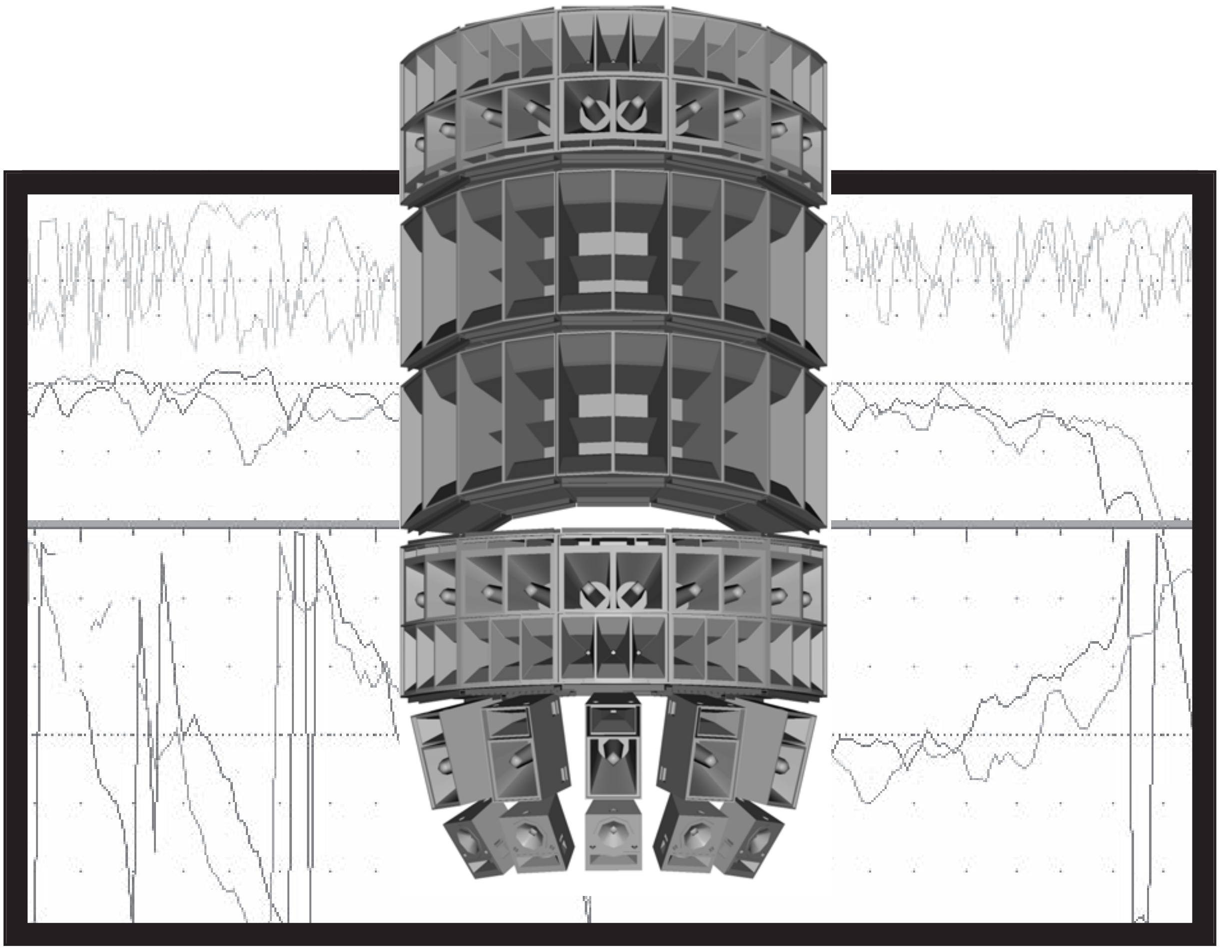
Meyer Sound Design Reference

For Sound Reinforcement



By Bob McCarthy

P/N 01.990.125.01

Meyer Sound Design Reference

for

Sound Reinforcement

by
Bob McCarthy

Meyer Sound • Berkeley

Meyer Sound Design Reference

To Jeanne,

for her patience and support.

Front cover illustration by Francois Desjardin, is reprinted courtesy of Solotech, Inc.

Technical specifications are from Meyer Sound Data Sheets, Operating Instructions and TechNotes. Speaker illustrations utilize drawings from the Meyer Symbol Library, created by Jamie Anderson, Scott Gledhill and Joe Rimstidt. SIM System II and Meyer CEU front panel illustrations are by Ralph Jones.

Meyer Sound
2832 San Pablo Avenue
Berkeley, CA 94702
Phone (510) 486-1166
Fax (510) 486-8356

First Printing: May 1, 1998

Table of Contents

Table of Contents

Section 3: System Design

Section 4: Verification

Section 5: Alignment

Table of Contents

Acknowledgments and Sources

This book evolved over the course of three years with the support of a great number of people. In particular, I would like to thank the following people their contributions to the project:

Jamie Anderson, Karen Anderson, David Andrews, John Bennett, Andrew Bruce, Jean Calaci, Mike Cooper, Jim Cousins, Dave Denison, Frank Desjardin, Peter Elias, Sharon Harkness, Lisa Howard, Andrew Hope, Roger Gans, Scott Gledhill, Ralph Jones, Dave Lawler, Akira Masu, Steve Martz, Tony Meola, Todd Meier, Helen Meyer, John Meyer, Joe Rimstidt, Charis Baz Takaro, Mark Takaro, Candace Thille, Hiro Tomioka, Lisa Van Cleef and Tim Wise.

Preface

Meyer Sound has exerted a powerful influence upon the audio industry since its inception in 1979. Many of the reasons behind this influence are revealed in this book, which brings together Meyer Sound's products, history, and philosophy.

This book is designed to serve as a comprehensive reference document for current and potential Meyer Sound users, going far beyond the scope of data sheets and operating instructions. I have made every effort to minimize the mathematics in favor of practical examples. Wherever possible, points are illustrated by field data accumulated from my extensive library of SIM® System II measurements.

In my capacity as SIM Engineer, I have been fortunate to have had the opportunity to align sound reinforcement systems, of virtually every shape and size, for some of the world's finest sound engineers and designers. Each system and venue present unique challenges that make each day a learning experience. Over the years, my goal has been to find repeatable solutions for these challenges by developing a methodology that can clearly differentiate the complex mechanisms that affect a sound reinforcement system. The result is an approach to sound system design and alignment that transcends a particular musical genre or type of venue. This is the essence of *Meyer Sound Design Reference*.

This book is divided into five major sections that flow in logical order from system conception to final alignment. A Meyer product revision history and appendix follow.

Section 1: Building Blocks describes the components that, when taken together, create a complete sound system. Each component is detailed with key factors that must be considered for optimum performance.

Section 2: Acoustical Factors describes the acoustical mechanisms that affect the performance of your installed system. The interaction of speakers with each other and with the room are covered in detail.

Section 3: System Design describes how to bring together the components into a complete system for your application. Complete Meyer product reference data is included to aid speaker selection.

Section 4: Verification details how to ensure that your installed system is working as designed. Checkout procedures and field data are included.

Section 5: Alignment describes the alignment process from start to finish, including extensive field data.

Each section is divided into a series of short subjects to allow for quick reference. Whenever possible the left and right pages are grouped together when covering the same topic, particularly when one page describes figures on the other.

The Goals of Sound Reinforcement

Meyer Sound has always been committed to creating the highest quality loudspeaker system products possible. Over the years there has evolved a core of principles among ourselves and our users. These principles, while not unique to Meyer Sound, serve as the guiding force behind this handbook.

These underlying principles are a commitment to:

- Provide the most accurate reproduction of the input signal's frequency and phase response, free of coloration or distortion.
- Maximize system intelligibility.
- Provide a consistent sound pressure level and frequency response over the listening area.
- Create realistic sonic imaging.
- Minimize the effects of poor room acoustics and make the best of the good ones.
- Optimize the power bandwidth of the system for the source signal.
- Maximize dynamic range.
- Minimize the noise floor.
- Make efficient use of limited time and budget resources.
- Maximize short- and long-term reliability of the system.
- Maintain compatibility of the system over time.
- Minimize destructive interference between speaker subsystems.
- Minimize downtime by efficient troubleshooting and repair.
- Operate all equipment safely.

And last but not least,

- Make it sound good and have a good time doing it!

The Challenges of Sound Reinforcement

It is one thing to list the goals of sound reinforcement. It is quite another to accomplish them. There are tremendous challenges presented by even the most simplistic sound designs once the system is installed in a space. Even if we assume a perfectly designed and manufactured loudspeaker system, the response of the sound system can be degraded by:

- Distortion
- Polarity reversals
- Wiring errors
- Interaction between speakers
- Compression
- Dynamic acoustical conditions
- Reflections
- Redundant coverage
- Delay offset between speakers
- Rattles and buzzes
- Component failure
- Crossover cancellation
- Gain structure errors
- Poor impedance matching and termination
- Improper grounding
- Insufficient power bandwidth
- Compromised speaker positions
- Insufficient time for alignment
- Lack of proper test equipment

This handbook is designed to address all of these challenges, enabling you to achieve the goals of sound reinforcement.

Meyer Sound's Total Solution™

The people of Meyer Sound are committed to providing the tools to achieve these goals. Since its inception Meyer Sound has engendered a comprehensive, systematic approach to sound reinforcement, in contrast to the component level approach of other speaker system manufacturers. Meyer's comprehensive approach began in the late 1970s with the manufacture of speaker systems, each including a dedicated Control Electronics Unit (CEU) that optimized the response of the speaker. More recent advances have led to the creation of a complete line of self powered speakers.

Each speaker and CEU manufactured by Meyer Sound is rigorously designed and tested using state-of-the-art measurement technology. The response of the system may be compromised by challenges in the field. The need for a comprehensive field solution led to the development of SIM® System II and the Remote Monitor System (RMS™).

SIM is a comprehensive measurement system dedicated to detecting and solving the challenges and problems that face a speaker system in the field, including, as the final step, verifying and fine tuning the system's response during a performance. SIM System II is run by Certified SIM Operators and Engineers trained to completely analyze and align a system on-site.

RMS is capable of continually monitoring the status of all self powered system speakers and amplifiers so that problems can be detected immediately. No other speaker system manufacturer can offer anything near this level of capability to verify and optimize the performance of the system for the end user.

This is Meyer Sound's Total Solution™ for Sound Reinforcement.

1.1 Meyer Speaker Systems

Every Meyer Sound professional loudspeaker product is designed as a fully engineered, integrated system incorporating the loudspeaker and an active line-level signal processing component. This active processor is termed the "Control Electronics Unit" (CEU). Each loudspeaker model requires a specific CEU and must not be operated without it. The function of the CEU is to replace a series of separate components made by various manufacturers—containing a large number of user-adjustable parameters—with a unit that is designed for the specific application of optimizing the performance of a particular loudspeaker enclosure.

At the time of Meyer Sound's inception the sound reinforcement industry's approach to speaker system design and alignment was very different from the current style. Most users assembled systems by mixing and matching various manufacturer's components into a custom designed enclosure. Virtually every company had their own self-designed system incorporating off-the-shelf or custom-built crossovers, equalizers, limiters, drivers, horns, delay networks and power amplifiers. Many companies staked their reputation on the fact that they were the sole source of a particular speaker system.

However, for a sound engineer on tour, encountering a different "custom" system every night meant that they would have very little idea of what they would encounter at each venue.

Meyer Sound changed the direction of the industry by introducing a complete, calibrated system, which created a standard, repeatable level of sound quality that was available to all levels of the industry *non-exclusively.* When mixing on a Meyer Sound system, the sound engineer knows what to expect, because the system has the same enclosures, drivers, crossover, and limit thresholds, wherever it is rented.

The consistent performance of these systems over time has given Meyer products a reputation for being a system that the mixer can count on night after night anywhere in the world.

The CEU is designed to be the final component in the signal chain before the power amplifier. No other signal processing equipment should be inserted between the CEU outputs and the amplifier inputs. If this were done, it is almost certain to disturb the CEU's performance by limiting its ability to protect, and may result in damage to the loudspeaker components.

The general functions of Control Electronics Units (CEUs) in Meyer Sound professional loudspeaker products are:

- Active crossovers that are optimized for the particular response characteristics of given drivers in their enclosure.
- Equalization to adjust for flat frequency response in free-field conditions.
- Phase correction through crossover for optimized addition and polar response.
- Driver protection for maximum long-term reliability (Peak and RMS limiting). The driver protection circuitry only engages at the point where the system would otherwise be at risk. Therefore, there is no excess compression.
- In some cases, dynamic excursion protection circuits act at the onset of overload to maintain reasonably linear response and protect the drivers from mechanical damage.

1.1 Meyer Sound Systems

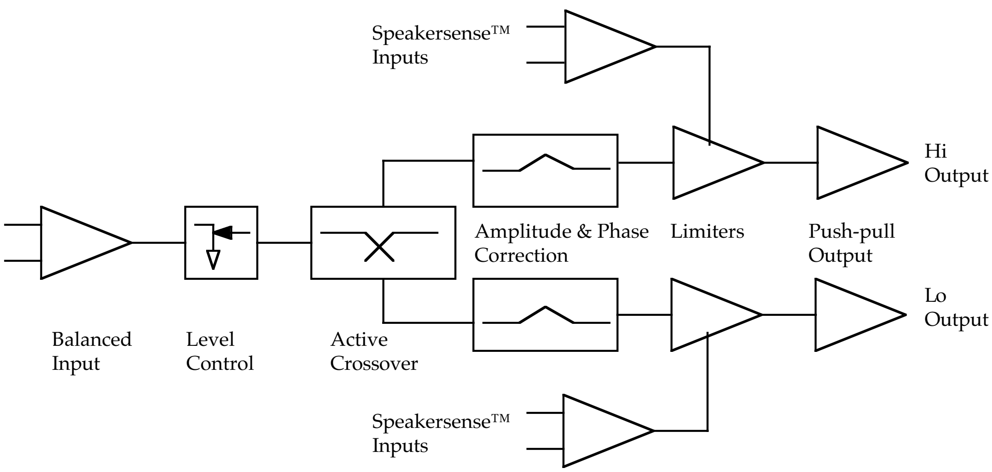


Fig 1.1a Basic flow block and connection diagram.

Basic Signal Flow

Each Meyer CEU has unique characteristics, but the general flow is shown in Figure 1.1a.

Balanced Input: The signal is actively buffered from its source. All inputs are high impedance (greater than 5kΩ). Some units incorporate the patented ISO-Input™ which ohmically isolates the source through a transformer.

Level Control: The level control follows the input buffer stage. The level control is used to set relative gains for the systems.

Active Crossover: This stage splits the signal into high and low frequency ranges. The filter topology and crossover frequency vary for each model CEU.

Amplitude and Phase Correction: Each model CEU differs markedly in this area to optimize the response for each speaker model.

SpeakerSense Inputs and Limiters: The signal from the power amplifiers is fed back into the CEU via the SpeakerSense inputs. If the level is too high, the limiters are engaged to reduce the signal flow into the power amplifier.

Push-pull Output Stage: The output drive for all CEUs is a balanced push-pull drive capable of driving loads of 600Ω or higher.

Fig 1.1b The Meyer Sound M-1A Control Electronics Unit.

1.2.1 Electronic and Acoustical Crossovers

The acoustical crossover point in multi-driver systems is defined as the frequency at which the drivers have equal amplitude response levels. In well-designed systems, this point will also coincide with its phase response.

> **The acoustical crossover points for Meyer Sound speakers are carefully selected to:**
>
> - optimize the power response of the system to maximize component reliability and linearity.
> - optimize the phase transition between components.
> - maintain uniformity of pattern control through crossover.

There are a number of audio texts describing the advantages of various crossover topologies. These describe the filter shapes such as Linquist-Reilly or Chebyshev and their relative slopes (6, 12, 18 and more dB/octave). A relatively recent trend is the promotion of digitally derived crossovers. This has been touted as a great advance due to their ability to create extremely steep slopes. However, a discussion of electronic crossovers in the abstract is misleading, because without factoring in the physical aspects of the acoustical components, there is no assurance whatsoever that the combined acoustical response will be satisfactory. For example, crossovers with extremely steep filters are prone to large amounts of phase shift, creating delay in the crossover region. In addition, dividing the system steeply may decrease the power capabilities of the system by depriving it of a region where the transducers can efficiently combine.

It is important to note that the *acoustical crossover is not necessarily the same as the electronic crossover* in a system. In fact, setting the crossover frequency of an off-the-shelf electronic crossover can be very misleading.

> **A generic electronic crossover** *does not* **factor in:**
>
> - Relative driver sensitivity at crossover.
> - The relative voltage gain from the crossover input to the amplifier output terminals.
> - Efficiency of the horn.
> - The phase relationship between the drivers.
> - The relative quantities of the drivers.

Meyer Sound products specify the electronic and acoustic crossover points. The electronic specification can be used to verify the response of the CEU. The acoustical crossover specification is important for polarity verification of the system and for the relative level setting of mid-bass speakers and subwoofers.

Figure 1.2a shows the electrical response of the S-1 CEU, designed to function as the controller for the MSL-2A speaker system. Observe that the electrical crossover frequency is 1300 Hz. The phase response of the Hi output indicates that there is a frequency selective delay network enacting phase correction in the crossover frequency range. This is indicated by the downward slope of the phase response.

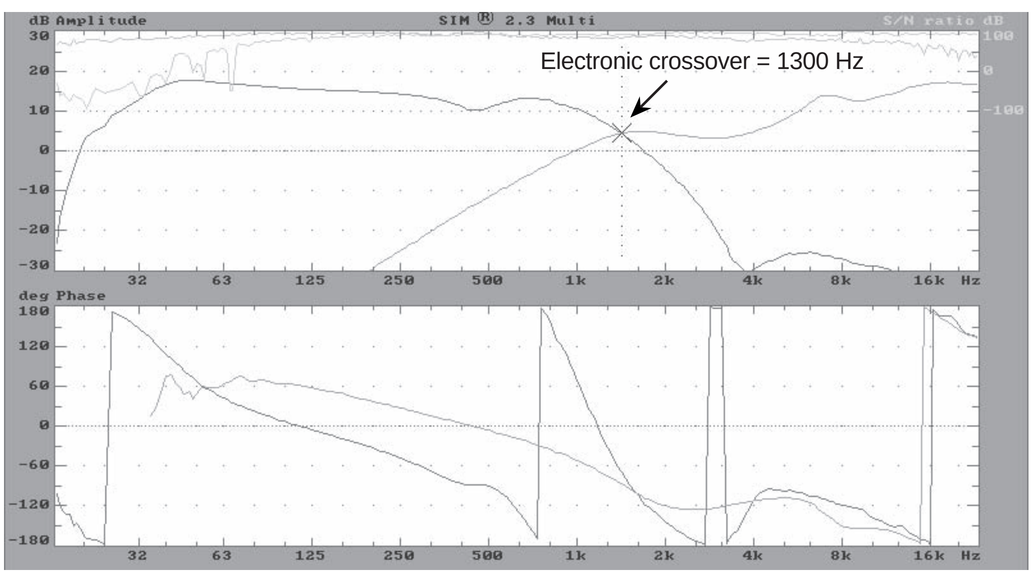


Fig 1.2a Electrical response of S-1 CEU.

1.2.1 Electronic and Acoustical Crossovers

Fig 1.2b shows the acoustical response of the MSL-2A speaker system when driven with the S-1 Control Electronics Unit. This plot shows the acoustical responses of the Hi and Lo channels measured individually. The upper screen shows the amplitude response and indicates an acoustical crossover of 900 Hz. The lower screen shows the phase response. Notice that the measured acoustical phase response differs markedly from the purely electrical response shown in Fig 1.2a. Most importantly, notice that the phase responses of the Hi and Lo channels converge in the crossover region, which will enable the transducers to combine with maximum efficiency. Above and below the crossover range the Lo driver leads the Hi driver as can be seen by their relative phase slopes.

Fig 1.2c shows the combined acoustical response of the MSL-2A speaker system when both Hi and Lo channels are driven. Notice the transparency of the crossover point in both the amplitude and phase traces. In addition, notice the integrity of the signal-to-noise ratio trace through the region. These three factors together indicate that the response of system is optimized.

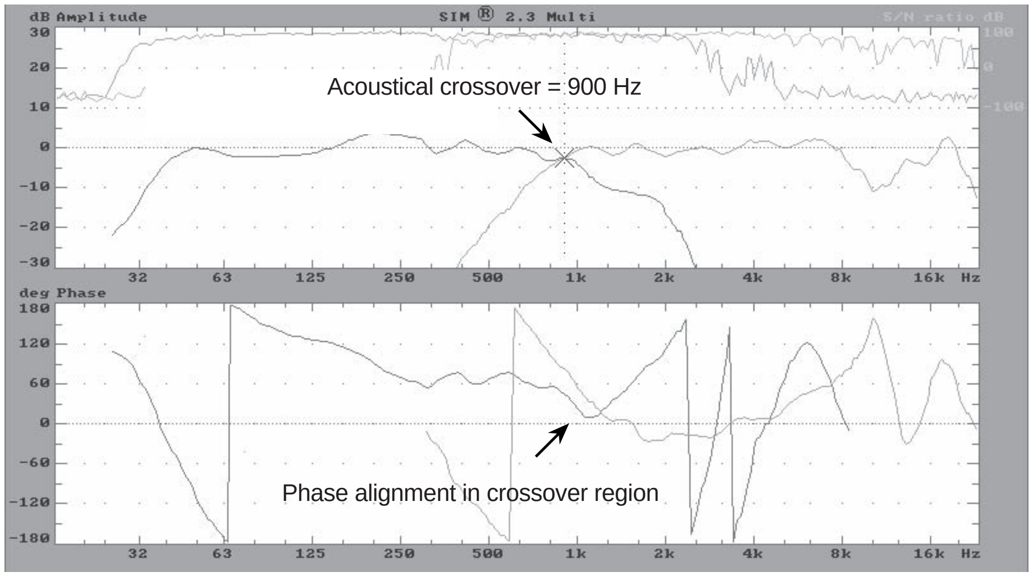


Fig 1.2b Acoustical response of the MSL-2A speaker system (with the S-1 CEU). Hi and Lo channels are measured separately.

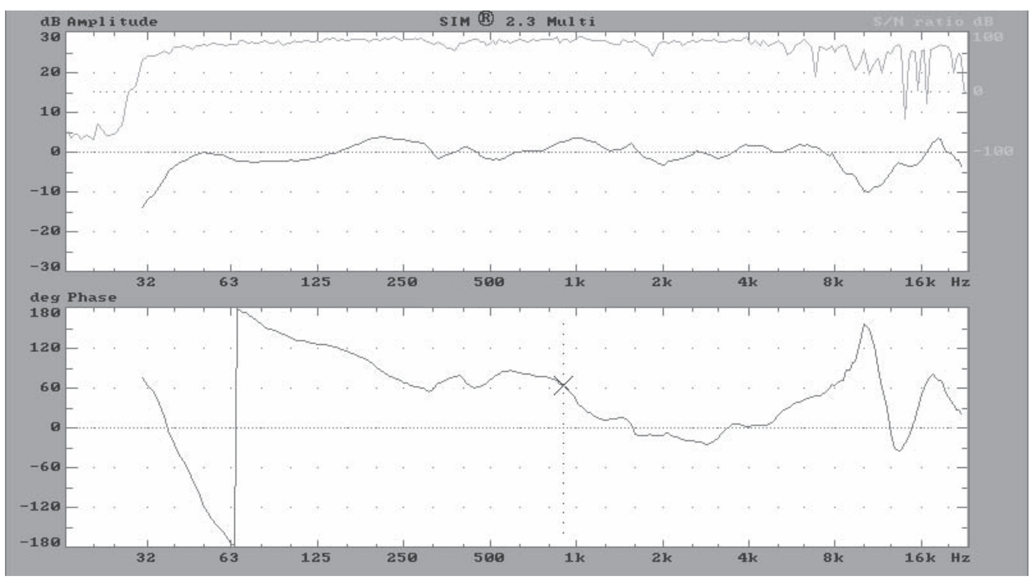


Fig 1.2c Combined acoustical response of the MSL-2A.[1]

[1] Not free field

1.2.2 Amplitude Correction

In an ideal world, we would have transducers that exhibit perfectly flat, free-field amplitude response over their entire usable band with no need for electronic correction. Each Meyer Sound speaker system begins with transducers that are exceptionally linear. However, practical considerations such as the enclosure tuning, horn shape and crossover point, to name a few, will have a substantial effect on each transducer's response. Each of these will cause peaks and dips in the system's response if not carefully measured and corrected. Meyer Sound's design approach is to optimize the amplitude response *as a system*, utilizing the best combination of physical and electronic means.

In order for this approach to succeed the speakers and electronics must provide repeatable results. At Meyer Sound, the free air resonance of each and every transducer is measured along with its frequency response when installed in the enclosure. The enclosure and horn dimensions are built to exacting standards to ensure repeatable tuning.

Fig 1.2d shows the amplitude response of the M-1A Control Electronics Unit used with the UPA-1C loudspeaker. The Hi and Lo channels are shown separately. Notice that the response of the M-1A contains substantial correction, particularly in the crossover region and above 8 kHz.

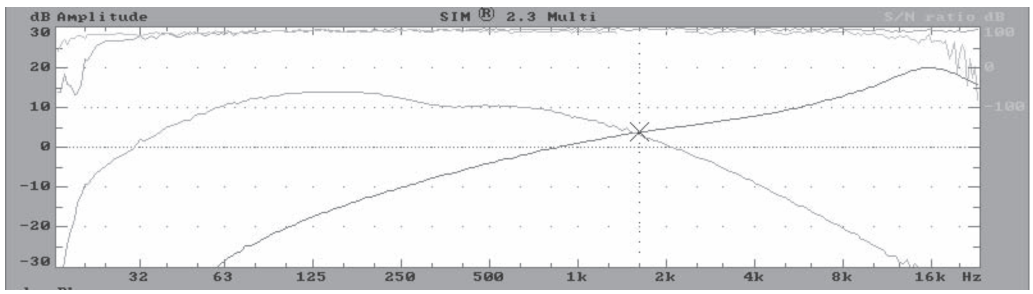


Fig 1.2d M-1A Controller Hi and Lo channels amplitude response.

Fig 1.2e shows the measured acoustical response of a UPA-1C loudspeaker driven by the M-1A. Notice that the final acoustical response is quite linear, indicating a successful optimization.

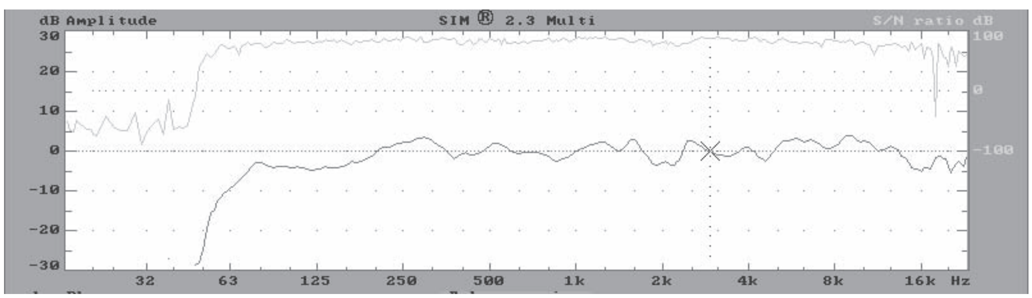


Fig 1.2e UPA-1C on-axis amplitude response.

1.2.3 Phase Correction

Phase correction is employed to ensure that the temporal relationship between frequencies remains intact, and to optimize the response of the system through crossover. Practical design considerations may cause the components in multi-way systems to be physically placed such that their phase alignment could potentially be degraded unless electronic phase correction circuitry is employed.

Phase response tends to be less well understood than its amplitude counterpart. While most audio engineers understand the importance phase response plays, many have never had the opportunity to measure phase response directly. This is due in large measure, to the fact that the most common audio measurement instrument—the real-time analyzer—cannot measure phase. SIM System II, however, has a phase display, allowing the phase response to be seen at 1/24th octave frequency resolution for the audible range.

1.2.3 Phase Correction

A fully phase-corrected loudspeaker system is one that is capable of reproducing its full range without any frequency-dependent phase shift (i.e. all frequencies will be reproduced with the same temporal origin). In actual physical loudspeaker systems, this is an extremely challenging endeavor. Real speaker systems exhibit a phase delay characteristic that is inversely proportional to frequency, which is to say that the low frequencies tend to lag behind the highs. This means that in order to synchronize high and low frequencies, high frequencies need to be delayed. Unfortunately, however, it is not as simple as adding some fixed amount of delay to the high end, because each frequency requires a different amount of delay. Frequency-selective delay networks are required to delay selected areas in order to achieve a net flat phase for the full system.

Figure 1.2f shows the amplitude (upper screen) and phase (lower screen) of the Meyer UPL-1 powered loudspeaker. This is an example of what is arguably the best phase-corrected, full-range sound reinforcement speaker in the world. The slope angle of the phase trace reveals that the system is fully corrected down to 250 Hz, with a gradual increase in phase delay below that. Notice, also, that the crossover region is completely transparent in both amplitude and phase, indicating a truly optimized crossover.

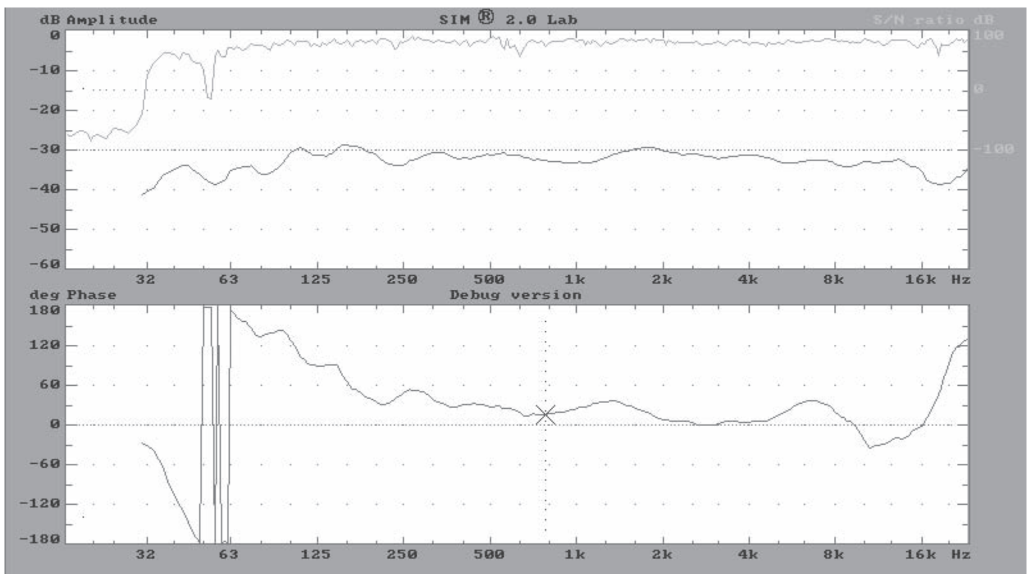


Fig 1.2f Meyer Sound UPL-1 powered loudspeaker system.

The competitor's system shown in Figure 1.2g shows the response of a very typical system without true phase correction. This four-way system is neither aligned at its crossover points nor over the full range. The Hi driver's phase response is flat for less than an octave and the mid and low drivers lag far behind. In contrast, with the UPL-1 it is a relatively simple matter to discern where the crossover points are in this system by viewing the sharp changes in phase angle and the corresponding dips in amplitude.

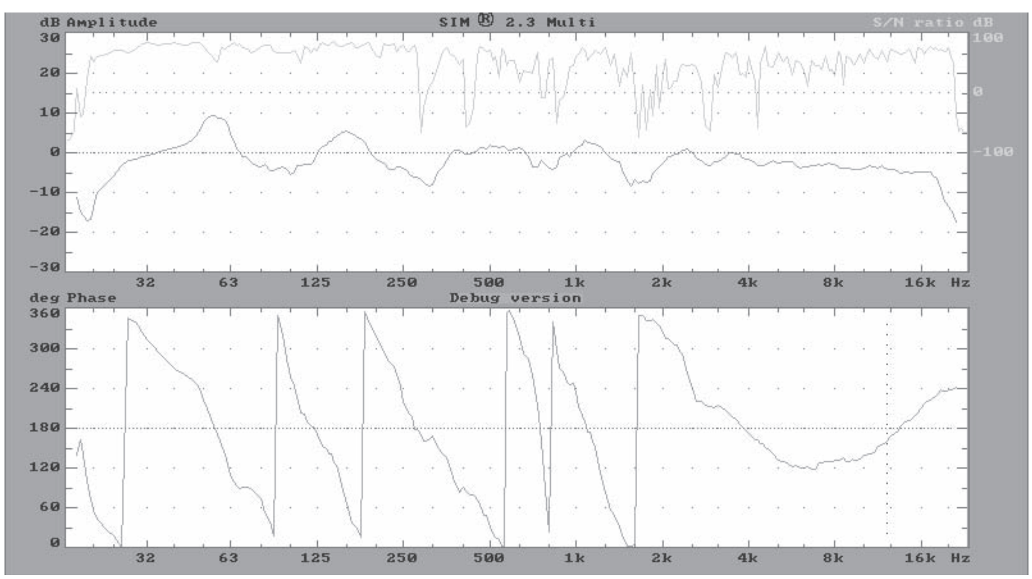


Fig 1.2g Competitor's system.

1.2.4 CEU Connections

Figure 1.2h shows the typical signal flow from mixer to speaker through the CEU. All of the line level connections are balanced line XLR connections.

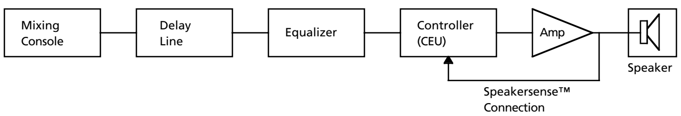


Fig 1.2h

Normal flow of signal from console to speaker. Delay line can be pre- or post-EQ.

How Many CEUs are Needed?

It is necessary to have a separate CEU for each signal channel (e.g., a stereo system needs a minimum of two CEUs). As systems grow in size, there are practical considerations to bear in mind when choosing the number of CEUs. The minimum load impedance presented by the power amplifiers must be greater than 600Ω. For a nominal 10kΩ input impedance amplifier channel, the limit would be sixteen units per CEU. Such a large number of amplifiers driven by a single CEU, however, leaves the system with minimal flexibility and renders it vulnerable to single point system failure. In order to optimize the system response, it is usually best to limit the load of the CEU to around six full-range systems and/or ten subwoofer systems. The primary factor in selecting CEU quantities is system subdivision for alignment, covered in Section 3.5.

A minimum of one CEU channel is required for each signal channel:

- Mono System one CEU channel.
- Stereo System two CEU channels.
- Quad System four CEU channels.

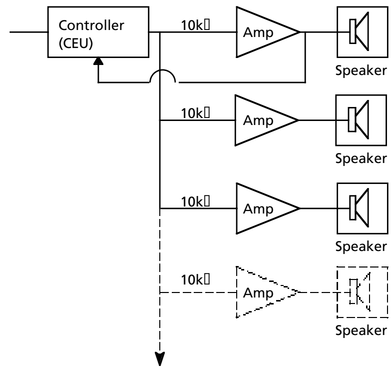


Fig 1.2i

The maximum number of amplifiers per CEU channel is limited by the load impedance of the amplifiers. The total load must be more than 600Ω.

Preventing Cost Overruns

Costs and system complexity can be kept down by preventing the addition of unneeded system components. There is no need for system outboard limiters in series with the main feeds. The limiting is handled by the SpeakerSense limiters. The additional compression of outboard limiters will reduce dynamic range and may actually endanger the system by causing excessively high RMS levels with reduced peaks.

There is no need to add line drivers between the CP-10 Parametric Equalizer and the CEU except for extremely long distances (over 100 meters).

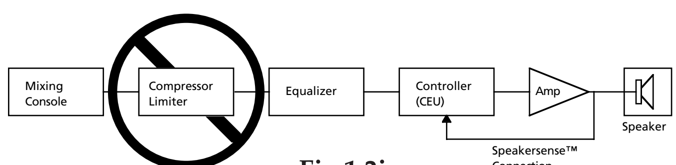


Fig 1.2j

A compressor or limiter is not needed to protect the speaker. Excess compression can actually endanger the system. See Section 1.3.6, Limiter Operation.

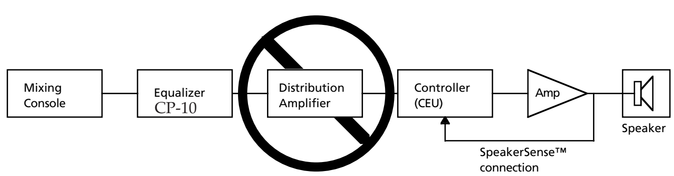


Fig 1.2k

A distribution amplifier is not needed to drive the CEU.

