 1968, and has continued to utilize it to present day. The EIA standards adopted a similar test for loudspeakers in 1975 (RS–426). If all manufacturers would subscribe to a standard noise test (such as the EIA test), the end user would have his life simplified considerably. Unfortunately, this is not the case at present, so undefined power ratings are worth about as much as confederate money.

EFFICIENCY VS. POWER CAPACITY OR WANT TO BUY A 400-WATT LOUDSPEAKER?

If all you plan to do when purchasing a loudspeaker is examine the power rating, someday some sharp salesman may sell you a light bulb. Light bulbs can handle lots of power, but they don't put out much sound. Remember what speakers really do? They convert electrical energy into (1) sound (great) and (2) heat (not so great). So, beware of buying a super-power 400-watt speaker on that consideration alone. You might be getting a device that takes all the power claimed but is less efficient than another speaker, therefore giving less actual sound output with a given power amp.

For example, start with a 200-watt speaker that delivers 120 dB at a distance of 4 feet with an input of 200 watts. Then go to a 400-watt speaker that puts out only 117 dB at 4 feet with 200 watts in. Although the second speaker takes more power, it requires a 400-watt input to get the 120 dB you got from the first, more efficient speaker with only 200 watts in. (Remember, to make up the 3-dB difference between 120 and 117 dB you need two times the amp power.) Therefore, before you buy a speaker with a big power rating – even if its a realistic rating and not just some number resulting from adding up the digits on an advertising guy's paycheck – consider its efficiency and see how much SPL it produces with a given power input.

HOW BIG AN AMP CAN I USE WITH MY SPEAKER?

Now that we have a realistic power test all you need to do is determine the power rating of the loudspeaker, buy an amplifier with the exact same size power rating and enjoy perfection – right? **WRONG!** If it were only that easy, everything would work perfectly and people who re-cone speakers would be in the bread line. That is not to say that a 100-watt amp won't work well with a loudspeaker rated for 100 watts (per some specification). It may very well work fine.

The matter of mating an amplifier rating with a loudspeaker system is frankly a pain in the posterior. We feel it would be very difficult for anyone who is intimately acquainted with the details of power testing procedures and loudspeaker failure modes to make absolute statements involving this subject with a clear conscience. Getting that off our chest, we will attempt to offer some guidelines that should be of help in making selections based on having a reasonable base to proceed from. That reasonable base is, testing the loudspeaker with a noise spectrum over a substantial portion of its operational range for a long period of time. It may be the EIA eight-hour form of noise test (RS–426) or some other test at least as rigorous. Unqualified ratings such as "100 watts" or "75 watts peak" are very difficult to deal with since so much is left open to question.

Assuming a reasonable rating to work from exists, we can divide almost all systems into two categories; those using full-range, single-cone loudspeakers (one-way systems) and those which are multi-way (two and three-way systems). We will consider multi-way systems first.

Multi-Way Systems

A. **To use a speaker system to full capacity,** skilled experts in sound system installation and operation will obtain the best results if the power amplifier is 2-to-4 times the long-term average noise power rating of the speaker system. (The woofer rating may be thought of as the system rating if a separate system rating is not given.)

   The caution cannot be made strongly enough, however, that **this arrangement is only for experts** or for those people who can discipline themselves against "pushing" the system for ever-higher sound levels and who can avoid "accidents" such as catastrophic feedback or dropping microphones. Dropping a mike causes a peak which can mechanically damage a speaker. Feedback can thermally destroy speakers, especially the mid- and high-frequency components which handle the frequency range where most feedback occurs. Big amps can destroy speakers with ease, and mishandling will be very expensive.

   The large amplifier in this recommendation is to permit driving the speaker system near to its long-term average rating with enough reserve power left to handle the typical short-duration program peaks which do not harm the speaker. (See "The Relationship Between Thermal and Mechanical Failure" section.) The amplifiers in this recommendation allow short-duration peaks of 3-to-6 dB above the long-term average capacity of the speaker. This makes it easier to approach with program material the speaker's long-term average rating without clipping the program peaks. However, if the level is pushed so that the amplifier starts clipping a substantial portion of program peaks (and you can stand the resultant harsh, irritable distortion), the thermal limit of the speaker may be exceeded.

   Non-experts can safely obtain the "best" or "expert" results by using amplifiers which have special circuitry to do the expert's job. Such circuitry allows the long-term average power output to be set to the speaker's rating, but still allows the entire muscle of the amp for short-duration program peaks. The power amplifiers made by E-V/TAPCO incorporate a powerlock feature which provides this performance.

B. **A more conservative, "nominal" amp size,** which will produce audible results nearly equal to those of the "expert" system, is one equal to the long-term average noise power rating of the speaker system.

   The caution here is to studiously avoid the amplifier clipping described above. Although the small amplifier is less likely to produce damaging long-term average power output, it is more likely to be driven into clipping on program peaks. The harsh, irritating distortion products generated are high in frequency and are thus fed to the high-frequency components of the system. This output is not present in normal program material and can overload tweeters and midranges – creating a one-way system without a tweeter. As a result, if you pick the relatively conservative "nominal" amp size – and end up driving the system into hard clipping – you may end up frying the tweeters sooner than if you had picked the large amp of the "expert" system (above). The problem of over-powering system high-frequency components is minimized if they are protected in some manner, such as relay interrupting devices (such as the Electro-Voice STR) or electronic limiting devices (such as the Electro-Voice High-Frequency Auto Limiting circuit). However, such devices are not foolproof, especially if they are continuously being cycled.

C. **To be very conservative,** one can use an amplifier rated at .5-to.7 times the long-term average noise power rating of the loudspeaker. This will provide an extra margin of safety while still making reasonable utilization of speaker capability. The comments made above on clipping and the generation of potentially damaging high-frequency distortion products apply here also.

One-Way Systems

In general, the suggestions and comments made about multi-way systems apply also to one-way systems. However, the high-frequency distortion products generated by amplifier clipping will not, in general, be damaging.

Musical Instrument Speakers. Special consideration should be given to the one-way speakers typically used in musical instrument applications, such as guitar and bass. Amplifier clipping is often part of the desired "sound," so that the peaks of such program are not very far above the long-term average. Thus, if you take the "expert" option and couple an 800-watt amplifier to a 200-watt guitar speaker you may be more likely to exceed the speaker's long-term average noise rating than if the program material were voice or miked acoustic instruments. The importance of this warning is heavily influenced by your personal playing style.

Bi-Amped and Tri-Amped Multi-Way Systems. Such systems become, in effect, one-way systems when considering amp-to-speaker matching. Be alert to the lower power ratings typical on mid- and high-frequency speakers, and avoid over-powering them with amps the size of those used to drive woofers. (The efficiency of horn-loaded midranges and tweeters is higher than that of a cone woofer, so that a smaller amp is usually appropriate.)

VITAMIN C FOR LOUDSPEAKER LIFE EXTENSION

Some comments on ways to keep your speakers performing for years to come seem to be in order after all this discussion of power testing, blown speakers, and big repair bills.

