



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.

Table of Contents

1. Is Electro-Voice Going into the Publishing Business? . . .	1
2. How to Read This Guide	1
3. What This Booklet Is About	1
4. What's Wrong With a Lot of P.A.'s	2
A. Low-Efficiency Speaker Systems	2
B. Not Enough Amplifier Power	2
C. Poor Frequency Response	2
D. Highs Miss Half Your Audience	2
E. Double-Distance Rule Gets You	4
F. Room Reverberation Swamps Your Voice	4
5. Basic Approach to System Design	6
A. Small Size Room	7
B. Medium Size Room	8
C. Large Size Room	13
D. Monitor Systems	13
E. Some Thoughts on Permanent Installation Systems	13

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.

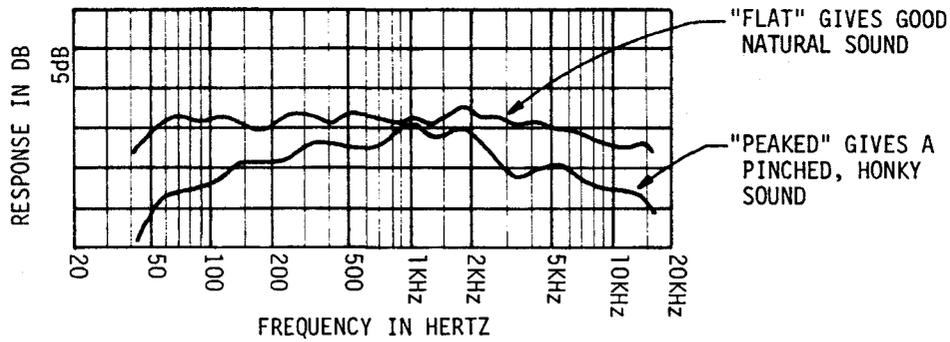


FIGURE 1 – Axial Frequency Response

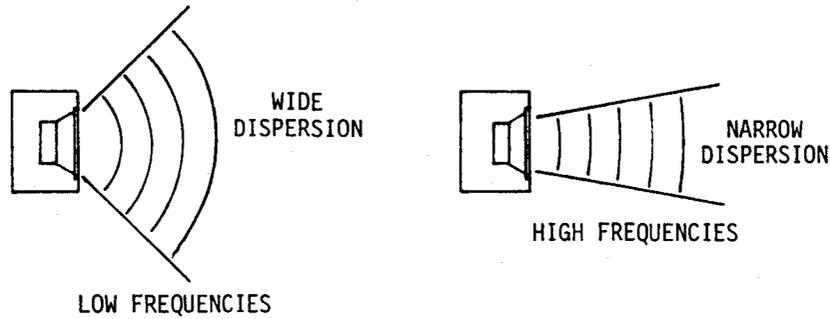


FIGURE 2 – 12-inch Speaker Dispersion

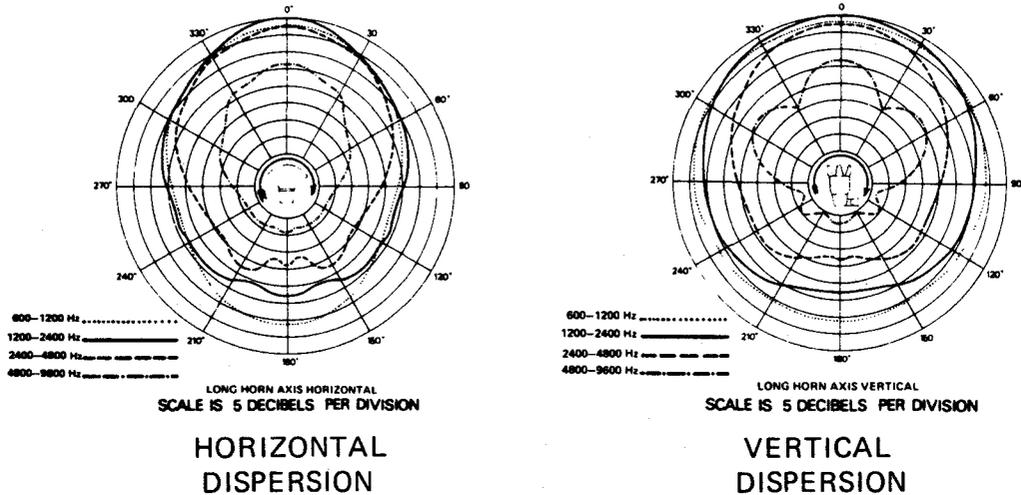


FIGURE 3 – Polar Response, Octave Bands of Random Noise, Horn in Normal Orientation

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-

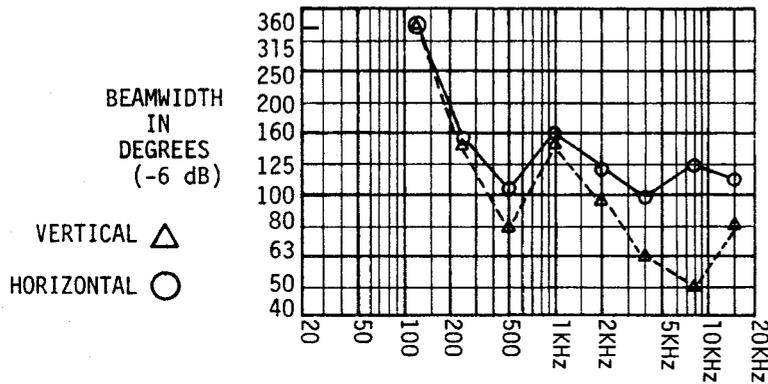


FIGURE 4 – Plot of Coverage Angle (beamwidth) versus Frequency

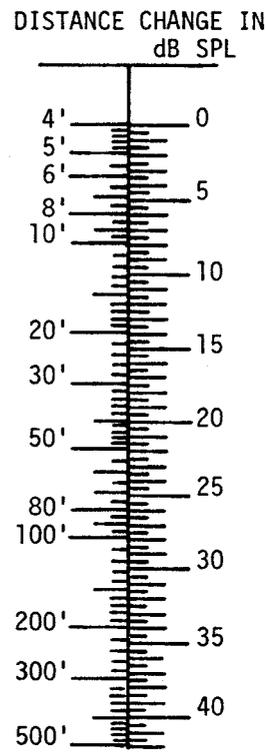


FIGURE 5 – Level Variation with Distance

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.