1.2.5 CEU Level Control

The primary purpose of the CEU level control is to trim the relative gain of subsystems and the relative levels of full range systems versus subwoofer systems. The level controls are always for the entire signal range of the speaker under the CEU's control. For example, the M-1 has no separate control for the LF and HF drivers of the UPA loudspeaker. However, the levels of full range speaker and subwoofer systems are set separately because the ratio of quantities of these systems is case dependent.

Controlling Noise

Keeping the noise floor under control is a major component of any installation. It is important to understand what role the processor and amplifier level controls play in this. The CEU level control follows a single low noise input buffer stage. The vast majority of the noise (and there isn't much) created in the CEU is in the crossover, amplitude and phase correction circuitry. Therefore, efforts to reduce system noise solely by turning down the CEU level control is ineffective. If turning down the CEU *does* significantly reduce the system noise, then chances are the noise is being generated by the devices that feed *into* the CEU. This can be verified by a simple test: Unplug the CEU input. If the noise goes away, it is from the devices that feed the CEU. If it does not, then it is from the CEU and/or power amplifier. In either case, this may be indicative of excessive gain at the power amplifier. Gains of 32 dB and more are now typical, making it harder to control noise.

Does it Matter Whether You Turn Down the CEU or the Amp?

Yes. Either one will reduce the noise, but the closer one gets to the end of the signal chain, the more effective it is to keep gains low. This is due to the accumulation of noise through the system. Second, and more importantly, the SpeakerSense circuitry is more effective when the amp gain is lower, affording better speaker protection. (See Section 1.3.2 on SpeakerSense and voltage gain.)

CEU Level Range

Typically the CEU should be operated with the level set between 0 to12 dB attenuation. Operating the system with more than 12 dB of attenuation creates the possibility of overloading the preceding devices. If settings lower than –12 dB are required for gain structure matching or noise considerations, reduction of the amplifier gain is recommended.

Log and Linear

Current CEU models use linear taper pots for level setting. This restricts the range of operation but improves the accuracy of the controls. These are marked in dB attenuation. Older models of CEUs had a log taper level control which was marked by a 1 to 10 numerical scale having no bearing on the number of dB attenuation.

The change from log to linear level controls was implemented because:

- The attenuation of the log taper pots did not track sufficiently well from unit to unit. This created difficulties in adjusting the relative levels of subsystems and subwoofers in multi-way systems.
- The number scale of the log pots gave no real indication of the attenuation level. As a result, users sometimes tended to arbitrarily set levels on the CEU too low. (The 12:00 setting is approximately –20 dB.) This created a loss of system headroom as described above.

All older CEU models can be upgraded. Linear pot upgrade kits are user-installable and available from your Meyer Sound dealer.

Fig 1.2l   Linear level control scale.

1.2.6 CEU User Controls

Each model of CEU has unique user adjustable features. The operating instructions for each unit detail each of these. However, a brief overview of two the most common features follows.

Lo Cut Switch

(M-1, M-3, P-1A, P-2 , MPS-3 and S-1 CEUs)

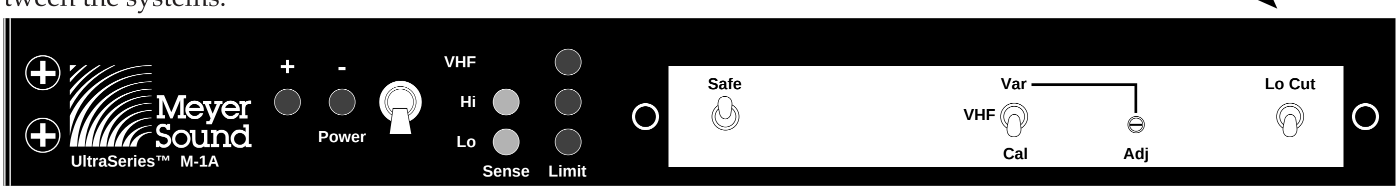
The "Lo Cut" switch is a user-insertable, first-order (6 dB per octave) shelf function with a corner frequency of 160 Hz. For three way systems using subwoofers, the switch acts as part of the crossover circuit to create an acoustical crossover of 100 Hz. This works well in arrays where the full-range enclosures are stacked directly on top of the subwoofers. In such cases, the full-range system becomes a mid-high system with power concentrated into a smaller bandwidth, reducing driver excursion and distortion. However, in cases where the full-range enclosures are separated from the subwoofers by a significant distance (more than six feet or two meters), there are distinct benefits to disabling the Lo Cut switch. If the Lo Cut switch is left in, the system will have distinct sonic origins for the low frequency and midrange, so that the sonic image becomes vertically disjointed. This creates an unnatural effect since musical instruments and other acoustic sources do not tend to propagate over frequency in this manner. Disabling the Lo Cut switch improves the sonic imaging of the system by spreading out the crossover vertically so that a gradual transition occurs between the systems.

A special note about the MSL-2A:

• The MSL-2A differs from other Meyer Sound biamplified products in that its low frequency range extends down to 40 Hz. Therefore, almost the entire range of an accompanying subwoofer system is also covered by the MSL-2A. In most cases the MSL-2A can be made to add very constructively to the subwoofer system, contributing additional acoustical power in the LF range. This may cause a peak in the low frequency response, which can be easily equalized. However, it adds significant LF acoustic power, which is always welcome. *Therefore, in most applications, do not engage the low cut control when adding MSL-2As with subwoofers.*

Note: The LF phase response of the MSL-2A is quite different from other biamplified products. Be careful to check that there is acoustical addition between the MSL-2A and subwoofers on a case-by-case basis. (See Section 4.9 on subwoofer polarity verification.)

The Lo Cut circuit is in when the switch is up.



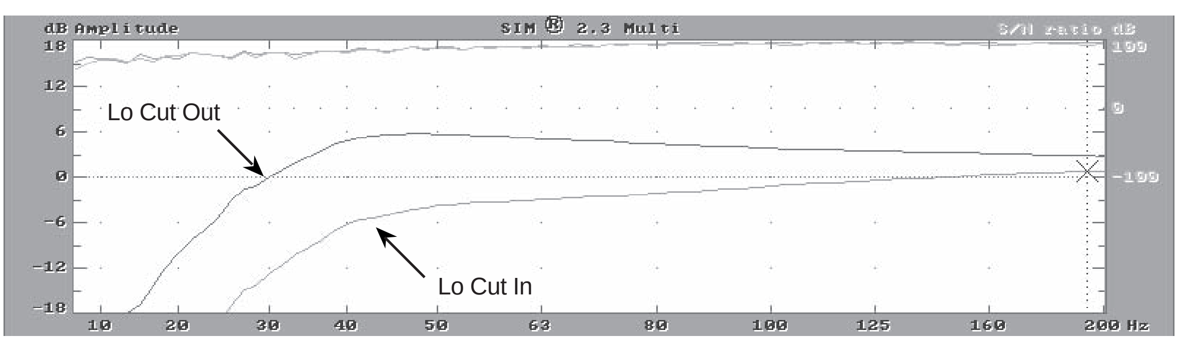


Fig 1.2m Lo Cut circuit response in the S-1 controller.

1.2.6 CEU User Controls

VHF/Cal Switch

M-1, M-3, and S-1 CEUs

The above controllers contain a filter circuit tuned in the extreme HF range. This circuit is intended to provide a simple pre-equalization to the response of the system based on the proximity of the listener. The response is tailored in the VHF range to compensate for distance and humidity related HF loss. Similar functions can be achieved from the system equalizer but this switch may save filters that could be used in other areas.

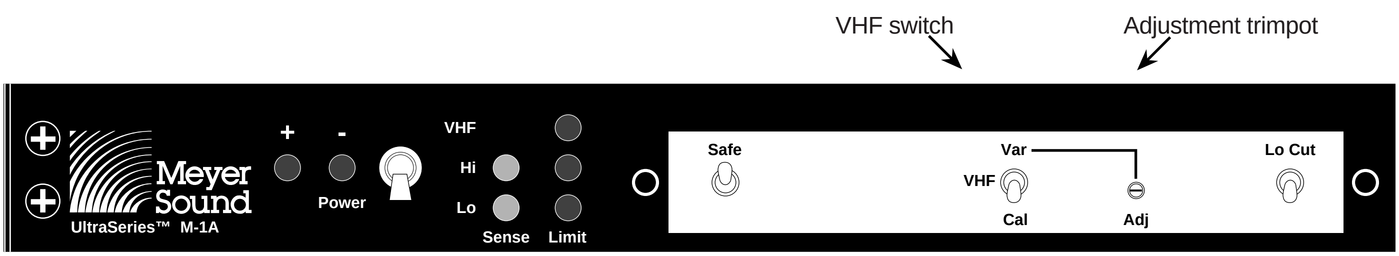
Using the VHF Switch

- If the coverage area is primarily in the near-field: The VHF circuit can be inserted and the VHF range attenuated.
- If the coverage area is primarily in the mid-field: The VHF circuit should not be inserted.
- If the coverage area is primarily in the far-field: The VHF circuit can be inserted and the VHF range boosted.

The strange truth about the VHF level control:

There are three different versions of the level control. The differences are as follows:

- M-1s: The M-1 has a ten-turn potentiometer. The orientation of the pot is reversed from what you would expect: *clockwise is cut and counterclockwise is boost*. The range is from –2 dB to +5 dB.
- M-1As: The M-1A has a single-turn potentiometer. The orientation of the pot is again reversed from what you would expect: *clockwise is cut and counterclockwise is boost*. The range is from –2 dB to +5 dB.
- S-1s: The S-1 has a single-turn potentiometer. The orientation of the pot is as you would expect: *clockwise is boost and counterclockwise is cut*. The range is from –3 dB to +3 dB.



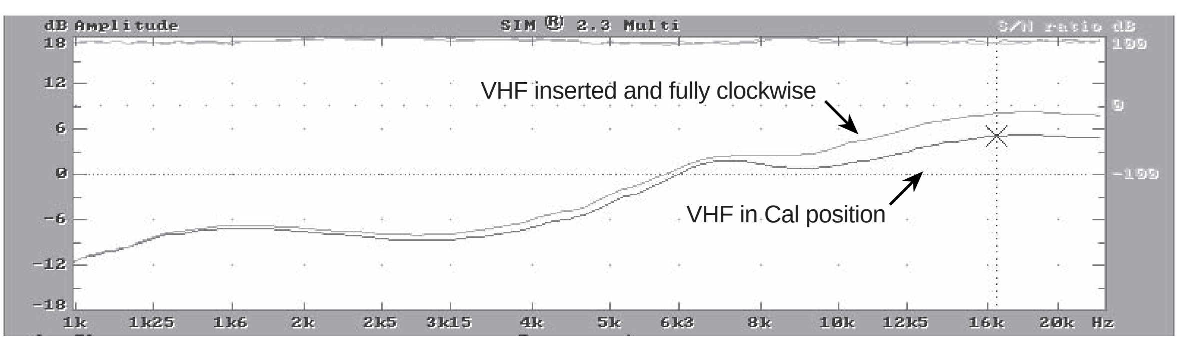


Fig 1.2n The VHF circuit response in the S-1 controller.

1.3.1 Introduction

All Meyer Sound Control Electronics Units employ SpeakerSense circuitry to protect the loudspeaker drivers from damage due to overheating and excessive excursion. Pioneered by Meyer Sound and incorporated into Meyer products since the company's founding, SpeakerSense is now widely imitated in the professional audio field.

The principle of SpeakerSense is relatively simple. Through a connection back to the CEU from the amplifier outputs (the "Sense" connection), the SpeakerSense circuit continuously monitors the power applied to the loudspeaker drivers. When the safe operating limits of the drivers are exceeded, signal limiters in the CEU act to clamp the signal level, protecting the drivers from damage.

Several types of limiters are found in Meyer CEUs:

- True RMS-computing limiters that act on the average signal level while allowing peaks to pass relatively unaltered.
- Excursion limiters that react quickly to protect the speakers from damage due to over-excursion.
- Peak limiters to control the peak signal level.

Limit Thresholds

SpeakerSense limiters are only engaged when the reliability of the system would otherwise be compromised. Every Meyer Sound speaker system is rigorously tested for both short- and long-term power handling. The limiting thresholds set for our products are set accordingly to allow the maximum levels with minimal sonic intrusion. These limits are not simply a matter of voice coil dissipation but must include the excursion limitations of the drivers and their mechanical limits. The complex acoustical impedance presented by an enclosure or horn will have a dramatic effect on excursion. Therefore, all limit thresholds are based upon the loudspeaker loaded in its enclosure. This is one of the factors behind Meyer Sound's approach to individual CEUs calibrated for each speaker model rather than "one size fits all" controllers with user adjustable limit thresholds. Such topologies can not factor in the precise short- and long-term power handling of different models.

Amplifier Loading

The SpeakerSense connection into the CEU presents a very high impedance (10 kΩ) to the power amplifier. The connections are opto-isolated so that there is no risk whatsoever that the sense connection will load down or otherwise compromise the reliability of the amplifier.

SpeakerSense and CEU Level Controls

The CEU level control has no effect on the limit threshold. The limit threshold is based on the actual power present at the speaker terminals. However, amplifier level controls (and amplifier voltage gain in general) will affect the system's protection capability. This is described in the next section, Amplifier Voltage Gain and SpeakerSense.

General SpeakerSense Rules

1. Do not insert any additional equipment between the CEU and amplifier.
2. Keep amplifier voltage gain between 10 and 30 dB.
3. If multiple amplifiers are driven from one CEU, sense the one with the highest voltage gain.

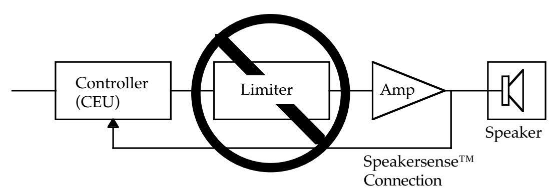


Fig 1.3a Additional limiters are not required for system protection and may actually compromise reliability as well as dynamic range.

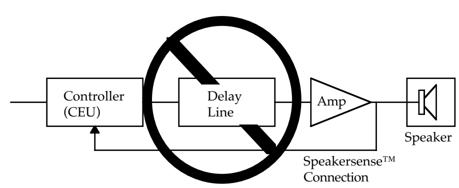


Fig 1.3b Delay lines should not be inserted here since it will disrupt the attack and release times of the limiters causing audible clipping and pumping.

1.3.2 Amplifier Voltage Gain and SpeakerSense

Power amplifiers must have a voltage gain of between 10 and 30 dB for proper operation of the protection circuitry. "Brickwall" limiters, despite their ability to limit voltage, are not used in Meyer CEUs because of their poor sonic characteristic. The RMS limiters used in Meyer Sound CEUs have a "soft" character allowing short-term peaks to go through without limiting, engaging only when required for long-term protection. This creates a graceful overload characteristic.

There is, however, a finite amount of compression available in the limiter circuit. Under normal circumstances this works perfectly well. However, if the amplifier voltage gain is excessive, the limiters can bottom out, endangering the speakers.

Figure 1.3c shows the basic flow of a system with SpeakerSense. Note that the power amplifier is within the feedback loop so that amplifier gain is seen by the controller. In addition, amplifier clipping, which doubles its output power, is seen by the CEU.

Example

Let's take a system with a limiter that protects a speaker that can dissipate 100 watts long term, and much more in the short term. Table 1.3d shows the power dissipation and compression for a system at various drive levels assuming a power amplifier with 23 dB voltage gain. Notice that as the limiting threshold is passed, the burst power is allowed to rise to 400 watts before the limiters engage.

Table 1.3e shows the same system with the amplifier gain increased to 32 dB. Notice that the CEU is capable of keeping the long-term power level delivered to the speaker at 100 watts. However, more compression is required in the limit circuit to achieve this. If the voltage gain is increased further, the compression required to protect the speaker will rise further, eventually overrunning the limiters and endangering the speaker.

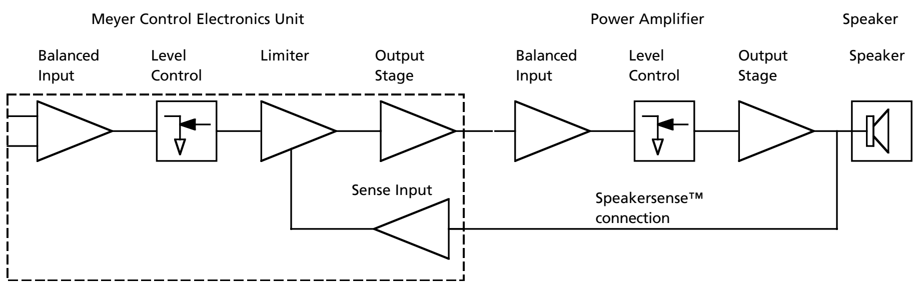


Fig 1.3c SpeakerSense signal flow block.

	Input	dB	CEU	Amplifier	8Ω Speaker
Limit Threshold = 100 watts	**Drive**	**Compression**	**Output**	**Output**	**Power level**
Amplifier Voltage Gain = 23 dB (14x)				**Voltage**	
Below limiting threshold	2V	0 dB	2V	28V	100 watts
Over threshold (Before onset of limiting)	4V	0 dB	4V	56V	400 watts
Over threshold (After onset of limiting)	4V	6 dB	2V	28V	100 watts
Drive level increased further	8V	12 dB	2V	28V	100 watts

Table 1.3d SpeakerSense limiting with amplifer gain at 23 dB.

	Input	dB	CEU	Amplifier	8Ω Speaker
Limit Threshold = 100 watts	**Drive**	**Compression**	**Output**	**Output**	**Power level**
Amplifier Voltage Gain = 32 dB (40x)				**Voltage**	
Below limiting threshold	.7V	0 dB	.7V	28V	100 watts
Over threshold (Before onset of limiting)	4V	0 dB	4V	160V	3200 watts
Over threshold (After onset of limiting)	4V	15 dB	.7V	28V	100 watts
Drive level increased further	8V	21 dB	.7V	28V	100 watts

Severe clipping!

Table 1.3e SpeakerSense limiting with amplifer gain at 32 dB.
Note that up to 21 dB of compression is now needed to fully protect the speaker.

Conclusion:
SpeakerSense circuitry is capable of accurately monitoring both short- and long-term power and amplifier clipping at voltage gains up to 30 dB without any need for user calibration. SpeakerSense allows you to take the speaker system to its full rated sound pressure level since it limits based *only* on the actual power limits at each speaker.

1.3.3 The Case Against Predictive Limiters

In order to cut costs, some manufacturers use predictive limiting instead of monitoring the signal at the speaker. This is similar to the old-style outboard limiters approach. This approach is not embraced at Meyer Sound due to its limitations in terms of dynamic range and protection. "Predictive" limiting is a form of limiting that assumes a given power level at the speaker for a given voltage at the controller output. This assumption relies on the amplifier voltage gain, which is an open variable. Any change in the amplifier level control moves the limiting threshold! If your amp gain is unknown, your limiter is de facto uncalibrated.

Figure 1.3f shows the basic flow of a system with predictive limiting. Note that the feedback is contained entirely within the controller and is not influenced by amplifier outputs.

Example

Let's take a system—with the limiter set to 2 volts at the controller output—that is charged with protecting a 100 watt speaker that would be destroyed by significantly higher, long-term power. Table 1.3g shows the power dissipation and compression for a system with a power amplifier of 23 dB gain. The results in this case would be similar to the SpeakerSense example shown previously.

Table 1.3h shows the same system with the amplifier gain increased to 32 dB. Notice that the compression occurs as before, but the actual power delivered to the speaker has increased to 800 watts. This, of course, would destroy the speaker. The inverse of this would occur if the amplifier gain was reduced, (such as when an amplifier level control is turned down) causing the limiters to engage prematurely.

In addition to the considerations outlined above, predictive limiting does not factor in the additional power generated by amplifier clipping since it does not monitor the amplifier outputs.

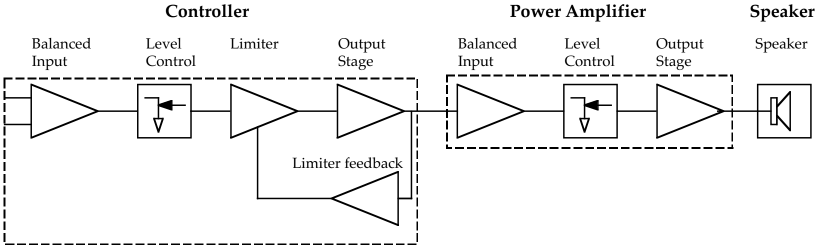


Fig 1.3f Predictive Limit signal flow block.

	Input	dB	CEU	Amplifier	8Ω Speaker
Limit Threshold = 2 Volt	**Drive**	**Compression**	**Output**	**Output**	**Power level**
Amplifier Voltage Gain = 23 dB (14x)				**Voltage**	
Below limiting threshold	2V	0 dB	2V	28V	100 Watts
Over threshold (Before onset of limiting)	4V	0 dB	4V	56V	400 watts
Over threshold (After onset of limiting)	4V	6 dB	2V	28V	100 watts
Drive level increased further	8V	12 dB	2V	28V	100 watts

Table 1.3g Predictive Limiting with amplifier gain at 23 dB.

	Input	dB	CEU	Amplifier	8Ω Speaker
Limit Threshold = 2 Volt	**Drive**	**Compression**	**Output**	**Output**	**Power level**
Amplifier Voltage Gain = 32 dB (40x)				**Voltage**	
Below limiting threshold	2V	0 dB	2V	80 V	800 watts
Over threshold (Before onset of limiting)	4V	0 dB	4V	160V	3200 watts
Over threshold (After onset of limiting)	4V	6 dB	2V	80 V	800 watts
Drive level increased further	8V	12 dB	2V	80 V	800 watts

Blown speaker

Table 1.3h Predictive Limiting with amplifier gain at 32 dB.
Note that the speaker is being driven to 800 watts.

Conclusion: In order to be effective, predictive limiters must be reset with every change in voltage gain, must know when an amplifier is clipping and must know exactly how much instantaneous and long-term power the speaker is capable of dissipating. *Any change in these parameters will require recalibration of the limiters* if the system's dynamic range and protection capability are to be preserved.

1.3.4 Standard Sense Connections

The SpeakerSense connection is shown in Fig 1.3i.

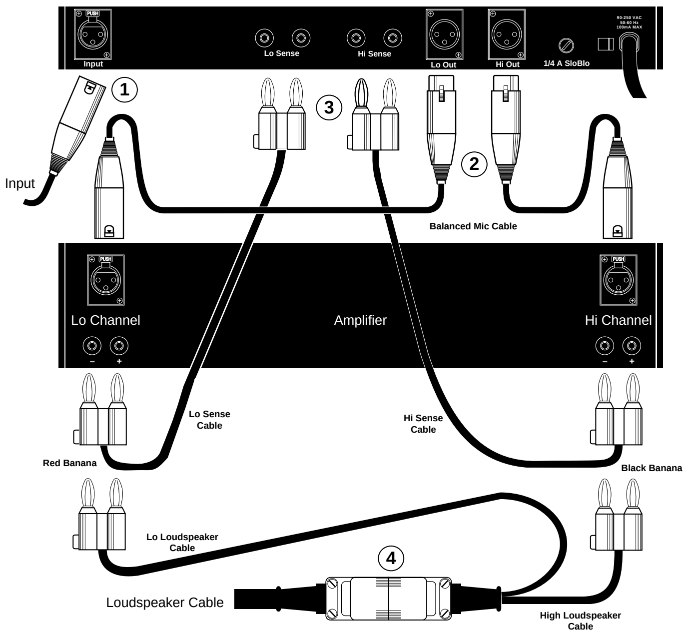


Fig. 1.3i Basic connections.

① Balanced input signal line into CEU.

② Balanced CEU outputs to drive the power amplifier.

③ SpeakerSense connection from amplifier to speaker.

④ Loudspeaker cable connection.

For CEU models with single sense inputs per channel the following rules apply:

- When multiple amplifier channels are driven from the same CEU, the Sense connection must come from the amplifier with the highest voltage gain. If the gains are all the same, then any channel could be used.
- Do not connect the sense lines together. This would create a short circuit between the amplifier output terminals.
- If the amplifier that is being "sensed" stops passing signal, then the system will no longer be protected. Therefore, it is vital to verify that the amplifier is working properly.

CEU models with single Sense inputs/channel:

B-1	B-2	B-2A	B-2Aex	B-2EX
P-1	P-1A	P-2	MPS-3	
M-1	M-1A	M-3	M-3T	M-3A

Driving Multiple Amplifiers

It is typical practice to drive several power amplifiers from a single CEU. When doing so, the sense line should be connected to the amp with the highest gain.

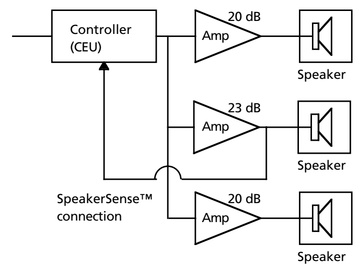


Fig 1.3j Sensing one of several amplifiers. Sense line connected to the correct amplifier.

This system block has three amplifiers with different voltage gains. The sense connection is made to the amplifier with the highest gain.

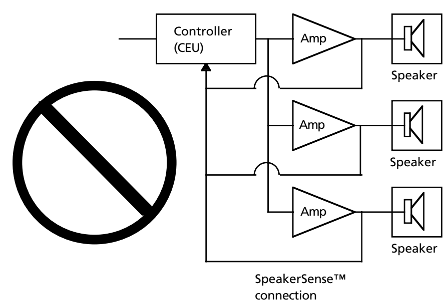


Fig 1.3k Sensing one of several amplifiers. Sense line connected to amplifier with lower gain.

Here the sense connection is made to the wrong amplifier (lower gain) and therefore will not fully protect the speaker. The limiters would engage to protect the speaker on that amplifier. However, speakers powered by the other amplifiers would limit at 3 dB higher power, effectively doubling the power allowed into the speakers.

1.3.5 MultiSense™ Connections

Newer model CEUs have incorporated an advanced sense circuit that is capable of sensing multiple amplifiers. This circuit automatically senses the amplifier with the highest voltage gain. This further enhances the reliability of the system in that a single amplifier failure will not compromise protection. In addition, the user does not have to monitor which amplifier has the highest voltage gain. An example of a MultiSense connection is shown in Fig. 1.3l.

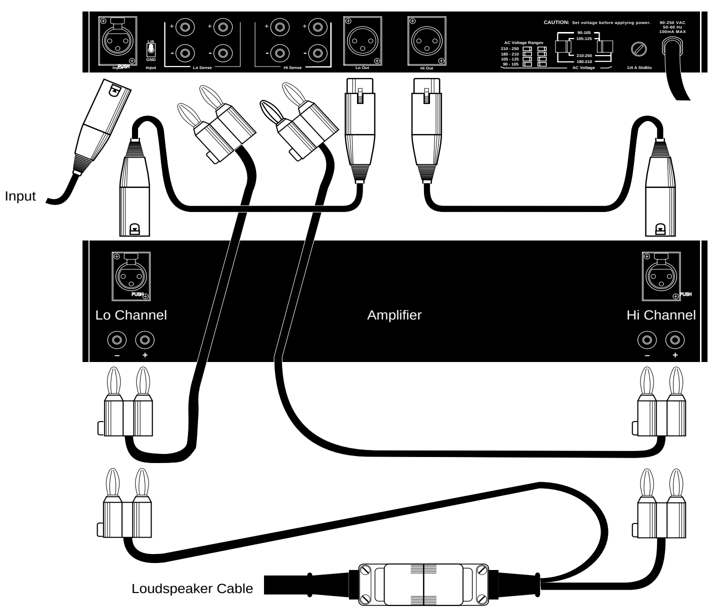


Fig. 1.3l MultiSense connection.

This CEU can accommodate two Hi and two Lo amplifier channels. The second amplifier plugs into the additional Hi and Lo sense connections.

> **For CEU models with multiple sense inputs per channel the following rule applies:**
>
> • Polarity of the sense connection must be the same for all channels.

> **CEU models with MultiSense:**
>
> **S-1 M-10A M-5 D-2**

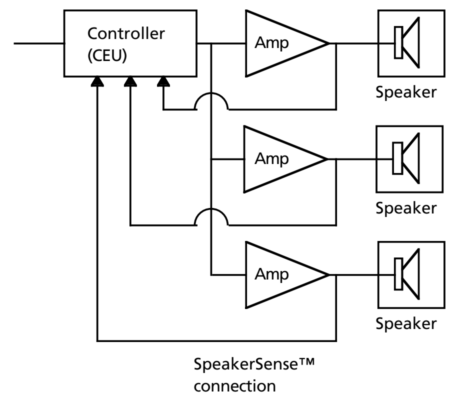


Fig. 1.3m MultiSensing one of several amplifiers. Sense lines connected to multiple CEU sense inputs.

Here the sense connections are made correctly. Each amplifier is returned separately to the CEU sense inputs. The CEU will look to see which of the amplifiers has the highest gain and will limit as required. Sense lines must all have the same polarity.

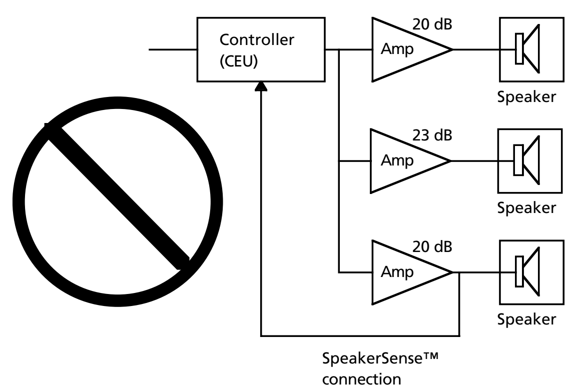


Fig. 1.3n Improper MultiSensing. Sense lines shorted together.

The sense connections are made incorrectly. The amplifier outputs have been shorted together. This will endanger the output devices of the amplifiers.

1.3.6 Limiter Operation

The "Safe" Switch (for all CEUs Except B-2EX)

Most of the Meyer Sound CEUs have a limiting threshold adjustment termed "Safe," which reduces the remote monitor system (RMS) limiting threshold by 6 dB. This reduces the system's maximum continuous level to one-fourth of full power. It is typically used in order to obtain the maximum system reliability in high power applications. However, it does not absolutely guarantee that the system cannot be overdriven, nor is the switch required for the system to operate safely. In other words, switching the "safe" circuit *out* does not set it to "unsafe," but rather to its standard setting, which is already *very* safe. It is not unusual to hear from users who have rarely used the full power setting for fear of blowing drivers or voiding the warranty. This should not be a concern. In retrospect, the switch labeling is somewhat of a misnomer. It would have been better named "–6 dB Limit" and "Full Power."[1] Meyer Sound speaker systems are designed to maintain continuous extreme power levels at their full power setting without failure. Therefore, one should consider the full power position the default setting rather than vice versa. It is critical to the satisfaction of mix engineers to obtain the maximum dynamic power from the system, and therefore users should not reduce the dynamic range of Meyer systems without just cause.

RMS Limiter Time Constants

The limiting circuits affected by the Safe switch are the RMS limiters. All peak and excursion limiters are independent of the switch. These RMS limiters are relatively soft, creating a graceful overload characteristic, unlike the brickwall-type limiters that give a hard sound. They are designed to act slowly, so that the short-term high power peaks are preserved since they pose no danger to the drivers. The attack and release times of the limiters are different for the two threshold settings, because there is less integration time required to actuate the limiters when in the "safe" position. The decay time will also be lengthened since the signal must decay further before it goes under the release threshold. This means that the system is likely to spend a much greater amount of time in limit, and that this limiting is likely to be much more audible.

Amplifier clipping versus limiting

With the exception of the MSL-5 and MSL-10A systems, the limiting action of the CEUs will not prevent amplifier clipping. This is done to preserve maximum dynamic range of the system as described above. Whereas the action of the limiters is more audible in "safe," amplifier clipping will be more audible with the "safe" switch out.* The lower threshold of the "safe" setting will tend to pull amplifiers out of clipping much faster than the full power setting. If the clipping is of short duration, such as with a snare drum signal, then it will probably be less objectionable than engaging the limiters. Therefore, the full power setting may be the best choice. Conversely, if continuous signals such as vocals are run into hard clipping, it will be fairly noticeable, and therefore the "safe" position may be a better choice. The decision of whether or not to use the "safe" setting can be based as much on the sonic quality of the system for the given program material as it is for system reliability on a case by case basis.

When Should "Safe" be Used?

The system is not in any significant danger unless the limiters are engaged for long periods of time. In this case the "safe" position is warranted. However, if the system is already in "safe" and you are seeing continual limiting action, try switching it out and observe. If the limiters are intermittently or no longer engaged, then the full power setting will return dynamic response to the system that had been compressed, which may be more satisfactory to the mix engineer. Conversely, if the limiters are still fully engaged, then a return to the "safe" setting is indeed warranted.

It is a fairly typical practice to open a show in "safe" as the engineer gets settled into the mix. The additional compression provides a cushion in case a channel jumps up in the mix. Later, the system can be opened up to maximize the dynamic range.

* This is more true of systems using Type 1 amplifiers (UPA, UM-1, etc.) than those with Type 2 amplifiers such as the DS-2 and MSL-2A.

[1] "Full Power" is reached when the system is driven hard in safe. This allows for a compressed sound to be generated by the processor.

1.3.6 Limiter Operation

The "Autosafe" Circuit of the M-5 and M-10A

The "safe" setting for these systems functions identically to those above. However, the alternate position, termed "autosafe" is different. These systems employ a circuit to monitor the long-term power dissipation of the system over several minutes. When switched to the autosafe position the system will run with the normal full power limiter settings. If the system is run into continuous overload over a long period of time, it will automatically switch itself into the safe position, reducing the limiting threshold by 6 dB. The Safe LED will light to indicate the change. After the system has sufficiently reduced its long-term dissipation, the threshold will reset to full power. Therefore, since the autosafe circuit effectively monitors whether the system is being overdriven in the long term, there is very little need to engage the standard safe setting.

Sense and Limit LED Indicators

The CEU front panel has separate "Sense" and "Limit" LEDs to indicate the SpeakerSense status.

Sense LED

If the Sense LED is green, this indicates signal presence at the Sense Inputs. On CEUs equipped with amplifier voltage gain checking, the Sense LED is bi-color. It will turn red if the amplifier voltage gain falls out of the required range. Typical causes of this are excess voltage gain, the amplifier turned off, or improper connection of the sense line. The gain sensing works by comparing the CEU output signal with the sense return from the amplifier. Therefore, it will only indicate red when a signal is present.

The following CEU models have gain sensing:

CEU	Allowable range
S-1	10–30 dB
D-2	10–30 dB
M-1E	10–30 dB
B-2ex	10–30 dB
M-5	15–17 dB
M-10A	15–17 dB

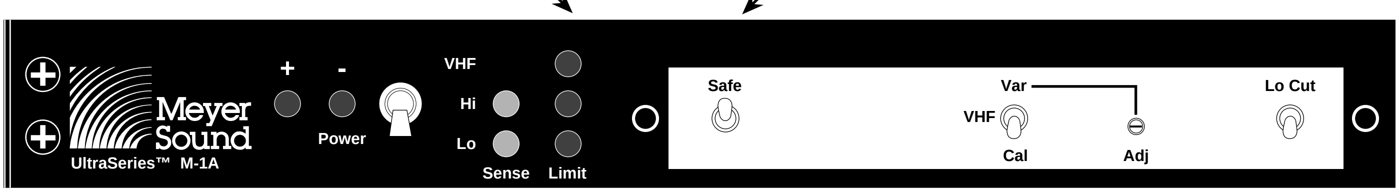
The gain sensing does not mute the speaker except on the M-3T (see Section 6 for details). It merely indicates the need to optimize the amplifier gain. In the case of the M-5 and M-10A the range is very tight (± 1 dB). Users should not be alarmed if the Sense LED occasionally turns red under dynamic operating conditions. This can be due to compression or distortion in the amplifier.

Limit LED

The Limit LED will only light when the limiters have engaged. Most of the limit LEDs indicate the action of a single limiter, such as the RMS or VHF. The Limit LEDs in the M-5 and M-10A CEUs show the action of multiple limiters, peak RMS and excursion together. Therefore, it is normal that these Limit LEDs will be lit more than you would expect to see with the single function Limit LEDs.

Limit LEDs indicate onset of limiting

Safe switch



Sense LED shines green indicating proper operation.

Sense LED shines red when there is a fault condition.

1.3.6 Limiter Operation

Excursion Limiters

Several CEUs incorporate peak excursion limiters. These are fast-acting, frequency-selective limiters that prevent the driver from over excursion. These have a fast attack and a slow release, and are admittedly the most audible of the limiters in Meyer CEUs. They are, however, highly effective in their protection.

The B-2EX Safe Circuit

The "safe" switch in the B-2EX controller has two functions: In addition to reducing the RMS limiting threshold by 6 dB (as with all of the others), it also switches in the excursion limiter. When the "safe" switch is out, the excursion limiter is disabled.