Actually, loudspeakers are fairly forgiving and dependable devices. When you stop to think, a loudspeaker cone is a piece of paper which may only weigh half an ounce, and the voice coil is sometimes made from wire only .01 inch in diameter! It is pretty impressive that speakers can withstand high power, cycling thousands of times a second without coming apart. If speakers are not mistreated, they can provide many years of excellent service. A few quick pointers may be of use in extending the life of your speaker:

1. Provide some means of high-pass filtering in the electronics to prevent sub-sonic frequencies from bottoming the speaker. Such frequencies are typically below the frequency which the system can reproduce but cause large cone excursions which mechanically strain the speaker mechanism and modulate ("muddy up") the bass which the system is capable of reproducing. A high-pass filter is a filter designed to pass all material above a certain frequency and reject everything below. For instance, if a system only has low-frequency response down to 50 Hertz, a high-pass filter at 45 Hertz would be desirable.

   Some power amps and mixers incorporate high-pass filters of this type; at least one separate high-pass filter is available. The Electro-Voice XEQ-1 active crossover (for bi-amped systems) features a 20-Hz high-pass filter.

2. Don't use speaker connecting cords with AC plugs on them, because some day the speakers will get accidently plugged into 120 volts AC. You might enjoy 60 Hertz for a millisecond, if you are lucky.

3. Look for power amplifiers that have DC (direct current) protection, turn on delay, and peak output indicators. Any or all of these features would be desirable. If an amplifier fails and puts out DC, the voice coil will go critical and melt-down will occur. Turn on delay will keep the amplifier from sending a large transient to the speaker which could cause many bad effects. Peak indicators are useful for operation. They will indicate when you have reached the maximum safe power from the amplifier (remember clipping?).

4. On the subject of power amplifiers, if a system uses a separate power amplifier and mixer, the mixer should be switched on before the power amplifier. If this is done in reverse sequence and the mixer puts out a turn-on transient, the amplifier will amplify it and send it to the speakers. When turning off equipment, turn the power amplifier off first, then the mixer, for the same reason. (This is great unless you left an extension cord where someone will catch his foot on it and unplug the whole system. Being a good guy, he quickly plugs the whole thing in at once and the system takes the lumps.)

CONTEST WINNER

We would like to congratulate Randy Cook of Charlotte, Michigan, for finally finding the mystery mistake. We made an error (obviously on purpose) on page 4, line 10 of "Double Distance Rule Gets You" in the "PA Bible." The word "decreased" should read "increased." Now we know who has been reading closely!

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THE PA BIBLE

ADDITION NUMBER ONE
DRIVERS AND HORNS

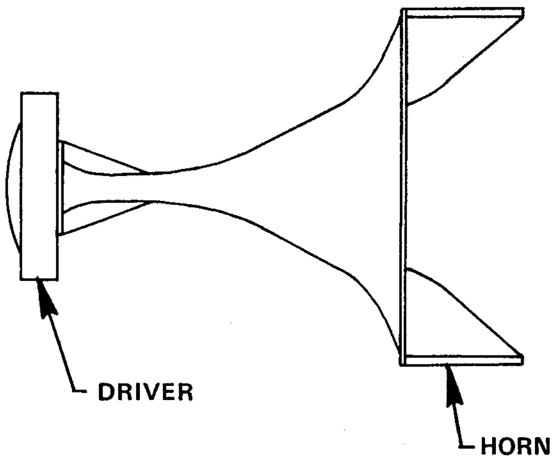

FIGURE 1 – Simple Driver and Horn View

No, this is not about a traffic jam in downtown Buchanan. This is the first supplement to the Electro-Voice "PA Bible." Although it is not absolutely necessary to have read the "Bible" to understand the following material, we believe you will find answers to many of your acoustics application problems in it. In the following material we will deal with the nature of horn drivers, the horns they must be attached to, and, most importantly, how to choose reasonable gear of this type for your sound system.

DRIVER INTRODUCTION

The driver, sometimes called a compression driver, high-frequency driver, or midrange driver, is the device attached to a horn which converts amplifier power into sound. See Figure 1.

The driver can produce sound without a horn, but it will sound bad and not be very loud. The horn "couples" the minute motion of the small diaphragm to the air, producing the characteristic high efficiency of the horn-type speaker. This means that a relatively high proportion of your amp's power is turned into the **acoustic** power your ears hear. Drivers are engineered to work best when connected to a suitable horn of the same connecting hole size.

In the material to follow, drivers will be split into two basic classes. These are "PA" or "midrange" units and "high-performance" units. In general, these two types are defined by the ability to reproduce frequencies above 5000-8000 Hertz. The high-performance units have usable response above this frequency range and represent a class of very sophisticated and difficult-to-design audio devices.

The Diaphragm

The heart of the driver is its "diaphragm," the surface which actually moves back and forth to produce the sound you hear. It is attached to, and moved by, the "voice coil" which is what gets connected to your amp. See Figure 2.

The diaphragm is often made of phenolic impregnated cloth on PA or midrange type drivers or aluminum on high-performance or wide-range type drivers. The phenolic diaphragms are usually more durable. However, they often weigh more than their aluminum counterparts and are not as mechanically rigid. This tends to produce limited high-frequency performance, most appropriate for police sirens, factory paging speakers and other places where the primary goal is intelligible voice communication.

Aluminum diaphragms are often used in high-performance drivers because their light weight enables them to reproduce a more extended frequency range than their phenolic counterparts. Some high-performance driver diaphragms are of "composite" construction with the main central part of the diaphragm made of aluminum and the outer suspension part (which allows movement to occur) made from a more durable material. This permits durability to be optimized while still retaining the desirable properties of the aluminum in the main part of the diaphragm. On a sophisticated high-performance driver the diaphragm and its associated voice coil can have a weight on the order of 1/30 of an ounce and yet it can sustain input powers of the order of 30 watts!

The Phasing Plug

In order to maximize efficiency and effectively permit energy originating at the diaphragm to reach the horn attached to the driver, a device called a "phasing plug" is an integral part of driver design. (See Figure 2 again.) The phasing plug in a PA type driver is a simple device which has a small number of openings or channels that will only allow proper channeling of frequencies up to the 5000-8000 Hz range. In a high-performance driver, the phasing plug becomes more complex in design with many openings which allow high-frequency performance to extend to as high as 20,000 Hertz.

High-Frequency Output Roll-off

It should be noted that even though high-performance class drivers may be very sophisticated in concept and design, they cannot produce a constant output of acoustic power at very high frequencies relative to the lower frequencies. This condition is masked by response curves run with the drivers connected to horns of a type which becomes more directive with increasing frequency. However, a high-frequency roll-off will be evident on response curves obtained on terminated acoustic tubes (a semi-standard way of measuring total driver output versus frequency, without the distorting effects of a horn) or (more important for you, the user) on horns which have the desirable characteristic of uniform directivity as frequency increases.