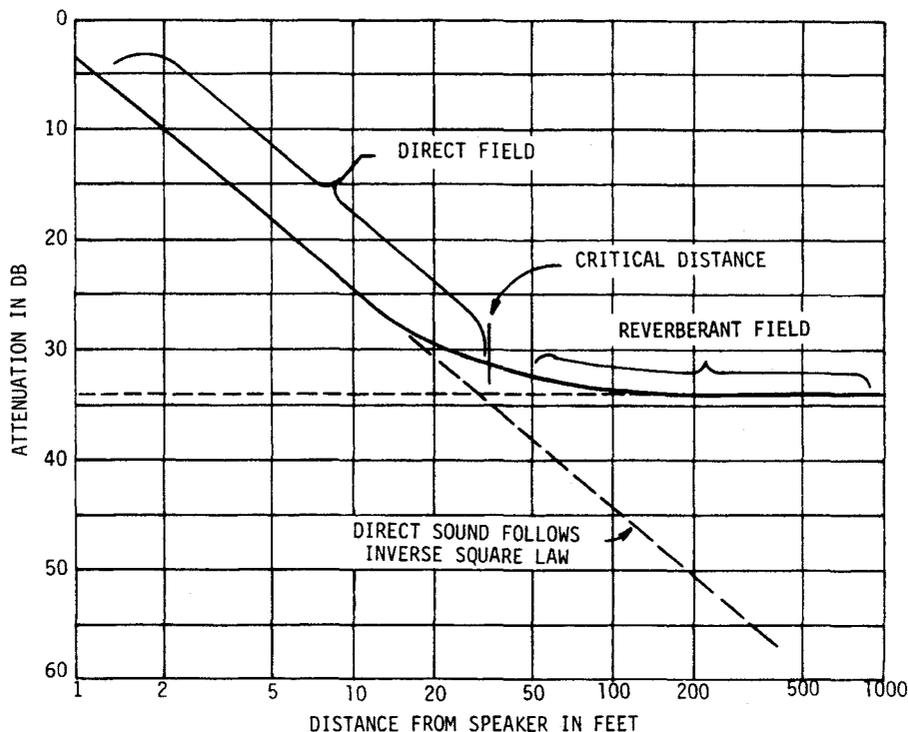


FIGURE 6 – Attenuation with Increasing Distance in Typical Large Room

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-foot room and the large 90,000-cubic-foot 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

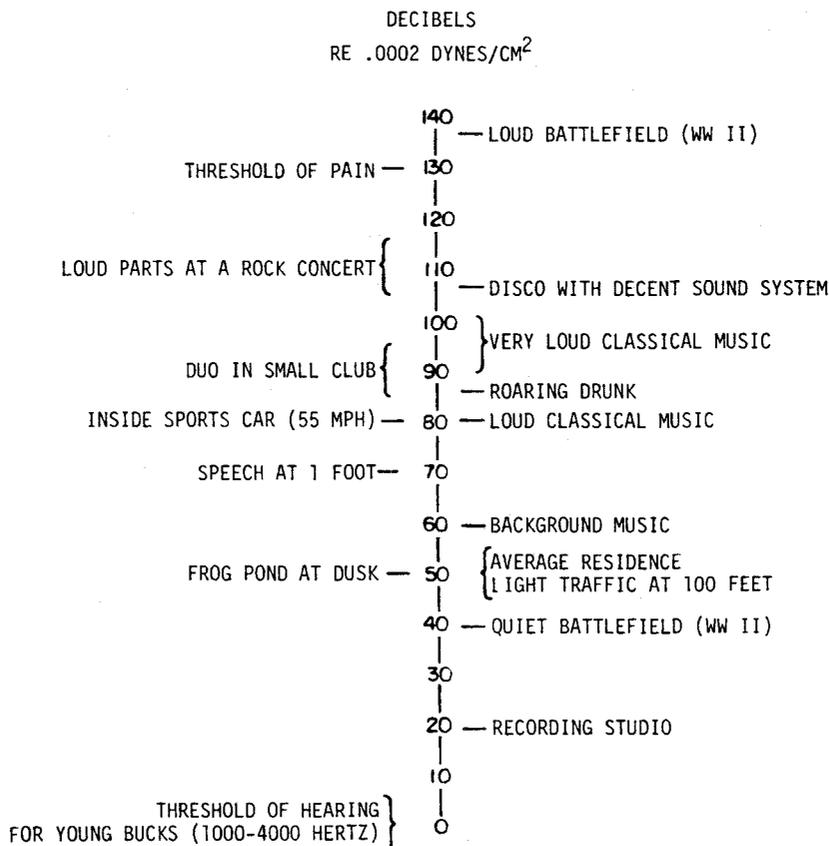


FIGURE 7 – Typical A-Weighted Average Sound Pressure Levels

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.