Some Comments About Outboard Limiting and Compression

In addition to the limiting action of the CEUs, most systems carry a variety of outboard compressor units, patched either into channels, subgroups or the main outputs. The following should be considered regarding their usage:

Compressor/Limiters on the main outputs will not increase the system reliability and, in fact, may significantly compromise it. Peak limiters, and "brickwall" types in particular, will degrade the system's performance and reduce system reliability. Stiff limiters will reduce the peak to average ratio with their fast acting attack. These are, of course, those very same peaks that the Meyer CEU is designed to let pass since they will not endanger the speakers. The removal of the peaks is then followed by an increase in drive level as the mixer strives for the feeling of dynamic power. This eventually leads to a dense, compressed and distorted signal that engages the CEU's RMS limiters continually because the dynamics are insufficient to allow the limiters to release. Removing all of the peak power capability of the speaker system, results in a 10 to 12 dB of peak pressure reduction (16 to 18 dB in "safe"). To make matters worse, you might guess what the reaction of some mixing engineers will be to having the system sound like this: They turn it up! Now there is real danger of burning voice coils.[1]

The same considerations are valid with outboard compressors placed on all the key channel and submasters. If these are all compressed, then the same result may occur.

These points are particularly relevant to the **MSL-10A** and **MSL-5** speaker systems. These systems have an extremely high peak-to-average ratio. Highly compressed drive signals into these systems will lead to disappointing results. Open it up—let Meyer Sound take care of the dynamics!

[1] The processor will limit power to the speaker independent of the type of signal used (however there is a tendency for the system to be pushed harder when the consoles are being clipped).

1.4.1 Output Power Classifications

There are three power amplifier classifications for use with Meyer Sound speaker products. The classifications are primarily distinguished by different maximum power output capability, but other features are considered as well.

Type 1 and Type 2 Power Amplifier Specifications

Types 1 and 2 are for use with the general speaker line, with the Type 2 generally having 3 dB more power. The Type 1 and 2 classifications do not include specific brands and/or models of power amplifiers, but rather serve as a guide for choosing the best model for your needs.

Voltage Gain: Must be a minimum of 10 dB to a maximum of 30 dB when measured from input to output.

Mains AC Power: The AC power inlet must be a three-circuit grounded plug with the earth (mains AC) ground permanently connected to the chassis. The amplifier must meet the power output criteria specified below over a line voltage range of 100V to 240V AC, 50/60 Hz (which may be split into selectable ranges).

Why can't I use Type 2 amplifiers on all of the products?

In order to accommodate the increased peak power of the Type 2 amplifiers, the CEUs must incorporate fast acting peak excursion limiters. This has been implemented in the S-1, D-2 and B-2EX CEUs but not in the M-1A or M-3A. If Type 2 amplifiers are used with these products, the reliability will be compromised due to the excess peak power.

How much louder would my UPA be if I used a Type 2 amplifier?

At first there would be an addition of 0 dB of continuous SPL and 1 to 3 dB of peak SPL. The continuous level is governed by the CEU rather than the amplifier. Then, there would be a reduction of 125 dB continuous when the drivers are blown.

Type 1 Power Amplifier

FTC rating at 8Ω: 150–350 watts

FTC rating at 4Ω: 300–750 watts

Type 1 power amplifiers are for use with:

UPA-1	UPA-2	UM-1
UPM-1	UPM-2	MSL-3A
MST-1	MPS-355	MPS-305

650-R2, MSW-2 or USW-1 when used with B-2 or B-2A Controller.

Type 2 Power Amplifier

FTC rating at 8Ω: 350–700 watts

FTC rating at 4Ω: 700–1500 watts

Type 2 power amplifiers are for use with:

USM-1	MSL-2A	DS-2

650-R2, MSW-2 or USW-1 when used with B-2Aex or B-2EX Controller.

1.4.1 Output Power Classifications

Type 3 Power Amplifier Specifications

The Type 3 class is used exclusively for the MSL-5 and MSL-10A systems, and is specifically designed to work with the specialized protection circuitry of the M-5 and M-10A CEUs. These amplifiers should not be used on other products, nor should the MSL-5 or MSL-10A speakers be powered by other amplifiers. Type 3 amplifiers are limited to those amplifiers which are "Meyer approved," having satisfied all of the stated criteria. Contact Meyer Sound for the current list of approved Type 3 amplifiers.

Voltage Gain: 16 dB, internally fixed.

Power Output

0.5 second burst at 4Ω	1800 watts
FTC rating at 8Ω	1100 watts

Nominal (235 VAC) Operation: With 4Ω resistive load, reproduce three specified burst waveforms[1,2,3] each continuously for 1 hour without shutdown or limiting.

High (255 VAC) Mains Operation: With 4Ω resistive load, reproduce a 400 msec sine wave burst at 255 watts, 2.8 second burst interval, continuously for 1 hour without shutdown or limiting.

General: Latch-up protection, indicators for clipping, limiting, thermal overload.

There are several questions regarding Type 3 amplifiers that require some explanation.

Why 16 dB voltage gain?

The M-5 and M-10A use a combination of SpeakerSense™ and predictive limiting. Because of the limitations of predictive limiting (see section 1.3.3), the gain must be fixed. Predictive limiting is used so that the limiting can engage before the amplifier has reached the clip point.

Can't I just use a different amplifier and turn it down to 16 dB gain?

It's not that simple. Many models of power amplifiers have pre-attenuator, balanced input stages. These would then be clipped by the CEU and the signal would distort.

Isn't 1100 watts too much power for the speakers?

This is a tremendous amount of power. But, since the amplifier is never allowed to clip, the power remains safely harnessed, maximizing the distortion-free dynamic range.

With the gain so low how can I drive the speakers to full power?

The M-5 and M-10A CEU have approximately 16 dB of throughput gain to make up for the low gain in the power amplifier. From the point of view of the mix outputs the system reacts as if it has the combined voltage gain of 32 dB. The mixer will have no trouble bringing this system to full power.

Can I use Type 3 amplifiers for the other products?

No. The voltage gain would be too low. The other CEUs have a throughput gain approaching unity. The 16 dB of gain at the power amp would not allow the system to be brought to full level with any degree of headroom.

1) Cycle consisting of 50 msec sine wave at 120V peak and 450 msec sine wave at 24V peak.
2) Cycle consisting of 25 msec sine wave at 120V peak and 975 msec sine wave at 41V peak.
3) Cycle consisting of 400 msec sine wave at 120V peak and 2400 msec sine wave at 0V peak.

1.4.2 Power Amplifier Voltage Gain

Power amplifiers increase line-level audio signals to a power level suitable for driving loudspeakers. Possibly the simplest and most basic components in an audio system, amplifiers may easily be taken for granted. Yet their electrical characteristics can affect both sound quality and reliability in reinforcement loudspeaker systems.

Why Control Gain?

The voltage gain of an amplifier determines the input signal required to drive the amplifier to a given output level. Amplifiers with high gain require less input voltage to reach full power that those with lower gain.

On the face of it, one might conclude that more gain is better. Wouldn't raising the gain increase the system headroom? In actuality, this is true only if the stages that feed the amplifier (the mixer outputs, for example) are clipping before the amplifier does. This is rarely the case. If the amplifier is the first component in the system to clip, then raising its gain further will be detrimental.

Every gain stage will amplify not only the audio signal, but also any unwanted noise that is generated by the stages which precede it. The power amplifier is the last component in the chain before the loudspeaker. The higher the gain, the louder the noise will be when your system is idling.

Excess gain means that the amplifier will likely spend more time in clipping. If the amplifier's power capability significantly exceeds the power handling capacities of the loudspeakers, the clipping can also damage the speaker components.

The most common misperception about amplifier gain is that amps with more gain have more power, and turning an amp down would be throwing power or headroom down the drain. In actual fact, moderate amp gain will optimize dynamic range by keeping the noise level low, while using the full rated power of the amplifier.

Recommended Amplifier Voltage Gain Range

The power amplifier voltage gain is the ratio of input to output voltage. This number determines the amount of input voltage required to bring the amplifier to full power and is independent of the amplifier's maximum power output capability. The effectiveness of SpeakerSense circuitry in protecting the speakers depends on both the amplifier's maximum output capability and the voltage gain, and must be between 10 and 30 dB for proper operation. (see Section 1.3.2). Consult the owner's manual of your amplifier to determine voltage gain, or measure it directly.

Voltage Gain Specifications

Amplifier specification sheets have three different ways of denoting voltage gain:

1) dB voltage gain.

2) Multiplier (ratio of output voltage to input).

3) Sensitivity (input voltage required to achieve full voltage swing at the output.

If the manufacturer specifies the multiplier:

The specification will read something like:

Voltage gain = 20X

Voltage gain and the multiplier are related below:

$$\mathbf{dB}_{\text{Voltage gain}} = 20 \log \left(\mathbf{V}_{\text{Output}} / \mathbf{V}_{\text{Input}}\right)$$

Which is to say,

$$\mathbf{dB}_{\text{Voltage gain}} = 20 \log \mathbf{V}_{\text{Multiplier}}$$

Formula 1.4a

e.g., 20 Log (20 volts output/1 volt input) = 26 dB

As an alternative, Fig. 1.4b can be used to determine the voltage gain in dB from the multiplier.

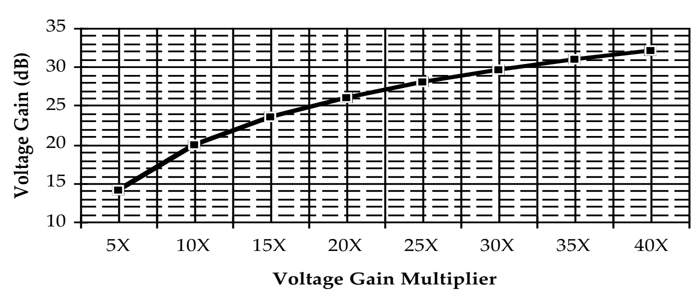


Fig. 1.4b Voltage gain multiplier versus dB gain.

1.4.2 Power Amplifier Voltage Gain

If the manufacturer specifies sensitivity, the specification will read something like:

Sensitivity = .775V input drive for full rated output.

Amplifiers that use a standard sensitivity for full power output have a different voltage gain for each model of amplifier since they each have different rated output power. Models that specify sensitivity require a more complex calculation since it is necessary to determine the voltage level at the output when the rated power is achieved. It is best to use the 8Ω power rating since the voltage will tend to sag under low impedance load conditions, yielding a slightly lower voltage gain number.

To determine the output voltage at rated 8 Ω power:

$$\mathbf{V}_{\text{Max Output}} = \sqrt{\text{rated } 8\ \Omega \text{ power (watts)} \times 8}$$

Formula 1.4b

For example an amplifier is rated at 313 watts into 8Ω with a sensitivity of .775V for full power. First we solve for the maximum output voltage:

$$\mathbf{V}_{\text{Max Output}} = \sqrt{313 \times 8}$$

$$\mathbf{V}_{\text{Max Output}} = \sqrt{2504}$$

$$\mathbf{V}_{\text{Max Output}} = 50 \text{ volts}$$

Having now determined the voltage at full power output, the multiplier can be found by dividing it by the input voltage (the sensitivity figure.)

$$\mathbf{V}_{\text{Multiplier}} = \mathbf{V}_{\text{Output}} / \mathbf{V}_{\text{Input}}$$

Formula 1.4c

The European Meyer standard for voltage gain is 23 dB (14x).

23 dB

The North and South American Meyer standard for voltage gain is 26 dB (20x).

26 dB

Continuing our example:

$$\mathbf{V}_{\text{Multiplier}} = 50 / .775$$

$$\mathbf{V}_{\text{Multiplier}} = 64.5\text{x}$$

Once the multiplier is determined the voltage gain can be determined using formula 1.4a.

$$\mathbf{dB}_{\text{Voltage gain}} = 20 \log\ 64.5$$

$$\mathbf{dB}_{\text{Voltage gain}} = 36.2 \text{ dB}$$

This is above the safe voltage gain limit of 30 dB and should be reset.

There is an important difference between amplifier models that are manufactured to a standard dB voltage gain and those set to a standard sensitivity. Fig 1.4c shows the relationship between these two standards. Notice that the sensitivity rated amplifiers have higher voltage gain for higher output power, whereas the dB voltage gain-rated units have a constant gain. Notice also that where the voltage gain has exceeded 30 dB, the effectiveness of the SpeakerSense circuit is compromised at the same time that amplifier power is increasing (see Section 1.3.2).

Since it is common practice for many users to build systems with a mix of different models of amplifiers (with different power ratings), those using sensitivity-rated amps may be unwittingly uncalibrating their system. This will also affect the crossover point in multi-way systems, causing them to shift in frequency and phase (see section 1.4.5).

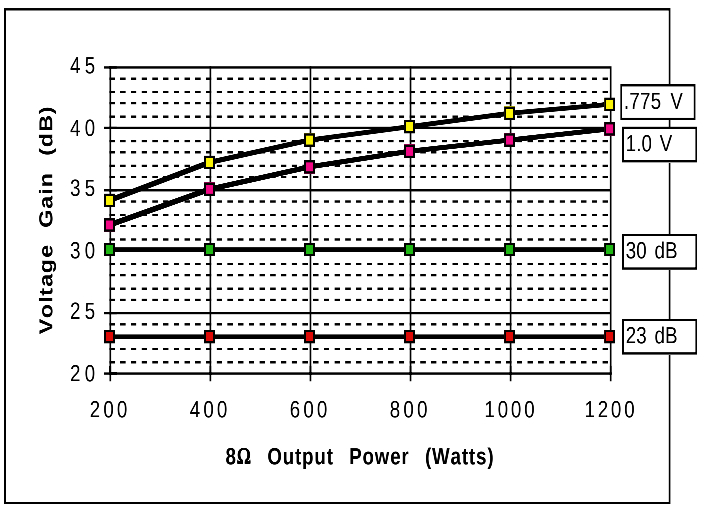


Fig. 1.4c Amplifier sensitivity versus dB gain.

1.4.3 Amplifier Level Controls

The standard markings for the level controls of power amplifiers is in dB attenuation. This is actually rather confusing when one considers the fact that these amplifiers are not attenuators at all. They are quite the opposite —they are amplifiers! The front panel markings refer only to dB reduction in voltage gain *relative* to the fully clockwise (maximum) setting.

The level control on a power amplifier:

A) Does not reduce the amplifier's maximum output power unless it is turned so low that the device driving the amplifier input clips before it can bring the output to full voltage swing.

B) Does not necessarily correlate between models of amplifiers—even between different models of the same manufacturer *unless* they have been set to the same maximum voltage gain.

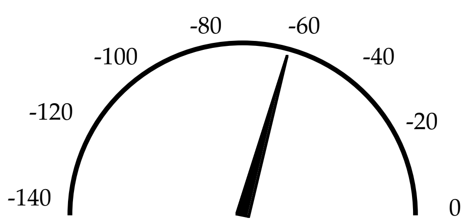


Fig 1.4d Amplifier level control logic applied to a speedometer.

If automobile manufacturers were to adopt the same logic as power amplifier designers, your speedometer might be rated in miles per hour *under* the maximum speed of your car as shown in Fig 1.4d.

A race car and a school bus are traveling at –60 m.p.h. according to their speedometers. How fast are they going?

To make the power amplifier level controls usable as relative level controls:

- Set all amplifiers in your system to a standard maximum voltage gain.
- If the levels of some models cannot be reset to the standard, then mark the attenuator position that correlates to your standard.

Will these 300 watt amplifiers give you the same output level? Not necessarily.

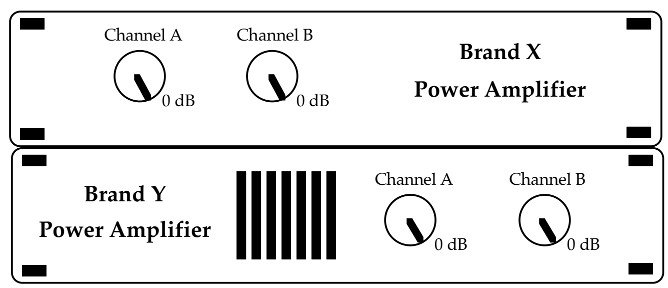


Fig. 1.4e Two 300-watt amplifiers with all channels set to 0 dB.

The front panel settings might lead you to believe that these amplifiers were matched. However, the gain may be completely different.

1.4.4 Drive Level Requirements

The CEU output must be capable of cleanly driving the amplifier input sufficiently to bring the amplifier to full power. Typical, CEUs can drive +26 dBu (+24 dBV), which is sufficient to drive most amplifiers well into clipping, even with relatively low voltage gains. Chart 1.4f shows the drive levels required to reach full power at various maximum power output ratings.

For example, a Type 2 amplifier rated at 400 watts into 8Ω, and set to the European standard of 23 dB, would reach full power with a drive level of +14 dBu. This would leave 15 dB of headroom in the system. In other words, the amplifier could be driven 12 dB into clipping before the CEU itself clipped.

Note: M-5 and M-10A systems have a unique gain structure and are not represented in the chart below. See Section 1.4.1 for an explanation regarding these systems.

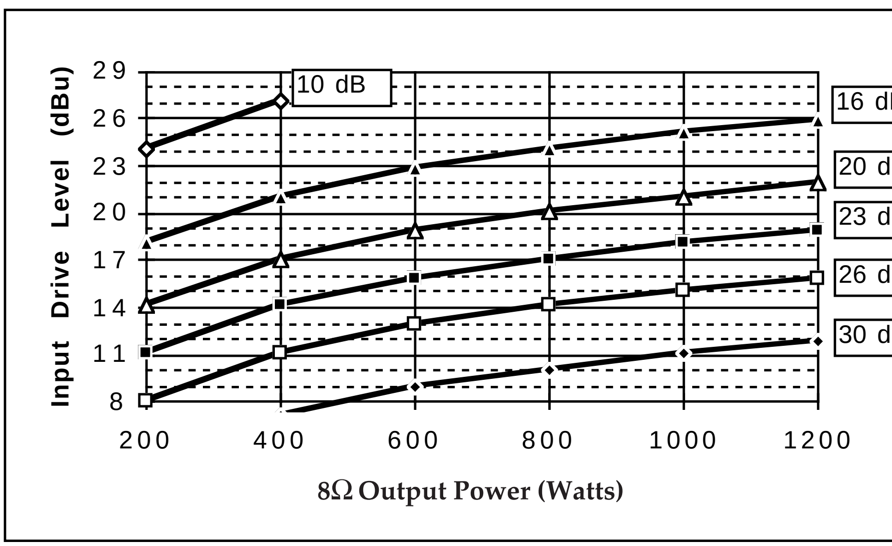


Fig 1.4f Drive level requirements for full power amplifier output.

1.4.5 Matching Amplifier Voltage Gain

One of the most common mistakes made with Meyer Sound systems is the operation of HF and LF amplifiers at different voltage gain settings. It seems simple enough. If there is too much low end, turn down the LF amplifier. If it is too bright, turn down the HF amplifier. People seem to feel much better if they can keep from using their equalizer. Unfortunately there are some serious side effects to this practice that should be considered. For example, is it better to save a filter in the low end if it means you are more likely to destroy your HF driver?

The most common practice is to turn down the LF amplifier.

Why turn down the LF amplifier gain?

- The speaker is coupling with the room in the low end. Turning down the LF amplifier can save filters in the low end.
- An array of speakers is coupling in the low end. Turning down the LF amplifier can save filters.
- The LF amplifier has twice the power capability of the HF amplifier (a mistake). Turning it down will reduce the power capability (a fallacy).

What are the side effects of unbalancing the drive levels to the speakers?

Reducing the LF amplifier gain:

- Decreases the LF buildup as desired.
- Shifts the acoustic crossover down in frequency.
- Requires the HF driver to carry more of the MF power response.
- Misaligns the phase relationship at crossover causing possible phase cancellation.
- Alters the directional response at crossover.
- Leaves a dip in the midband because the LF coupling rarely reaches up to the crossover region.
- Increases MF distortion. The HF driver distortion is worst below its passband.

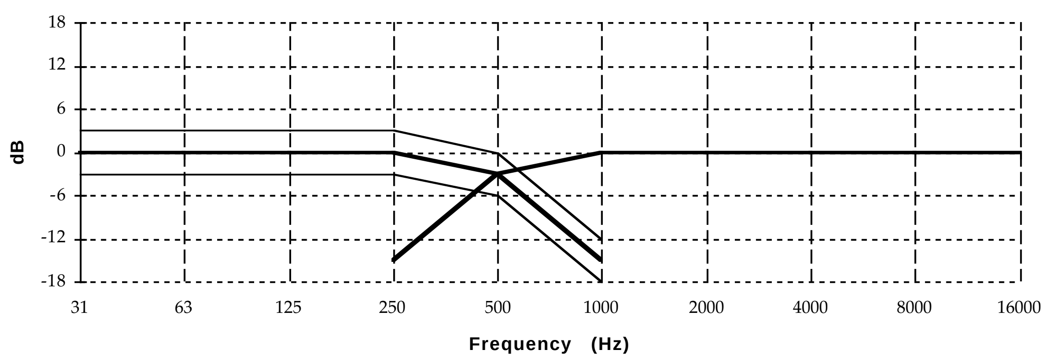


Fig. 1.4g Unbalancing the acoustic crossover.

The acoustical crossover point of any speaker system is affected by the relative amplifier gain. In this example, the crossover is 500 Hz when the gains are the same. If the gain of the Lo channel is raised by 3 dB, the crossover rises to 630 Hz. If it is lowered 3 dB, it drops to 400 Hz.

1.4.5 Matching Amplifier Voltage Gain

Any given program material requires a certain amount of midrange power. What will supply it? Will it be the LF and HF drivers coupling together with a phase aligned crossover as shown in Fig 1.4h? Or will it be the HF driver alone, running below its passband and out of phase at the actual acoustic crossover as shown in Fig 1.4i?

The coupling of LF drivers, either to the room or each other, is pure efficiency gain. LF coupling means more power for less drive, creating more headroom and less distortion. It can easily be equalized, if desired. If the LF amplifier is turned down you will throw away the benefits of coupling and penalize the HF driver for not coupling by requiring it to handle more of the midrange power.

! **All biamplified Meyer loudspeaker systems use the same power amplifier voltage gain for the HF and LF channels.**

There is a huge difference between equalizing the coupled energy and turning down the LF amplifier.

If the amplifiers remain matched and the coupling is equalized:

- LF buildup is reduced as desired.
- The acoustic crossover is maintained.
- LF and HF driver power response is optimized.
- Phase relationship at crossover remains optimized.
- Directional pattern remains optimized.
- There is no dip in the midband.
- Midrange distortion is minimized.

Note: The above caution is particularly true of the MSL-3 due to its low acoustic crossover. Turning down the LF drivers will seriously endanger the MS-2001A.

Energy is shared in crossover region. Crossover is centered at 1200 Hz.

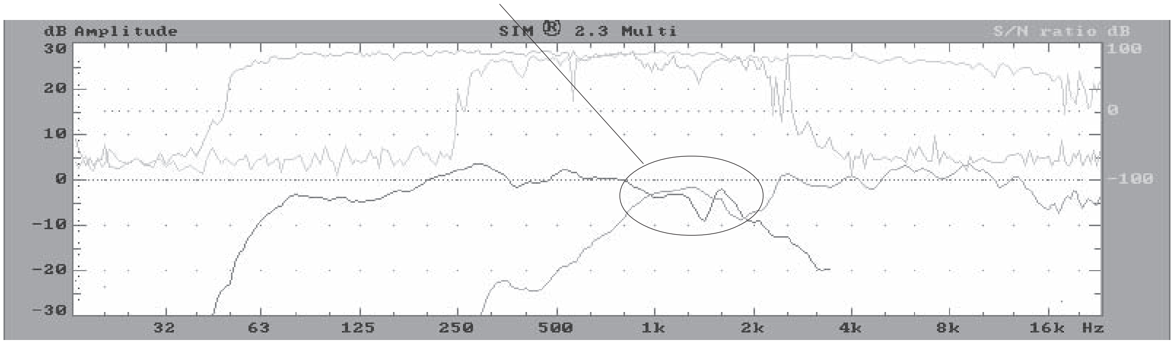


Fig 1.4h UPA-1C with matched voltage gains at crossover.

The HF driver must supply the acoustic power down to 900 Hz.

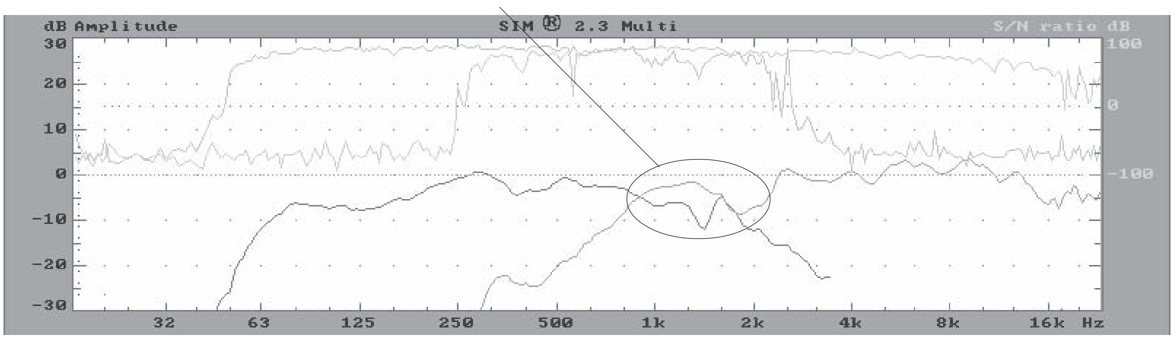


Fig 1.4i UPA-1C with unmatched voltage gains at crossover.

1.4.6 Matching Output Power

HF and LF channel amplifiers for all Meyer Sound biamplified systems should have the same power rating. Forget the old school practice of powering the HF driver with a lower wattage power amplifier. The power handling capability of the individual components has already been factored into the design of the speaker system's protection circuitry. Lower wattage power amplifiers will clip earlier, creating distortion and effectively doubling their average power output. As a result, this will noticeably degrade sonic quality and may compromise reliability.

The whole issue of wattage clouds things. The key point here is voltage swing. The HF driver must have sufficient voltage swing to follow the crest of the input waveform.

Figure 1.4j shows the impulse[1] response of a speaker system. This is the transient response of the system as you would see on an oscilloscope, indicating the type of waveform created when a pulse signal (not unlike a snare drum) is put into the speaker. The highest part of the peak is the high-frequency content. If the HF amplifier does not have sufficient voltage swing the transient will be clipped, resulting in lost dynamic range.

The protection of the driver from excess long-term power will be handled by the SpeakerSense™ limiting.

There is a recent trend in amplifier manufacturing where models are created that have two channels of different output power (e.g. 600 watts and 150 watts). These are marketed as appropriate for LF and HF drivers respectively. *These amplifiers are not recommended for biamplifying Meyer speakers!*

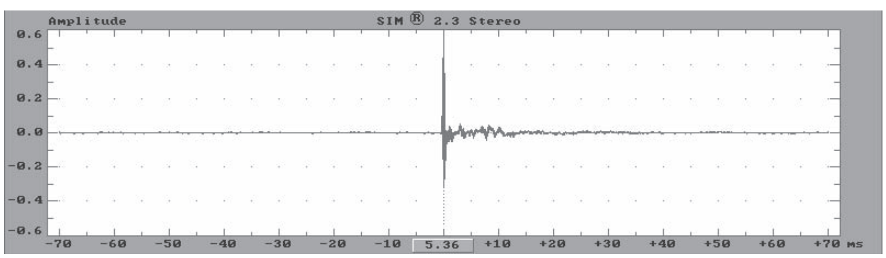


Fig 1.4j

Impulse response of a loudspeaker. This impulse can only be reproduced if both the HF and LF amplifiers have sufficient voltage swing to follow the crest of the waveform. Low wattage amplifiers (often mistakenly applied to the HF driver) will prevent the crest from being reproduced.

! **All biamplified Meyer loudspeaker systems use the same power amplifier type for the HF and LF channels.**

[1] This is a bandwidth limited impulse response which corresponds to -3 dB at 12 kHz.

1.4.7 Amplifier Polarity

The input section of a power amplifier is typically the last balanced drive stage. The output section is typically single-ended with a "hot" pin and a reference ground. The hot pin can swing either positive or negative over time as it tracks the input voltage. Because the output is single-ended it will track the polarity of only one of the pins (2 or 3) of the balanced differential input drive, making the amplifier either "pin 2 hot" or "pin 3 hot" respectively. The AES standard is pin 2 hot.Unfortunately, however, it was adopted some twenty years too late and various manufacturers had established their own standards and are understandably reluctant to change.

Meyer Sound speakers and CEUs will work equally well with either polarity standard, provided of course that all units are driven with the same polarity.

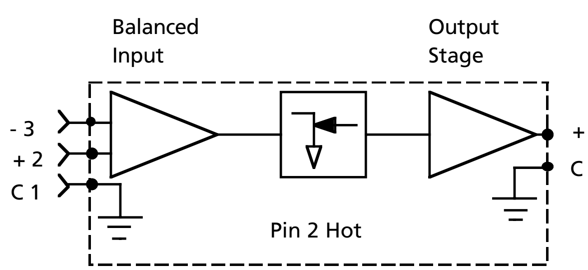


Fig 1.4k

A power amplifier with "pin 2 hot" polarity. The output signal tracks the voltage present at pin 2 and is the opposite to the signal present at pin 3.

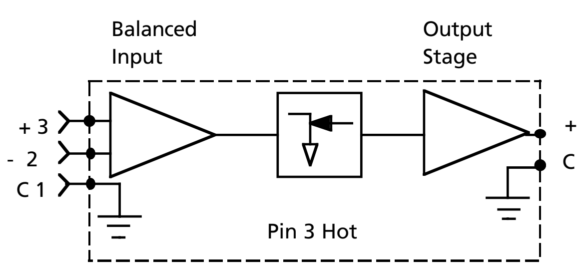


Fig 1.4l

A power amplifier with "pin 3 hot" polarity. The output signal tracks the voltage present at pin 3 and is the opposite to the signal present at pin 2.

1.4.8 Bridged Mode Operation

When placed in "bridged" mode, two amplifier output sections are configured as a push–pull output drive. The speaker is then loaded across the "hot" output terminals of the respective channels, doubling the maximum voltage swing across the load. With today's high power amplifiers, the bridged mode is capable of providing hazardous voltage levels across the output terminals. Therefore, extreme caution is advised. Bridged mode can, in the best case, give a four-fold power boost across the load. This is usually not the case, however, since the load impedance seen by the power amplifier is effectively halved. Therefore, the amplifier's current limits are reduced. In other words, a 4Ω speaker is seen as a 2Ω load in bridged mode, a load that is more likely to be limited by current capability than voltage swing.

Bridged mode not only increases the maximum output power but, in addition doubles, the voltage gain (+ 6dB). Check to make sure that both the maximum power output and voltage gain specifications are within the limits of the speaker before using bridged amplifiers. Determine the polarity of the bridged amplifier (which channel is hot) by checking manufacturer's specifications.

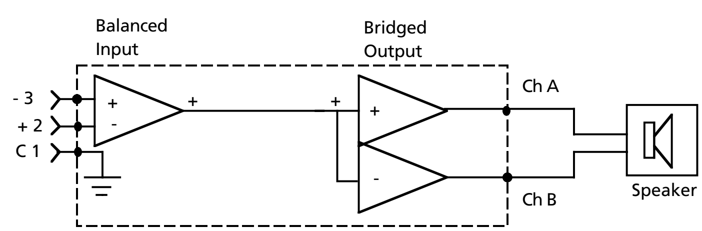


Fig 1.4m

Bridged mode operation.

> **Before operating in bridged mode:**
>
> - Check that the maximum power output capability does not exceed the speaker system's maximum rating.
> - Verify that the voltage gain is below 30 dB.
> - Verify the polarity of the amplifier (which channel is hot.)

1.5.1 Self-Powered Speaker Systems

The previous chapters of this book have detailed the concepts behind Meyer Sound's speaker systems with external CEUs and amplifiers. These design concepts, while quite radical some years back, are now standard among current professional speaker system manufacturers. When compared with custom component designs, this has the obvious advantage of repeatable system results, system compatibility, high quality and reliability. However, this approach has two equally obvious limitations: the high variability of power amplifier performance, and the possibility of inadvert mis-wiring. These were two important factors in Meyer Sound's decision to create the self-powered line of speakers.

One of the best things about Meyer Self Powered Speakers:

The information in Sections 1.1 through 1.4 of this book is critical to achieving optimized reliable performance from a conventionally powered system. With the self-powered series systems it has all been taken care of inside the box.

The Generic Power Amplifier

Imagine for a moment that the automotive industry had evolved such that cars were sold complete with body, transmission and drive train, but *without* the engine. Buyer's would then shop around for what they thought would be the best engine based on maximum horsepower and best price. After all, who wouldn't want the maximum acceleration and speed? However, the installed engine could exceed the capabilities of the drive train and destroy the transmission—but that would not be the engine manufacturer's problem. The acceleration may be so great that the car manufacturer would have to create increasingly sophisticated and costly safety systems to protect itself. In the end, the maximum *safe* speed and acceleration would be based on the entire car as a system—not any single component.

Sound insane? Welcome to the audio industry.

Generic amplifiers (i.e. an amplifier that is used with any speaker) have always been the industry standard. Let's look at the major market factors in amplifiers. To compete, the manufacturer must differentiate their product.

Each manufacturer tries to create an amplifier that has:

- The most power.
- The lowest price.
- The smallest, lightest package.
- The highest reliability.
- The most bells and whistles.

Consider the following:

1) Amplifier power capability has steadily risen every year. There is every indication that this trend will continue. As amplifier power goes up, the life span of speakers attached to them goes down. Speaker systems require ever more sophisticated limiters to protect themselves from the excess power.

2) Amplifier manufacturers design their protection circuitry *to maximize the reliability of the amplifier*—not the speaker.

3) Increased power amplifier output does not necessarily make the speaker system louder. After the maximum capability of the speaker is reached, increasing the amplifier power only reduces reliability and increases the cost.

4) Generic amplifiers have unknown loads. Is it 8Ω? 4Ω? 2Ω? A short circuit? Is it a reactive load? A long cable run? Therefore, they must have all of the circuitry to be ready for any condition. This amounts to huge banks of parallel output transistors, and a bucketful of other components—any of which can fail.

5) The cost per watt is lower at loads of 4Ω or less. An amp rated at 250 watts into 8Ω may give you 400 watts at 4Ω. Naturally, the user would power two 8Ω speakers with a single channel loaded to 4Ω. While economy points to loads of 4Ω or less, the sonic performance and reliability of power amplifiers is superior at 8Ω (better damping factor and slew rate, less current draw).

1.5.1 Self-Powered Speaker Systems

6) Speaker cable offers no sonic advantage. Amplifier efficiency is reduced. The losses are variable with frequency due to the variations in speaker impedance. Heavy-gauge wire can be prohibitively expensive as are multiple runs.

7) What does an amplifier sound like? There have been countless attempts to listen to amplifiers in blind tests, however, the results are always inconclusive since inevitably you are listening to a speaker. An amp that works well with one model may work poorly with another.

The New Standard: Fully Integrated Systems

In 1989 Meyer Sound set a new standard for sound reproduction with the HD-1 High Definition Studio Monitor. Each speaker contains an integrated power amplifier and control electronics, creating a speaker system with unparalleled quality. There were initial requests for Meyer to offer the HD-1 with generic amplifiers until people realized that it would be impossible to achieve the same level of quality and consistency without keeping the amplifier inclusive. This is now the currently accepted standard approach to studio monitoring and, naturally, the competition has followed.

With six years of experience in building powered speakers, Meyer has implemented this technology for the sound reinforcement industry. In order to move sound reinforcement quality to the next level, any unnecessary obstacles between the music and listener must be eliminated. The self-powered series minimizes this path by streamlining the power amp and removing cable loss from the equation. Listening tests continue to confirm the conclusions made earlier with the HD-1: Generic systems cannot match the sound and power of the self-powered series.

The sonic advantages to the self-powered series are:

- Total optimization of the system amplitude and phase response.
- Maximum power efficiency due to known speaker and amplifier.
- Optimized damping and slew rate (all loads are 8Ω)
- Low distortion class AB/H amplifier.

Reliability

Meyer Sound has never lost sight of the fact that it is reliability that is the number one priority for a sound system. It does not matter how good it sounds in the first half of the show if it does not make it to the end.

The self-powered speakers are more reliable than conventional speakers and amplifiers because:

- The amplifiers were designed specifically for the peak and continuous capability of the speakers.
- True Power Limiters (TPL) provide long-term protection based on the actual power dissipation of the speaker.
- The speakers are virtually impossible to miswire.
- The Intelligent AC™ automatically senses the line's voltage so that it can be used anywhere in the world.
- Meyer Sound has been manufacturing powered speakers since 1989.
- Meyer Sound has been building amplifiers since 1986.
- Remote monitoring of the system status. Heatsink temperature and power output can be continually monitored to assure safe operation. Component failure is detected within seconds and the operator is alerted at the house position.

1.5.1 Self-Powered Speaker Systems

Logistics

The self-powered series is also the fastest and easiest high-quality sound reinforcement system. Roll it out of the truck, place it, plug in the AC and plug in the signal. Your PA is ready. For permanent installations there is no amplifier room or speaker cable runs required.