Under these conditions smooth high-frequency power response roll-offs will begin to take place above the 3000-6000 Hertz range. This condition is natural and is caused

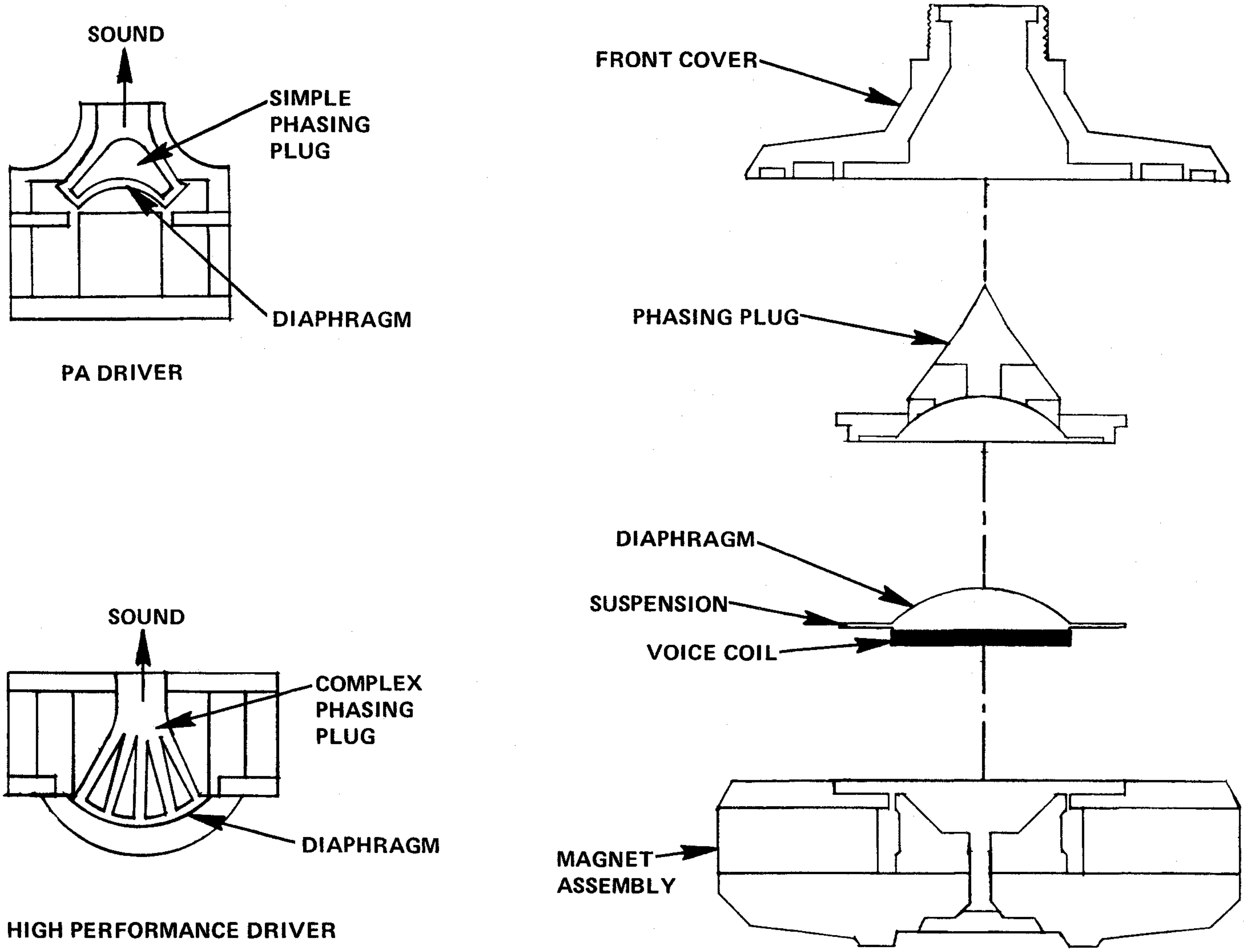

FIGURE 2 – Parts of a Driver
(actual geometry can vary on specific units)

chiefly by the extreme difficulty of making the mass of what must be a durable diaphragm vanishingly small. In practice the natural power roll-off, if smooth, can easily be compensated for by electrical equalization in the form of high-frequency boost. Most well designed drivers are electrically durable enough to take this measure in stride.

Choosing Your High-Performance Driver

You can now see why high-performance drivers are highly desirable for full-range music reproduction. (Two E-V drivers which are high-performance types are the DH1012 and DH1506.) A puzzling matter to many users considering the use of high-performance drivers is how to select an appropriate unit from several offerings. Many driver lineups can be subdivided into large and small units in terms of diaphragm diameter and driver weight (occasionally with the option of all-phenolic or all-aluminum diaphragm construction on a given driver.) A general rule of thumb is that the larger units will be usable to lower frequencies, handle more power, and have more restricted high-frequency output than the smaller models. These characteristics are a consequence of the basic design trade-offs made when the driver was conceived. If low crossover frequencies and/or brute acoustic output are of paramount importance, then the larger driver is probably the best choice. If extended high-frequency output is of more importance relative to the other matters, then the smaller driver is probably indicated. Individual manufacturers' data needs to be consulted in order to make an intelligent selection in any case. In the case of E-V high-performance drivers the significant performance matters are displayed in Table 1.

From these features you can see that the DH1012 is a somewhat more durable driver which is capable of going lower in frequency at the expense of some very-high-frequency output. This does not mean that the DH1012 cannot be crossed over at 800 Hertz or higher. A good rule of thumb is that the highest crossover frequency that will do the job is preferable in order to maximize driver durability in actual usage.

The high-performance drivers are the best choice for maximum efficiency systems which are two way. A two-way system is one which uses the driver to cover the mid and high frequencies (such as 800 Hertz to 20,000 Hertz) with a woofer covering the low frequencies (such as 75 Hertz to 800 Hertz). Although we have not primarily concentrated on this type of unit in this paper, drivers with phenolic diaphragms and simple phasing plugs (such as the Electro-Voice 1824M) are good choices for midrange devices in three-way systems where higher frequencies are handled by a separate tweeter.

Recommended Minimum Crossover Frequency,
- **DH1012:**
  - 500 Hz
- **DH1506:**
  - 800 Hz

Power Handling Capacity (see spec sheet of driver)
- **DH1012:**
  - 40 watts
- **DH1506:**
  - 30 watts

Horn Mounting Method,
- **DH1012:**
  - Bolt-on
- **DH1506:**
  - Screw-on

Highest Usable Frequency,
- **DH1012:**
  - 10,000 Hz
- **DH1506:**
  - 20,000 Hz

Table 1 – Major Specifications for "Sorting" Drivers (E-V)

HORN INTRODUCTION

A horn is a device which performs the function of acoustically coupling sound generated from a driver to the air. Horns are necessary in order to form a complete horn-type transducer.

Horns, unlike direct radiators (cone speakers on a flat baffle), have the unique property of being able to control directionality. By this we mean that it is possible, over a fairly wide range of frequencies, to have the zone of coverage of the energy eminating from the horn to be uniform and controlled. This can be a very useful characteristic in the design of high-quality systems especially when articulate and natural reproduction of the human voice is of importance. (See the "Bible" for more particulars on complete system design.) Horns can be designed to provide various coverage angles such as 90° horizontal by 40° vertical (E-V HR9040A and HR90 horns).

Basic Forms of Horns

The most rudimentary horn designs take advantage of the high efficiency characteristic inherent in horns and have little or no coverage control. These horns are usually relatively small, simple-appearing units which have been designed primarily to perform the basic function of attaining high efficiency. It should be noted that even in the case of simple horns it is possible to do limited forms of directional control – especially if they are intended for midrange frequencies only. Such control is in the form of fairly uniform wide-angle coverage if careful thought is given during the design process. However, control over a wide frequency band, especially for narrow coverage zones, can only be practically achieved through more sophisticated and larger horn structures.

An additional embellishment of the simple design incorporates some directional control in addition to providing the basic horn function. One of the earliest attempts which can still be seen today is the multicell horn (see Figure 3). The multicell horn is usually quite large and is made up of several (sometimes 10 or more) small exponential horns. This is a method of attempting to obtain controlled directionality. This works fairly well at low to mid frequencies, but it creates many finger-like "lobes" at high frequencies as each small section has its own "beaming" problem which eventually shows up at high frequencies.