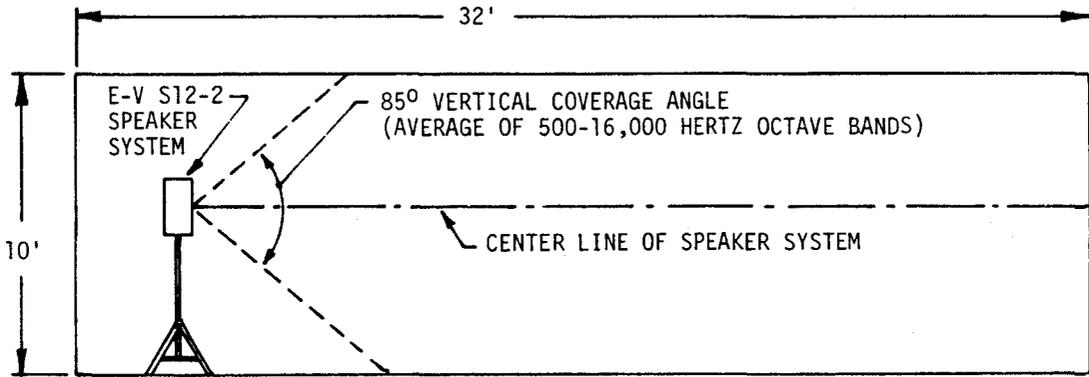


FIGURE 8 – Small Size Room Elevation View

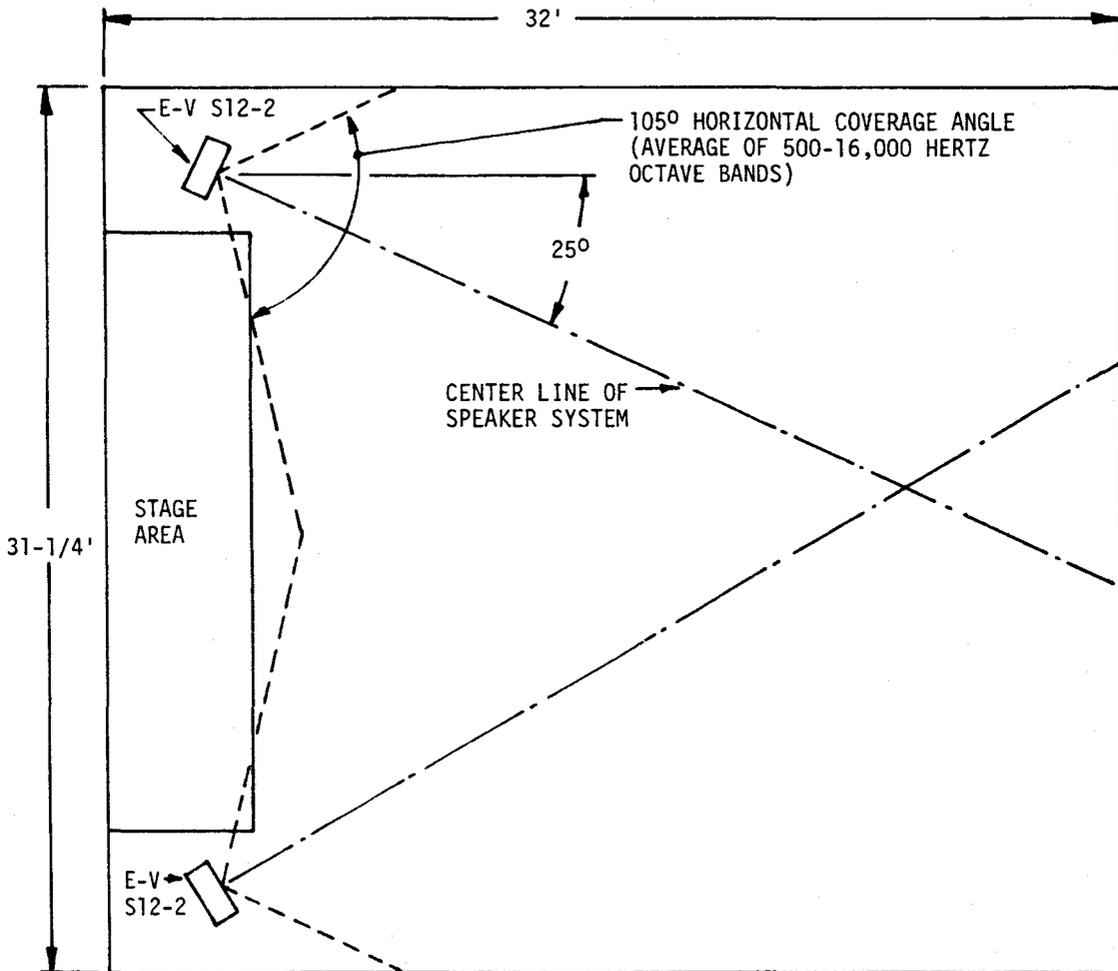


FIGURE 9 – Small Size Room Plan View

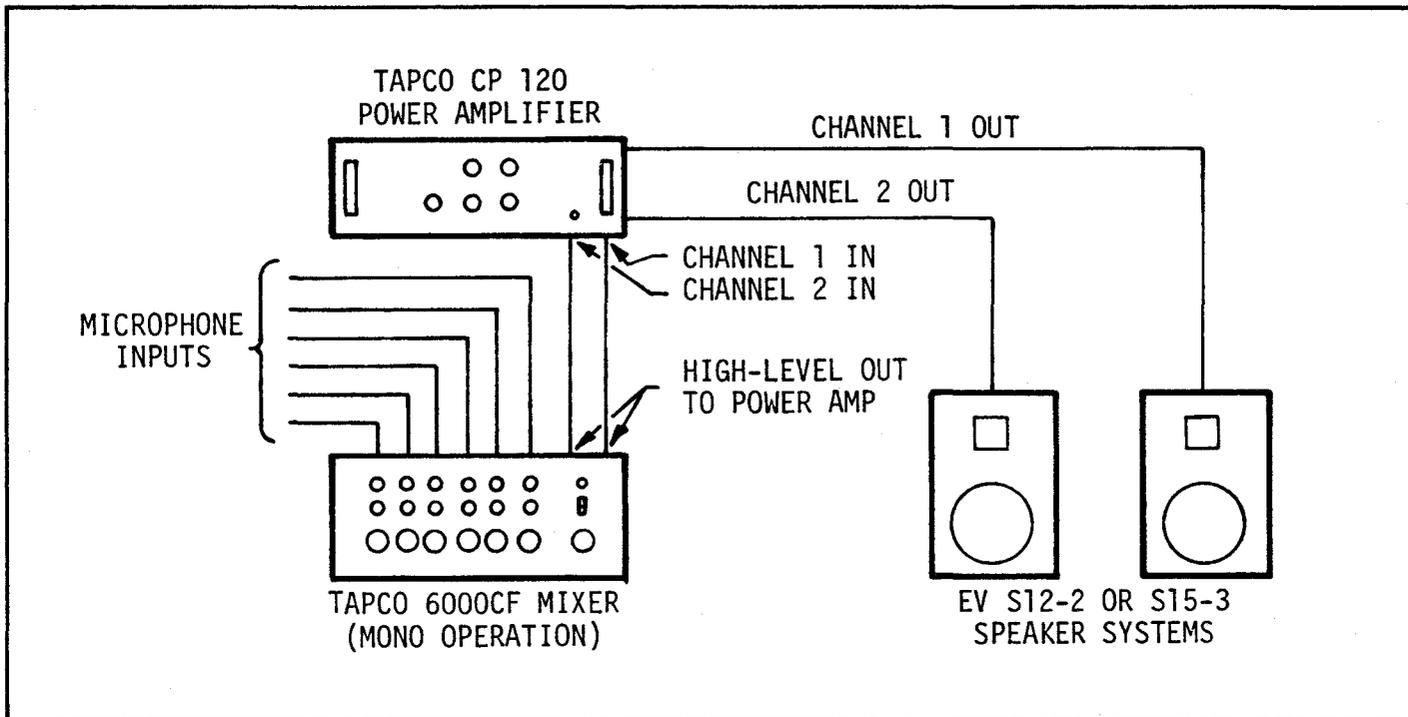


FIGURE 10 – Block Diagram for System Shown in Figures 8 and 9

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.

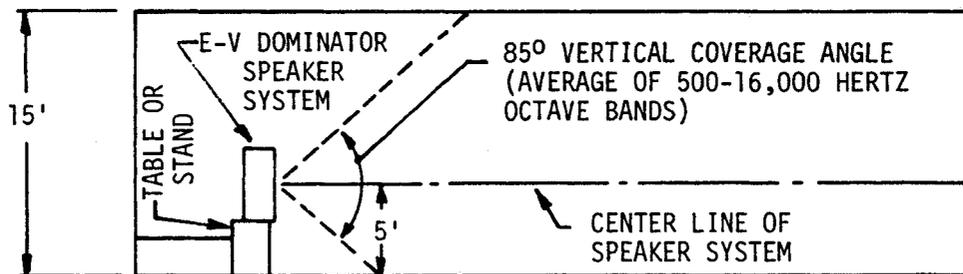
For more headroom and higher sound pressure level capability a two-channel power amplifier capable of producing 150 watts per channel with both channels driving an 8 ohm load (such as the TAPCO CP500) could be used in place of the 50-watt-per-channel power amplifier previously described. With the 150-watt-per-channel 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.