In addition, there are a large number of logistical advantages to the self-powered series, such as :

- Ease of setup and minimal hookup.
- Ease of system design. No CEUs, power amplifiers, interconnect cables, sense lines, speaker cables, custom rack panels, or amplifier racks.
- Ease of multichannel operation. Each speaker is its own channel.
- Ease of supplementing a system. Just add more speakers.
- Full control of relative levels (LD-1) at the house mix position.
- Remote monitoring of the system (RMS™) at the mix position.
- Less truck space and weight.
- Ease of CEU and amplifier service replacement.
- Reduced training time for operators and customer rental.
- Can be rented without fear that the customer will rewire your racks or overdrive the system.
- More room on stage.

Cost

When considering cost, it is important to remember to factor in the auxiliary expenses with conventional systems.

Self-powered speakers are self-contained, however, other speaker systems will need the following additional items:

- Power amplifier(s).
- Control Electronics Unit(s).
- Speaker cable(s).
- Interconnect cables.
- Amplifier rack.
- Custom interconnect and speaker panel(s).
- Wiring and fabrication labor.

Conclusion

The Meyer Sound self-powered series is in the process of transforming the sound reinforcement industry's view of speakers and amplifiers by making it easier and more cost-effective for engineers to achieve consistent high-quality sound.

There is only one known disadvantage to the self-powered concept:

The amplifiers will be more difficult to service in instances where the speakers are hard to reach, for example when hung in the air. Weigh this against all of the other advantages. Be sure to consider that you will be less likely to have to climb up and service the drivers because they are less likely to need service.

1.5.2 Remote Monitor System (RMS™)

The Remote Monitor System (RMS™) is a PC-based computer network that allows the user to monitor all of the significant status parameters of the self-powered speakers. This gives the user a more comprehensive view of the system's operational status than could be achieved with conventional CEU/amplifier/speaker type systems.

The monitored parameters include:

- Input drive level.
- Amplifier output level, clipping.
- Driver continuity.
- Limiting.
- Speaker polarity.
- Heatsink temperature.
- Fan speed.

1.5.3 The LD-1A

The LD-1A is a two-rack space device that controls up to eight speaker subsystems. The LD-1A replaces the key user controls that disappeared with the CEU, such as level, Lo cut, polarity and DS-2 crossover functions. The LD-1A has two main channels that are set up to run the subwoofers, DS-2s and main full-range speakers. In addition there are six auxiliary channels that provide level and Lo Cut capability for additional subsystems.

Is the LD-1A required for *all* self-powered series applications?

No. The LD-1A is best suited for applications where separate level, delay and eq are being used for subsystems driven with the same signal. If the DS-2P and 650-Ps are used together the LD-1A is required.

The LD-1A would not be required if either:

- The main system was a simple mains plus subwoofers without downfill or sidefill subsystems.
- Each subsystem were driven off of its own matrix output.

With SIM®, RMS™ and the LD-1A, Meyer Sound provides everything required to analyze, align and monitor the response of a self-powered sound system without leaving the booth.

RMS™ and the LD-1A are described very completely in the existing Meyer Sound literature. Contact Meyer Sound to obtain these brochures.

1.6.1 Equalizer: The CP-10

The CP-10 Complementary Phase Parametric Equalizer was developed to compensate for the frequency response anomalies encountered with installed speaker systems. The CP-10 was developed concurrently with SIM (Source Independent Measurement) and its features grew from the needs of actual measured sound systems.

To compensate for the anomalies that result when speaker systems are installed, the equalizer must have:

- Adjustable center frequency.
- Adjustable bandwidth.
- Adjustable level.

Why Parametric Equalization?

High-resolution frequency response measurements of installed systems very quickly revealed that the frequency response anomalies have no respect whatsoever for ISO standard center frequencies and bandwidth. In actual practice, peaks and dips can be centered at any frequency whatsoever—as wide or narrow as they choose. It is absolutely essential to have independent control of center frequency, bandwidth and level for each filter section, so that filters may be placed precisely. It is a proven practice that a few carefully chosen parametric filters are capable of providing superior correction to the usual array of twenty-seven fixed 1/3-octave filters.

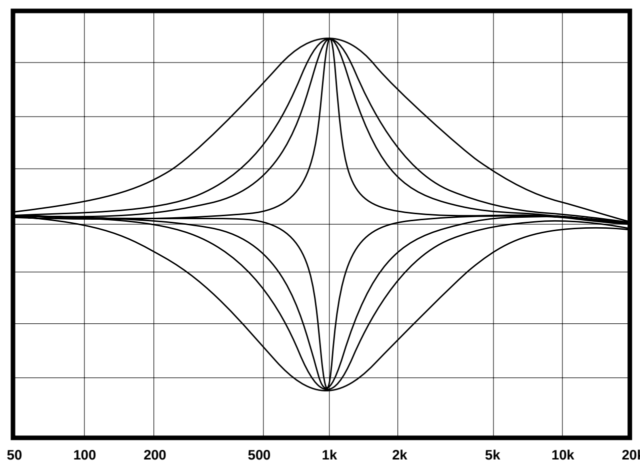


Fig 1.6a The CP-10 parametic equalizer section family of curves.

Complementary Phase

In order to provide a true correction for the effect of the speaker system's interaction with the room, the system's frequency response must be restored in both amplitude and phase. A truly symmetrical, second-order filter topology provides the best complement for correctable minimum-phase phenomena in installed systems. The CP-10's complementary phase circuitry helps to restore the system's original amplitude and phase response by introducing an equal and opposite complementary characteristic.

Shelving Filters

The Lo Cut and Hi Cut shelving functions provide gentle, first-order rolloffs of the system extremes. The Lo Cut circuit was strategically designed to compensate for the type of LF buildup encountered when speakers are combined in arrays. The Hi Cut filter effectively prevents an overly bright presence when using speakers at close range.

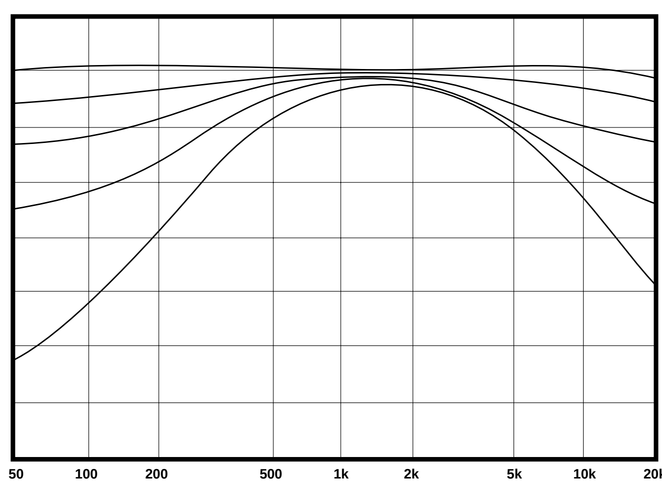


Fig 1.6b The CP-10 shelving section family of curves.

1.6.2 Advantages of Parametric Equalizers

The most popular equalizer in professional audio is the 1/3-octave graphic equalizer, a parallel bank of equal percentage bandwidth filters spaced at 1/3-octave intervals. It is the worldwide accepted standard for equalizers, and is typically used for the correction of installed speaker systems. Since it is called an "equalizer" one might assume that it is capable of creating an equal but opposite response (a complement) to that of the speaker system in the room. The term "graphic" indicates that the front panel fader positions give a graphical indication of its actual response.

Unfortunately, both of these assumptions are wrong.

Graphic Equalization versus Complementary Equalization

There is no mechanism in the interaction of speakers that causes logarithmically spaced peaks and dips. There is nothing that governs these interactions and compels them to adhere to ISO standard center frequencies such as 500, 630 and 800 Hz. There is no mechanism that causes successive peaks and dips of equal percentage bandwidth, the type shown in Fig 1.6c. What actually results in interactions is shown in Fig 1.6d. In short, the only interaction that a 1/3-octave graphic equalizer can complement is that of another graphic equalizer.

To create a complement of the types of responses actually occurring in room and speaker interaction requires a parametric equalizer that can independently control center frequency, bandwidth and level.

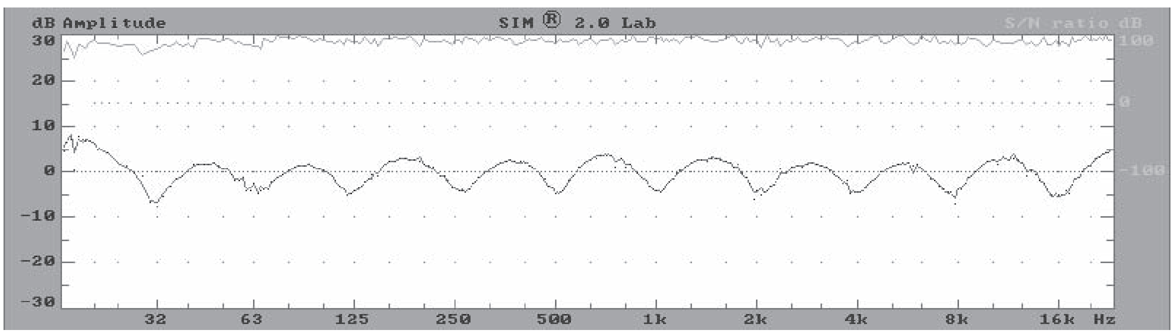


Fig 1.6c The way it would be if comb filtering had log frequency spacing on ISO center frequencies.

A graphic equalizer could create a complementary response to this. Unfortunately, there is no known mechanism in the interaction of speakers and rooms that will cause this type of response.

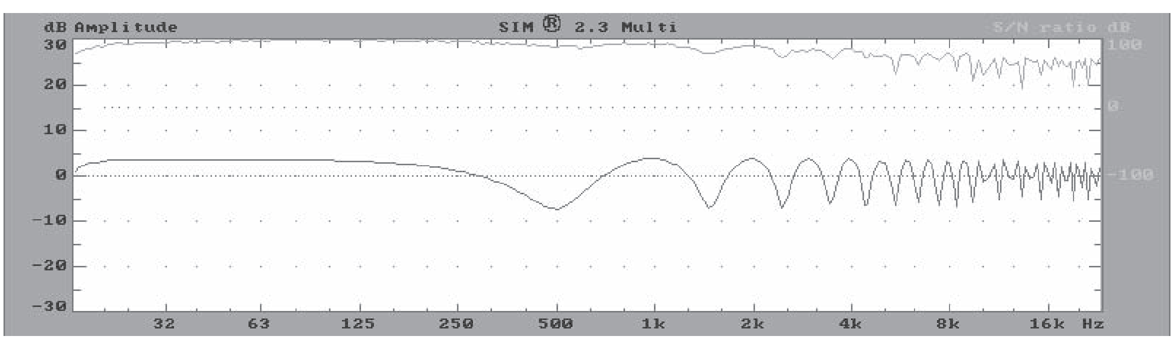


Fig 1.6d Linearly spaced comb filtering as created by the interaction of speakers with rooms and other speakers. See Section 2 for complete details. A parametric equalizer with adjustable center frequency, bandwidth and level is capable of creating a complement to this type of response.

1.6.2 Advantages of Parametric Equalizers (cont.)

Graphic Response vs. Actual Response

The front panel indicators do not take into account the interaction of neighboring bands. The summation response that occurs when the boosts and cuts are added together will often vary more than 10 dB from the response indicated on the front panel.

Since a graphic equalizer cannot correct room and speaker interaction and gives false indication of its response, what should it be used for?

A tone control for mix engineers to create artistic shaping of the system's overall response.

The electrical response of the Graphic EQ with 15 filters inserted and of the CP-10 with 1 filter inserted.

The graphic equalizer front panel settings are superimposed upon its measure response. A total of 15 filters were used ranging from 1-5 dB of cut. The panel setting shows a 10 dB error from the actual curve generated by the equalizer.

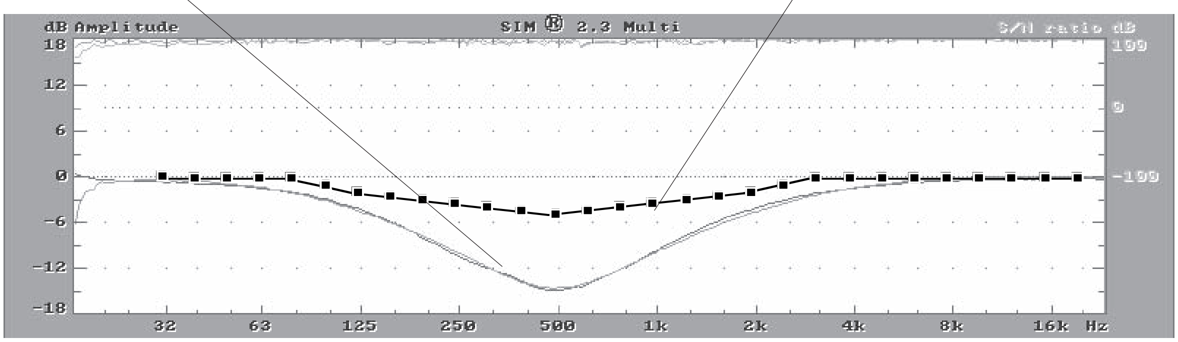


Fig 1.6e Contrasting a graphic equalizer and the CP-10 Parametric

Graphic equalizers provide a visual indication of their response. (Hence the term "graphic"). Unfortunately though, the visual does not necessarily correspond to the actual response. This is especially true when multiple filters are engaged, since they may be highly interactive. Fig 1.6e shows the difference between the graphic front panel settings and its actual response.

The most common reservation about parametric equalizers is the perception that the front panel setting does not help the engineer visualize the response.

Another reservation is the perception that the 27-31 bands of a graphic equalizer are more flexible than the 5 or 6 bands of a parametric. Note that in the above figure, it took 15 graphic EQ filters to create the response that the CP-10 made with one filter.

1.6.3 Complementary Equalization

To create the complement of a given response the equalizer must create an inverse amplitude and phase response. If this occurs the resulting response will have have a flat amplitude and no phase shift.

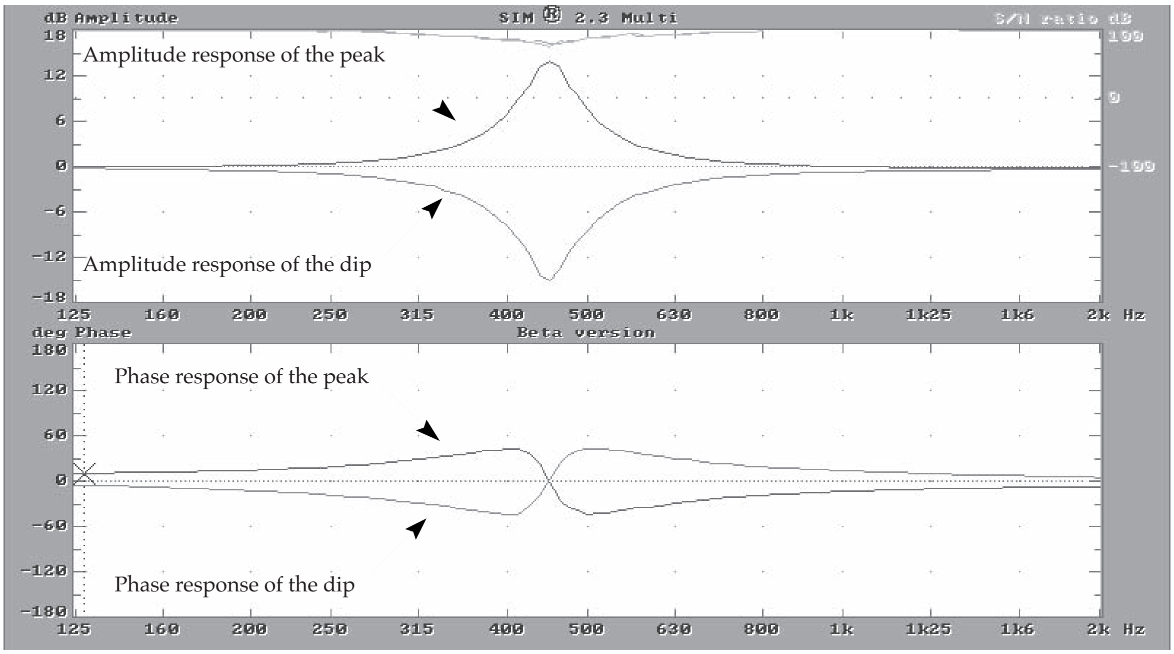


Fig 1.6f Peak and dip are complementary in amplitude and phase. The center frequency, bandwidth and level of the peak and dip are matched.

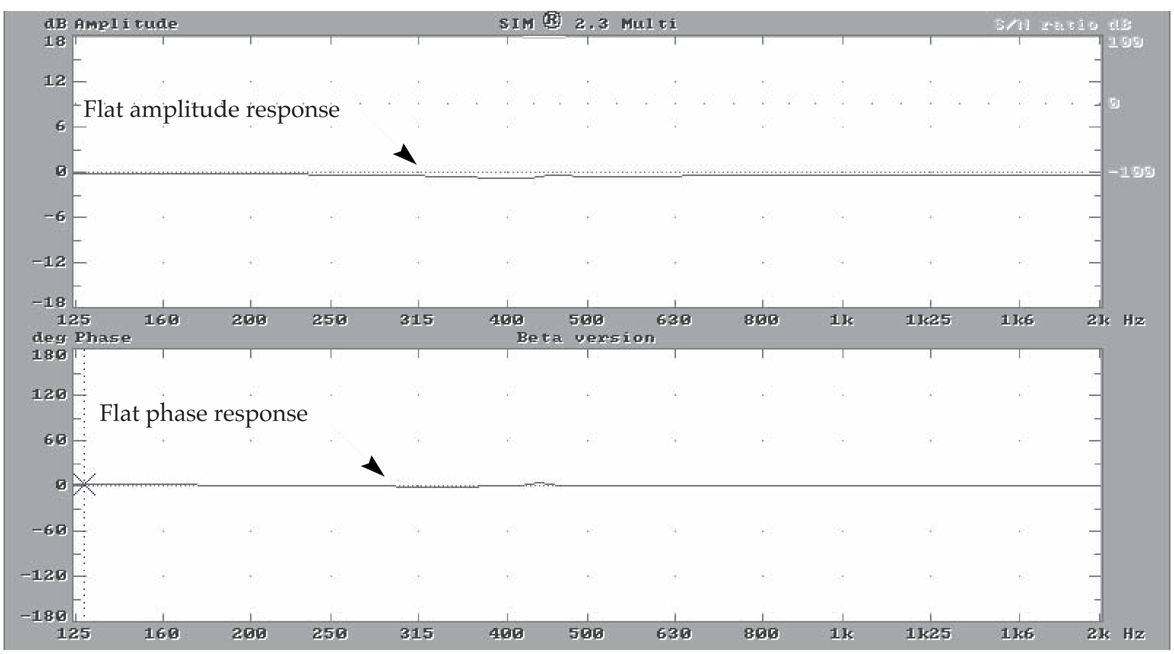


Fig 1.6g The result of complementary phase equalization. Amplitude and phase responses are flat.

1.6.4 Error in Center Frequency

If the center frequency is not correct the resulting response will contain leftover peaks and dips as well as phase response anomalies.

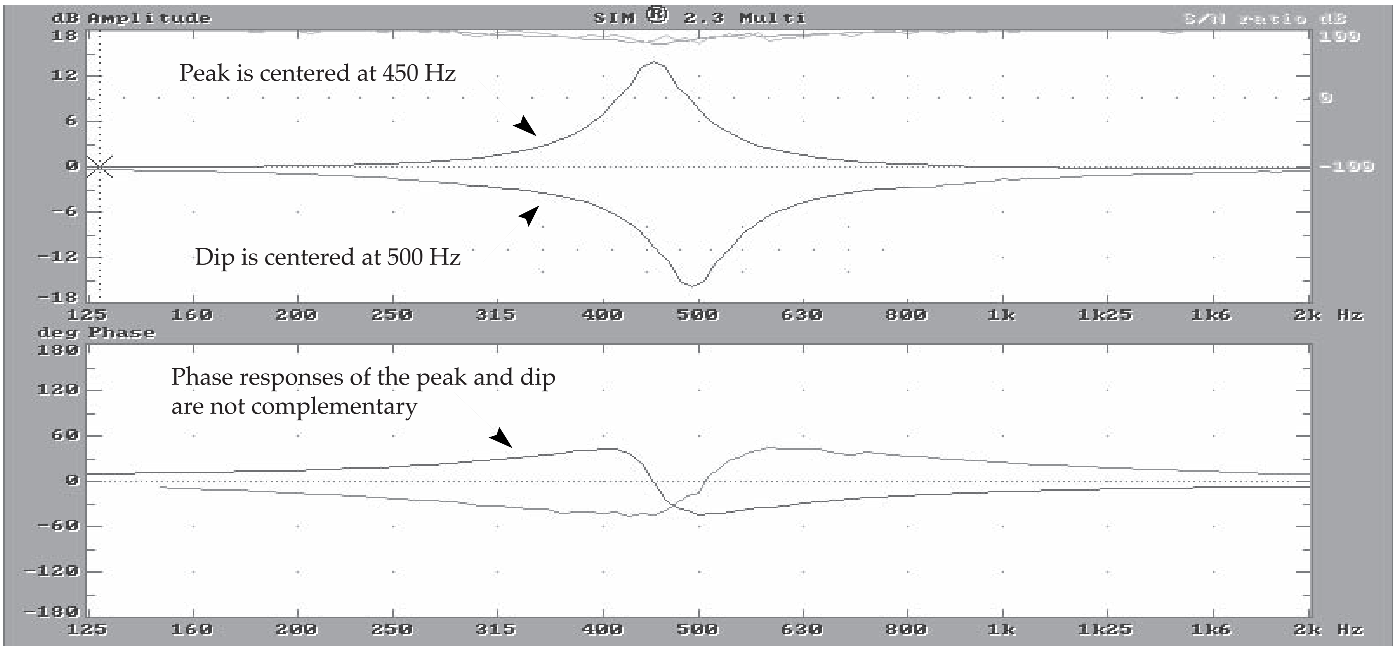


Fig 1.6h Peak and dip do not have the same center frequency.

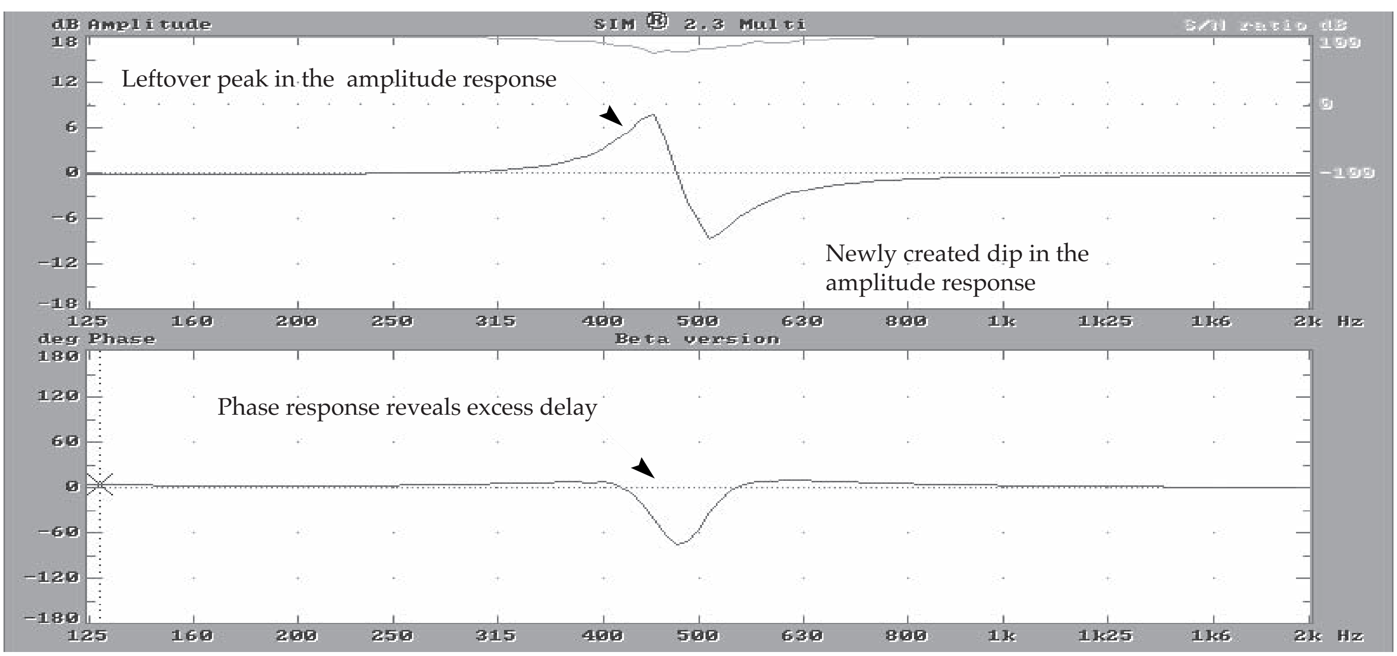


Fig 1.6i Result of error in center frequency.

1.6.5 Error in Bandwidth

If the bandwidth is not correct, the resulting response will contain leftover peaks and dips as well as phase response anomalies.

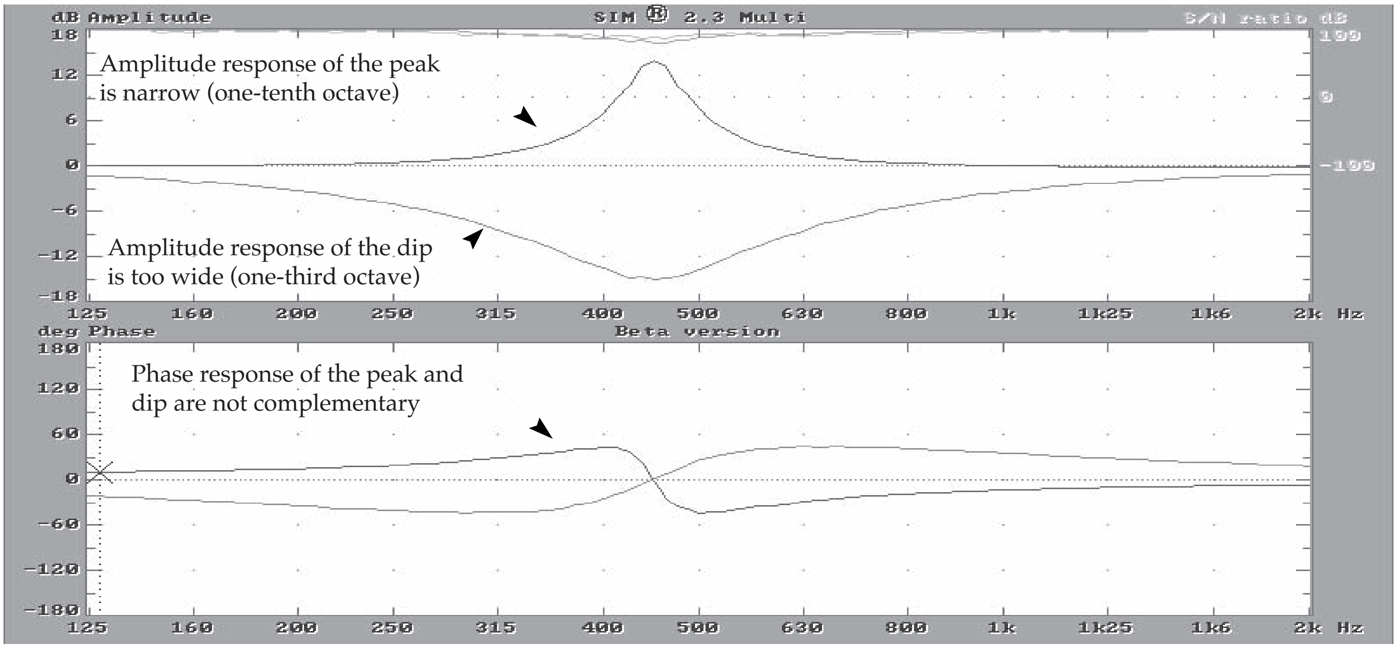


Fig 1.6j Peak and dip do not have the same bandwidth.

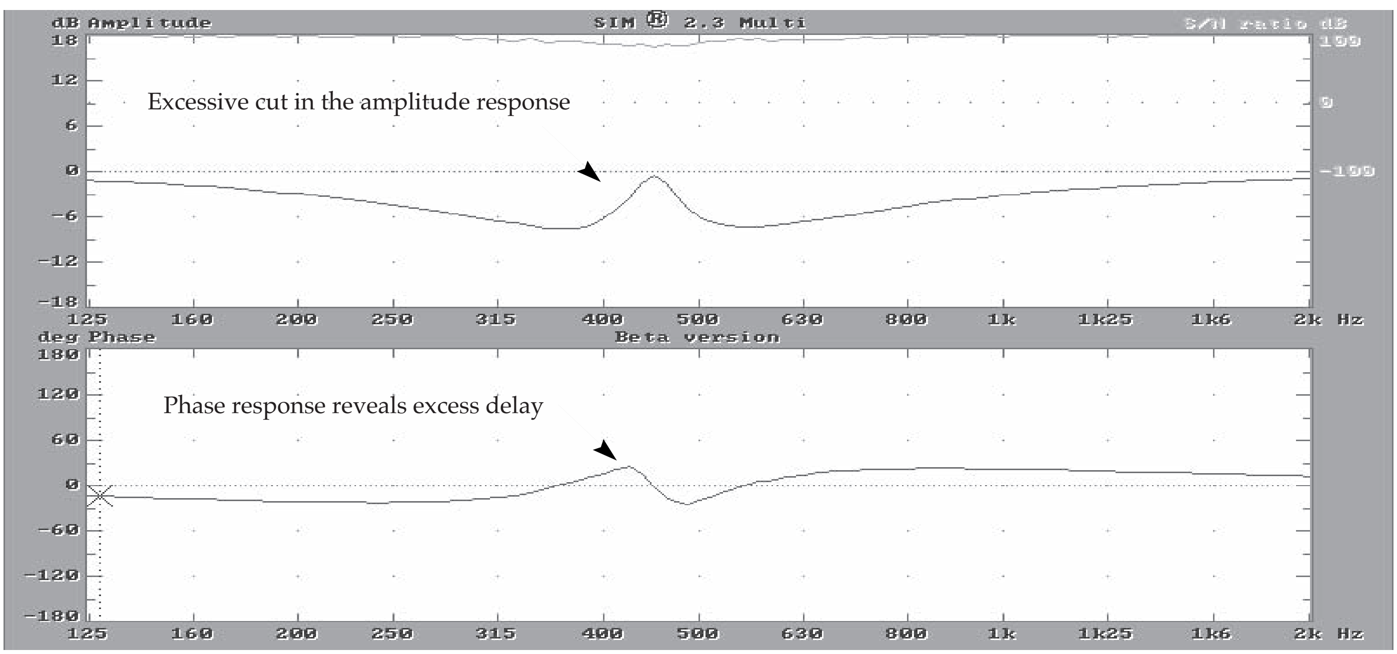


Fig 1.6k Result of error in bandwidth.

1.7.1 Line Level Connections

All CEUs and equalizers manufactured by Meyer Sound utilize balanced inputs and push-pull balanced outputs. XLR 3-pin connectors are used in all of these and in some cases is accompanied with a Tip-Ring-Sleeve one-fourth-inch phone jack.

In all cases the signal is configured as follows:

XLR	Phone Jack	Function
Pin 1	Sleeve	Common
Pin 2	Ring	Signal
Pin 3	Tip	Signal

Notice in the above chart that the two signal pins are not designated as either + or – . This is because all of these devices are balanced from input to output and, therefore, are neither pin 2 nor pin 3 hot.

The polarity of the input signal will be maintained in the same way that a microphone cable is neither pin 2 or 3 hot. Figs 1.7a and 1.7b show the signal flow through a standard mic line and an active balanced device, respectively. Non-inverting polarity is maintained in both cases.

Are Meyer Sound CEUs pin 2 or pin 3 hot?

Neither. They are balanced in and balanced out. Polarity of the original input signal is preserved.

I checked my CEU with a phase popper and it says that LF output is normal but the HF output is reversed. Is there a problem?

No. The phase correction circuitry of the CEUs acts to create linear phase of the loudspeaker system. Frequency-selective delay networks are used to optimize the crossover and correct for anomalies in the speaker. Therefore, measuring the CEU alone can be confusing in terms of polarity.

Which type of amplifier should be used? pin 2 or pin 3 hot?

It doesn't matter as long as all of them are consistent.

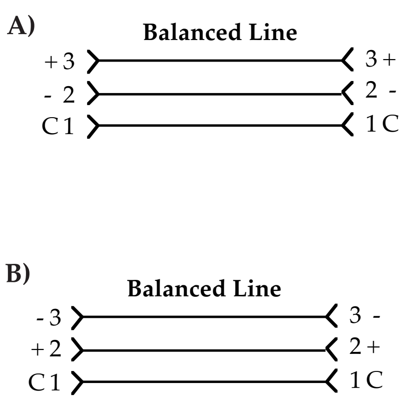


Fig. 1.7a Balanced mic line.

A) Positive input at pin 3 gives positive output at pin3.

B) Positive input at pin 2 gives positive output at pin 2.

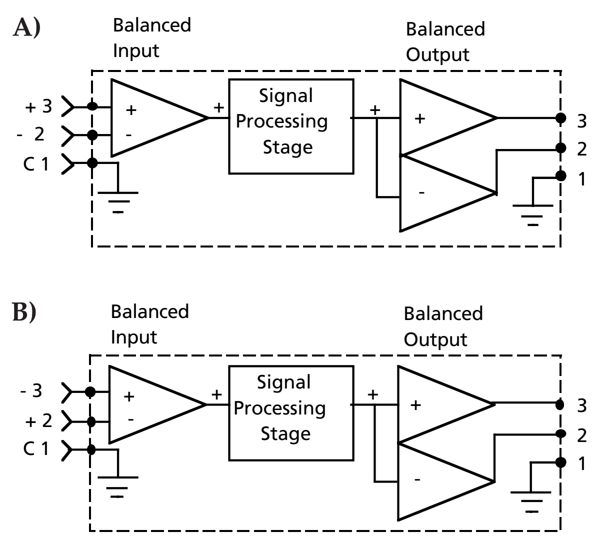


Fig. 1.7b Active balanced inputs and outputs.

A) Positive input at pin 3 gives positive output at pin3.

B) Positive input at pin 2 gives positive output at pin 2.

1.7.1 Line Level Connections

Input Connector Wiring*

This section explains the capabilities and advantages of the ISO-Input circuit, and describes the connector wiring practices that will enable you to best utilize those attributes. The connection configurations given here should be particularly attractive to those who are intending to use the ISO-Input in permanent installations such as theatre sound and studio monitoring.

The Meyer Sound ISO-Input

Meyer's patented ISO-Input circuit constitutes a three-port, floating, balanced signal input system.

The primary advantages of this circuit are:

- True transformer isolations without the drawbacks normally associated with transformer-coupled designs.
- Maximum flexibility of input connector pin assignment with no change in gain.

The ISO-Input circuit makes use of specially designed custom transformers that have a high-inductance nickel core and Faraday shield. The circuit achieves a full 500 volts of common-mode voltages without danger to the input components.

The transformers used in the input are designed specifically for voltage sensing rather than power transfer. In contrast to conventional audio transformers, they operate in the microwatt power range. For this reason, they do not exhibit the core eddy losses, hysteresis problems, ringing and phase shift normally associated with transformer designs. As a direct result, distortion in the ISO-Input stage is held to under .01% (even at 20 volts), and phase shift at 20 kHz (without TIM filter) is less than 10°.

ISO-Input circuits are also virtually insensitive to variations in source impedance (a major concern with conventional audio transformers) and, since they employ a humbucking design, do not require costly, heavy external shielding in order to maintain immunity from hum. The ISO-Input thus offers all the advantages of active balanced circuits, but with the far superior electrical isolation characteristics that only transformers can provide.

Perhaps most important from the standpoint of professional audio applications, however, is the fact that the ISO-Input will accept a wide variety of input pin connections, with *no change in gain*. Fig 1.7c is a truth table that shows all the input connection combinations that will work with the ISO-Input. In every case, the gain of the input stage will be the same: given equal input signal drive levels, every connection listed in the table will produce the same output level from the CEU. Only the output polarity will vary. (Note, however, that push-pull output drivers provide 6 dB greater drive level than transformer-coupled or unbalanced outputs, all other factors being equal.)

Notice that there is no input connection that will short the output of the signal source—other than connecting the hot lead directly to the input connector shell. In fact, driving any two input pins will work, and the gain of the amplifier will remain the same: only the signal polarity will be affected. This unique attribute allows the ISO-Input to accommodate virtually any 3-pin connection "standard," and permits the user to employ a variety of types of phase-reversing adapters without the fear of shorting out the signal source or suffering an unwanted change in gain.