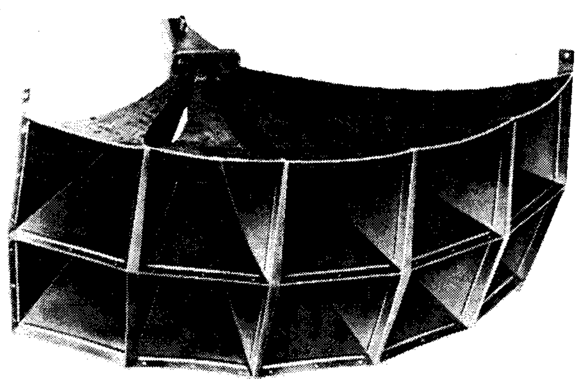

FIGURE 3 – Multicell Horn

Another form of horn is the **simple** sectoral (sometimes called radial) type shown in two views in Figure 4. This horn is made in the form of a sector (like a piece of pie) in the top view. The angle of the sector is intended to define a coverage zone for the horn in a horizontal direction. (In practice this can happen with reasonable precision – especially if care is taken in important secondary design details involving particular attention to the size and shape of the small and large end.) In the other view (side view) the shape is typically determined by some mathematical formula (there are several) which spells out the rate at which the horn flares out in going from the small (driver) end to the big (mouth) end. Because of this, the horn surfaces shown in the side view take on the shape of a smooth curve. The curved surface is not conducive to controlling the coverage angle of the horn in the vertical (up-and-down) direction with the result usually being that in this direction the coverage angle changes depending on what frequency the horn is handling. The coverage angle decreases with increasing frequency. Because of this the potential uniform coverage in the horizontal direction is not matched by uniform coverage in the vertical direction.

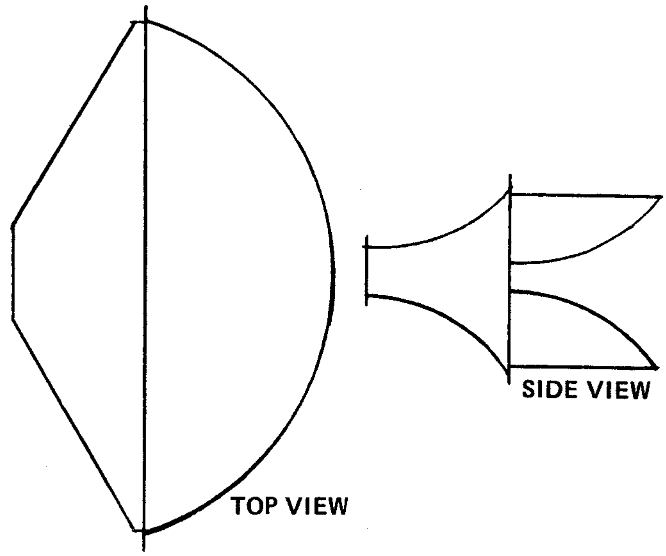

FIGURE 4 – Simple Sectoral Horn

Constant-Directivity Horns

It is possible to achieve more uniform directional control in both the horizontal and vertical directions by using a combination of flare rates. In doing this, enough of the horn must have an appropriate shape so as to preserve the basic function of assuring that the high-efficiency characteristic of the horn (mentioned in the section of rudimentary horns) is maintained over an appropriate range of frequencies. Additional flared sections can then be incorporated to assure tighter directional control in both directions of interest. Horns designed by these techniques can offer well-controlled directionality characteristics that can be an important part of predictable, sophisticated system design for specific application. A product example is the E-V HR series which in 1974 introduced the concept of constant directivity and which uses combinations of exponential and conical flares (U.S. patent number 407112).

Selecting Your Horn

The E-V HR series horns are offered in seven varieties and which one to use for a specific application is a question. Examining one feature will quickly break the group into two categories:

Recommended Crossover Frequency
(Beamwidth Limited)

Models	Crossover
HR9040A, HR6040A, HR4020A	500 Hertz
HR120, HR90, HR60, HR40	800 Hertz

Note that the "beamwidth limited" determines the recommended crossover point. What this means is that because of their relatively small physical size the HR120, HR90, etc., are not capable of controlling their coverage patterns below 800 Hertz. The large HR horns (HR9040A, etc.) can maintain their coverage angles (especially in the vertical plane) approximately one octave lower in frequency. This makes the large horns the best choice for highly refined systems where their size is acceptable.

Note, however, that the small HR horns do provide proper driver diaphragm loading down to 500 Hz so they can be crossed over at 500 Hertz if the decreased pattern control can be tolerated and the driver itself has capabilities down to that frequency.

Appropriate horn coverage angles depend on the application. If long-throw, "tight" patterns are needed to project sound to the back of a large room (consult the "PA Bible" for more particulars about this important matter), the HR40 or HR4020A (40° x 20°) would be recommended. For very wide coverage, the HR120, HR90, or HR9040A (90°/120° x 40°) would be a good choice. For medium-size rooms where some projection is necessary, the HR60 or HR6040A (60° x 40°) would perhaps be indicated. Obviously, this relatively brief discussion will not tell you everything to consider when choosing a horn, but some evaluation of the room (or rooms) you play in and the appropriate specification sheets and, perhaps most importantly a copy of the E-V "PA Bible" can guide you along the right path.

SPECIAL NOTE TO THE READER

The E-V "PA Bible" and this addition have been prepared to help you solve your PA problems. Let us know if you have any other areas in mind for us to tackle.

E-V "PA Bible"
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THE PA BIBLE

ADDITION NUMBER NINETEEN
CONDENSER MICROPHONES

Introduction

In prior additions of the PA Bible, we have covered microphone types (addition number three), microphone techniques (addition number eight) and barrier miking (addition number fifteen). This addition is an expansion of Addition Number Three's discussion of condenser microphone types; what the major differences are, why they are important and when each type would be used.

The Principle of Condenser microphones: Changing Capacitance and Biasing

All condensers work by converting a changing capacitance to a changing electrical signal. A condenser microphone consists of two parallel plates, the diaphragm and the backplate. A polarizing voltage is applied to the plates; which gives a fixed static charge between them. When the diaphragm vibrates in response to sound, the distance between it and the backplate changes. This movement produces a change in the capacitance and therefore the voltage. This changing voltage is the audio signal.

The differences in the types of condenser microphones are in how the charge is developed between the plates. Condenser microphones, in general, exhibit very desirable performance characteristics over other types of microphones. Because the diaphragm has very little weight, it can respond to transients (rapidly changing sound) quickly, giving it excellent high-frequency response.

What is Biasing?

Biasing is the application of voltage to the diaphragm and backplate. It is how this biasing voltage is produced that distinguishes the four major types of condenser microphones. In each case, the changing capacitance is too weak to produce a usable output, so additional gain stages will follow this conversion process.

The Four Types

- diaphragm electret
- back electret
- voltage-biased (true condenser)
- rf biased

How the Different Types of Condensers are Biased: Electret Designs — Two Types

The required bias voltage for electret designs comes from a long-term electrically charged stable film of plastic placed between the conductive diaphragm layer and the conductive backplate. When this plastic is laminated to make the diaphragm, it is referred to as a diaphragm electret. (see Figure 1) When it is laminated to the surface of the backplate it is referred to as a back electret. (see Figure 2)

Although electret designs offer significant size, sensitivity, and cost advantages, there are some compromises. The diaphragm design increases the thickness and mass of the diaphragm which may compromise the high-frequency response of the transducer. The back-electret compromise is a little more complicated, because the design engineer must strike a balance between the best acoustic porting and the optimum electret-biasing charge. The condenser microphone backplate requires acoustic ports in the form of small holes. In the case of back electrets, the size and number of these holes is limited because the bias field is generated from the flat area between these holes. As the number and/or size of these acoustic ports (holes) increase, the areas biased by the electret decreases. Thus, neither the acoustic performance nor the bias voltage (which stabilizes the sensitivity and signal-to noise) can be at their optimum.