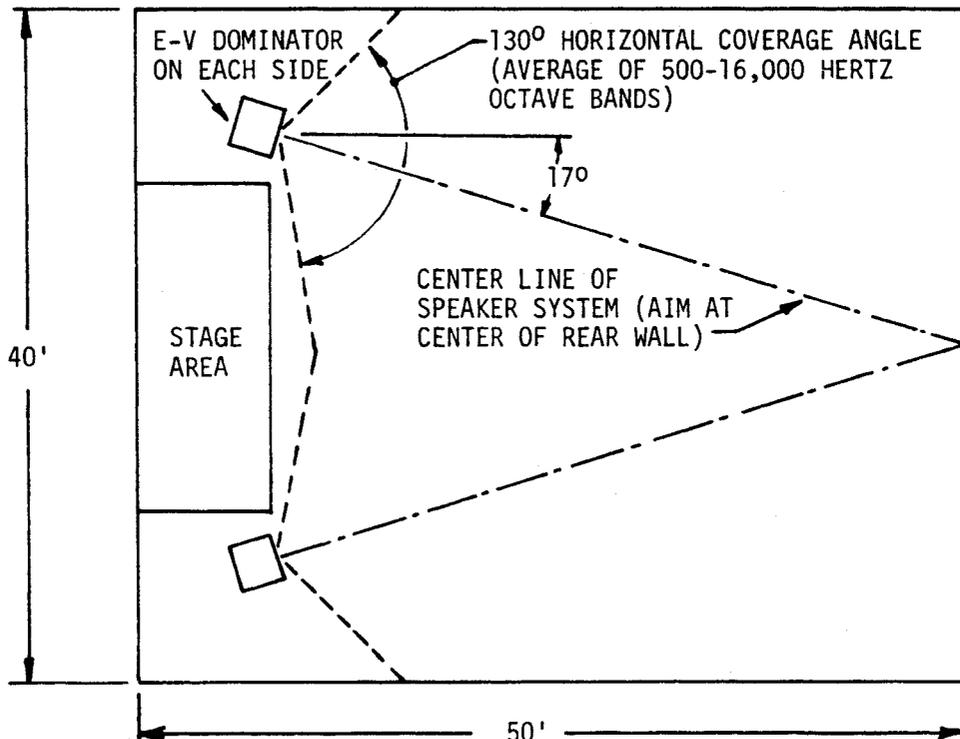
Medium Size Room. To provide sound to a medium size room, approximately 40 feet x 50 feet x 15 feet high, or a room volume of 30,000 cubic feet, a system such as the one shown in Figures 11 and 12 could be used. It uses E-V Dominators and will produce 106 dB average midband levels and peaks of 116 dB. 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.



ELEVATION VIEW



PLAN VIEW

FIGURE 11 – Medium Size Room, Setup Number 1

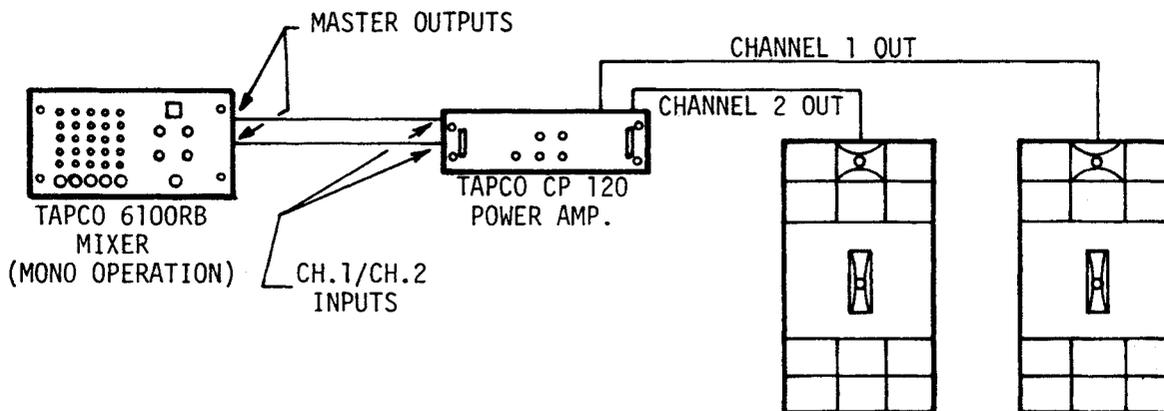


FIGURE 12 – Block Diagram for Medium Size Room, Setup Number 1

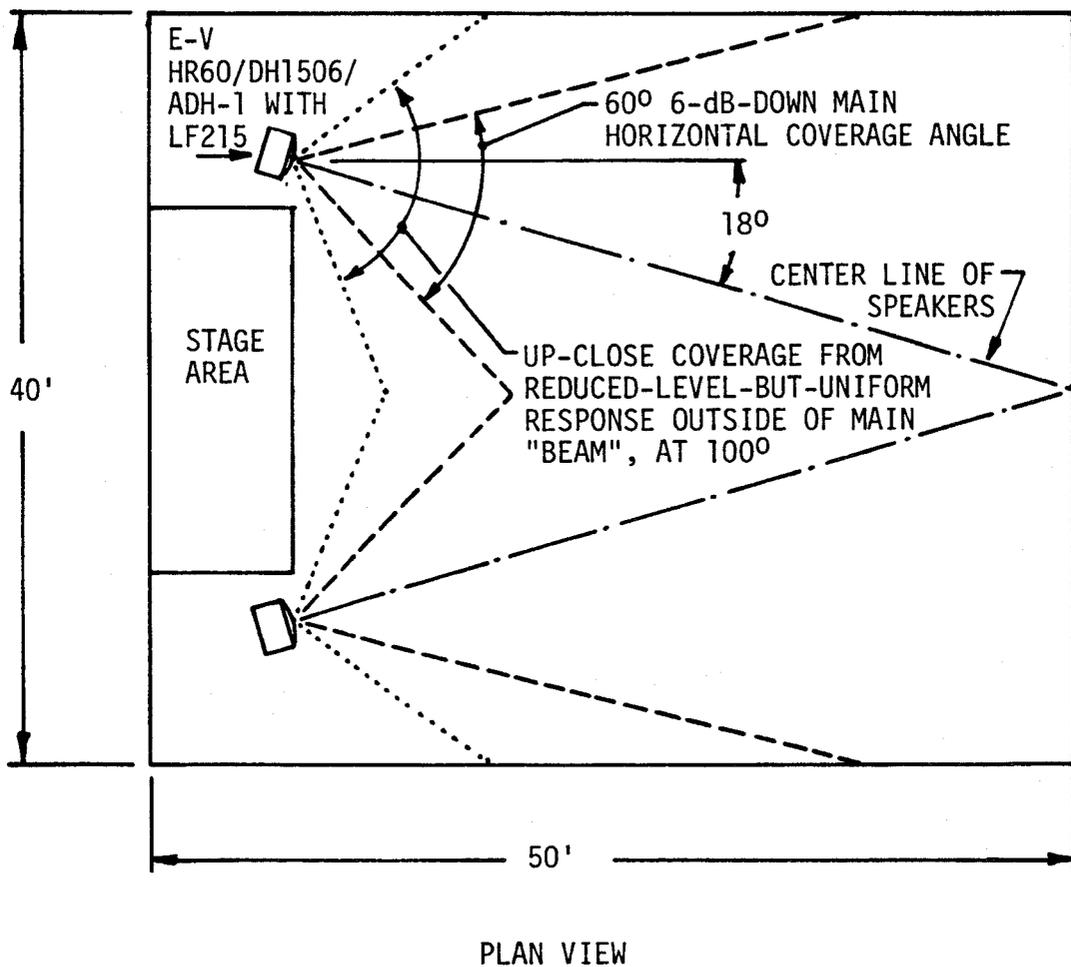
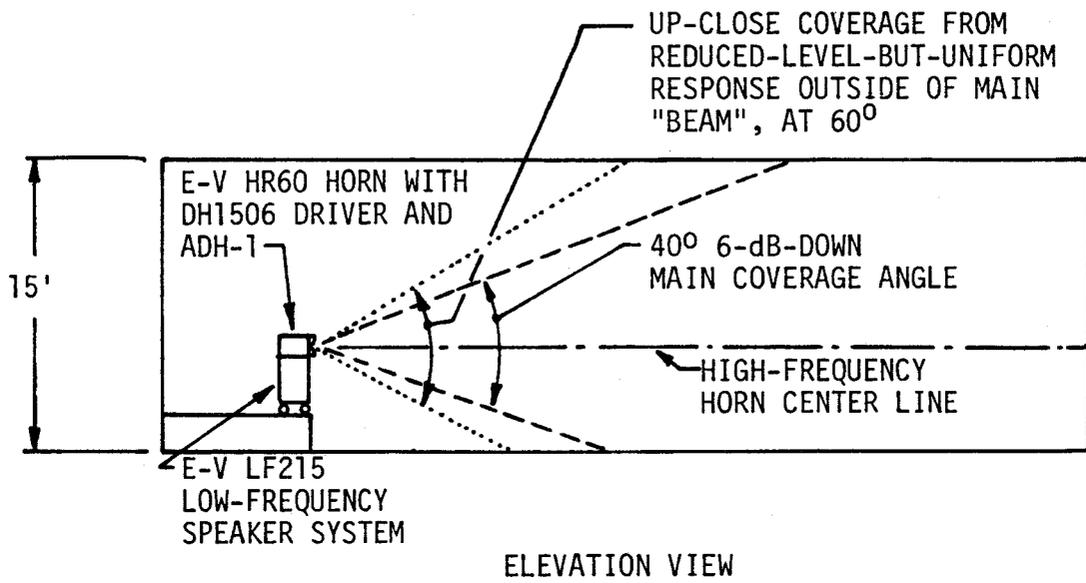