Source Output Configuration	Wiring of ISO Input				
	Pin 1	Pin 2	Pin 3	Polarity	Comments
Balanced	n/c	-	+	+	Best CMRR
	n/c	+	-	-	Lowest Hum
	C	-	+	+	
	C	+	-	-	
	-	-	+	+	
	-	+	-	-	
	-	n/c	+	+	
	-	+	n/c	-	
	+	n/c	-	-	
	+	-	n/c	+	
Unbalanced	n/c	C	+	+	Best performance
	n/c	+	C	-	unbalanced
	C	C	+	+	
	C	+	C	-	
	C	n/c	+	+	
	+	n/c	C	-	
	C	+	n/c	-	
	+	C	n/c	+	

Fig 1.7c ISO-Input wiring truth table.

*The remainder of Section 1.7.1 was written by Ralph Jones.

1.7.1 Line Level Connections

Hum-Free System Design

One of the most frustrating and difficult problems in audio system design and operation is line-frequency hum injection. This phenomenon is most often caused by ground loops—duplicate signal common paths carrying circulating currents which modulate the audio signal.

Ground loops can be eliminated by conventional transformer isolation schemes, of course, and well-engineered transformers with excellent performance characteristics have been available for some time. But well-engineered transformers are very costly. Frequently, therefore, audio professionals, deprived of the benefits of transformers by budgetary limits, are forced instead to design systems using only active balanced inputs and outputs—or worse yet, unbalanced inputs and outputs.

In such systems, signal common must be brought through with every interconnection in order to force all the system power supplies to the same common potential. Grounding must be handled with great care in order to avoid the formation of ground loops, while still maintaining protection against shocks, RFI and static potentials. Every system design then, becomes a compromise—and a very complicated one at that.

By contrast, the ISO-Input is completely isolated and floats both signal lines with respect to the chassis (which is connected to earth). This attribute greatly simplifies the design of the hum-free audio systems: *as long as no pin of the ISO-Input connector is linked to the connector shell, it will be literally impossible for ground loops to form.*

Several CEUs can be driven in parallel from a single audio source using "standard" connection cables, and no ground lifting adapters will be necessary as long as the signal common is kept separate from earth at the input connector. Even in relatively complex systems, the isolation between the components will be as good as that provided by opto-isolators, and each ISO-Input can operate as a self-contained, floating unit.

Connections to Standard Audio Equipment Outputs

This section details the input connector wiring practices that must be followed in order to implement the principles discussed in the previous section. These wiring practices differ from those that are normally used today, having more in common with traditional transformer-isolated designs.

Particularly notable is the use of "telescoping shields." When shields are connected at only one end of the cable, and are not used for carrying common between the two devices, the potential for ground loops is greatly diminished. The connection is most ideal when a telescoping shield is connected only to mains earth, and not to signal common in either device. That way, static potentials and RFI are kept entirely separate from the signal path.

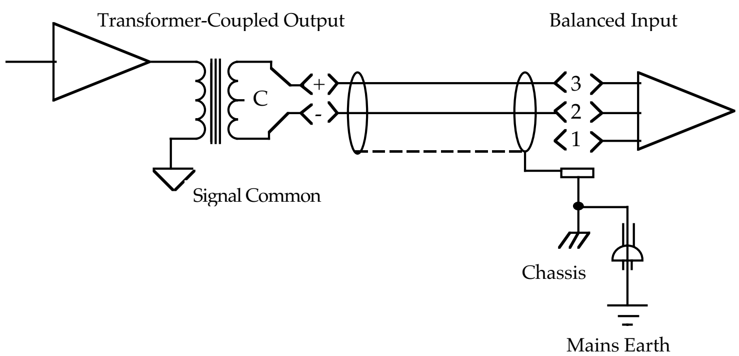


Fig 1.7d

Transformer-coupled output stage connection.

1.7.1 Line Level Connections (cont.)

Note: In all cases, the following connection instructions assume that the CEU chassis is connected to earth ground. If a three-wire grounded mains source is not available (which is the case, for example, in Japan and some European countries), then the chassis *must* be earthed by an external connection between the rear-panel chassis ground terminal and a reliable earth ground point.

Since the ISO-Input works very well with "standard" audio cables (again, so long as no pin is linked to the connector shell)—and since cables wired as described here will not be interchangeable with standard cables—it may be more practical to use standard cables for portable systems. In permanent installations, however, the benefits of wiring the system as described here are substantial.

Fig 1.7d illustrates the cable wiring scheme to use when the CEU is to be driven from a source having a transformer-coupled output. (The transformer center tap may or may not be present depending upon the design of the source equipment—in any event, it is not used.) The ISO-Input is wired in a floating differential configuration (sometimes called an "instrumentation input"). This figure shows the signal input pins may be used with no change in gain. The connection shown in the diagram yields the best performance, however.

Notice that the cable shield is connected only to the shell of the input connector, so RFI and static potentials in the shield will drain directly to earth. There is no ground loop path, regardless of whether or not the signal common of the source is connected to earth.

The following Meyer CEUs utilize the Iso-Input:

M-3A	**S-1**	**VX-1**
M-10A	**M-5**	**D-2**

Fig 1.7e shows how the same connection scheme may be used for source equipment having a push-pull output (as do all Meyer Sound electronic products). The same observations apply to this figure as to Fig 1.7d. The push-pull output stage provides 6 dB greater drive than the transformer output (all other factors being equal). This may be compensated for, if necessary, by dropping the gain of the CEU. Again, regardless of whether or not signal common of the source is connected to earth, there is no ground loop path.

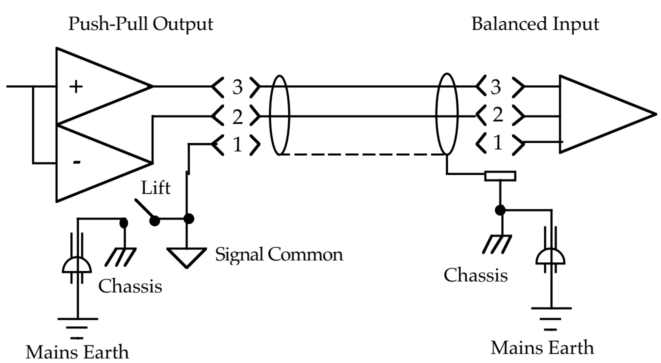


Fig 1.7e

Push-pull output stage connection.

1.7.1 Line Level Connections

Unbalanced Lines

When connecting unbalanced inputs using single-conductor shielded cable, wire the connectors as shown in Fig 1.7f. Notice that the shield is connected to pin 1, and *there is no connection between pin 1 and the shell.* In this case, the connection between the signal common of the source and earth provides the path by which RFI and static potentials in the shield are drained to earth. This connection scheme may be used with any unbalanced equipment that has a grounding AC plug (such as pro mixers or tape recorders).

A different, and more optimal method for handling unbalanced equipment is shown in Figure 1.7g. This treatment is similar to that shown for balanced drivers, above, and yields equivalent performance. Unbalanced equipment that is battery-operated (or for other reasons floats from earth) should be connected in this manner, so that there is a path from the shield to earth. (This scheme is particularly effective with battery-operated compact disc players and other high quality, floating, unbalanced equipment.)

Even if the signal ground of the source is connected to earth, however, there still is no ground loop path. This connection scheme can therefore be used for all unbalanced equipment. It will yield balanced performance, since the shield is not connected at the source output.

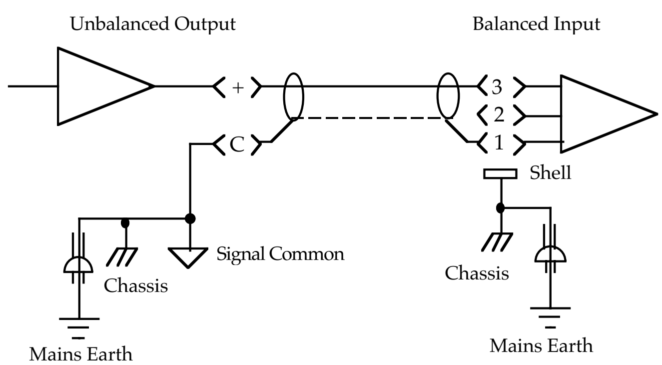


Fig 1.7f

Unbalanced output stage connection, single-conductor shielded cable.

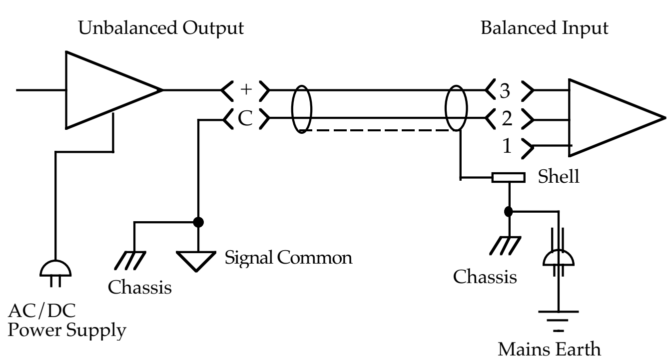
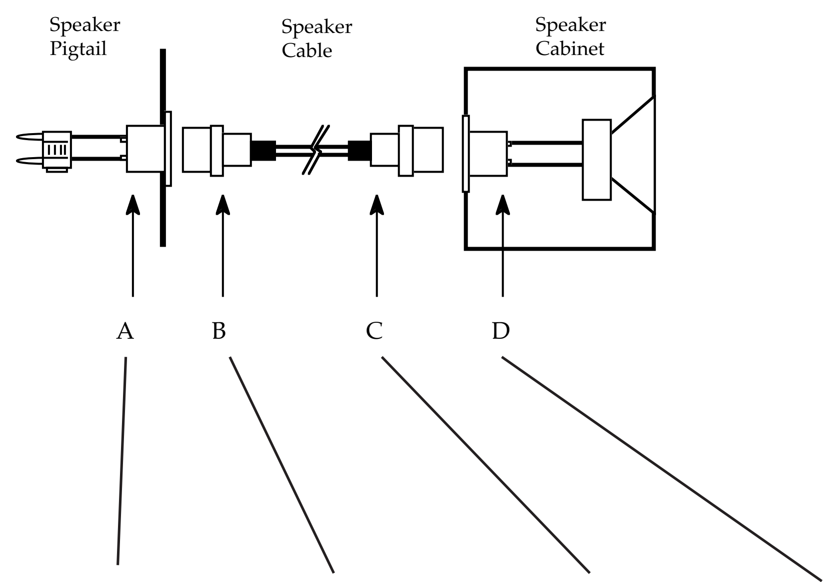


Fig 1.7g

Achieving balanced performance with unbalanced equipment.

1.7.2 Speaker Cables

The typical speaker connectors are Canon EP-type. The four different types (male, female, inline and chassis) are detailed below for reference, along with the Pyle National used in the MSL-3A.



Speaker Connectors	Rack-panel Female	In-line Male	In-line Female	Speaker Female
Cannon EP-4	EP4-13	EP4-12	EP4-11	EP4-14
Cannon EP-5 (Europe)	EP5-13	EP5-12	EP5-11	EP5-14
Pyle Natl. (MSL-5 & MSL-10A)				
Pyle Natl. (MSL-3A)	ZPLP-12-311SN	ZRLK-1212-311PN	ZPLK-1212-311SN	ZRLP-12-311PN

Fig 1.7h Speaker cable reference chart.

1.7.2 Speaker Cabling

Speaker cables should be made of high-quality stranded copper. For portable applications, flexible cable is essential.

One of the most pressing considerations in regard to speaker cabling is the selection of the proper gauge. No one wants to see the system's power used to warm up the cables. On the other hand, thick cable is costly and heavy, and adds to installation and travel related costs.

There are three primary factors that determine the amount of power lost in cable runs: length, load and gauge (wire thickness).

The proportion of power lost in the cable will increase:

- As the cable length increases.
- As the load impedance decreases.
- As the wire diameter decreases.

Most references to cable loss refer to the amount of power loss in watts or percentage of the signal. Losing 250 watts may sound like a terrible waste but is actually only 1 dB in the case of a 1200 watt power amplifier. The reference charts on this page show the losses directly in dB for different length cable runs for 4, 8 and 16Ω loads respectively.

For example, refer to Fig. 1.7.i. You will find that a 14 AWG cable run will lose 3 dB when loaded to 4Ω at a length of 300 feet (91 meters).

These charts can be used in conjunction with tables 1.7l and 1.7m, which include the load impedance for each speaker model.

14 AWG (1.6 mm)

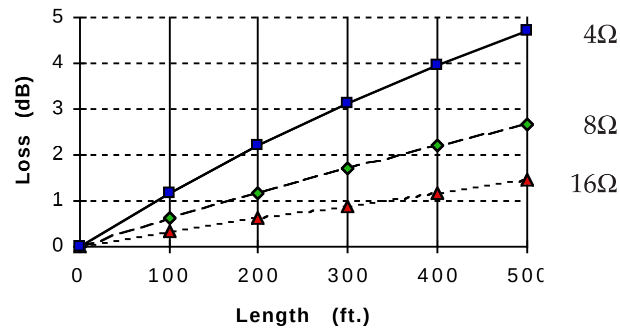


Fig. 1.7.i Cable loss over distance for 14 AWG (1.6 mm).

12 AWG (2.0 mm)

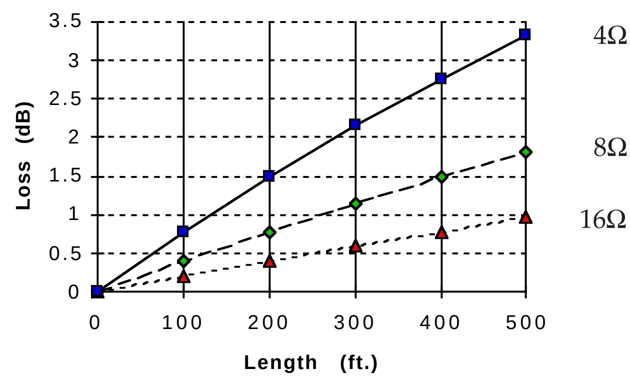


Fig. 1.7.j Cable loss over distance for 12 AWG (2.0 mm).

10 AWG (2.5 mm)

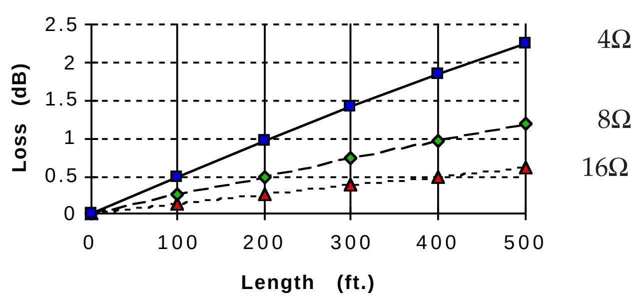
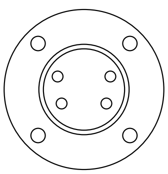


Fig. 1.7.k Cable loss over distance for 14 AWG (2.5 mm).

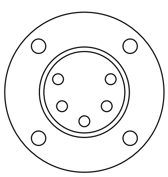
1.7.3 Speaker System Standard Cable Reference

Table 1.7l is designed to provide the user with the information required to properly wire and verify amplifier racks and speaker cables. Included is information on the pigtail (see next page), connector type standard wire color codes, pin designations and load impedance.

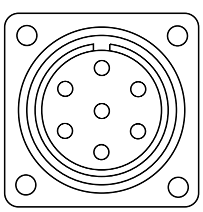
Please note the following: the nominal load column denotes the load on a particular wire pair and can be used for the cable loss charts on the previous page. In cases where the drivers are run in parallel (such as the 650-R2) the total load impedance of the cabinet is 4Ω but the individual cable runs are 8Ω each.



EP-4



EP-5



PYLE NATIONAL

Speaker Wiring Chart						
Product	Connector	Pigtail	Pin #	Color	Function	Nominal Load (Ohms)
MSL-10A	Pyle Natl.		1	Black	Low 1&2+	
			2	White	Low 1&2-	4 Ω
			3	Red	Low 3&4+	4 Ω
			4	Green	Low 3&4-	
			5	Orange	High+	4 Ω
			6	Blue	High-	
			7		N/C	
MSL-5	Pyle Natl.		1	Black	Low-	4 Ω
			2	Red	Low+	
			3		N/C	
			4	Green	High-	4 Ω
			5	White	High+	
			6		N/C	
			7		N/C	
MSL-5	EP-4, EP-5	Full-Range	1	Red	Low+	4 Ω
			2	Black	Low-	
			3	Green	High-	4 Ω
			4	White	High+	
			5		N/C	
MSL-3A	Pyle Natl.		1	Black	Low-	4 Ω
			2	Red	Low+	
			3		N/C	
			4	Green	High-	12 Ω
			5	White	High+	
			6		N/C	
			7		N/C	
MSL-3A	EP-4, EP-5	Full-Range	1	Red	Low+	4 Ω
			2	Black	Low-	
			3	Green	High-	12 Ω
			4	White	High+	
			5		N/C	
MSL-2A USM-1	EP-4, EP-5	Full-Range	1	Red	Low+	8 Ω
			2	Black	Low-	
			3	Green	High-	12 Ω
			4	White	High+	
			5		N/C	
UP-1C UM-1C UPA-2	EP-4, EP-5	Full-Range	1	Red	Low+	8 Ω
			2	Black	Low-	
			3	Green	High-	12 Ω
			4	White	High+	
			5		N/C	
650-R2 USW-1 DS-2	EP-4, EP-5	Subwoofer	1	Black	Low 2-	8 Ω
			2	Black	Low 1-	8 Ω 4 Ω
			3	Red	Low 1+	
			4	Red	Low 2+	
			5		N/C	
MSW-2	EP-4, EP-5	Subwoofer	1	Black	Low-	
			2		N/C	8 Ω
			3		N/C	
			4	Red	Low+	
			5		N/C	
HF-3	EP-4, EP-5	Full-Range	1		N/C	
			2		N/C	
			3	Green	High-	12 Ω
			4	White	High+	
			5		N/C	

Table 1.7l

Speaker Wiring Reference (A).

1.7.3 Speaker System Standard Cable Reference

Table 1.7m is designed to provide the user with the information required to properly wire and verify amplifier racks and speaker cables. Included is information on the pigtail (see below), connector type standard wire color codes, pin designations and load impedance.

Please note the following: Speaker models UPM-1J, UPM-2J, MPS-355J and MPS-305J are a special pinout for Japan.

Speaker Wiring Chart

Product	Connector	Pigtail	Pin #	Color	Function	Nominal Load (Ohms)
MST-1	EP-4, EP-5	Full-Range	1		N/C	
			2		N/C	
			3	Green	High-	4 Ω
			4	White	High+	
			5		N/C	
UPM-1	EP-4, EP-5	Full-Range	1		N/C	
			2		N/C	
			3	Black	-	16 Ω
			4	Red	+	
			5		N/C	
UPM-1 MPS-355	XLR		1	Black	-	
			2	Red	+	16 Ω
			3	Red	+	
UPM-1J MPS-355J	XLR		1		N/C	
			2	Red	+	16 Ω
			3	Black	-	
MPS-305	XLR		1	Black	-	
			2	Red	+	8 Ω
			3	Red	+	
MPS-305J	XLR		1		N/C	
			2	Red	+	8 Ω
			3	Black	-	
MPS-355	Speak-on		1	Black	-	
			2		N/C	16 Ω
			3	Red	+	
			4		N/C	
MPS-305	Speak-on		1	Black	-	
			2		N/C	8 Ω
			3	Red	+	
			4		N/C	

Table 1.7m Speaker Wiring Reference (B).

1.7.4 Speaker Pigtails

The output connectors for most power amplifiers are five-way binding posts, commonly known as "banana plugs". The connection to the speaker cable is accomplished by a "pigtail" adaptor, which has banana plugs on one end and the EP type connector on the other. There are two different types of pigtail configurations: Full-range (for biamplified systems) and subwoofer. The pigtails are designed so that, in the event of an inadvertent mispatch, no signal will flow through the speakers. This prevents potential damage to HF drivers if hooked up to a subwoofer feed.

Table 1.7n contains complete pinout information on the pigtails.

Product	Banana Plug	Wire Color	Connector	EP Pin #	Function	Nominal Load (Ohms)
Full Range Pigtail	Black- (Gnd)	Red	EP-4, EP-5	1	Low+	8 Ω
	Black +	Black		2	Low-	
	Red- (Gnd)	Green		3	High-	12 Ω
	Red +	White		4	High+	
				5	N/C	
Subwoofer Pigtail	Black - (Gnd)	Black	EP-4, EP-5	1	Low 2-	8 Ω
	Black - (Gnd)	Black		2	Low 1-	8 Ω 4 Ω
	Black +	Red		3	Low 1+	
	Black +	Red		4	Low 2+	
				5	N/C	

Table 1.7n Pigtail Wiring Reference.

1.8.1 Speaker Power Ratings

Maximum SPL Ratings

Meyer Sound speaker maximum SPL ratings are derived from measurements taken on-axis using the Bruel & Kjaer 2209 Sound Level Meter with a 4133 half-inch capsule. The 2209 is calibrated using the Bruel & Kjaer calibrator before each measurement cycle.

The Meyer Sound measurements series, TechNotes, use a full bandwidth pink noise input and the SPL meter set to the linear response setting.

The data sheets for the various products differ in that the input signal is "A," weighted pink noise.

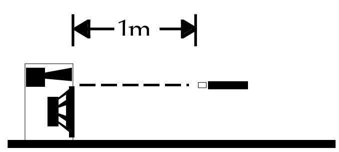


Fig 1.8a

Maximum SPL measurement technique.

All maximum SPL measurements are made in half-space conditions. The distance at which the measurements are conducted varies depending upon the focal distance of the system. The data is then scaled as appropriate to obtain a one meter rating. The exception is the MSL-5 and MSL-10A which are rated for 100 feet (thirty meters). For two-way systems, the mic is usually aligned to the point between the HF and LF drivers.

Sensitivity

We are often asked by users about the sensitivity rating of our speaker systems. A sensitivity rating denotes the SPL at a given distance generated by the speaker for a given input power. This is typically 1 watt input measured at a distance of one meter. Much has been made in the past about the fact that a given model of speaker may be a few dB more sensitive than another. However, the sensitivity rating gives no indication that one speaker has a higher maximum SPL rating than the other. This specification has some use when working at a component level with standard loudspeakers, but when applied to an integrated system, it falls short. We are, of course, concerned that the speaker system be efficient, so that we can make best use of our power amplifier resources.

There are two factors that Meyer is concerned with regarding sensitivity:

- What is the maximum SPL capability of the speaker?
- How much power is required to achieve it?

For example, the UPA-1C is rated at 125 dB SPL at one meter when driven with a 250 watt/8Ω amplifier. This gives you all the information you need in order to ascertain the power amplifier needs and costs. If, for some reason, you need to specify sensitivity, it can be approximated by prorating the above specifications down to 1 watt.

Maximum dB SPL – (10 log Power amp rating in watts)

For our UPA-1C example:

125 dB SPL – (10 log 250 watts)

125 dB SPL – (24 dB) = 101 dB 1 watt/1 meter

Bear in mind that this rating is for two power amplifier channels.

1.8.3 Coverage Angle Specifications

All coverage angle specifications are referred to the on-axis point for single speakers, and between the splayed cabinets for arrays. The TechNote™ series gives specific ranges of frequencies for which the published number represents the average coverage angle. The data sheets vary on this point and generally describe the high frequency range only (for full-range systems), and therefore tend to show a narrower angle than TechNotes would.

TechNotes specifications:

Speaker	Range
UPA-1C	125 Hz to 8 kHz
MSL-2A	125 Hz to 8 kHz
MSL-3A	125 Hz to 8 kHz
DS-2	60 Hz to 160 Hz

TechNote Measurement of Single Speakers

TechNotes measurements were made by utilizing the multiple microphone capability of SIM System II. The mics were placed in an arc around the speaker and response was compared to that of the on-axis position. The position where the average response reached (–6dB compared to the on-axis response) was designated as the edge of the coverage angle. For single speakers, the focal point is considered to be approximately the throat of the HF horn. For layout purposes this should be considered as the origin. *The accuracy of TechNotes coverage angle specifications is estimated to be ±10 degrees*

Horizontal

The horizontal pattern of all Meyer speakers is symmetrical between left and right sides. Therefore only a single side was measured. The stated pattern was 2x the coverage angle for one side.

Vertical

The measurements were conducted as above with the exception of the fact that both negative (below horn axis) and positive (above horn axis) reading were done for the vertical patterns. The stated coverage angle is the difference between the two points.

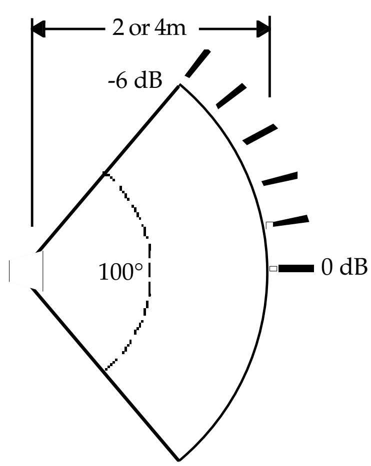


Fig 1.8d

Horizontal coverage for individual speakers as measured for TechNotes™.

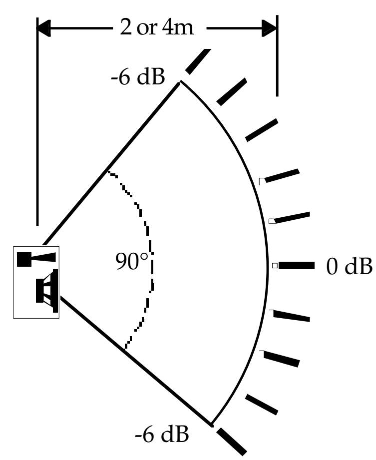


Fig 1.8e

Vertical coverage for individual speakers as measured for TechNotes™.

1.8.3 Coverage Angle Specifications

TechNote Measurement of Speaker Arrays

Array measurements were conducted as measurements of the single speakers. For arrays, the focal point is considered to be approximately the geometric origin of the virtual point source created by the array. This point is typically some distance behind the speaker array, and varies for different angles and quantities. For layout purposes this point should be considered as the origin (not the throat of a single HF driver). Ideally, the measurements would have been conducted in a free field setting and the actual doubling distance ascertained for each array configuration to obtain a true acoustic source. Unfortunately, this was beyond the scope of the TechNotes measurements. In case you were wondering why we would use the multiple microphones instead of a single mic and turn-table, imagine a table large enough to rotate six MSL-3As off center. It's too bad there are none of those locomotive turntables left!

The accuracy of TechNotes coverage angle specifications is estimated to be ±10 degrees.

Horizontal

The horizontal pattern of all Meyer speakers is symmetrical between left and right sides. Therefore, only a single side was measured. The stated pattern was 2x the coverage angle for one side.

Vertical

The measurements were conducted as above with the exception of the fact that both negative (below horn axis) and positive (above horn axis) readings were done for the vertical patterns. The stated coverage angle is the difference between the two points.

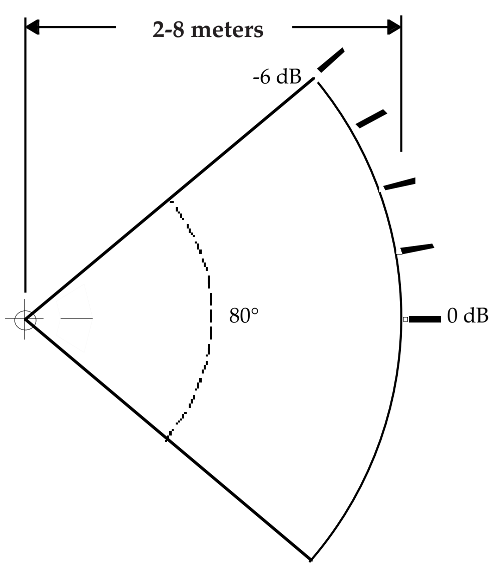


Fig 1.8f

Horizontal coverage for speaker arrays as measured for TechNotes™.

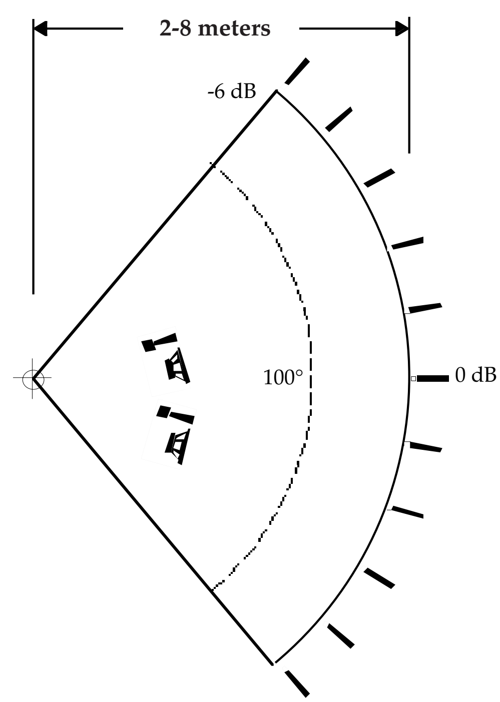


Fig 1.8g

Vertical coverage for speaker arrays as measured for TechNotes™.

1.8.4 Internal Networks

All enclosures with HF drivers contain an internal network that provides various functions.

The internal networks:

- Provide DC protection for the HF driver.
- Protect against LF signals generated by intermodulation distortion.
- Protect the HF driver in cases where the HF and LF amplifier feeds have been swapped.
- Provide optional frequency response modification.

Each model of HF driver has its own unique internal network to optimize its response. Upgrade versions of drivers often require a corresponding network change. The HF driver for the UPA has had four different driver/network iterations.

Driver	Network
1401	Y-1P
1401A	Y-1PB
1401B	Y-1PC (for UPA-1B speaker)
1401B	Y-1PD (for UPA-1C speaker)

More information on this subject can be found in the driver cross reference chart in Section 6, Revision History.

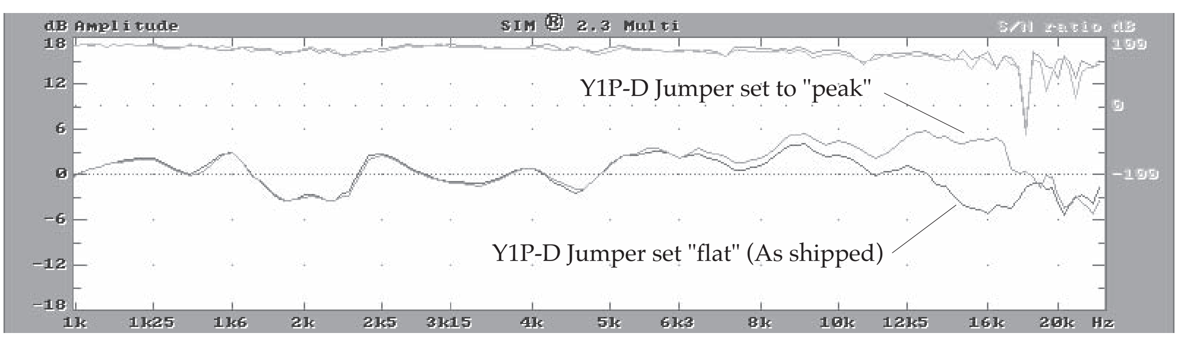


Fig 1.8h UPA-1C response with Y1P-D network.

1.8.5 Driver Components

All Meyer Sound driver components are exclusively manufactured by Meyer Sound. Most of these are, in all aspects, proprietary designs. Other driver components are remanufactured from units originally built by outside vendors. All components are carefully designed, manufactured and rigorously tested.

The grading process routes the drivers into the enclosures where they will provide optimal performance. For example, the MS-15 (fifteen-inch LF driver) is used in both the MSL-2A and USW-1 systems. The MS-15 for MSL-2A requires a high degree of linearity from 40 Hz through the midband, whereas the USW-1 only needs to reach 100 Hz. The MS-15 for the MSL-2A is graded "Silver."

Every component of every loudspeaker manufactured by Meyer Sound is analyzed to verify that its frequency response, phase response and distortion characteristics fall within our specifications. There are no exceptions.

! **An important note regarding remanufactured components:**

Meyer Sound remanufactures HF driver components originally manufactured by Yamaha (MS-1401A) and JBL (MS-2001A). If you examine these drivers you will plainly see the identification marks of these companies. Do not be confused. *The end product is not compatible with these original parts and can not be substituted.* These units are customized by Meyer Sound to achieve greatly improved performance and reliability.

Component testing includes:

- Overnight burn-in.
- Flux density analysis.
- Driver polarity verification.
- Frequency response analysis.
- Phase response analysis.
- Distortion analysis.
- Free air resonance verification.

Component modifications include:

- Ferrofluid™ injection to prevent coil overheating.
- Adhesive overhaul to improve immunity to heat and acceleration.
- Compliance modification to decrease distortion and extend mechanical life.
- Weather resistance to improve immunity to moisture.

Note: The M1 Driver is built on site in Berkeley, CA.

1.8.6 Rigging Options

Ring and Stud

The ring and stud system was developed for the aircraft industry. The ring has a safe working load of 600 pounds (272 kilograms) and is the limiting factor in the load limits for cabinets larger than the UPA. The 600 pounds (272 kilograms) is based on a straight vertical pull. If the cabinet is angled, the working load is derated slightly. When angling cabinets, it should be further noted that the weight is shifted unevenly across the points and can eventually cause the full load to be borne by a single point. It is for this reason that all enclosure working load limits are specified under the assumption that a single point can bear the full load with a 5:1 margin above the breaking point.

Stud

While the stud is rated higher than the ring for a straight vertical pull, it derates rapidly when pulled at an angle. For this reason the stud is less common.

Nut Plate

The standard Meyer nut plate is not "rated" for a safe working load. The Meyer nut plate should only be used to suspend the cabinet to which it is attached with no additional load. The nut plate is usually available only on cabinets weighing less than 100 pounds (46 kilograms).

MSL-5 and MSL-6

The MSL-5 and MSL-6 utilize a new triangular welded "ring" that is rated for 2000 pounds (920 kilograms) safe working load. The MSL-5 and MSL-6 are cases where the enclosure strength is less than the rigging point, giving them a net safe working load of 1500 pounds (681 kilograms).

MSL-10

The MSL-10 is primarily designed for permanent installation. As a result, the rigging fixtures are essentially holes in one-quarter-inch steel. The rigging is done with steel plates and bolts and shackles. The rigging for the MSL-10 is thoroughly described in its operating instructions.

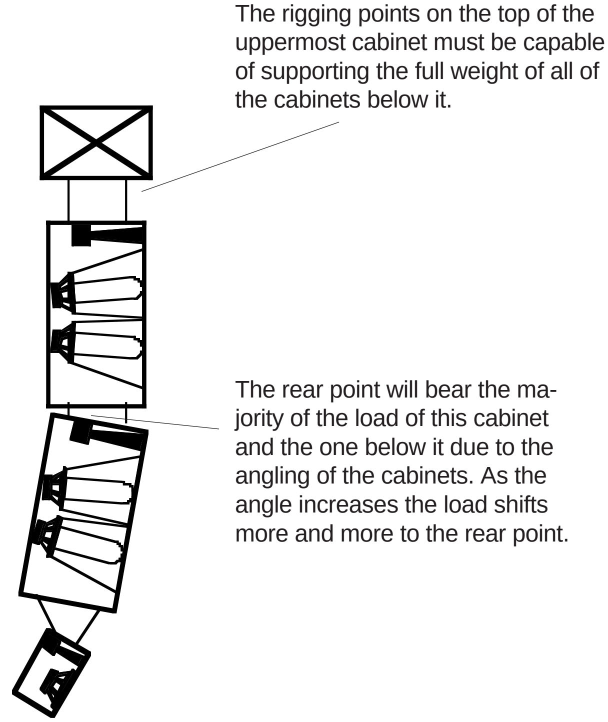


Fig 1.8i

Rigging points load distribution.

Do Meyer Speakers always require external rigging frames?

No. The commercially available frames are not required to simply hang a single speaker or column. The speaker can be hung directly from the standard rigging fixtures. Additional speakers can be underhung and their angle adjusted by the length of the front and back fittings that join the top and bottom cabinets.

The commercially available frames are useful when constructing horizontal point source arrays. The function of the frames is to fix the horizontal splay angles between rows of speakers. Additional rows can then be underhung.