In addition to the biasing-charge compromise, all electret designs also include a wider space between the diaphragm and the back plate to make room for the electret material. Increased space lowers the capacitance, which in turn increases the noise floor of the transducer and reduces the efficiency of the basic conversion of sound energy into varying capacitance.

The charge level also varies across the electret surface and can decrease with time or events such as p-pops and wind noise. These events drive the diaphragm into the back plate, significantly reducing electret field strength. Situations that expose the electret material to unusually high temperatures (such as leaving the microphone in the trunk of a car parked in the sun) can also permanently impact the strength of the electret field strength.

Even with these disadvantages, high-quality acoustical performance, nearly that of a true condenser, is achievable with well-designed electret microphones. Electrets often offer high sensitivity in a small package, wide frequency response and inexpensive electronics circuitry. For all but the most demanding applications, electret-condenser microphones offer performance that is quite adequate.

Figure 1: Diaphragm-Electret Condenser Generating Element

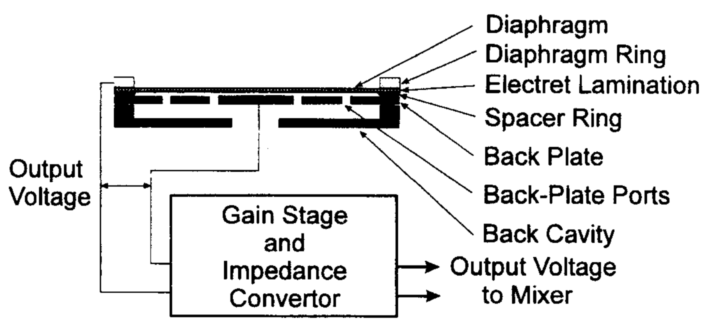
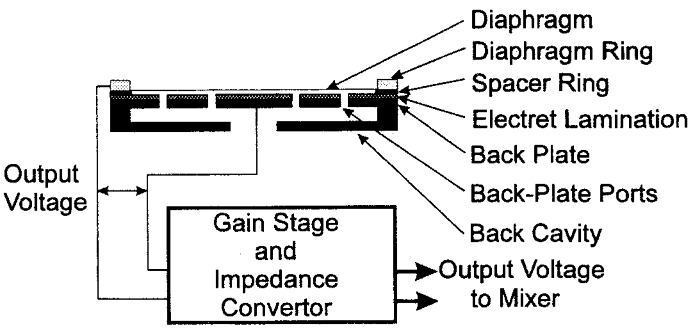


Figure 2: Back-Electret Condenser Generating Element

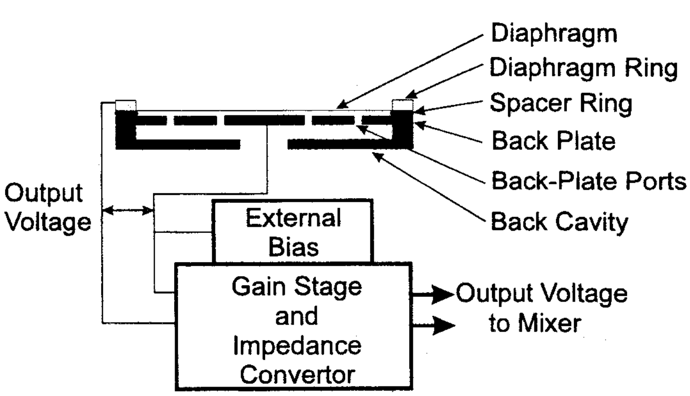


Voltage Biased

The voltage-biased condenser microphone is also referred to as true condenser, in recognition of their design being the basis for the condenser microphone's excellent reputation. This method was first employed before the introduction of stable electret materials in the early 1960's. Voltage-biased condensers develop their charge from external electronic circuits, either by battery, dc/dc converter or directly from the console's 48-volt phantom supply. The true-condenser microphone typically offers better stability, sensitivity and performance characteristics over the "more-modern" electret design.

First, the sensitivity is directly related to the uniformity and stability of the polarization voltage between the diaphragm and the back plate. Unlike the electret design, the polarization voltage for the true-condenser design is developed by an external circuit and is uniform across the transducer's entire back-plate surface. Since the entire surface of the back plate is active in converting sound pressure into electrical energy, sensitivity of each microphone and even between microphones is very consistent. Second, the back plate can then be optimally designed for acoustic performance without concern for electret requirements. Without additional laminating materials occupying the air gap the plates of the true-condenser element can be placed closer, increasing the total capacitance. This ultimately lowers the self-noise generated by the microphone and reduces the problem areas that generate stray capacitance that, in turn, reduces the output of the transducer.

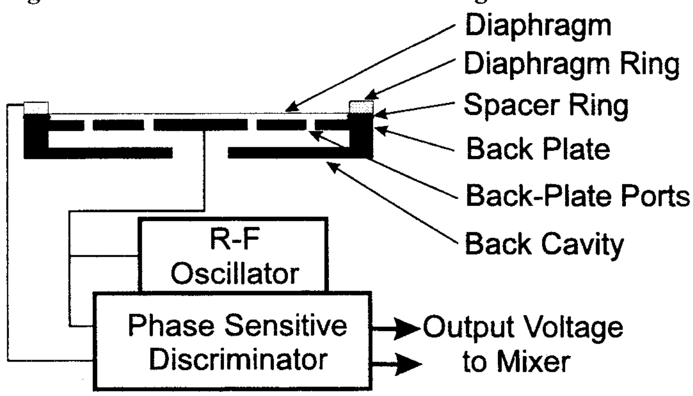
Figure 3: Voltage-Biased True-Condenser Generating Element



rf Biased

The rf-biased condensers have the advantage of no electrostatic attraction between the diaphragm and the back plate. This means that the tension on the diaphragm can be adjusted to increase that microphone's sensitivity and extend the low-end roll-off frequency of the transducer. Long-term stability, however, can often be a problem because the transducer is actually a part of the rf circuit; so anything that could detune the circuit, such as dropping the microphone or temperature extremes, will effect the stability, sensitivity and linearity (which will increase distortion levels) of the microphone. Aging or mechanical changes in the rf-biasing components can also effect performance. rf designs are very expensive to manufacture, but can offer good sensitivity and low self-noise.

Figure 4: rf-Biased Condenser Generating Element



Condenser Microphone and the Environment

Condenser microphones can be sensitive to temperature and humidity changes. Reasons for this can include (1) temperature changes effecting the diaphragm tension, and thus the tuning or sound of the element (2) humidity changes causing condensation between the plates (the diaphragm and the back plate). This condensation can cause arcing between the plates, producing audible snaps and crackles.

High humidity can significantly impact voltage biased and electret condensers. In the case of voltage-biased condensers, the effects last as long as the transducer's insulators are moist.

Operation of a voltage-biased condenser in this moist state can produce exceptionally high noise, low-end roll-off and significant changes in the frequency response. These effects totally cease once the transducer is dry.