FIGURE 13 – Medium Size Room, Setup Number 2

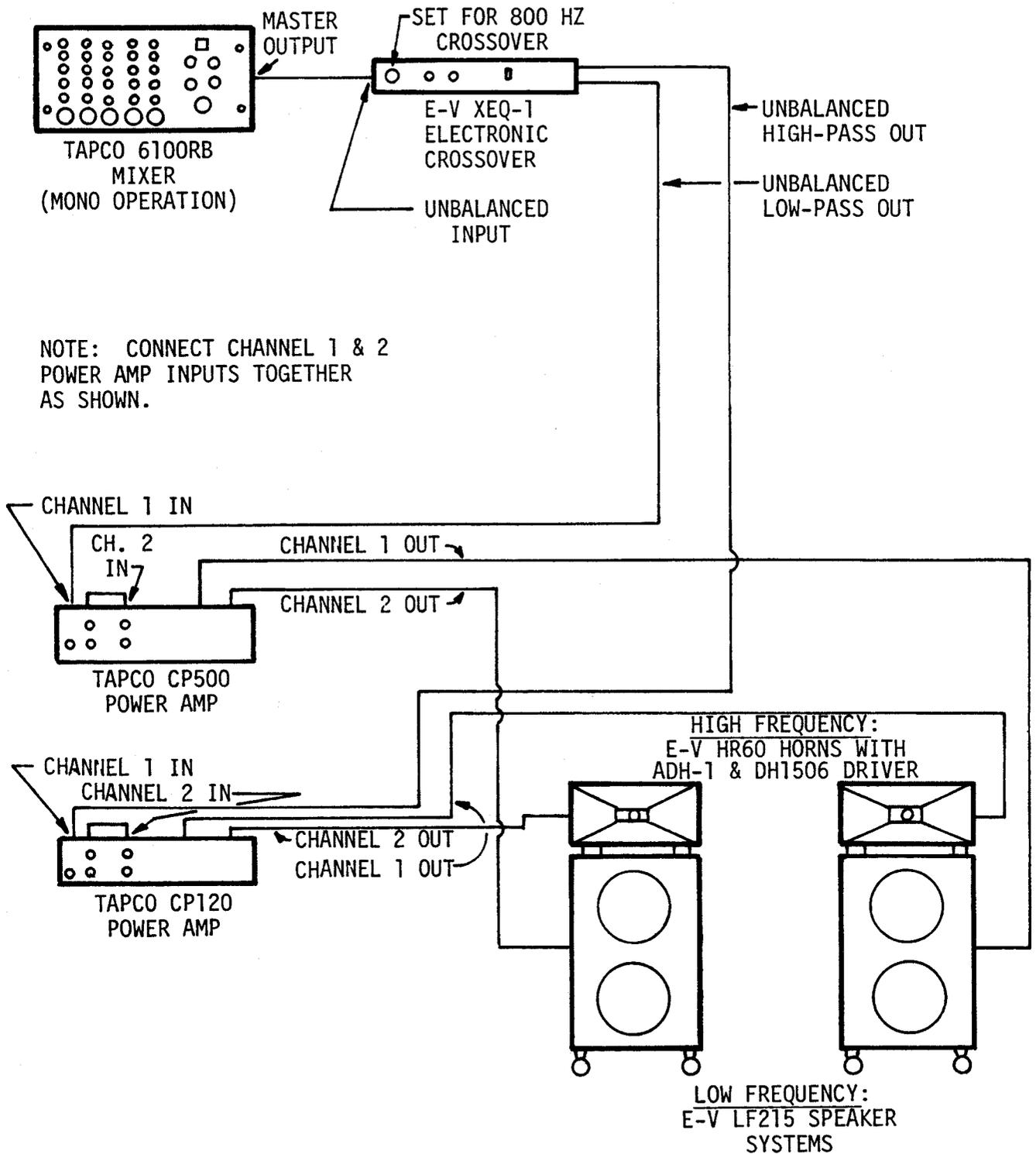


FIGURE 14 – Block Diagram for Medium Size Room (Setup Number 2)

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 $60^\circ \times 40^\circ$ 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 re-

quired. 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 amp 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