1.8.6 Rigging Options

Speaker	Rigging	Weight	Safe Working Load	Max Units Deep	Row 1	Row 2	Row 3	Row 4
					Example Safe Hang			
MSL-10A	8 points, 3/4" rigging holes in steel cradle	700 lbs. (318 kg.)	1700 lbs. (772 kg.)	2	MSL-10A MSL-10A	MSL-10A	MSL-4	
MSL-6	12 points, pivoting lift rings, 1500 lb. safe load capacity	510 lbs. (232 kg.)	1500 lbs. (681 kg.)	3	MSL-6 MSL-6	MSL-6 2x DS-2P	2x DS-2P MSL-4	 CQ-2
MSL-5	12 points, pivoting lift rings, 1500 lb. safe load capacity	500 lbs. (227 kg.)	1500 lbs. (681 kg.)	3	MSL-5 MSL-5	MSL-5 2x DS-2	2x DS-2 MSL-4	 MSL-2A
MSL-4	Aircraft pan fittings or M10 x 1.5 nut plates	180 lbs. (82 kg.)	600 lbs. (272 kg.)	3	MSL-4 MSL-4	MSL-4 DS-2P	CQ-2 CQ-2	 CQ-1
MSL-3A	Aircraft pan fittings	241 lbs. (109.3 kg.)	600 lbs. (272 kg.)	2	MSL-3A MSL-3	MSL-3A DS-2	MSL-2A MSL-2A	
MTS-4	Aircraft pan fittings or Blank Plates	280 lbs. (127 kg)	600 lbs. (272 kg.)	2	MTS-4 MTS-4	MTS-4 MSL-2A		
CQ-1	Aircraft pan fittings, 3/8"-16 or M-10 nut plates,	130 lbs. (58.6 kg.)	600 lbs. (272 kg.)	4	CQ-1 CQ-1	CQ-1 UPA-1C	UPL-1	
CQ-2	Aircraft pan fittings, 3/8"-16 or M-10 nut plates,	130 lbs. (58.6 kg.)	600 lbs. (272 kg.)	4	CQ-2 CQ-2	CQ-2 CQ-1	CQ-1 UPL-1	
MSL-2A	Aircraft pan fittings, 3/8"-16 or M10 x 1.5 nut plates	82 lbs. (37 kg.)	420 lbs. (191 kg.)	5	MSL-2A MSL-2A	MSL-2A MSW-2	UPA-1C MSL-2A	
USM-1	Aircraft pan fittings or 3/8"-16 or M10 x 1.5 nut plates	82 lbs. (37.3 kg.)	420 lbs. (191 kg.)	5	USM-1			
UM-1C	Aircraft pan fittings or 3/8"- 16 nut plates	67 lbs. (30.4 kg.)	420 lbs. (191 kg.)	6	UM-1C			
UPA-1C	Aircraft pan fittings or 3/8"- 16 nut plates	67 lbs. 81.675	420 lbs. (191 kg.)	6	UPA-1C UPA-1C	UPA-1C UPL-1		
UPA-2C	Aircraft pan fittings or 3/8"- 16 nut plates	67 lbs. (30.4 kg.)	420 lbs. (191 kg.)	6	UPA-2C			
UPL-2	3/8 "-16 nut plates	70 lbs. (32 kg.)	* *	1		* Rigging for these units is intended for single-cabinet use only.		
UPL-1	3/8 "-16 nut plates	70 lbs. (32 kg.)	* *	1		* Rigging for these units is intended for single-cabinet use only.		
UPM-1	3/8 "-16 nut plates	16 lbs. (7.3 kg.)	* *	1	UPM-1	* Rigging for these units is intended for single-cabinet use only.		
UPM-2	3/8 "-16 nut plates	16 lbs. (7.3 kg.)	* *	1	UPM-2	* Rigging for these units is intended for single-cabinet use only.		
MPS-355	3/8 "-16 nut plates	6.6 lbs. (3 kg.)	* *	1	MPS-355	* Rigging for these units is intended for single-cabinet use only.		
MPS-305	3/8 "-16 nut plates	11 lbs. (5 kg.)	* *	1	MPS-305	* Rigging for these units is intended for single-cabinet use only.		
DS-2	Aircraft pan fittings	250 lbs. (113.6 kg.)	600 lbs. (272 kg.)	2	DS-2 DS-2	MSL-3A MSL-3A	MSL-2A UPA-1C	
DS-2P	Aircraft pan fittings	243 lbs. (110 kg)	600 lbs. (272 kg.)	2	DS-2P	MSL-4	CQ-2	
PSW-4	Aircraft pan fittings, 3/8"-16 or M-10 nut plates,	205 lbs. (93 kg)	600 lbs. (272 kg.)	2	PSW-4	MTS-4	CQ-1	
USW-1	Aircraft pan fittings or 3/8"-16 or M-10 nut plates	115 lbs. (52.2 kg.)	420 lbs. (191 kg.)	3	USW-1			
MSW-2	Aircraft pan fittings or 3/8"-16 or M-10 nut plates	66 lbs. (30 kg.)	420 lbs. (191 kg.)	6	MSW-2 MSW-2	MSL-2A UPA-1C	MSL-2A UPA-1C	
650-P	N/A	201 lbs. (91.3kg)	No Points		650-P			
650-R2	N/A	176 lbs. (79.8 kg.)	No Points		650-R2			
HF-3	Aircraft pan fittings or 3/8"-16 or M-10 nut plates	50 lbs. (22.7 kg.)	420 lbs. (191 kg.)	8	HF-3 MSL-3A	MSL-2A MSL-3A		
MST-1	N/A	17 lbs. (7.7 kg.)	No Points					

! **Working load limits are rated at one-fifth (1/5th) of the minimum breaking strength.** Unless otherwise specified all ratings are based on a straight tensile pull. Load directions other than straight can result in a significant reduction in breaking strength. All ratings are for products in new condition. Age, wear or damage to the product can greatly reduce its rating. All speaker enclosures and rigging fixtures must be inspected on a regular basis. All worn, deformed or damaged enclosures or rigging equipment should immediately be removed from service and replaced.

All Meyer speakers must be used in accordance with local state and federal, and industry regulations.

It is the owner's and/or user's responsibility to evaluate the suitability of any rigging method and product for their particular application. All rigging should be done by competent professionals.

1.8.7 Weather Protect Option

The weather protection option for sound reinforcement products consists of loudspeaker cabinet, hardware and driver treatments which greatly enhance reliability in outdoor installations.

Designed to protect the cabinet and speaker components from inclement conditions, these treatments retard moisture intrusion and increase the physical strength of the cabinet joints. Electrical terminals and hardware are plated to resist corrosion, and a weather-resistant paint finish allows for expansion or contraction of the wood without cracking.

The Weather Protection Option is available for the Meyer Sound UPA-1A, UPA-2C, UM-1A, MSL-2A, MSL-3A, MSL-5 650-R2 and USW-1. Self-powered products with the weatherproof option include the CQ-1, CQ-2, MSL-4, MSL-6, PSW-2, PSW-4, and 650-P

Weather Protect Specifications

Enclosure	
Treatment	Epoxy sealant impregnation
Bonding	Structural epoxy glues
Hardware	Stainless steel
Finish	Flexible exterior-grade coating
Rigging	
Treatment	Sealed to prevent moisture penetration
Finish	Powder coated to resist corrosion
Interior Bracket Treatment	Finished to resist corrosion
Grill	Powder coated punched metal screen with acoustically transparent foam inside and out
Transducers	
High Frequency	Electrical terminals plated to resist corrosion
Low Frequency	Electrical terminals plated to resist corrosion Cone impregnated to resist moisture
Connector	Electrical terminals gold plated
Self-Powered	Optional rain hood attachment

1.9.1 Measurement Tools: Microphones

The analysis of loudspeaker systems requires a high-quality measurement microphone with flat frequency and phase response and low distortion. Acoustical testing at Meyer Sound is performed almost exclusively with calibrated microphones manufactured by Bruel & Kjaer (B&K) of Denmark, the worldwide leader in such products. A variety of models are used, with the most common being the Model 4133, an omnidirectional, "free-field" type.

Omnidirectional microphones are preferred for their linear response and freedom from the "proximity effect" of cardioid microphones. Proximity effect is the tendency of cardioid microphones to boost the low frequency response as the sound source approaches the microphone. This makes them impractical for measurement because the source distance would have to be factored into the analysis of the frequency response.

Free-field calibrated mics are used because their response is representative of the human ear's perception of a coherent sound source. This is in contrast to "random incidence" types used for measuring random noise.[1]

Field Measurement

The B&K 4007, also an omnidirectional "free-field" type, is a 3-pin XLR type, 48 volt phantom powered model which is quite practical for field use. This makes the 4007 the preferred choice for alignment of systems using SIM System II in studios and concert halls. The Multichannel version of SIM utilizes large quantities of 4007s. Because of the need for matched sensitivity between microphones B&K has created a "SIM selected" version of the 4007. Meyer Sound has published the criteria for SIM measurement microphones, which is available on request.

Cardioid Mic Measurements

The most popular measurement microphones for aligning sound reinforcement systems are hand held vocal microphones with the test signal of "Test 1, 2" directly coupled to its input in various styles and languages. The frequency response of the most popular of these microphones is shown in Fig 1.9b.

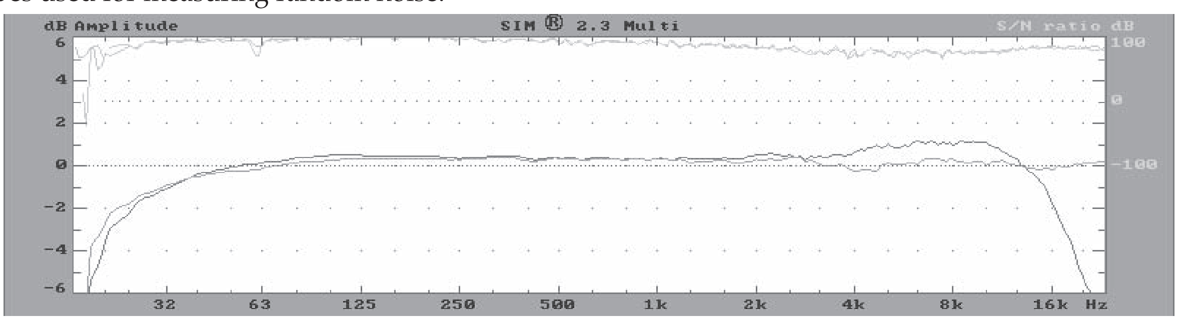


Fig 1.9a Frequency response of a typical Bruel & Kjaer Model 4007 microphone, compared to that of an alternative omnidirectional measurement mic suitable for multichannel sound reinforcement analysis.

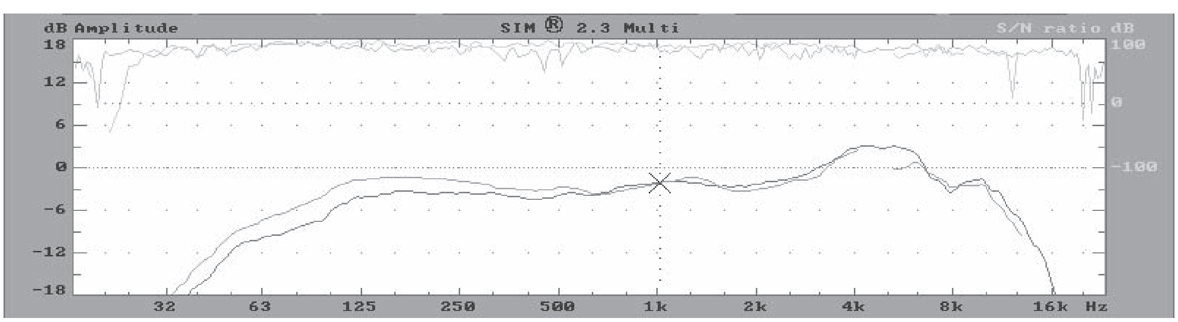


Fig 1.9b Frequency response of the industry standard vocal microphone, complete with proximity effect, presence peak and 20 dB loss at 16 kHz. Note the amplitude scale is 6 dB/division in contrast to the 2 dB/division of the above screen (Fig 1.9a).

[1] Free-field mics are to be pointed at the sound source where a random (pressure mic) has been oriented 90° to the source.

1.9.2 Real-Time Analyzer (RTA)

Real-Time Analyzers (RTAs) are the most common form of frequency response measurement in the sound reinforcement industry, and can be found at the mix console of virtually any concert. It seems, however, that the RTA is only reluctantly embraced by most mix engineers. On one hand, the RTA is always present, on the other, its readings are only given marginal credibility, and virtually no one sets up a system so that it conforms exactly to the RTA's readings (at least not more than once). One of the side effects of this, unfortunately, is that it creates a general distrust for frequency analyzers.

RTAs were originally developed in the 1960s as a simplified method of measuring sound systems. The previous technique of sinewave sweeps generated frequency response charts that were relatively high in resolution, and therefore not visually acceptable since they showed lots of peaks and dips. This method was slow and tedious and it was difficult to create an inverse filter set for the displayed response. Real-time analysis was developed around a parallel set of 1/3-octave-spaced filters. This led to the later development of 1/3-octave equalizers with the same ISO standard center frequencies. Systems could be equalized by simply adjusting the filters to create the desired response for the system. There is one thing that cannot be denied about equalizing by this method. It is the easiest method conceivable. You can teach anyone to do it. Modern technology can and does automate the process. Unfortunately, real sound systems are not that simple. What is it that the RTA is missing?

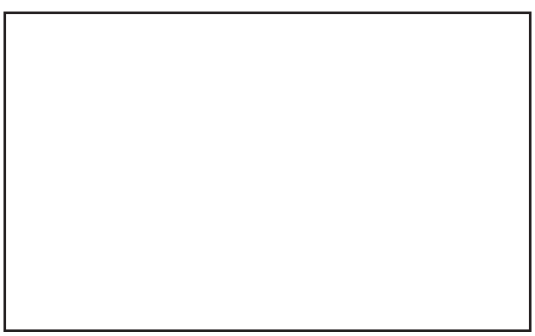


Fig 1.9c The phase response display of an RTA (really).

When using an RTA the following cautions apply:

1) Frequency Resolution

The frequency resolution of 1/3-octave is too low to see peaks and dips that are easily audible. Problems that are obvious to the listener may occur at frequencies that do not conform to the ISO standard center frequencies and are missed by the analyzer. Unfortunately, all of the factors that cause correctable deviations in an installed loudspeaker's frequency response manifest themselves as *linearly spaced comb-filtering*, which the RTA is ideally suited to ignore.

2) Phase Response

The RTA has no ability to measure phase. This precludes the RTA from providing critical information regarding the interaction of speakers, the alignment of crossovers and delay lines. Without knowledge of the phase response, combining speakers is, at best, an educated guess.

3) Signal to Noise Ratio

The RTA gives no indication as to whether the displayed response is derived from the sound system under test or from contamination. This leads to the common practice of voicing systems at ear-splitting levels in an attempt to suppress the potential for contamination. The side effect of this, though, is that distortion and compression are then factored into the analysis, giving erroneous readings.

4) Temporal Discrimination

The RTA has no ability to discriminate between the direct sound, early reflections and late arrivals. Large bodies of research show the human ear's discrimination toward the direct sound and early reflections in characterizing the response of a signal. The RTA, by contrast, displays the summation of all the energy integrated over its display rate time constant (typically 250 ms or more) with no knowledge of whether the signal at the measurement mic ever passed through the system equalizer and, if so, when.

1.9.2 Real-Time Analyzer (RTA)

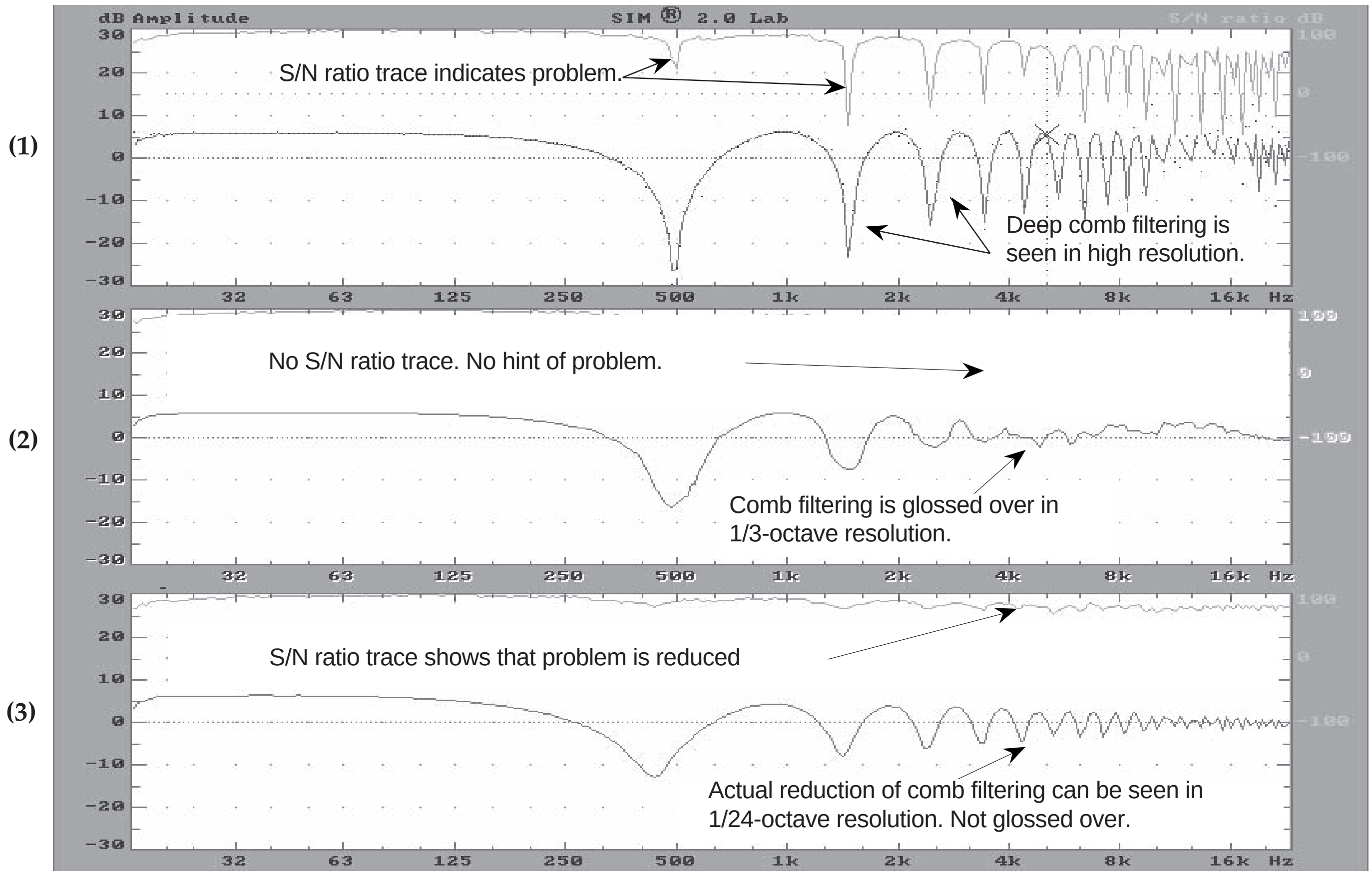


Fig 1.9d A comparison of high and low resolution for frequency response measurements.

(1) High resolution view of comb filtering (1ms delay).

With a resolution of 1/24-octave, we can clearly see the frequency response ripple caused by the comb filtering. If we are to propose solutions for such problems (and this is a problem!) we must first be able to see it. At this resolution we can see that equalization would be a poor solution since it would require hundreds of filters.

(2) Low resolution view of the same system as above.

At the decreased resolution of 1/3-octave, we have lost sight of the magnitude of the problem. In the MF and HF range there appears to be little problem. It also misrepresents the problem as one that could be equalizable by a moderate number of filters. Furthermore, note that this trace is a 1/24-octave representation (242 frequency lines) with 1/3-octave smoothing applied. An RTA has only thirty-one bands and therefore would be coarser than this.

(3) High-resolution view after acoustic absorption.

This is a view of the same system as above after the ripple in the HF range has been dampened by reducing the HF content of the echo as would be done by placing absorptive materials on a surface.

Notice the similarity between this trace (3) and (2). In this case, however, the response of the system has been dampened acoustically, opening the way towards equalization of the remaining ripple.

At this decreased resolution, we have lost sight of the magnitude of the problem. In the MF and HF range there appears to be little problem. It also misrepresents the problem as one that would be equalizable by a moderate number of filters.

1.9.3 Phase Poppers

The limitations inherent to the RTA (see Section 1.9.3) led to the development of tools intended to verify the polarity of system components. These tools are marketed under a variety of names and are generally referred to as "phase poppers." The name springs from the pulse sound emitted from the sender module and the resulting polarity indication at the companion receiver device. This underscores the general misunderstanding regarding the concepts of polarity and phase.

Phase poppers attempts to discern the polarity of a given device by analyzing the voltage orientation (+ or –) of the received pulse in contrast to that emitted. Like the RTA, these devices are simple to operate but again the results are dubious.

Most phase poppers work under the assumption that the device under test (DUT) has a flat frequency response from DC to light. This works well for mic cables and mixing desks and, for testing these items, the poppers are quite reliable. However, when testing loudspeakers, these assumptions are invalid due to the phase delay associated with band pass filtering, acoustical loading, resonance and nonlinearity. Each speaker will have unique characteristics in these regards.

Phase poppers tend to agree generally with the DC polarity of speakers. DC polarity verification is of limited use as discussed in sections 4.9 and 4.10. However, this does not tell us anything about the polarity through crossover, and therefore should not influence decision-making in this regard. An example of the phase response at an acoustical crossover is shown in Fig 1.9e. Notice that the phase response of the individual components changes over frequency. For what frequency range is the phase popper response valid? Notice that at 1100 Hz both the HF and LF phase responses are at 0°. However, by an octave above or below this they have moved to 180°. Can your phase popper single out this relevant range? If not, you may be misled.

A second example found in Section 4.10 is the crossover between the MSL-2A and the MSW-2. The DC polarity (and most likely the phase popper reading) would be identical for the these two components. However, these units are 180° apart over the majority of their overlap range, and therefore would cancel substantially. The fact that the MSW-2 should be reversed in this case would be easily detected by even a modest frequency analyzer or listening test.

If you are going to use a phase popper despite its limitations:

• Do not bother checking any of the Meyer CEUs. The phase correction circuitry in these units will confuse the popper, giving erroneous and misleading results. The phase response of every Meyer Sound CEU is calibrated and verified using FFT analysis at the factory.

• Do not attempt to "pop" your system with multiple transducers driven at the same time. Individual component polarity cannot be measured as such.

• Do not attempt to "pop" your system through the CEU for the same reasons as above.

• Conduct the tests at a moderate and consistent level.

• Replicate, as close as possible, the measurement conditions for each speaker.

• Disconnect the CEU outputs and drive the individual amplifier channels. This will give the least unreliable reading. *Caution*: The SpeakerSense circuit is no longer capable of protecting the drivers under these conditions. Do not overdrive the system.

• The reading on the popper may be green or red depending on the polarity of your power amplifier. What is important is that they are consistent.

• If you pop the speaker directly (no amplifier), all Meyer Sound components should indicate a positive pressure for a positive voltage. This is usually indicated by a green LED. The only exception to this is the HF driver in the UPA-1B and UM-1B which should read the opposite when operating normally.

• If you choose to make changes based upon popper readings, be sure to check the system's response with an analyzer, or by ear, to verify that you have indeed remedied a problem instead of creating one. Techniques for this are described in Section 4.

1.9.3 Phase Poppers

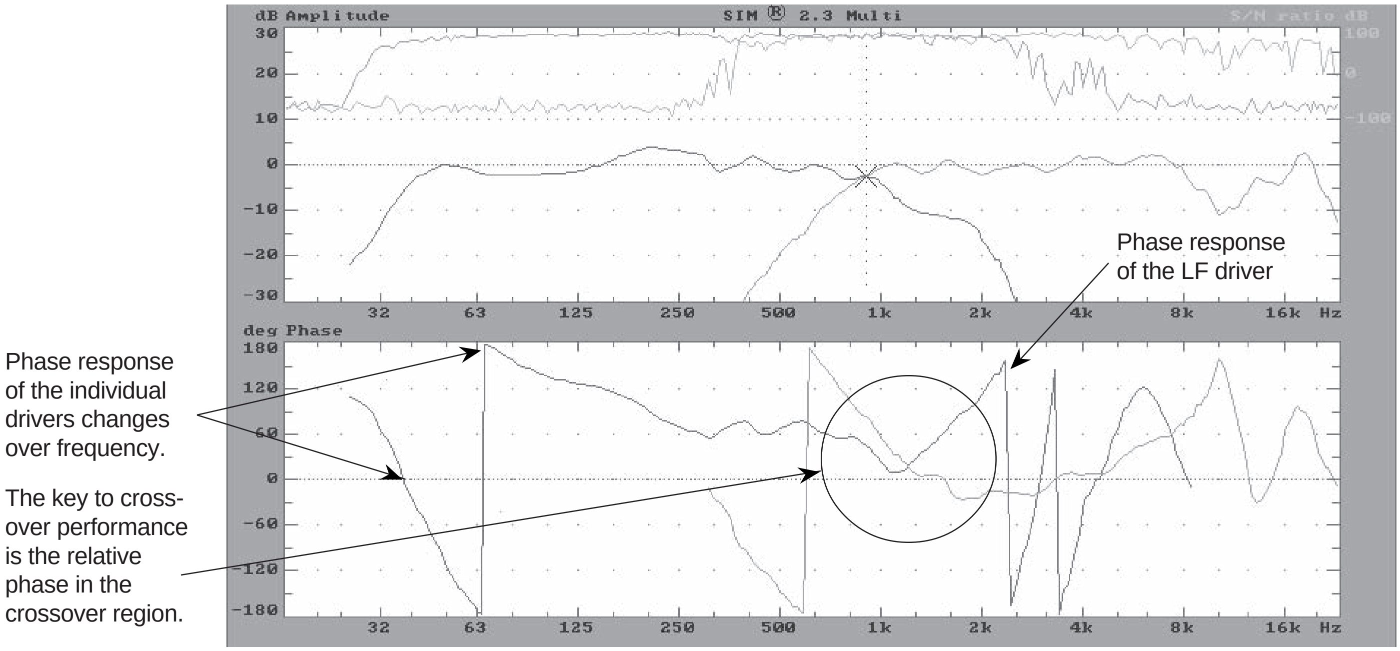


Fig 1.9e The phase response of the acoustical crossover of the MSL-2A loudspeaker. Notice the speakers are "in phase" through the crossover region, yet other regions are "out of phase." A simple "in phase" or "out of phase" reading is not valid for such a system.

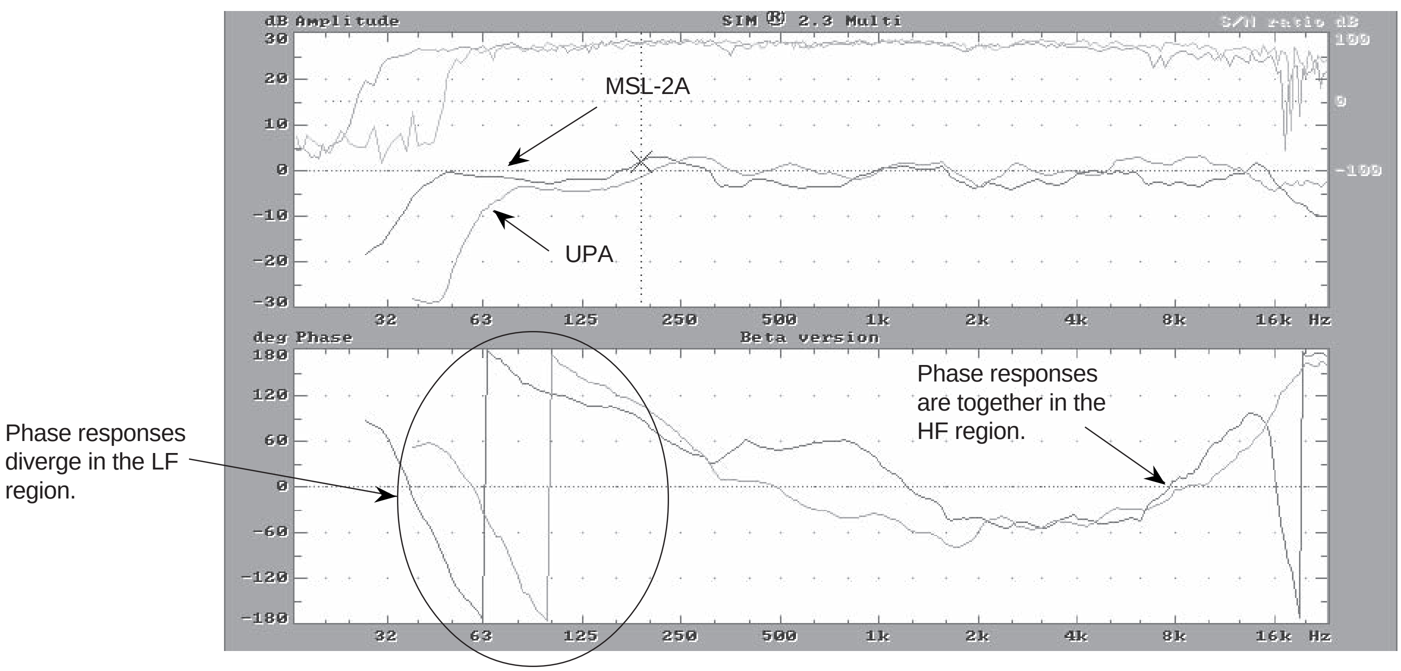


Fig 1.9f The phase responses of the UPA-1C and MSL-2A. The speakers are matched in phase throughout the HF range. However, as the LF cutoff is approached the phase responses diverge. Which of these is "in phase" and which is "out?" A phase popper could give conflicting results depending on which frequency range it reads. This difference in phase response becomes critical when subwoofers are added to these systems.

1.9.4 Source Independent Measurement (SIM)

Meyer Sound's preferred measurement tool is SIM System II. This system was developed specifically to enable audio professionals to get laboratory accuracy in a field setting. SIM System II allows the engineer to see the system status in great detail and point to real solutions. SIM is a significant investment for any user. However, you have to ask yourself, "How much is my sound system investment worth if it's not properly maintained and aligned?"

SIM stands apart from conventional audio analyzers, which focus primarily on the final acoustical response or on individual components. Such analyzers, even high power FFT analyzers developed for the aerospace industry, are not well-suited for sound reinforcement work because the user is required to construct a complex interface to access the system, or to spend time continually repatching. You can imagine the look on the mixer's face as you repatch your analyzer into the main system feeds during a concert.

SIM integrates itself into the heart of a complex sound reinforcement system with a simple interface. It provides data for the verification and alignment of the microphones, mixing console, outboard gear, banks of equalizers, delay lines and speaker subsystem at various positions in the hall without repatching. Most of these operations can be done while the music is playing and with the audience in place.

Because SIM is such a quantum leap forward from conventional analysis practices, there are some misunderstandings regarding its functions and capabilities. These range from focusing on only a single aspect (typically using music as the test signal) to rampant exaggeration into some kind of automated "Fix-your-PA-even-though-it's-a-badly-designed-and-poorly-installed-magic-machine."

Let's set the record straight.

SIM® Source Independent Measurement

SIM System II:

- Is an analyzer capable of providing accurate high-resolution data regarding the response of whole sound systems and components.
- Analyzes frequency response, phase and S/N ratio at 1/24-octave resolution and delay offset between devices (or echoes) at an accuracy of ± 20 µs.
- Provides easy methods for accurate equalizer, delay line and level setting.
- Displays the interaction between speakers so that their effects can be minimized.
- Verifies polarity.
- Analyzes total harmonic distortion (THD).
- Uses music as the test signal.
- Displays the frequency response of the direct signal and early reflections.
- Can access up to sixty-four equalizers and microphones without repatch.
- As with anything, requires a skilled operator to obtain the best results.

SIM System II:

- Is not an automated equalization system.
- Does not advocate equalization as a solution for all problems.
- Is not an acoustical prediction program.
- Does not continually change the system's response.
- Is not just a program you can buy for your PC.
- Does not undo a mixer's equalization.
- Is not exclusive to Meyer speakers and equalizers.
- Does not remove all of the low end from a speaker system.

1.9.4 Source Independent Measurement (SIM)

Reading the Spectrum Screen

Spectrum: This is the amplitude (dB) versus the frequency response of the independently measured input and output signals. This is the screen most recognizable to users of RTAs (although it is 1/24-octave).[1] As levels rise, the traces move upward on the screen.

The spectrum screen can be used for distortion analysis, maximum output level testing and noise floor analysis.

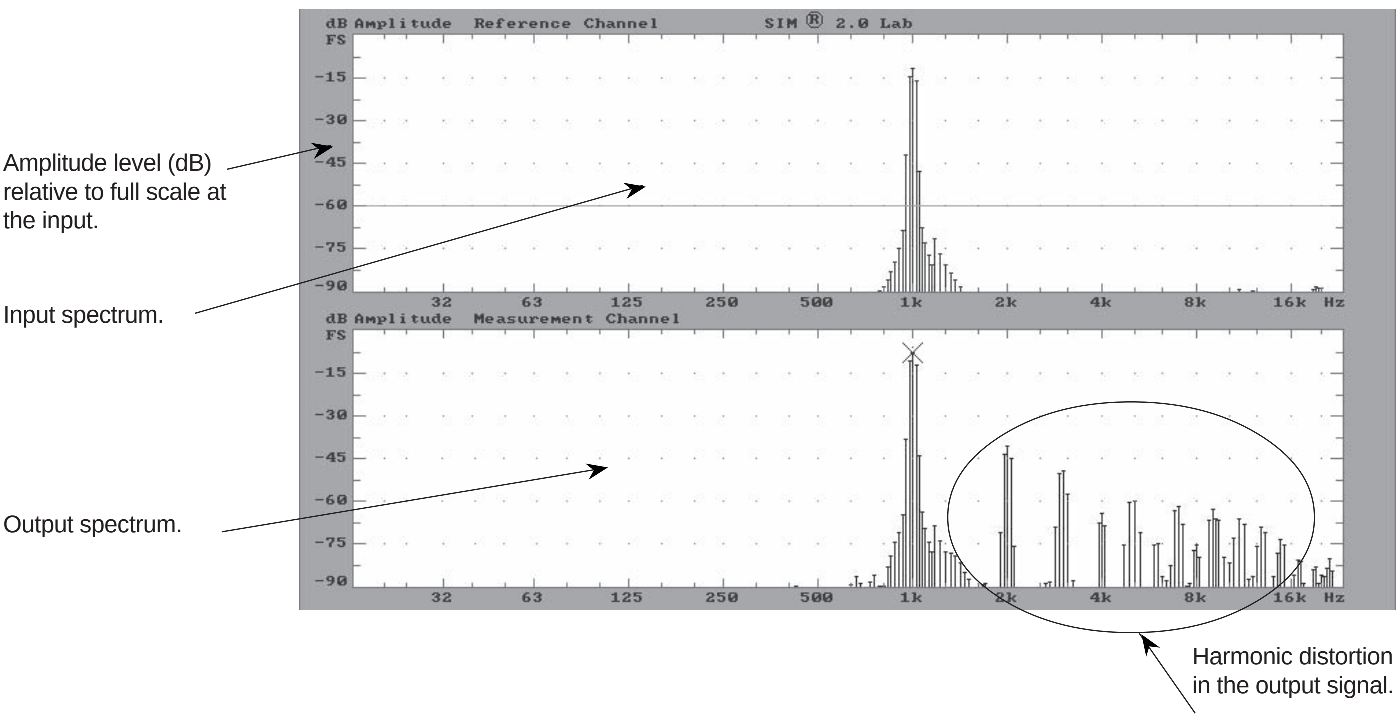


Fig 1.9g Reading the spectrum screen.

[1] One-sixth octave in spectrum mode. See SIM Data Sheet.

1.9.4 Source Independent Measurement (SIM)

Reading the Delayfinder Screen

Delayfinder: The delayfinder shows the transient response of the system over time. This is the measured response of the system calculated to appear as it would on an oscilloscope (amplitude versus time) when excited by a pulse. This trace shows the time offset between input and output. The delayfinder trace is used to set delays, identify echoes and characterize speaker interaction.