Exposure of an electret to high humidity, whether in operation or not, can result in a permanent bias-field loss. The higher the required field and the easier the entry of moisture, the greater the likelihood of permanent loss.

Operating electrets are somewhat less likely to produce extreme noise than voltage-biased transducers when operated moist. The biasing supply of the voltage-biased condenser can support conduction across insulators. The electret layer, being an insulator, effectively stops this conduction.

Moisture can effect the tuning somewhat and the frequency response of rf mics even more, but it won't make them noisy or severely roll-off the bass response.

Conclusion

The first condenser microphones were true condensers and because of their size, cost and fragility were used mostly in recording and broadcast studios. Electret transducers gave up some performance, but the reduced expense and size of the biasing circuitry provided some advantages. The performance advantages of the true condenser microphones have always made them the choice of recording studios. With the evolution of surface-mount technology, circuits have become smaller in size, less expensive, and more rugged. Electro-Voice engineers have adopted this advance in electronics making true condensers suitable for other applications. This has given the RE series of condenser microphones an exceptional marriage of performance and durability.

The PA Bible has been prepared to help you understand audio principles and solve difficult applications. Electro-Voice intends to expand on the PA Bible in future additions, covering other topics of interest to our customers. Let us know if you have any specific subjects you would like us to tackle. If you are reading a friend's copy of the "Bible" and would like to get your very own

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History of *The PA Bible*, Its Reincarnation, Plus Document List and Subject Outlines

By Jim Long, Telex Communications, Inc.
January 2007

Note from the author: in July of 1963, I started with Electro-Voice, Inc., in Buchanan, Michigan, as an engineering tech "intern." Other than continuing my education, I never left the company. In the late 1960's, I moved from engineering to marketing and have been involved in sales or marketing ever since. I recall with some vividness the creation of *The PA Bible*.

EV's *The PA Bible* is a 16-page document issued in 1979, followed by 19 "additions" issued over a 19-year period on various pro-audio product and applications subjects. The documents were printed in EV's in-house print shop and mailed all over the world at no charge. A few years after the last addition, a bound version of the documents was made available. All versions have long been out of print, but requests continue to come in. Some documents have appeared on the Internet from time to time, of varying quality and longevity, some free and some not. We are happy to now make them all available on the Electro-Voice Web site (www.electrovoice.com). We have discussed from time to time the updating of the documents, since the products used as examples are no longer available. In the end, we decided to reissue them "as is," feeling that the information was in nearly all cases as relevant now as it was some years ago. A few updating and clarifying comments are in the outline section starting on page 3.

Musicians who needed PA's were the target audience for *The PA Bible*. Quoting from the original document, "In this guide we'll address those basic problems, annoyances and questions that plague every musician at one time or another." However, *The PA Bible* and many additions were relevant to those involved in fix installations as well.

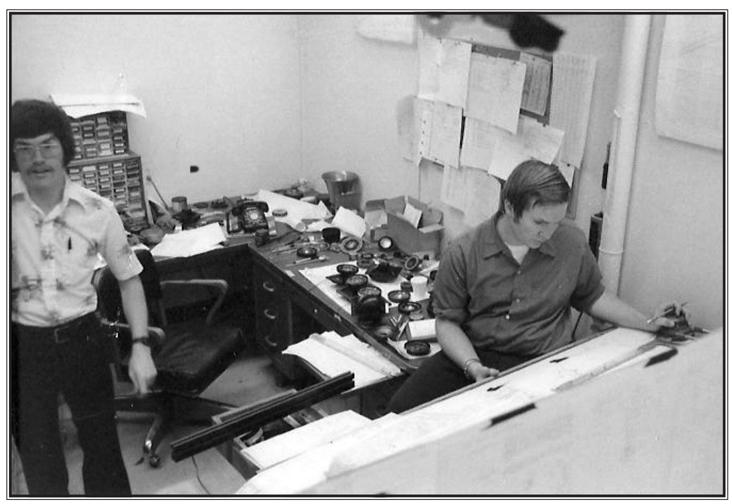


Figure 1. Jeff White (left), author of *The PA Bible*, with Kent Frye, key compression-driver engineer, circa 1977.

The PA Bible, covering "what's wrong with a lot of PA's" and a "basic approach to system design," was largely written by Jeff White, a young loudspeaker engineer who worked at EV from 1976 to 1981. See Figure 1. Jeff worked for Ray Newman (1940-1996), then EV's chief loudspeaker engineer who made a number of major contributions to our industry, including the basic concepts for the design of the now-ubiquitous constant-directivity high-frequency horns and the application of the analyses of A. N. Thiele and R. H. Small to the design of vented low-frequency loudspeaker systems. Jeff set the lighthearted-yet-informative tone of the document and subsequent additions, and wrote a number of them. I also recognize the writing style of Ray in some. I recall being pretty heavily involved in the editing process of many of the documents. Others contributed too.

Recalling the creation of the *The PA Bible*, Jeff, now a loudspeaker consultant in Southern Indiana, wrote in a December 2006 e-mail (slightly edited by the author):

I still have my **original** text that was hand typed 78 pages…with hand drawn images that were recreated by the EV art dept. …The original working title of the thing was:

Using Electro-Voice Components to
Improve Your P. A.

A guide from Electro-Voice Engineering on applying our building-block group of
horns, drivers, bass-boxes and crossovers to make
high-performance sound reinforcement systems.

Well, the main thing that comes to mind when remembering the nights at home drafting this was sitting at my dining room table typing it out and listening to Supertramp's 1979 LP (vinyl!!!) release of "Breakfast In America." I think I was turned onto Supertramp by you with the Stan Ricker re-master of Crime of the Century ("Bloody Well Right") at one of the AES Conventions. Oh well, lots of good memories now. I'm pretty sure I did some cursing at the time with the extra work involved.

Whose idea was *The PA Bible*? This may be lost in audio history, but in recent communication both Jeff White and I recall that EV's president of the day, Bob Pabst, was very high on the idea of educating our dealers and end users with white-paper-type documents so they would want EV components for their PA. Jeff and I don't recall who came up with the name, but it sounds like something Ray would do. Jeff says for certain that Ray was motivation, inspiration, support and concept. Jeff characterizes himself as author/grinder and basic layout of material. He labels me as renovate, rearrange, revise and transform.

Document List

All documents are listed below in Table 1. Following that, a subject outline with selected comments is given for each document. This may facilitate the choice of documents to download.

Titles	Pages	Date
The PA Bible	16	1979
Addition Number One: Drivers and Horns	4	1979
Addition Number Two: Power Handling Capacity	4	1979
Addition Number Three: Microphone Types	4	1980
Addition Number Four: Understanding Equalization and the Various Types of Equalizers	8	1980
Addition Number Five: System Interconnection	6	1980
Addition Number Six: The Con'·stănt Di·rec·tiv'i·ty White Horn Paper	4	1980
Addition Number Seven: Crossovers and Biamping	4	1981
Addition Number Eight: Microphone Techniques	6	1981
Addition Number Nine: Mixing for the Live Performance	4	1982
Addition Number Ten: A Central Cluster System for Rock and Roll	4	1982
Addition Number Eleven: Portable Sound Systems for Small Clubs	4	1982
Addition Number Twelve: "Force®" Boxes for Music Systems	3	1983
Addition Number Thirteen: The Electric Guitar Loudspeaker, a Unique Design	3	1983
Addition Number Fourteen: Loudspeaker System Types	4	1984
Addition Number Fifteen: Barrier Miking	3	1985
Addition Number Sixteen: Mismatching Drivers and Horns	4	1986
Addition Number Seventeen: What is Manifold Technology™?	3	1987
Addition Number Eighteen: Controlled Systems	3	1991
Addition Number Nineteen: Condenser Microphones	3	1997

Table 1. Titles, pages and dates for *The PA Bible* and additions.