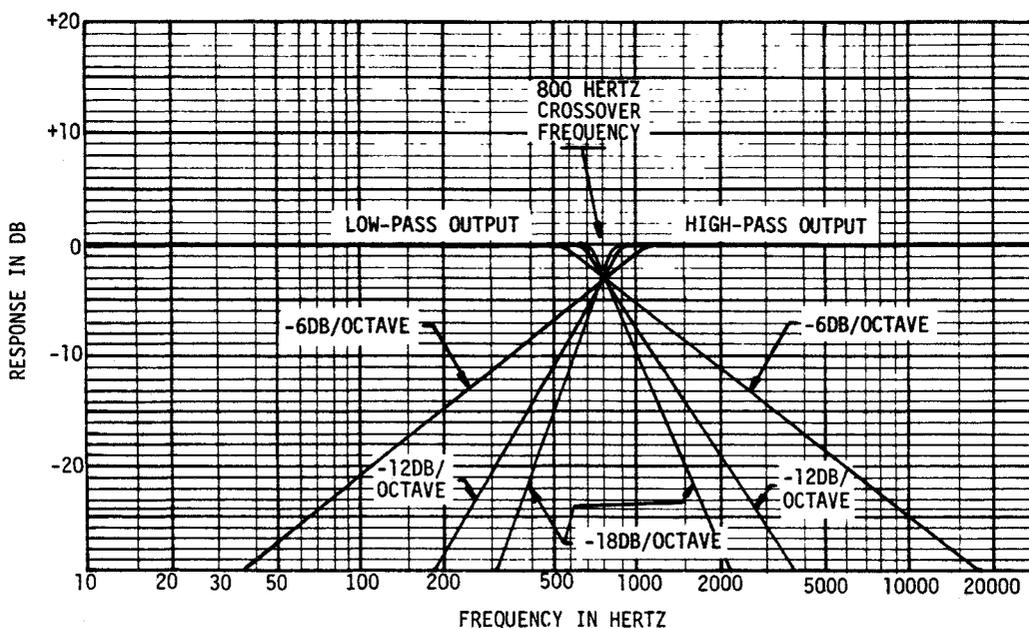


FIGURE 15 – Ideal Crossover Frequency Response

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 addition 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