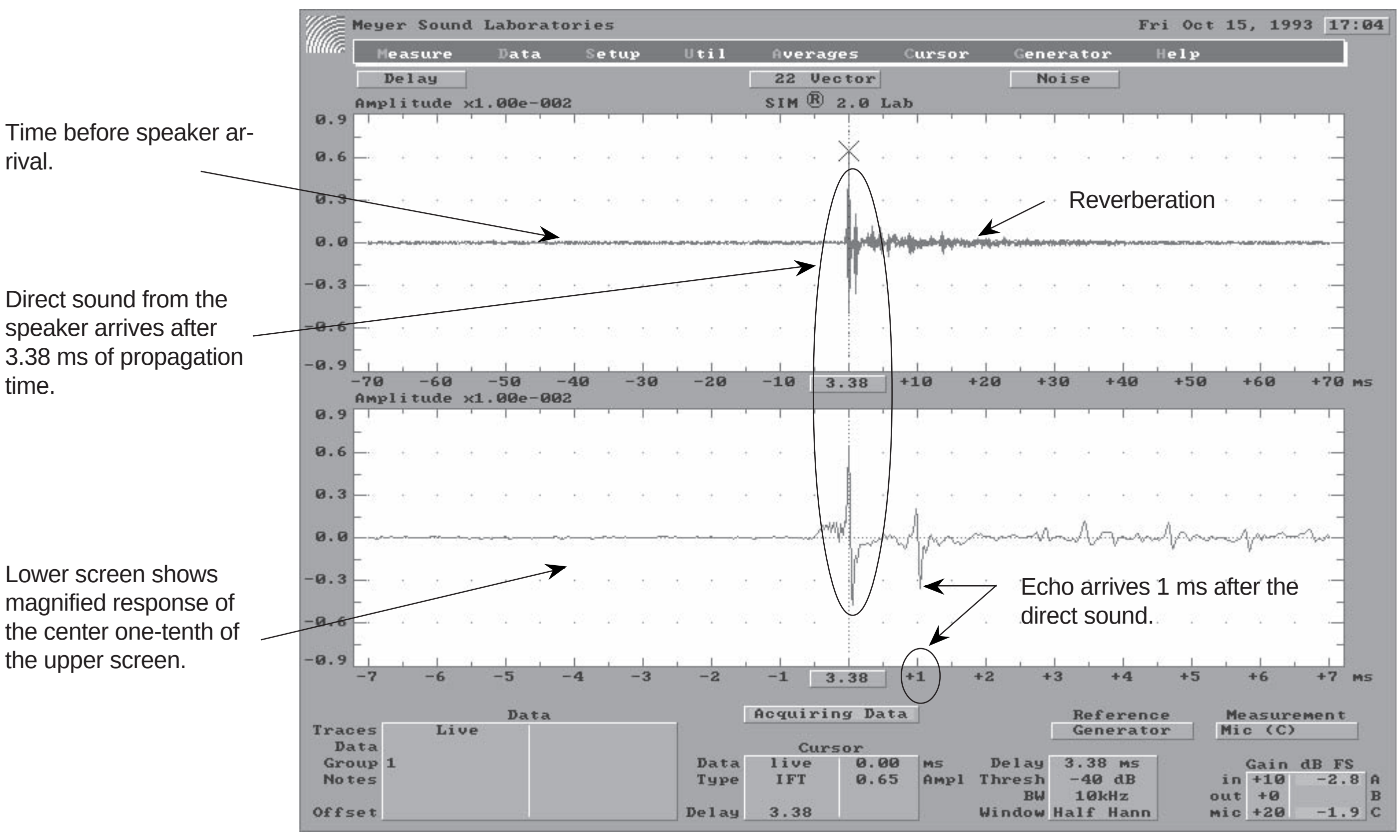


Fig 1.9h Reading the delayfinder screen.

1.9.4 Source Independent Measurement (SIM)

Reading the Frequency Response System

Relative amplitude: This is the *difference* in level (dB) versus frequency between the input and output signals. Unity gain is at the screen center. Because this is a differential measurement, it can be made with any source signal (a Source Independent Measurement).

S/N ratio: This corresponds to the amount of stability in the measured system. If the relationship between the input and output signals is not constant, this S/N ratio will degrade. Anything that causes the input and output signals to decorrelate is considered noise. Some of the typical factors that degrade the S/N ratio include thermal noise, distortion, reverberation, compression and air handling noise. Another mechanism for decreasing the S/N ratio is the reduction of the signal by cancellation, such as in the case of the interaction of reflections and multiple speakers. The S/N ratio is expressed in dB.

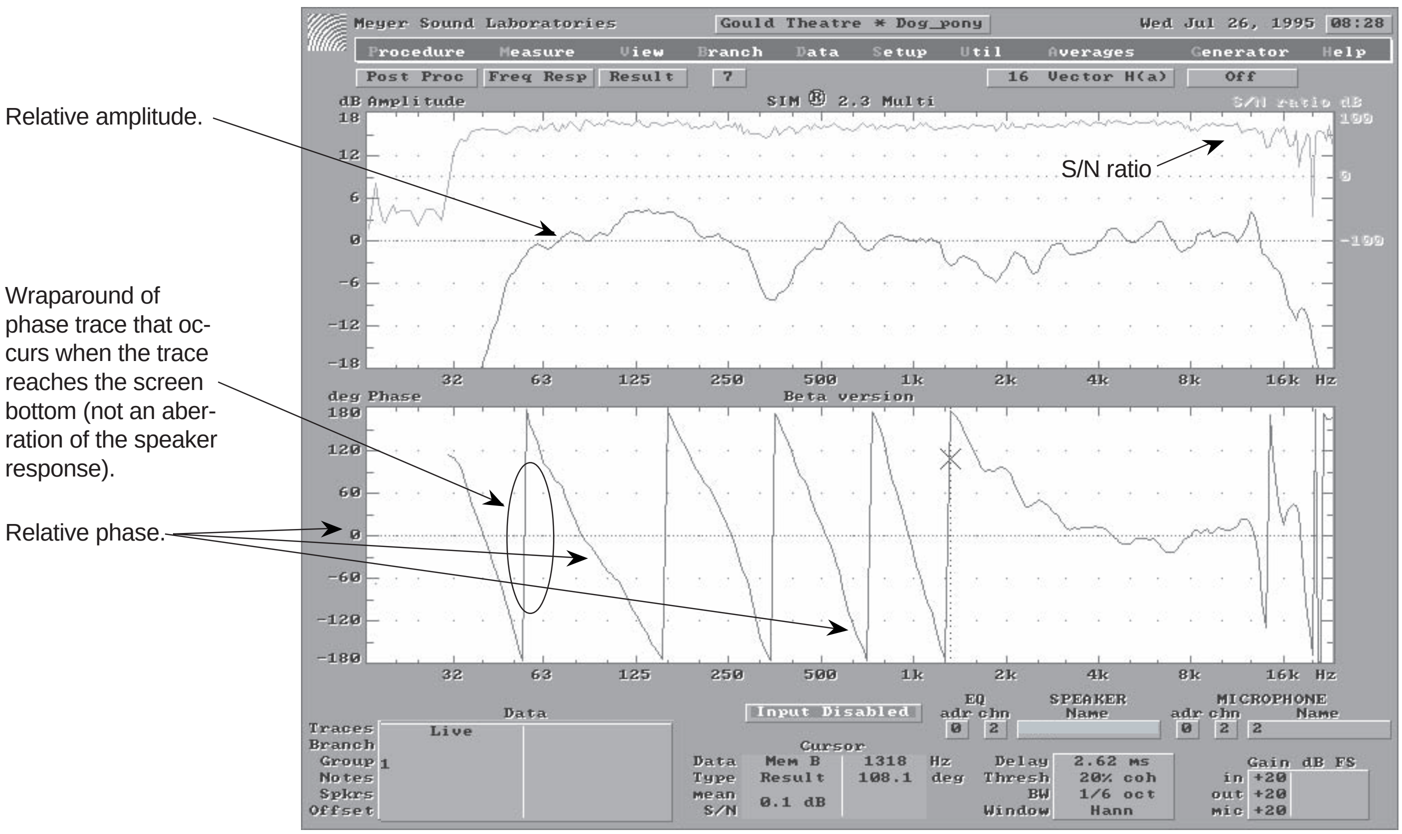


Fig 1.9i Reading the frequency response screen.

Relative phase: This is the difference in phase (degrees) over frequency between the input and output signals. This corresponds to the amount time delay versus frequency. A downward slope (from left to right) indicates delay, while an upward slope indicates lead. When the trace reaches the screen edge (±180°), it is redrawn at the opposite edge.

LF and MF ranges are delayed behind the HF range. This is seen by the downward phase angle. The response at 500 Hz lags behind that of 4 Khz by 2 ms. By 100 Hz it has fallen back 6 ms. (Not a Meyer speaker!)

1.9.4 Source Independent Measurement (SIM)

Do We Need High Resolution Complex Analyzers?

Yes. You already have two of them: Your ears. However, you will benefit greatly by having additional objective information about a system. If you have a poor analyzer, you will inevitably discard its information when it does not fit the reality perceived by your ears.

Bear in mind that the human ear does not hear sound as low-resolution amplitude only. It is obvious that we can perceive the order of things (phase response) and the direct-to-reverberant ratio as well as the amplitude response. If your analyzer cannot discern between these, you may be seriously misled.

The example responses shown in Figs 1.9j –1.9l will clarify the need for high-resolution complex analysis.

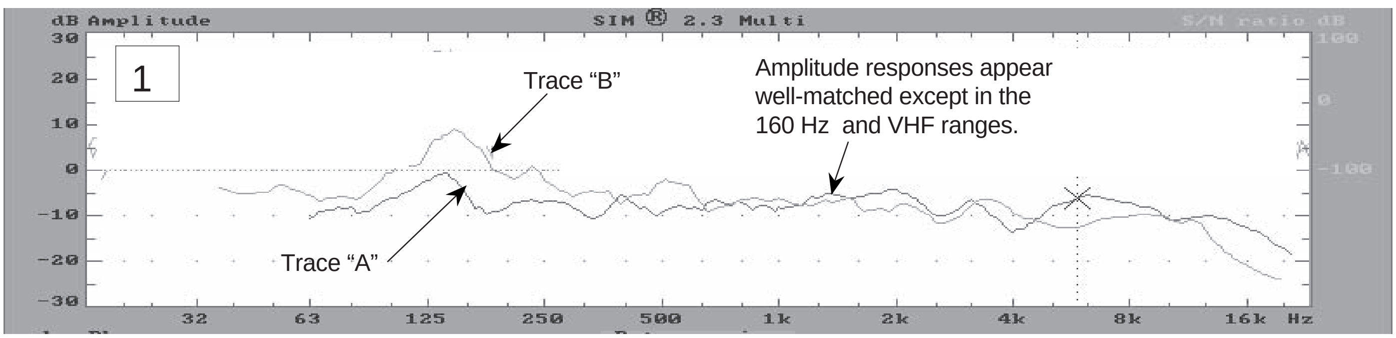


Fig 1.9j (1) The importance of high-resolution amplitude complex measurements.

(1) A comparison of two speaker systems when displayed with low-resolution amplitude responses only. The only substantial differences appear in the low-mid region where response "B" has a peak, and the extreme HF region where "B" has dropped off. There is nothing shown here that would indicate that one of the two systems is utterly unintelligible and the other crystal clear. There is also nothing shown that would indicate that an equalizer would be totally incapable of correcting the differences between them.

(2) This screen shows the amplitude, phase and S/N ratio of system "A" in 1/24-octave resolution. Notice that the S/N ratio appears near the top of the screen at most frequencies. Such areas have high intelligibility and a high direct-to-reverberant sound ratio. The areas where the S/N ratio dips down are frequencies where cancellations have occurred due to echoes or interaction between speakers. The presence of these echoes is confirmed by the phase response.

(3) This screen shows the amplitude, phase and S/N ratio of system "B" in 1/24-octave resolution. Notice that the S/N ratio appears near the middle of the screen throughout virtually the entire MF and HF ranges. This indicates that the system is unintelligible and has a low direct-to-reverberant ratio. The overwhelming strength of these echoes is confirmed by the phase response which shows huge variations. The amplitude response is so poor that most of the data has been blanked out because the S/N ratio is too low to make an accurate calculation. In actual fact, the data here comes from an underbalcony listening area with no direct sound path for the HF horn. The conclusion of all this is that differences that are obvious to our ears are not seen by conventional low-resolution amplitude-only analyzers. They are visible with SIM System II, enabling its operator to make competent decisions.

1.9.4 Source Independent Measurement (SIM)

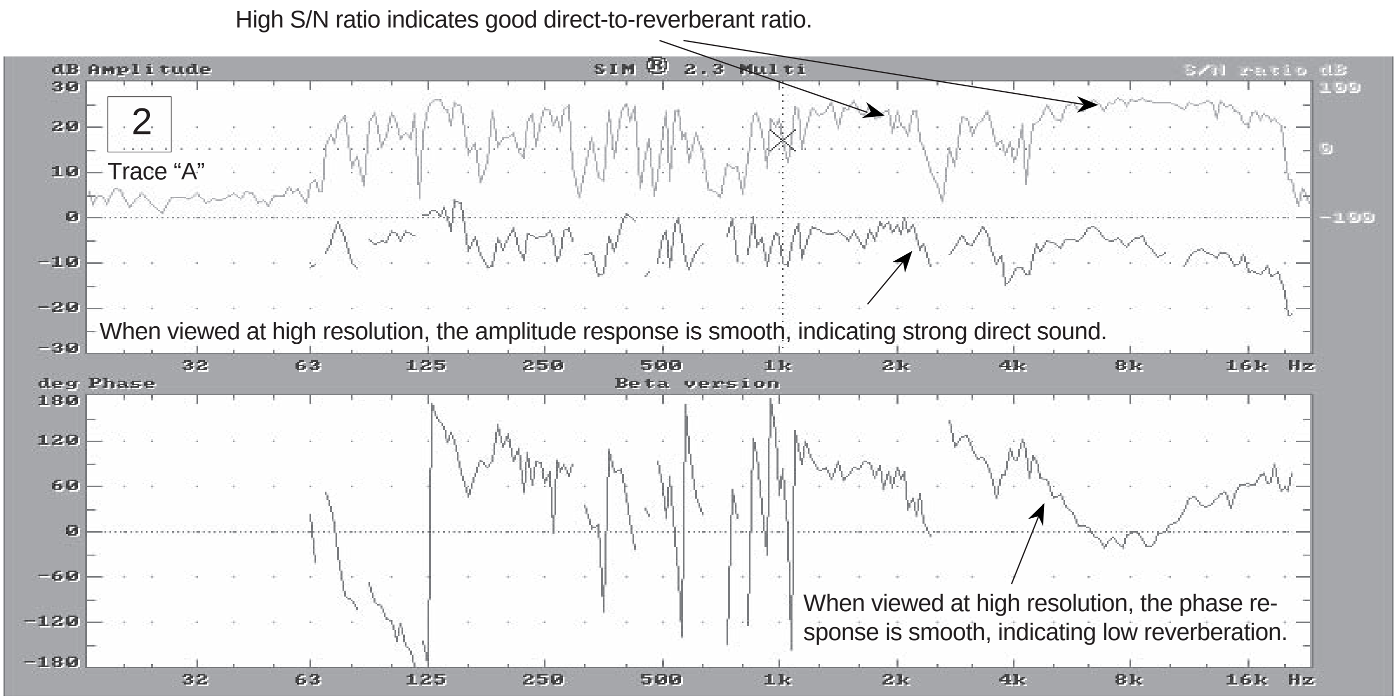


Fig 1.9k (2) High resolution view of amplitude, phase and S/N ratio of speaker "A" alone.

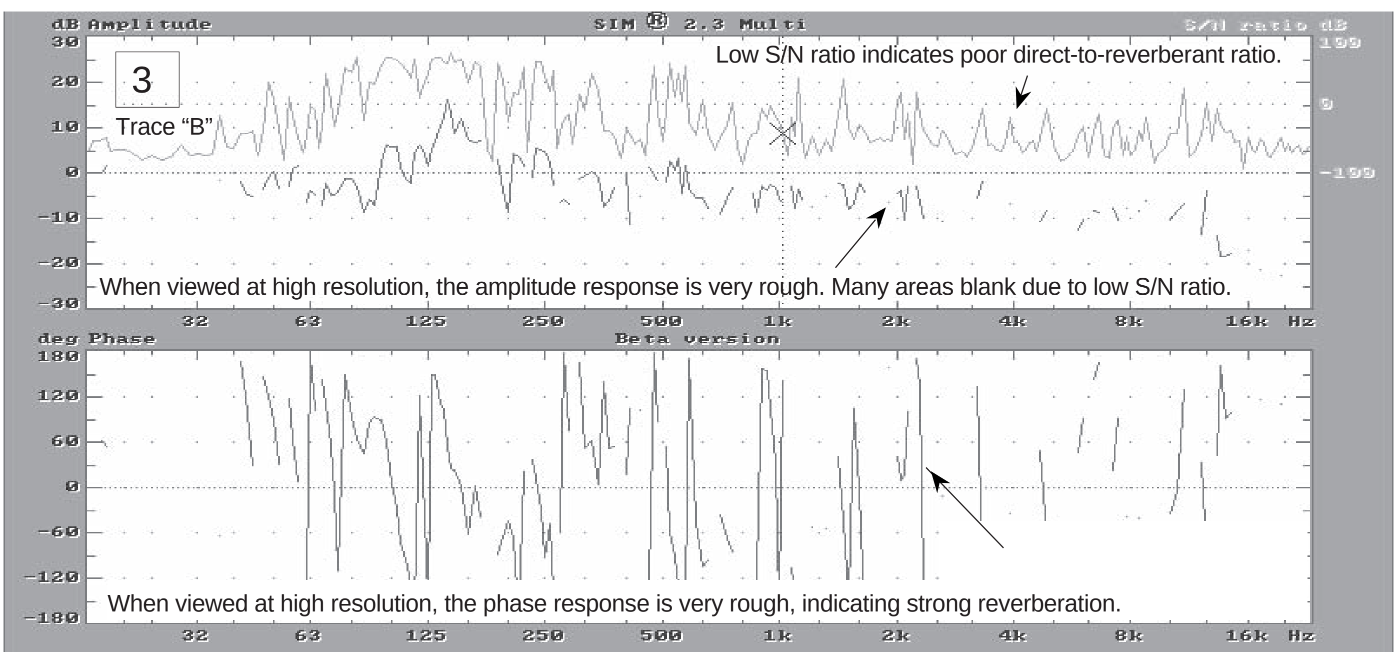


Fig 1.9l (3) High-resolution view of amplitude, phase and S/N ratio of speaker "B" alone.

1.9.4 Source Independent Measurement (SIM)

v2.0 Lab

The simplest version of SIM System II is v2.0 Lab. This version can be described as a dual channel Fast Fourier Transform analyzer. This stands in contrast to the other versions which are geared towards field alignment. The v2.0 version utilizes a single DSP card and gives a single transfer function. All connections are made to the front panel. It is capable of either electronic or acoustic measurement, using the internal generator or an external unknown source.

v2.0 Standard Functions

- Spectrum response: Distortion analysis, maximum output level, noise.
- Frequency response: Amplitude, phase, S/N ratio.
- Impulse response: Find delay offset.
- 1/24-octave frequency resolution.
- Capable of using unknown source for measurement.

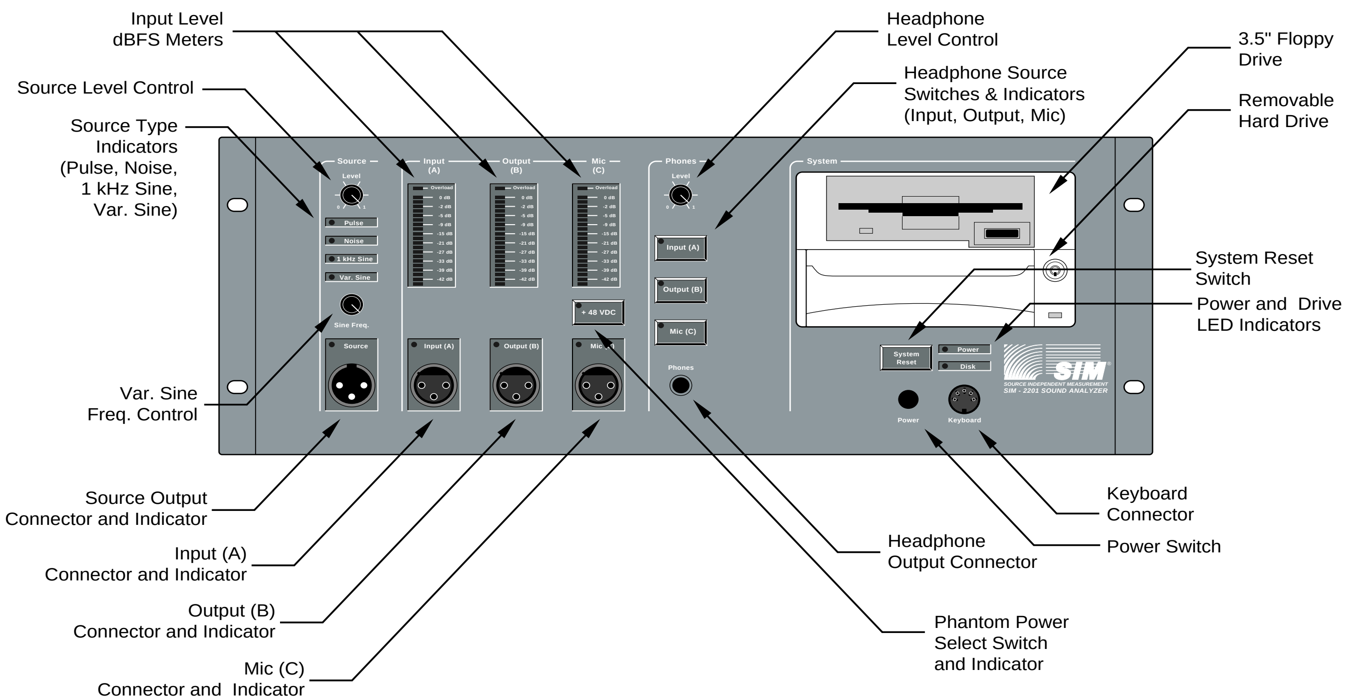


Fig 1.9m SIM 2201 sound analyzer.

Drawing by Ralph Jones

1.9.4 Source Independent Measurement (SIM)

There are a great variety of applications for v2.0. Below are a few samples.

v2.0 Lab Standard Applications

- Research: This is the central analysis tool for the research and development of Meyer Sound loudspeaker systems.
- Manufacturing test: All HF driver production testing and final loudspeaker testing for Meyer Sound uses this version.
- Rental stock test: v2.0 is used by rental companies to verify the performance of speakers and electronics before and after road use.
- Checkout of system components and racks: v2.0 is used by rental companies and installers to verify wiring and check electronics prior to, and during, installation.
- Microphone testing: Used for viewing the amplitude and phase response of any microphone. Measure the axial response of microphones to find the best feedback rejection.

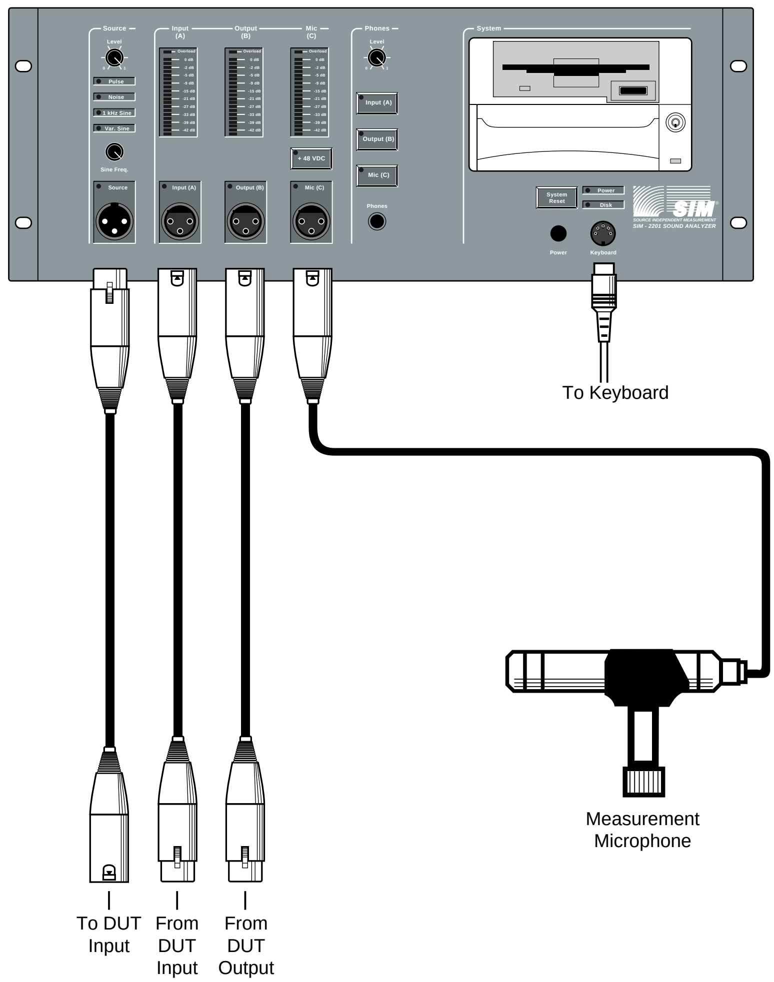


Fig 1.9n v2.0 lab hookup.

Drawing by Ralph Jones

1.9.4 Source Independent Measurement (SIM)

v2.3 Stereo

The stereo version goes way beyond a simple analyzer. This is an alignmnent system for audio systems that are optimized for precise equalization, delay and level setting. The hardware accomodates a stereo equalizer through an interface snake and a single measurement mic via the front panel. This system utilizes three DSP cards to measure the room+speaker, equalizer and result transfer functions simultaneously, allowing the user to precisely create the inverse response of the room on the equalizer. This is the easiest and *most accurate* method of EQ setting in the world.

v2.3 Stereo Standard Functions

- Contains all v2.0 Lab features.
- A procedure menu to guide the alignment process.
- Group view of room+speaker, equalizer and result transfer functions for precise equalization.
- Time windowing that allows the analyzer to see the frequency response as it is perceived, by allowing the direct and early reflections. This is also the range that will respond best to equalization.
- Delay setting procedure (takes under 5 seconds).
- Precise level setting (± .1 dB) at a frequency resolution of 1/24-octave.
- High immunity from outside noise allows analysis at low sound levels and the accommodation of people who need to work in the hall.
- S/N ratio function alerts operator when equalization will not be an effective solution.
- Phase response function provides key data for the alignment of crossovers.
- Storage and easy recall of 128 groups of data for comparison.
- Silent equalization procedure for when a lighting crew has to have total silence.

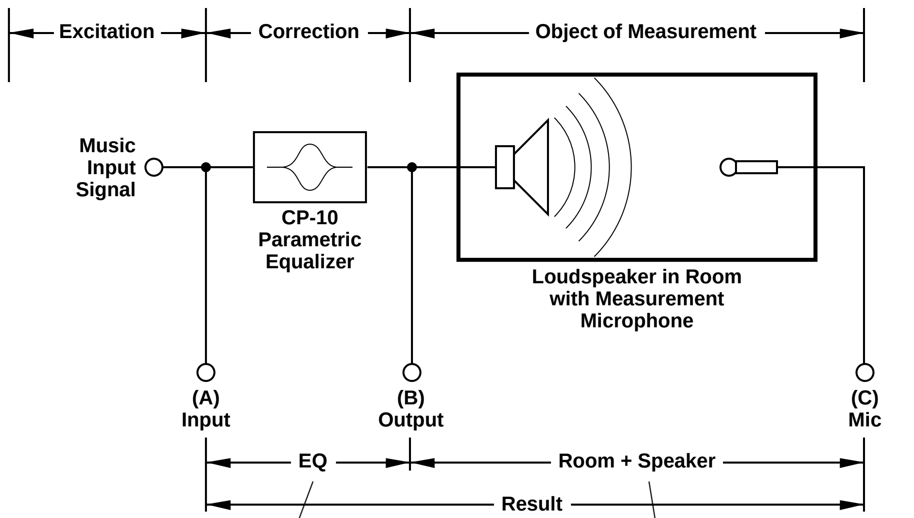
There are a great variety of applications for v2.3 Stereo. Below are a few samples:

v2.3 Stereo Standard Applications

- Contains all v2.0 Lab standard applications.
- Recording studio monitor verification and alignment.
- Stereo sound reinforcement system verification and alignment.
- Multichannel sound system alignment. (Repatching is required.)
- Checkout of system components and racks: v2.0 is used by rental companies and installers to verify wiring and check electronics prior to and duriing installation.
- Microphone testing: Used to view the amplitude and phase response of any microphone. Measure the axial response of microphones to find the best feedback rejection.

1.9.4 Source Independent Measurement (SIM)

The stereo (v2.3)and multichannel (v2.3m) versions of SIM System II measure three simultaneous transfer functions: room+speaker (the original response of the speaker in the space), the equalizer, and the result (the response of the speaker system and equalizer combined). The connections for this are shown in Fig 1.9p. This unique function is utilized in the "group view" screen shown below. The user equalizes the system by creating a complement to the room+speaker system. This process can be further eased by visually inverting the equalizer trace (1/EQ) and simply matching the response of the room+speaker system.



Drawing by Ralph Jones

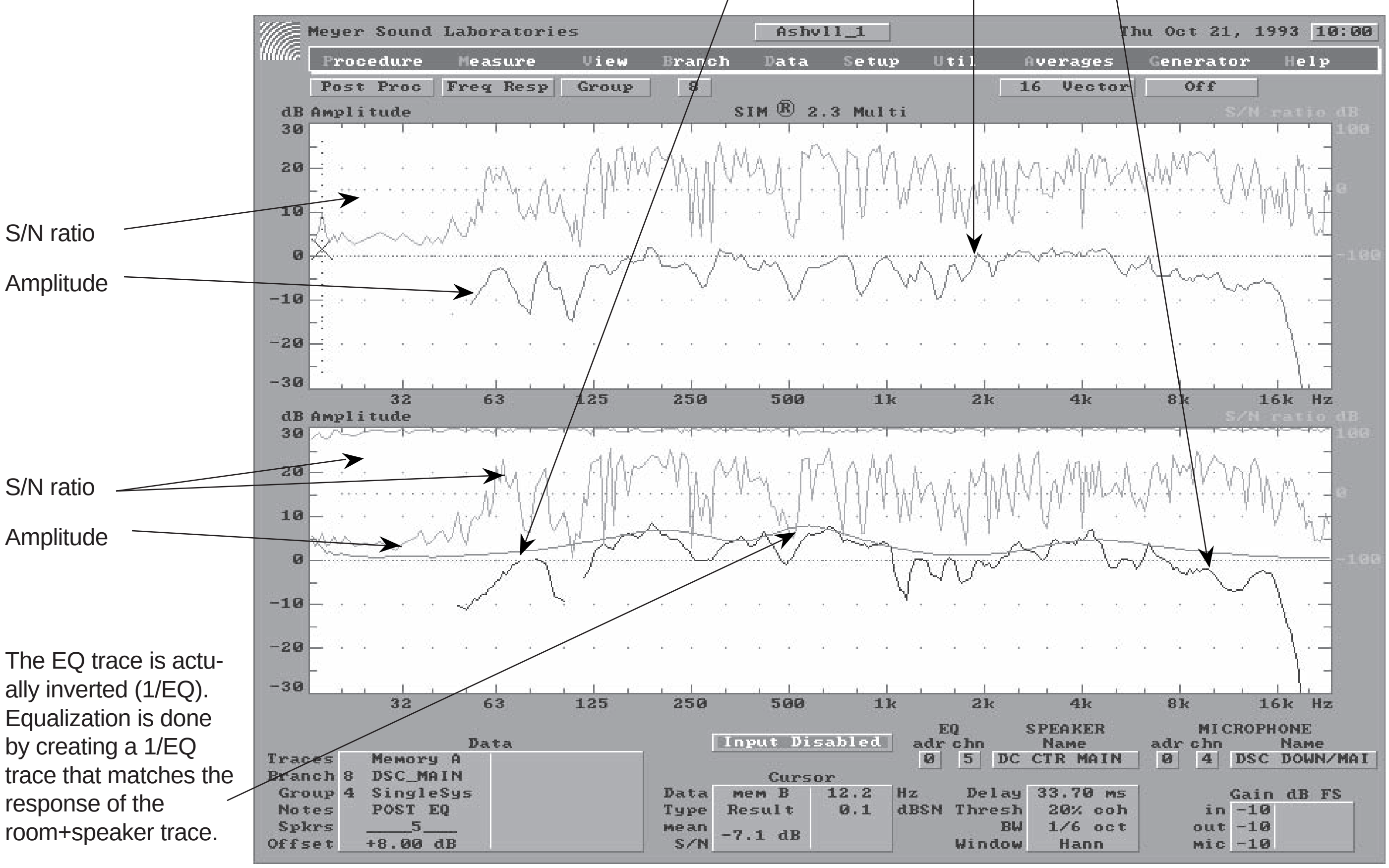


Fig 1.9p The v2.3 family of measurements (room, EQ, result).

1.9.4 Source Independent Measurement (SIM)

The stereo version can be interfaced directly into the sound system as shown below. The Stereo Interface Snake routes the signal at the EQ inputs and outputs into the SIM 2201 Sound Analyzer for measurement. The single measurement microphone is patched into the front panel mic preamp.

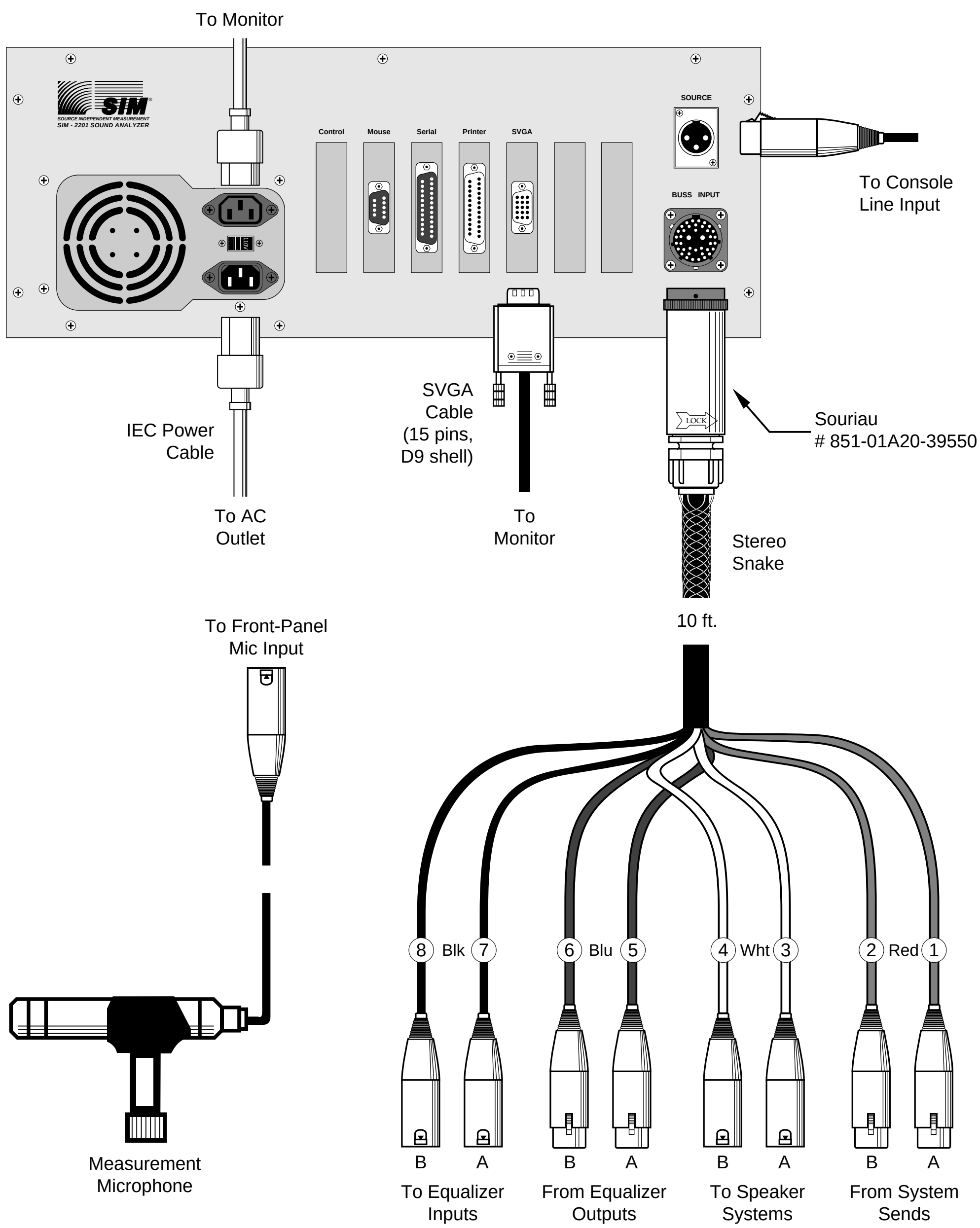


Fig 1.9q The v2.3 stereo hookup.

Drawing by Ralph Jones

1.9.4 Source Independent Measurement (SIM)

v2.3m Multichannel

The multichannel version (v2.3m) further expands upon the previous systems by accommodating multiple microphones and equalizers. This is the system of choice for complex system alignment on the road, or for permanent installation. The multichannel accesses the system via an external SIM 2403 Interface Network. Each 2403 has eight microphone and equalizer channels. Multiple units can be used to access up to sixty-four channels. What sets the multichannel system apart from the stereo version is the time saved in data acquisition, by being able to align different subsystems without repatching or moving microphones. Once the multichannel system is interfaced, the entire alignment of the system can proceed at a furious pace.