Detailed Subject Outline with Selected Comments

The PA Bible (16 pages, 1979)

In the "Double Distance Rule Gets You" section on page four, line ten, "decreased" should be "increased." This mistake was mentioned at the end of the second addition but never corrected in *The PA Bible* itself. Don't miss the two dudes in Figure 16:

1. Table of Contents
2. Is Electro-Voice going into the publishing business?
3. How to read this guide
4. What this booklet is about
5. What's wrong with a lot of PA's
    A. Low-efficiency speaker systems
    B. Not enough amplifier power
    C. Poor frequency response
    D. Highs miss half your audience
    E. Double-distance rule gets you
    F. Room reverberation swamps your voice
6. Basic approach to system design
    A. Small size room
    B. Medium size room
    C. Large size room
    D. Monitor systems
    A. Some thoughts on permanent installation systems

Addition Number One: Drivers and Horns (four pages, 1979)

1. Driver introduction
    A. The diaphragm
    B. The phasing plug
    C. High-frequency output roll-off
    D. Choosing your high-performance driver
2. Horn introduction
    A. Basic forms of horns
    B. Constant-directivity horns
    C. Selecting your horn

Addition Number Two: Power Handling Capacity (four pages, 1979)

1. Loudspeaker parts and operation
2. How power destroys loudspeakers
3. The relationship between thermal and mechanical failure
4. The rating game
5. Test signals
6. A meaningful test
7. Efficiency vs. power capacity or want to buy a 400-watt loudspeaker?
8. How big an amp can I use with my speaker?
    A. Multi-way systems
        i. To use a speaker system to full capacity
        ii. A more conservative, "nominal" amp size
        iii. To be very conservative
    B. One-way systems
        i. Musical instrument speakers
        ii. Bi-amped and tri-amped multi-way systems
9. Vitamin C for loudspeaker life extension

Addition Number Three: Microphone Types (four pages, 1980)

This Addition was based on *The Microphone Primer*, written for musicians circa 1968 by Jim Long and an executive of the Ludwig Drum Company, who at the time was distributing EV products to music dealers. Even as basic as it was, Lou Burroughs, EV's "Mr. Microphone," handed out this primer in his professional broadcast and recording seminars.

1. Microphone types and operation
    A. Ceramic and crystal generating elements
    B. Ribbon (or "velocity") generating elements
    C. Dynamic generating elements
    D. Condenser generating elements
2. Microphone pickup patterns

A. Omnidirectional pickup pattern
B. How does an omnidirectional microphone work?
C. Why an omnidirectional microphone?
D. Unidirectional pickup pattern
E. How does a unidirectional microphone work?
F. Why a cardioid microphone?
G. Two vastly different types of cardioid microphones

3. Microphone frequency response
4. Microphone impedance
   A. Choosing between low-Z and high-Z microphones
5. How to choose the right microphone
6. Operating tips
   A. Impedance matching for dynamic microphones
   B. Connecting the microphone to the mixer input
      i. Hi-Z cable
      ii. Lo-Z cable and inputs
   C. Avoiding multiple-microphone interference

Addition Number Four: Understanding Equalization and the Various Types of Equalizers (eight pages, 1980)

Material for this addition was supplied by Larry Blakely, a well known audio and recording industry consultant and writer of the day. His material was edited and augmented by Jeff White and Ray Newman. Page 8, very briefly, mentions room equalization using a real-time analyzer, which can only measure the total sound field in a room. 1980 was of course a long time before the currently popular PC-based analysis systems, which can under the proper conditions provide quasi-anechoic information, with a more useful predominance of the loudspeaker's direct field.

1. Introduction
2. Frequencies
3. Octaves
4. Fundamentals and harmonics
   A. Sub-sonic frequency range – 1 Hz to 20 Hz (approximately 4 octaves)
   B. Very low bass frequency range – 20 Hz to 40 Hz (1 octave)
   C. Bass frequency range – 40 Hz to 160 Hz (2 octaves)
   D. Lower mid frequency range – 160 to 315 Hz (1 octave)
   E. Mid frequency range – 315 Hz to 2,500 Hz (3 octaves)
   F. Upper mid range or presence frequency range – 2,500 Hz to 5,000 Hz (1 octave)
   G. High frequency or brilliance frequency range – 5,000 Hz to 10,000 Hz (1 octave)
   H. Extreme high frequency range – 10,000 Hz to 20,000 Hz (1 octave)
5. Equalizer characteristics
   A. Shelving
   B. Peak/dip
6. Basic types of equalizers and their applications
   A. Tone controls
   B. Two knob equalizers (selectable frequency)
   C. Three knob (three frequency) fixed frequency equalizer
   D. Four knob equalizers (fixed and selectable frequency)
   E. Graphic equalizers
      i. ISO center frequencies
      ii. One octave graphic equalizer
      iii. 1/3 octave graphic equalizer
      iv. 1/2 octave graphic equalizer
   F. Sweepable frequency equalizers
   G. Parametric equalizers
   H. Paragraphic equalizers
7. Basics for the use of equalizers
8. Know the fundamental frequencies of voices and musical instruments
   A. Vocals
   B. Electric bass
   C. Drums
      i. Bass drum
      ii. Snare drum
      iii. Tom toms

        iv. Cymbals
9. Room equalization (room voicing)
10. Do not use equalization to excess

Addition Number Five: System Interconnection (seven pages, 1980)

Some topics are perhaps by now superfluous, e.g., (1) discussion of high-impedance microphones, which have essentially disappeared from the scene, (2) the overloading of microphone inputs, largely a thing of the past given today's input trim pots and (3) the use of 1/4-inch phone plugs for portable speakers, largely replaced by the Neutrik Speakon® connectors. Also, the discussion of decibels does not describe the dBu, today's most common method or rating the voltage output of mixers and line-level electronics (0 dB = 0.775 volts).

1. Interconnecting the system
2. Low level connections
    A. Attenuators
    B. Impedance
    C. Phantom connection
    D. Connectors
3. Line level connections
    A. Balanced versus unbalanced
    B. Interference pickup
    C. System ground
    D. Crossovers
4. Power amplifier connections
    A. Load impedance
    B. Polarity
    C. Output impedance matching transformers
5. Definitions
    A. Decibel (dB)
    B. dBV
    C. dBm
    D. Active devices
    E. Passive devices
    F. Impedance
    G. Ground loop
    H. RFI
    I. SCR hash
    J. Signal

Addition Number Six: The Con'·stănt Di·rec·tiv'i·ty White Horn Paper (four pages, 1980)

This addition is a great way to find out how horns can be made to spread sound evenly over a wide frequency range. The statement in the last paragraph of the "'Constant Directivity' Defined" section, "varies from here to Newport, Tennessee," refers to the distance between the EV headquarters in Buchanan, Michigan, and the loudspeaker plant in Tennessee (high-performance compression drivers were made in Buchanan). A Jeff White contribution.