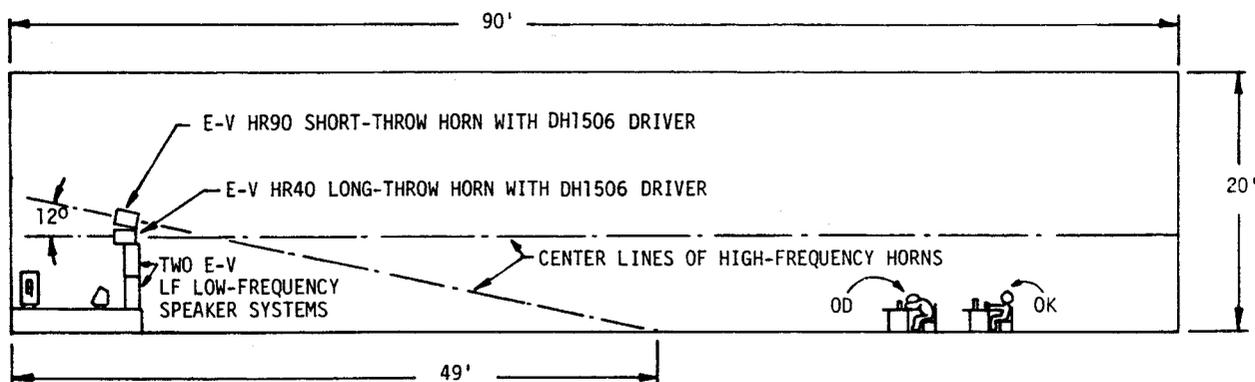


FIGURE 16 – Elevation View of Large Size Room

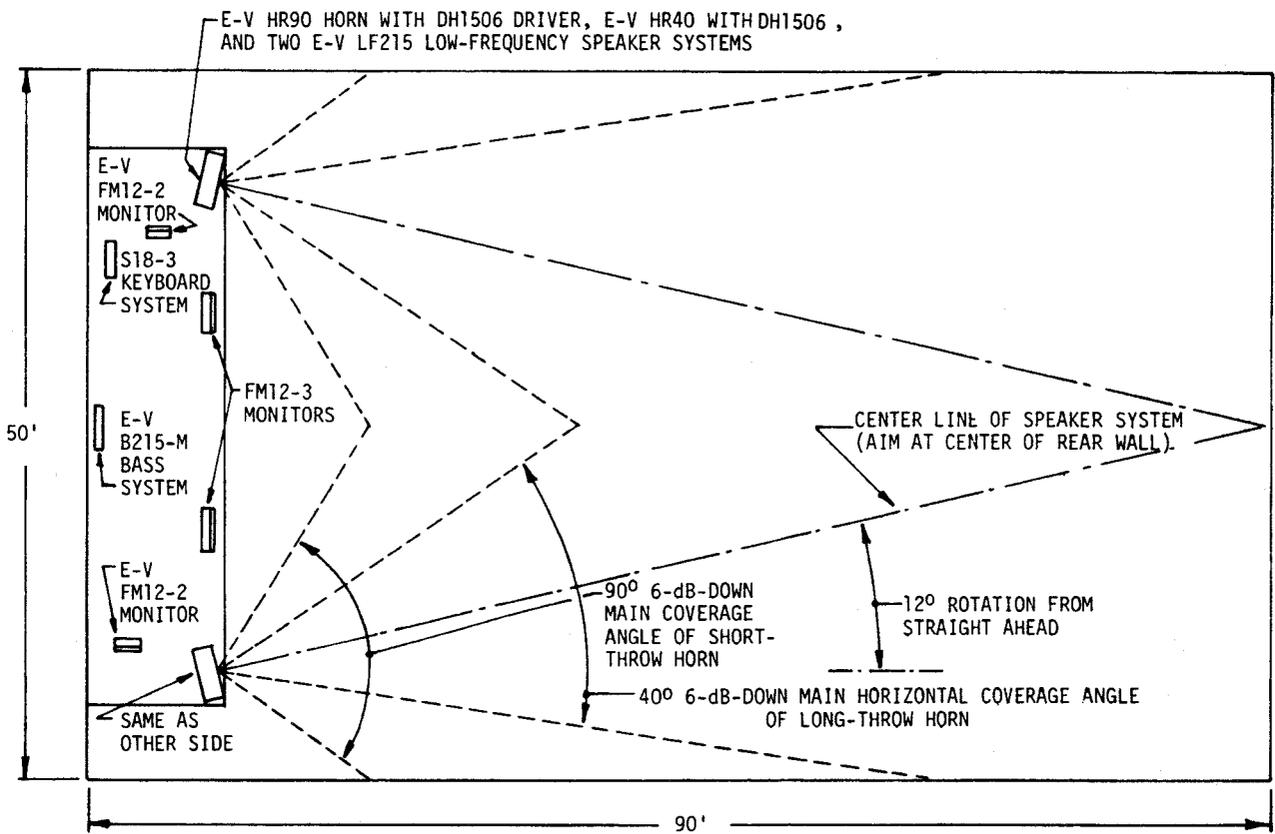


FIGURE 17 - Plan View of Large Size Room

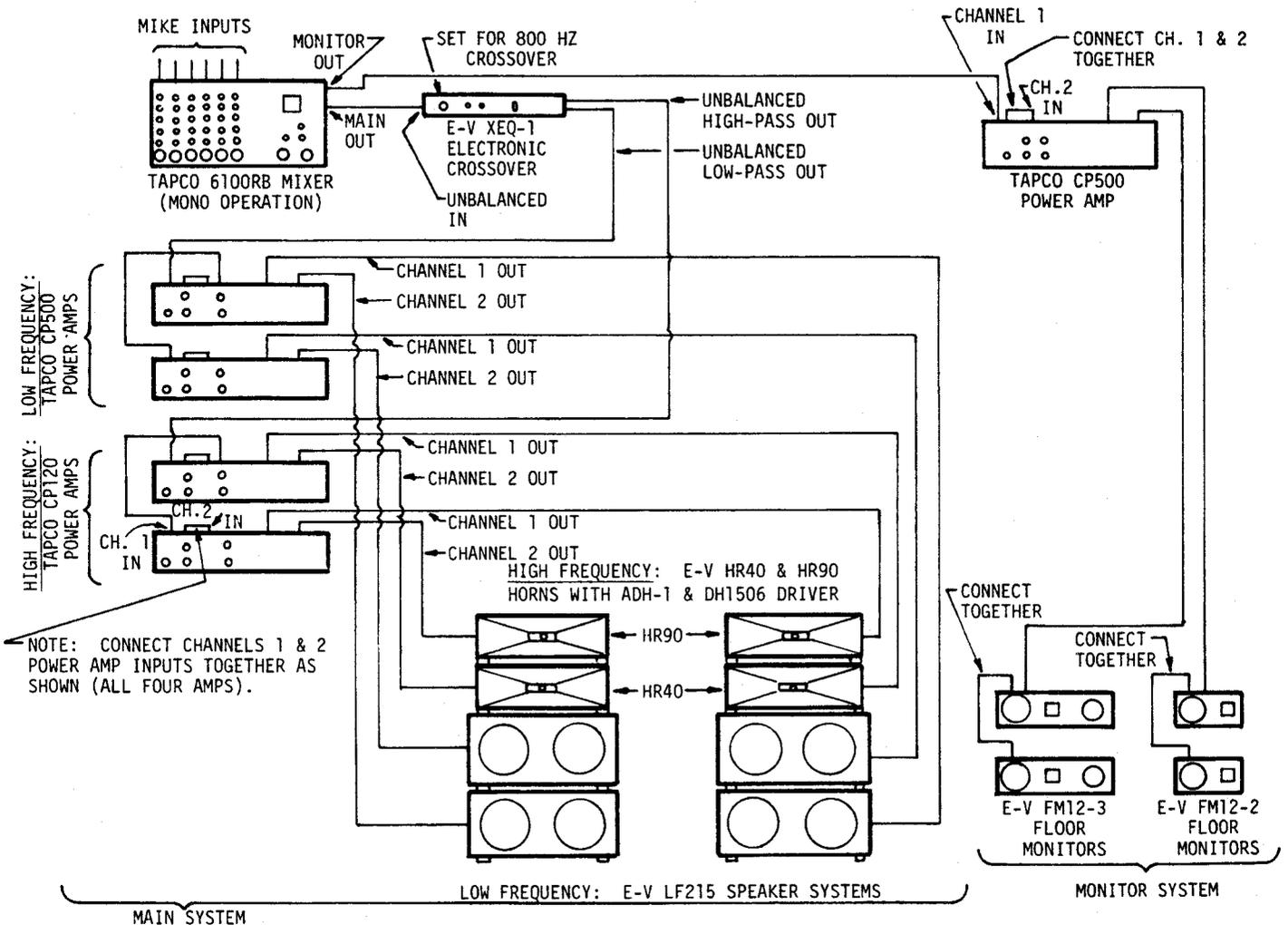


FIGURE 18 - Block Diagram for Figures 16 and 17

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 road-able 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.

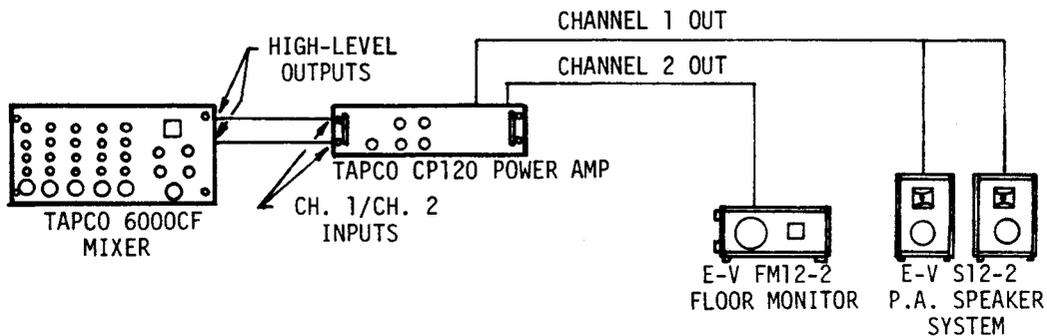


FIGURE 19 – Simple Monitor System

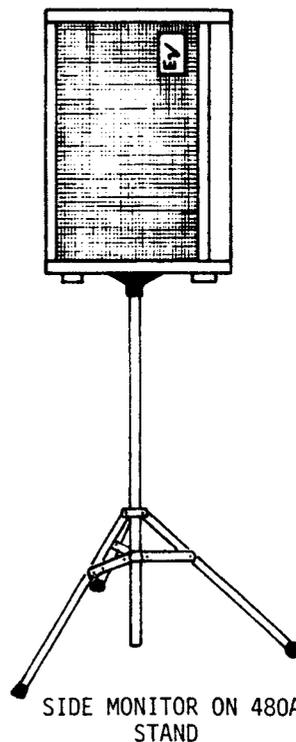
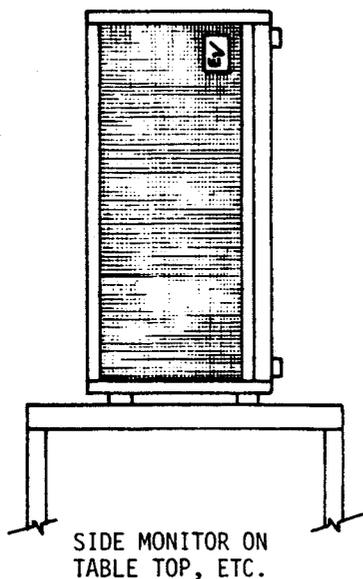
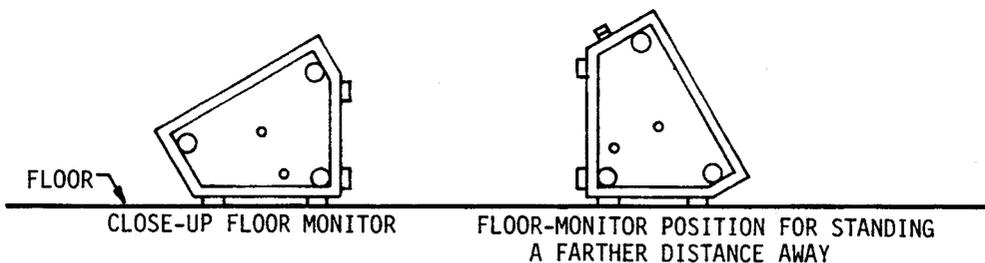