1. "Constant directivity" defined
2. What constant directivity horns do for you
    A. Constant directivity horns give a well defined zone of coverage that you can count on
    B. You can use fewer constant directivity horns
3. What makes a constant directivity horn?
    A. A constant directivity horn is fed by a small opening
    B. A constant directivity horn has straight sidewalls over a major portion of its length in both the horizontal and vertical planes
    C. Constant directivity horns have an additional wide-flare section near their mouth openings
    D. Constant directivity horns are usually bigger than conventional horns
4. A driver on a constant directivity horn needs equalization
    A. The Newman Criteria for drivers
    B. Horns affect driver output
    C. Utopia by equalization!
5. Do constant directivity horns require any other special equipment?
6. Why are there a number of different constant-directivity horns?
    A. Coverage angle
    B. Minimum crossover frequency
    C. Minimum frequency for coverage angle control

Addition Number Seven: Crossovers and Biamping (four pages, 1981)

The last paragraph on page 2 tells how the high-frequency distortion products of power-amplifier clipping can destroy the high-frequency sections of passively crossed over speaker systems. This is probably much less likely today because internal protection devices are now so common. Page 4 talks about over equalization sending speaker parts to Hobart, Tasmania (third paragraph). That's not an old EV plant location but the southernmost city in Australia. Another Jeff White contribution, along with sending the speaker recone guy to Hawaii (page 3, third paragraph).

1. Crossover frequency and slope rate
2. Types of crossovers
   A. "Passive" crossovers
   B. "Active" crossovers
3. What to use – active or passive?
4. Why biamp?
5. Pointers for speaker biamping
6. Connecting thoughts

Addition Number Eight: Microphone Techniques (six pages, 1981)

Section I

1. Directional microphones
2. Non-directional microphones
3. Frequency response
4. Proximity effect
5. Placement
   A. Angle
   B. Distance
6. Number of microphones
7. Hints and other miscellany
   A. Elimination of distracting signals
   B. Special microphones
   C. Some general guidelines

Section II

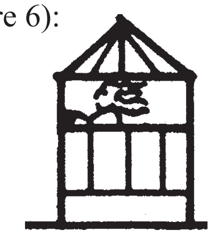
1. Vocalist microphones
2. Electric/bass guitar pickup
3. Piano microphone
4. Acoustic guitar
5. Drum microphones
6. Conclusion

Addition Number Nine: Mixing for the Live Performance (four pages, 1982)

1. Introduction
2. The mixer
3. Planning
4. The set-up
5. The performance

Addition Number Ten: A Central Cluster System for Rock and Roll (four pages, 1982)

In the System Adjustment section, his addition describes loudspeaker level setting and equalization done with a three-mic level averager operating in real time, a useful technique mostly lost in these days of software-based analysis systems. According to Jeff White, the end result had quite an effect on the sound guy's hair (Figure 6):



1. Split stacks
2. Single cluster
3. The equipment
4. System adjustment
   A. Step 1 (setting levels and EQ of short throw)
   B. Step 2 (adding mid-throw horn)
   C. Step 3 (adding long-throw horn)
5. Overall coverage and response uniformity
6. Sound pressure levels

Addition Number Eleven: Portable Sound Systems for Small Clubs (four pages, 1982)

In the "A System for Today" section, this addition describes the EV Entertainer system consisting of two compact, lightweight two-way speakers systems (100S) and a powered mixer. The 100S was the precursor of today's Sx100+ and Sx300.

1. Some history
2. The needed system
   A. What are the criteria of portable equipment?
   B. How much sound is enough?
   C. What frequencies should be covered?
   D. How can we get this good sound to all seats?
3. A system for today
4. Addendum

Addition Number Twelve: "Force®" Boxes for Music Systems (three pages 1983)

"Force" was the moniker applied to EV's "high value" line of musical instrument loudspeakers, less expensive than the original SRO® and subsequent EVM® lines. Designs were based on the vented-box analyses of A. N. Thiele, which first appeared in the US in 1971 issues of *Journal of the Audio Engineering Society.*

1. Introduction
2. Enclosure designs
3. Construction suggestions

Addition Number Thirteen: The Electric Guitar Loudspeaker, a Unique Design (three pages, 1983)

1. The guitar sound approach
2. The special sound quality
3. Cone
4. Coil
5. Dome
6. Gap structure
7. Power handling capacity
8. Power test
9. Venting
10. Cabinets
11. Conclusion

Addition Number Fourteen: Loudspeaker System Types (four pages, 1984)

This addition reads like Ray Newman. Ray was very fond of the "system interrelationship equation" discussed on page 3, second column.

1. Introduction
2. What is a system?
3. Common types of systems
   A. Dipoles
   B. Sealed boxes
   C. Vented boxes
   D. Horns
   E. Combination boxes (horn midrange and vented bass)
   F. Combination boxes (horn bass and sealed box midrange)
4. The Thiele/Small connection
5. Comparing system types
6. Summary and application comments

Addition Number Fifteen: Barrier Miking (three pages, 1985)

1. Introduction
2. Early discoveries
3. Solving the problem
4. Further developments
5. Common questions
6. What is barrier miking?
7. When should barrier miking be considered?
8. What are some typical instrument miking applications?
   A. Piano miking
   B. Drum miking
   C. Miking stringed instruments
9. I've heard that a microphone mounted on a barrier is more sensitive than the same microphone mounted on a stand. How is that possible?
10. What effect does carpeting have on the performance of a barrier-mounted microphone?
11. I've read about using Plexiglas and

plywood panels to improve mic performance.
12. What type of microphones can be used in barrier miking?

Addition Number Sixteen: Mismatching Drivers and Horns (four pages, 1986)

This addition includes a table showing some popular competitive compression drivers of the day (1986), showing diaphragm construction, compression ratio, coil diameter and throat (exit) diameter. *Note: the graphs in Figures 12 and 13 do not appear to match the text just above them.*

1. Introduction
   A. In a jam and/or the show must go on
   B. Availability and "equivalent units" on a sound system "spec"
   C. Listener preference
   D. Some existing equipment on hand
2. Loudspeaker mechanisms
3. "Small format" drivers
4. Large-format drivers
5. Throat adapters
6. Conclusions

Addition Number Seventeen: What is Manifold Technology™? (three pages, 1987)

This addition applies to EV's first "concert sound" systems, the MT-4 two-box, four-way system consisting of the MTH-4 HF box and the MTL-4 LF box.

1. Manifold Technology benefits: more from less
2. Inherent engineering problems
3. Some solutions
4. Some specific realizations
5. Where does all this lead?

Addition Number Eighteen: Controlled Systems (three pages, 1991)

This addition applies to EV's DeltaMax™ compact controlled or processed systems, analog precursors to the contemporary EV Xi-1122A/85F and Xi-1152A/64F systems with digital signal processing.

1. What are controlled systems?
2. Why are such systems of interest and where do they apply?
3. What is being controlled and what are the consequences?
   A. Amplifier gain
   B. Peak limiting
   C. Dynamic frequency response tailoring
   D. Crossover frequency shifting
4. The future of this technique
   A. Box characteristics
   B. Woofer characteristics
   C. High frequency section characteristics

Addition Number Nineteen: Condenser Microphones (three pages, 1997)

This "last hurrah" addition was written to highlight the advantages of EV's then new RE2000 large-diaphragm voltage-biased "true condenser" mic, introduced by EV to compete in the high-end recording market.

1. Introduction
2. The principle of condenser microphones: changing capacitance and biasing
3. What is biasing?
4. The four types: diaphragm electret, back electret, voltage-biased (true condenser) and rf biased
5. How the different types of condensers are biased: electret designs – two types
6. Voltage biased
7. rf biased
8. Condenser microphone and the environment
9. Conclusion