FIGURE 20 – Various Possible Placements of FM12-2 and FM12-3 Floor Monitors

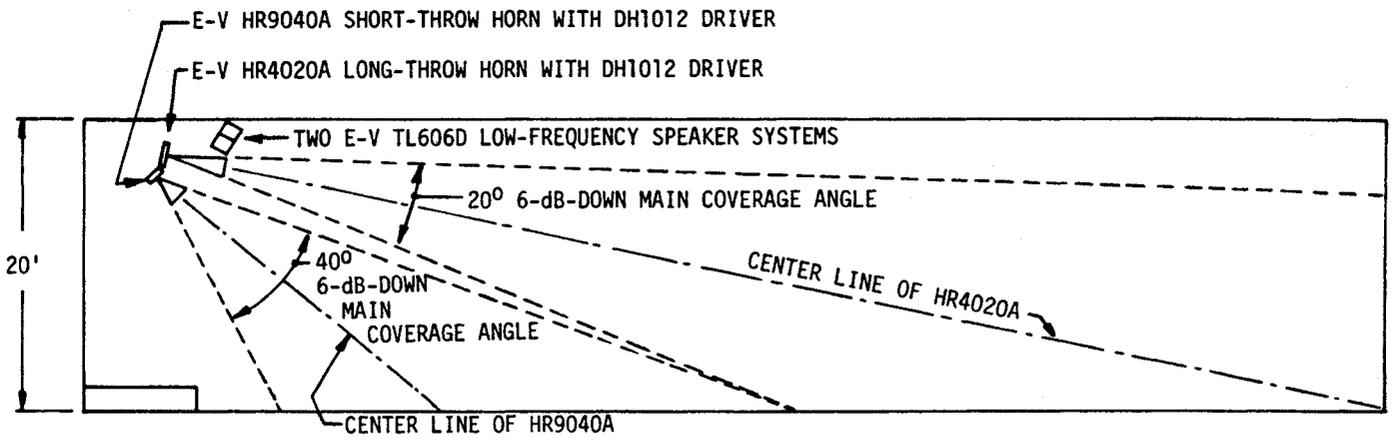


FIGURE 21 – Elevation View of Large Size Room, Permanent Installation

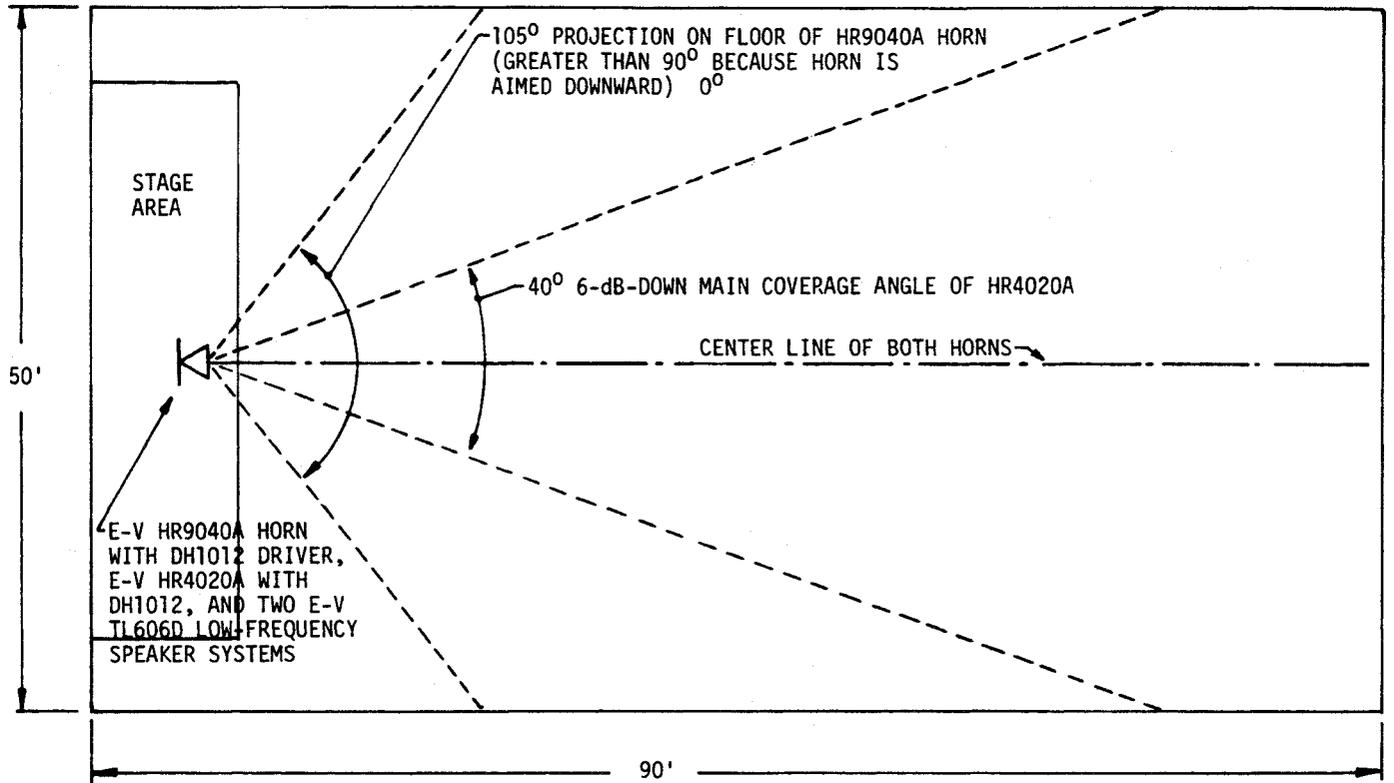


FIGURE 22 – Plan View of Large Size Room, Permanent Installation

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.



Electro-Voice®
 600 Cecil Street, Buchanan, Michigan 49107
 7473 Avenue 304, Visalia, California 93277
 Electro-Voice Div., 345 Herbert Street,
 Gananoque, Ontario • Electro-Voice, S.A.,
 Romerstrasse 3, 2560 Nidau, Switzerland

E-V "PA Bible"
 Electro-Voice, Inc.
 P. O. Box 186
 Buchanan, Michigan 49107