Multichannel sound systems differ from simple stereo systems in that there are complex interactions between the subsystems (as described in Section 2.2). These interactions require careful analysis at various mic positions so that corrective action can be taken on the various equalizers, delays and level controls. It is during this type of analysis that repatching or moving mics slows things down too much to be of practical benefit. Time is usually the most precious commodity in the course of a system alignment. This is where the v2.3m version goes beyond all other alignment systems and stand-alone analyzers.

v2.3m Multichannel Standard Functions

- All v2.0 Lab and v2.3 stereo features.
- Mic compare procedure to check the response of mics.
- Data panel spreadsheet keeps data sorted for easy comparison.
- Interface network allows access to multiple mics and EQs without repatch.
- Lobe study procedure indicates the degree of isolation between subsystems.
- Combined systems procedure shows the interaction between speakers.

There are a great variety of applications for v2.3m Multichannel. Below are a few samples.

v2.3m Multichannel Standard Applications

- All v2.0 Lab and v2.3 stereo standard applications.
- Alignment of touring shows.
- Musical theatre alignment.
- Permanent installation alignment.
- Reconfigurable system maintenance and alignment.

1.9.5 Source Independent Measurement (SIM)

The multichannel version can be interfaced directly into a sound system as shown below. The SIM 2403 Interface Network routes the signal at each of eight EQ inputs, outputs and measurement microphones into the SIM 2201 Sound Analyzer for measurement. The SIM 2403 can mute any combination of speakers to aid in the alignment process. Additional 2403 networks can be cascaded to provide additional channels (up to sixty-four) for measurement. The record holders (at time of writing) are Sean Glen and Phil Harris of STG Entertainment who have used three 2403s with a total of twenty-four EQs and mics for several large-scale movie openings.

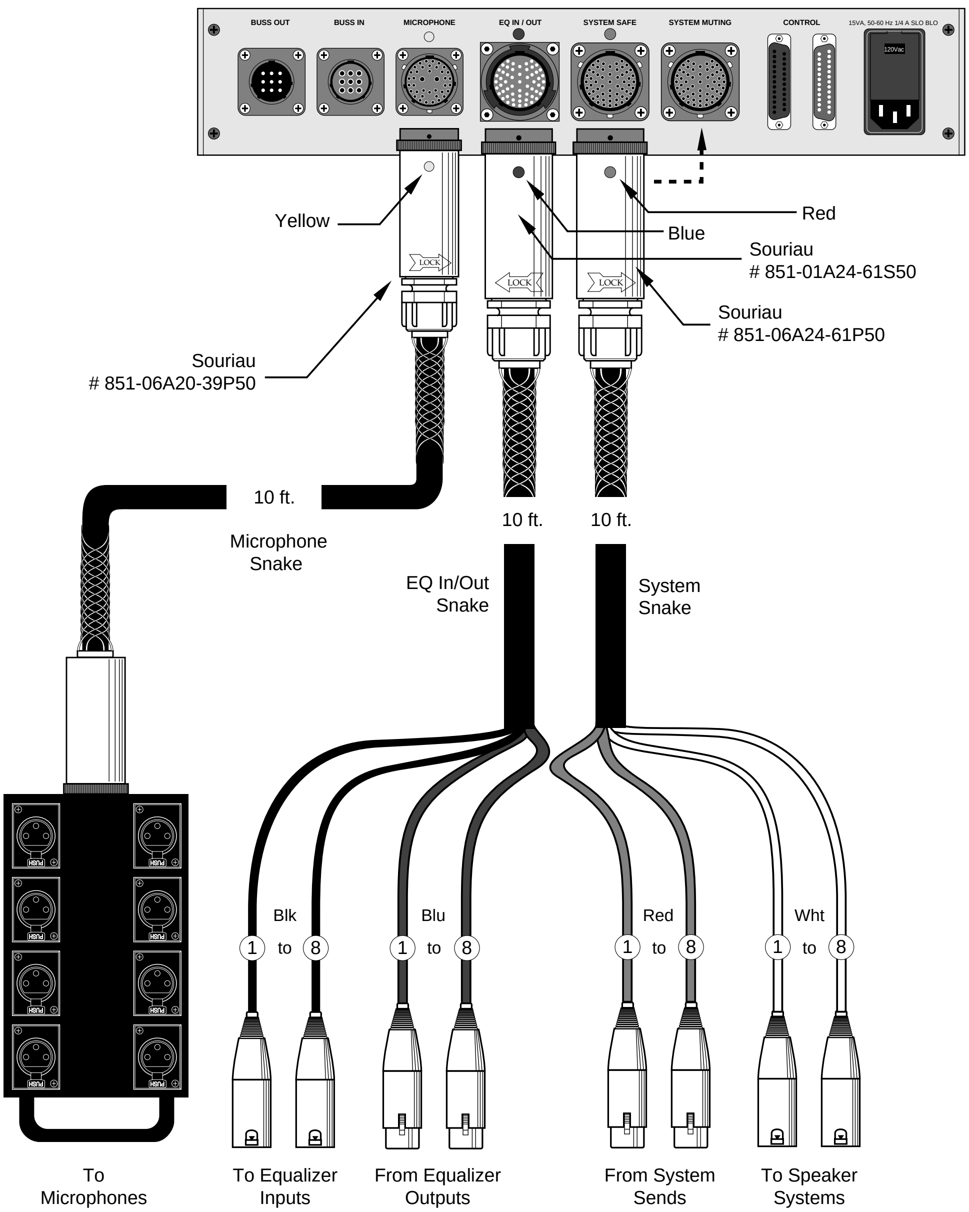
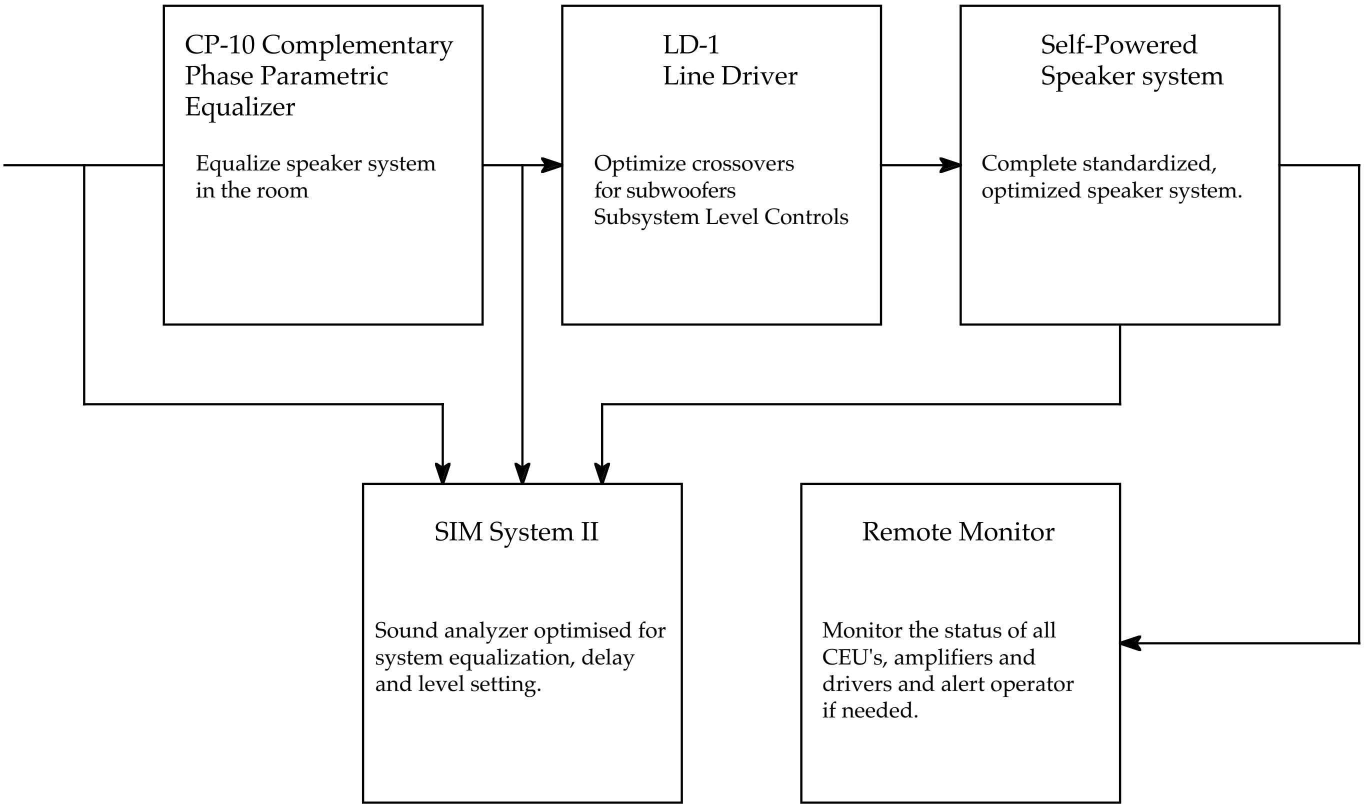


Fig 1.9r The v2.3m multichannel hookup.

Drawing by Ralph Jones

1.10 Meyer Sound's Total Solution



The self-powered speaker systems serve to complete Meyer Sound's Total Solution ™ for sound reinforcement. Meyer Sound provides all of the vital components for engineers to install, align and verify the performance of their system.

2.1.0 Introduction

In a typical acoustics text you might find the section on comb filtering in the neighborhood of such subjects as the Doppler Effect. The Doppler Effect is interesting, of course, but has no practical application for the sound reinforcement professional.

Comb filtering, by contrast, is the principle method of coloration of sound system response, present in virtually every interaction with a sound system. Hours are spent repositioning speakers and adjusting equalizers and delays trying to tame comb filtering. Many esteemed audio professionals are experts at intuitively disarming comb filters, in spite of never having seen them on an analyzer, and having only a vague notion of their cause.

The resolution of Real-Time Analyzers (RTAs) is too coarse to see comb filtering. It is little wonder that this subject is so poorly understood.

Fluency with the concept of comb filtering can improve your ability to design efficient, coherent sound systems in which the speakers work together and the effects of the room are minimized.

There are a variety of ways in which comb filtering is introduced into a sound system. Four of the more typical methods are shown in Fig 2.1a. The end result is the same in all cases: frequency and phase response ripple and signal-to-noise ratio loss.

Two independent factors determine the magnitude and frequency response of comb filtering: The relative *level* and *phase* (time offset) between the signals.

1) As the relative levels approach unity, the magnitude of the peaks and dips increases.

2) As the time offset increases, the frequency range (where the combing is most audible) decreases.

The formula for comb filter calculation is:

comb frequency = 1/time offset

time offset = 1/comb frequency

The first cancellation will occur at one-half the comb frequency

2.1.0 Introduction

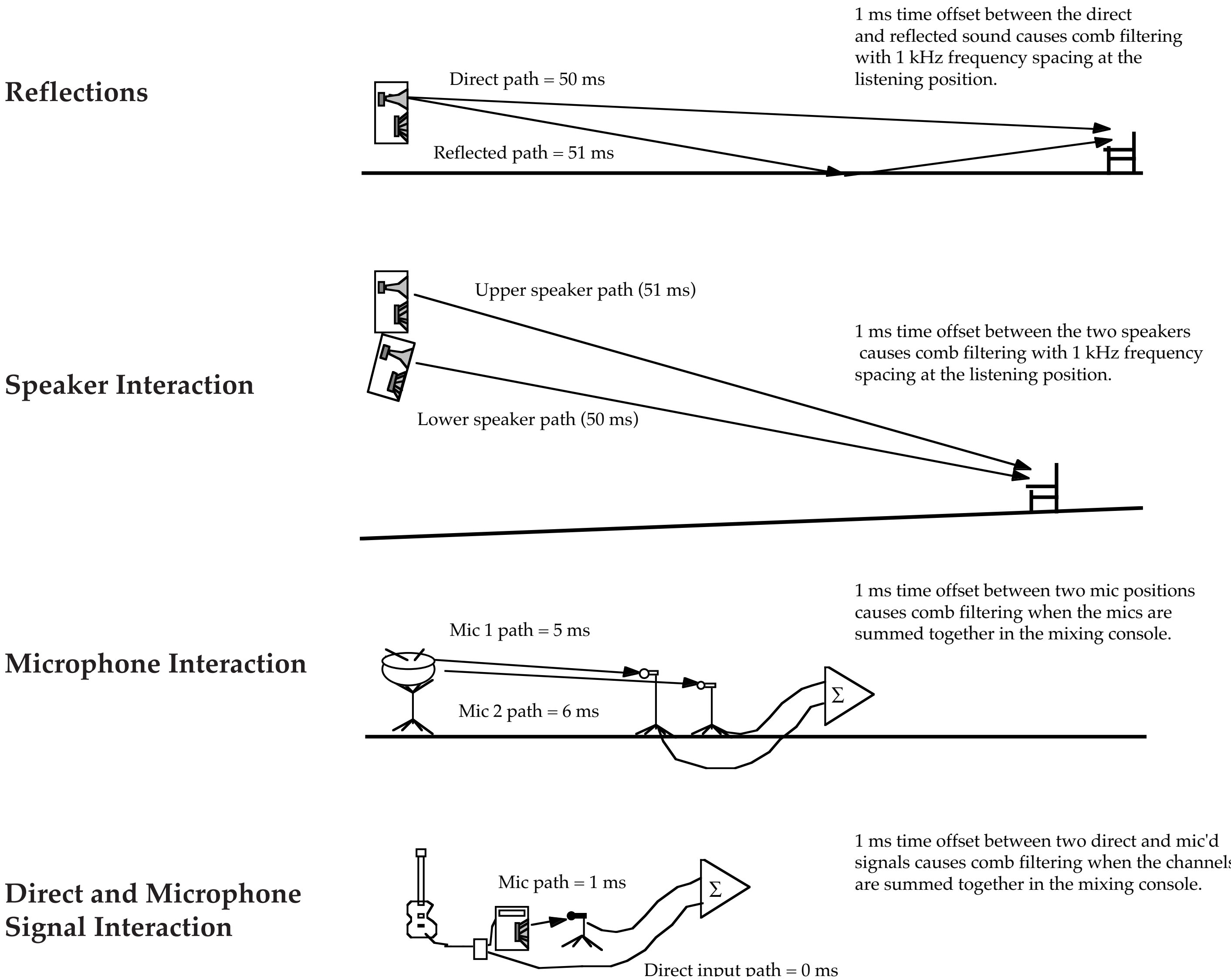


Fig 2.1a Four examples of typical causes of comb filtering.

2.1.0 Introduction

What is the mechanism that creates comb filtering? For example, a 1 ms time offset illustrated in Fig. 2.1a (previous page) will result in a comb frequency spacing of 1000 Hz.

1/time offset = comb frequency spacing

1/.001s = 1000 Hz

The response for such a system is shown in Figs 2.1b-2.ld.

Fig 2.1b shows the delayfinder response of a summation of two signals with 1 ms of time offset. The arrival of the direct and reflected signals is shown as discrete peaks displaced horizontally by 1 ms. The delayed signal is slightly reduced in amplitude, as can be seen by its relative height.

The frequencies which will have the maximum addition are integer multiples of the comb frequency—in this case, every 1000 Hz. The addition occurs because the phase relationship between the two signals is a multiple of 360°, resulting in phase addition. The maximum cancellation will occur at the half-way point between additions—in this case 500 Hz, 1500 Hz, 2500 Hz, etc. This is due to the phase cancellation that occurs when the signals are 180° apart.

Fig 2.1c shows the amplitude and phase response of two signals measured separately before summation. The high frequency range of the delayed signal has been gently rolled off. This is similar to the absorption that might occur from a soft boundary. The phase response reveals the time offset between the signals and where the summation and cancellation will occur can be clearly seen. Maximum addition will occur where the phase responses come together. When the phase responses are 120° apart there will be no addition or cancellation. Maximum cancellation will occur when the phase responses are 180° apart.

A Note About Reading the Phase Trace

The phase response trace of the delayed signal moves suddenly from the bottom of the screen to the top at 500 Hz, 1500 Hz, etc. This is a display function of the analyzer. When the phase response reaches –180° (screen bottom) it rolls over to +180° (screen top) and onward. At 360° of phase shift the trace has returned back to the screen center (0°). For this application such a display is preferable since it is possible to always view the relative phase responses in the way that illustrates where the addition and cancellation will occur.

The summed response is shown in Fig 2.1d. Note that the positions of the maxima and minima correlate to the frequencies where the phase responses come together and apart, respectively. Note also that the "ripple" in the amplitude response (its deviation above and below 0 dB) diminishes as frequency increases. This is due to the HF rolloff of the delayed signal, which increases the level offset between the direct signal and delayed signals.

2.1.0 Introduction

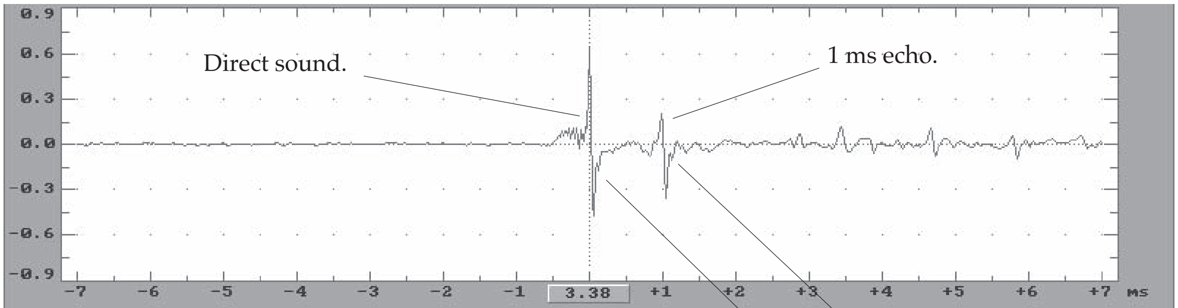


Fig 2.1b

Delayfinder response of direct and delayed signals.

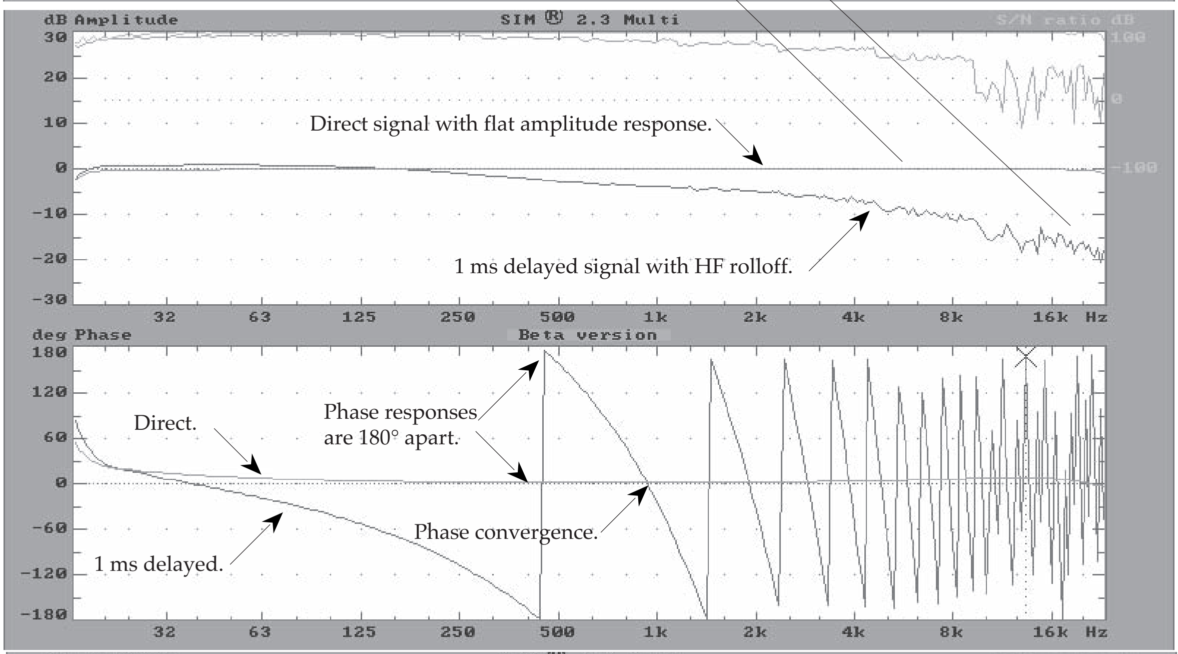


Fig 2.1c

Frequency response of direct and delayed signals measured separately.

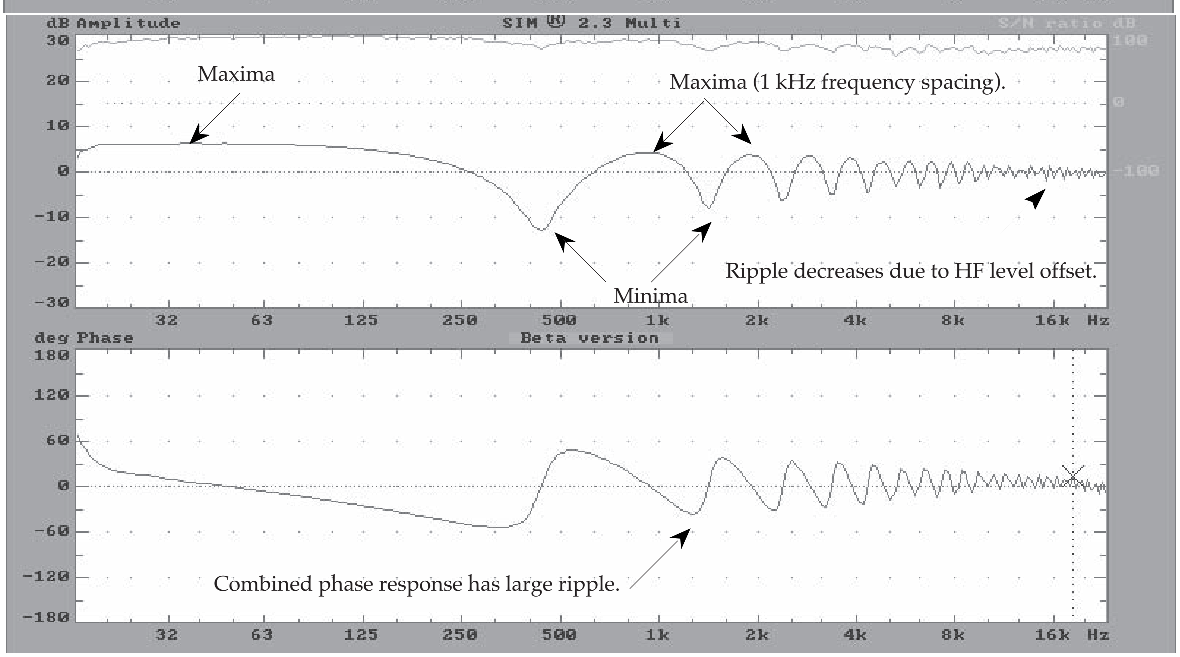


Fig. 2.1d

Comb filtering caused by summation of the direct and delayed signals.

2.1.1 Comb Filter Frequency

The term “comb filtering” comes from the fact that the peaks and dips look like the bristles of a comb on a linear frequency axis display. But since human hearing responds logarithmically, the image of a “comb” is misleading when visualizing the sonic effect of comb filtering. To our ears the spacing between the peaks and nulls is not even at all. When viewed on a log scale (Fig 2.1d), we see it as we hear it, with wide peaks in the lower frequencies and steadily compressing as frequency rises. The frequencies where comb filtering will begin is dependent upon the time offset. As the time offset *increases*, the frequency of the first null *decreases*. This is shown in Figs 2.1d–2.1f. It is only the start frequency (where the first null occurs) that changes with time offset. Above the first null the shape of the response is the same, illustrating how the same sonic effect moves through the audio range as the time offset changes. In each case the second peak (this is the peak between the first and second nulls) is an octave wide. The succeeding peaks are 1/2, 1/3, 1/4 octave, etc.

The term "accordion filtering" would probably be more descriptive, since as the time offset increases the peaks and dips are compressed further to the left resembling the movement of an accordion, but it is doubtful that this term will catch on as an industry standard.

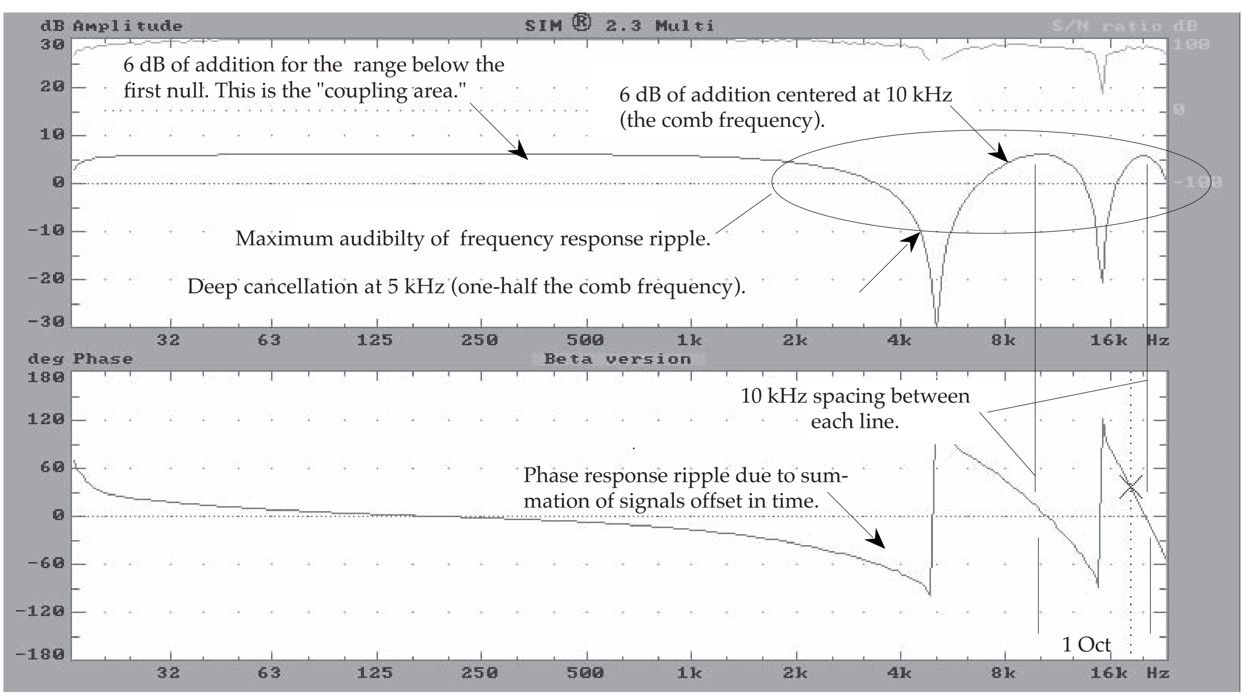


Fig 2.1e Comb filtering with .1 ms time offset between signals with 0 dB level offset.

2.1.1 Comb Filter Frequency

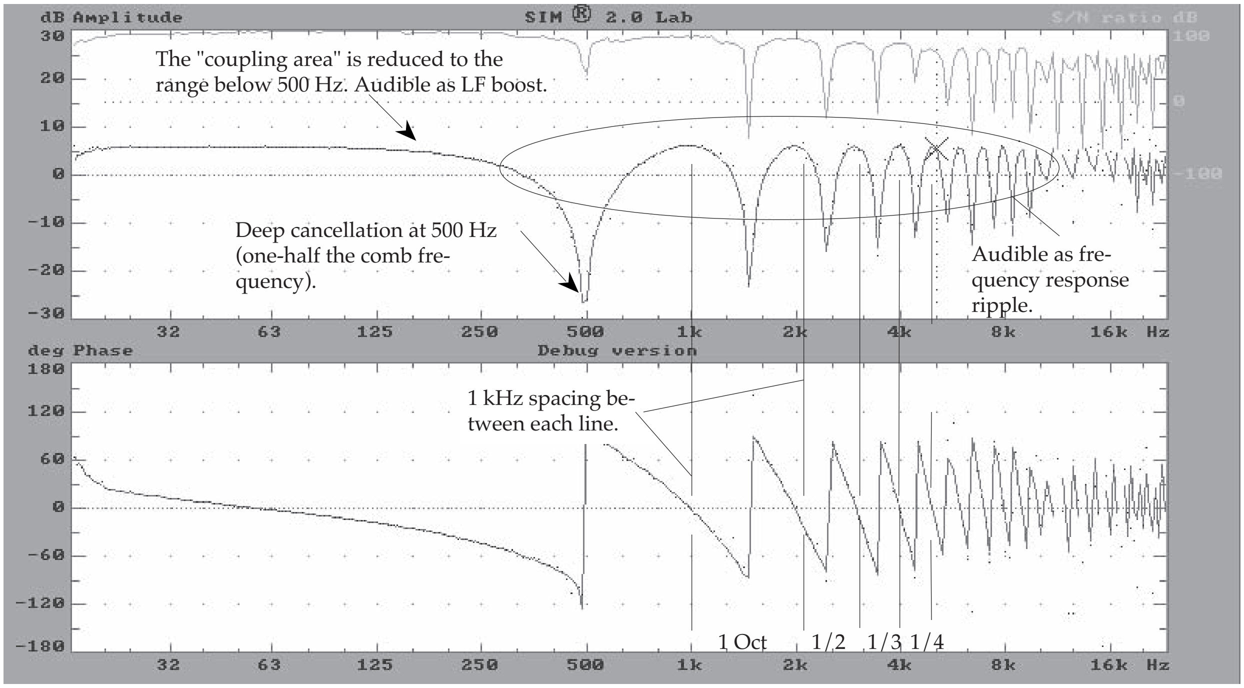


Fig 2.1f Comb filtering with 1 ms time offset between signals with 0 dB level offset.

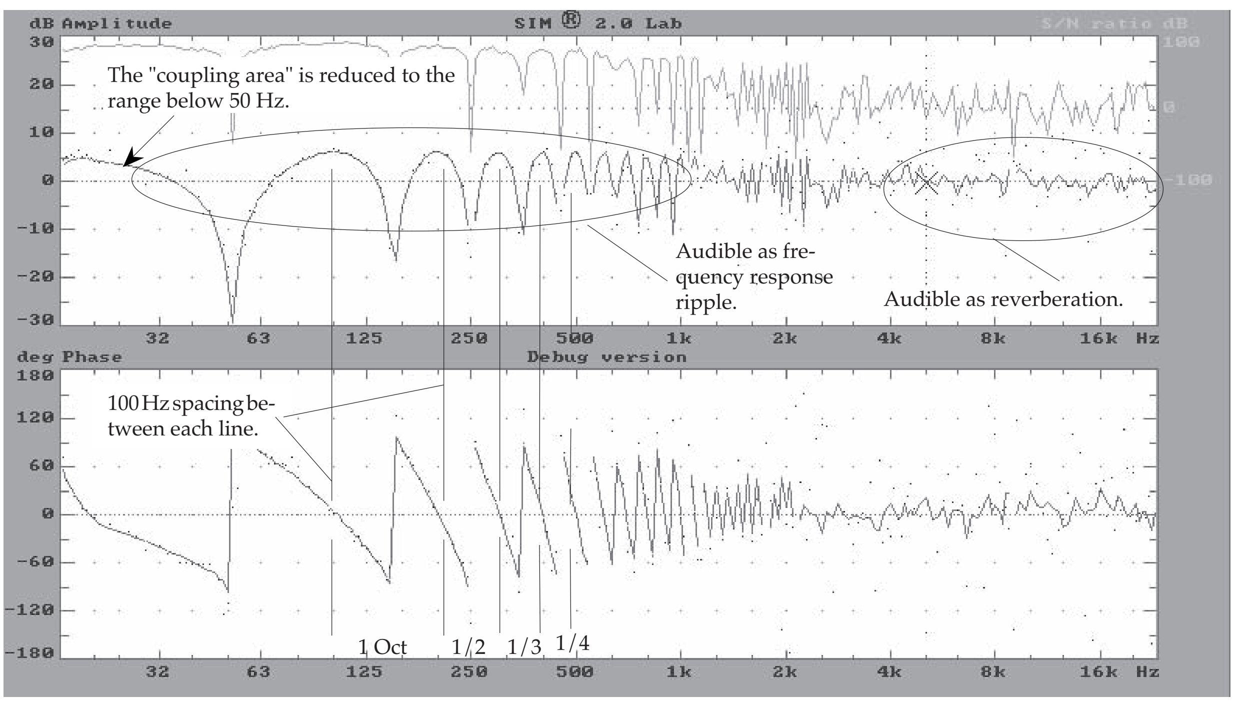


Fig 2.1g Comb filtering with 10 ms time offset between signals with 0 dB level offset.

2.1.2 Comb Filter Level

The maximum addition of 6 dB occurs when two signals of equal level are combined. This is the most potentially positive aspects of comb filtering. On the other hand, the cancellation effect is extremely deep. The percentage bandwidth of the peaks and dips are not the same, with the peaks being broader than the dips. While the size of the peaks may be reduced by equalization, the dips are often too deep and narrow to be practically equalized.

As the level offset increases, the magnitude of the peaks and dips is reduced and the filter slope decreases. This allows the system to be more adaptable to equalization. This is shown in Figs 2.1h–2.1j.

Recall from the previous page the progressive narrowing of the percentage bandwidth of each succeeding peak, from an octave to hundredths of an octave. For equalization to be effective, the equalizer must have adjustable bandwidth and center frequency. It is often said that comb filtering can't be fixed with an equalizer. But since comb filtering is creating virtually all of the frequency response problems (if we have a linear speaker to start with) what else are you doing with that equalizer? Once all of the other means of system alignment have been exhausted (such as repositioning, delay setting and level adjustment) you will have minimized the time offsets and maximized the level offsets. The response ripple that is left will be dealt with by equalization. It is true, however, that you can't fix comb filtering with fixed center frequency and bandwidth devices such as a 1/3 octave graphic equalizer. These can only have, at best, one frequency range where its bandwidth matches the system response. If you are lucky it might fall on one of the ISO standard center frequencies. The phase response of a graphic equalizer is often blamed for its poor end result. While this may be true in some cases, it is more often the graphic equalizer's inability to create the complementary amplitude and phase response of the system to be equalized. For this reason, parametric equalizers such as the Meyer Sound CP-10 have been employed exclusively by users of high-resolution alignment systems such as SIM System II.

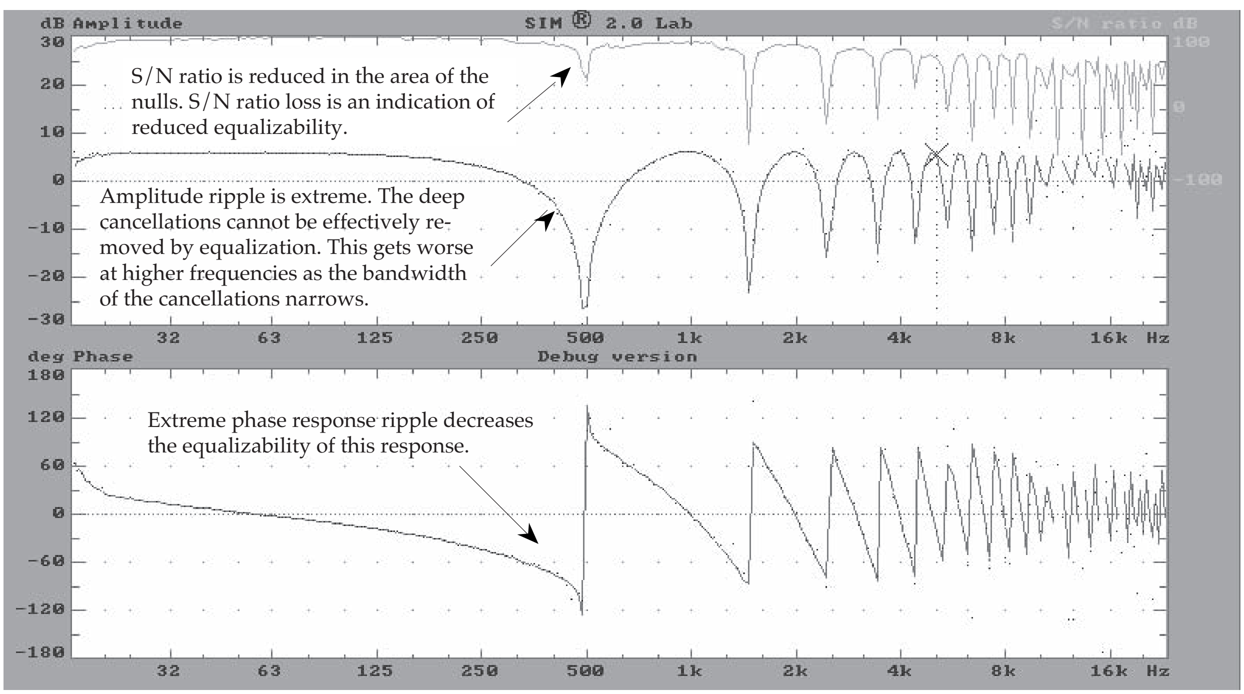


Fig 2.1h Comb filtering with 1 ms time offset between signals with 0 dB level offset.

2.1.2 Comb Filter Level

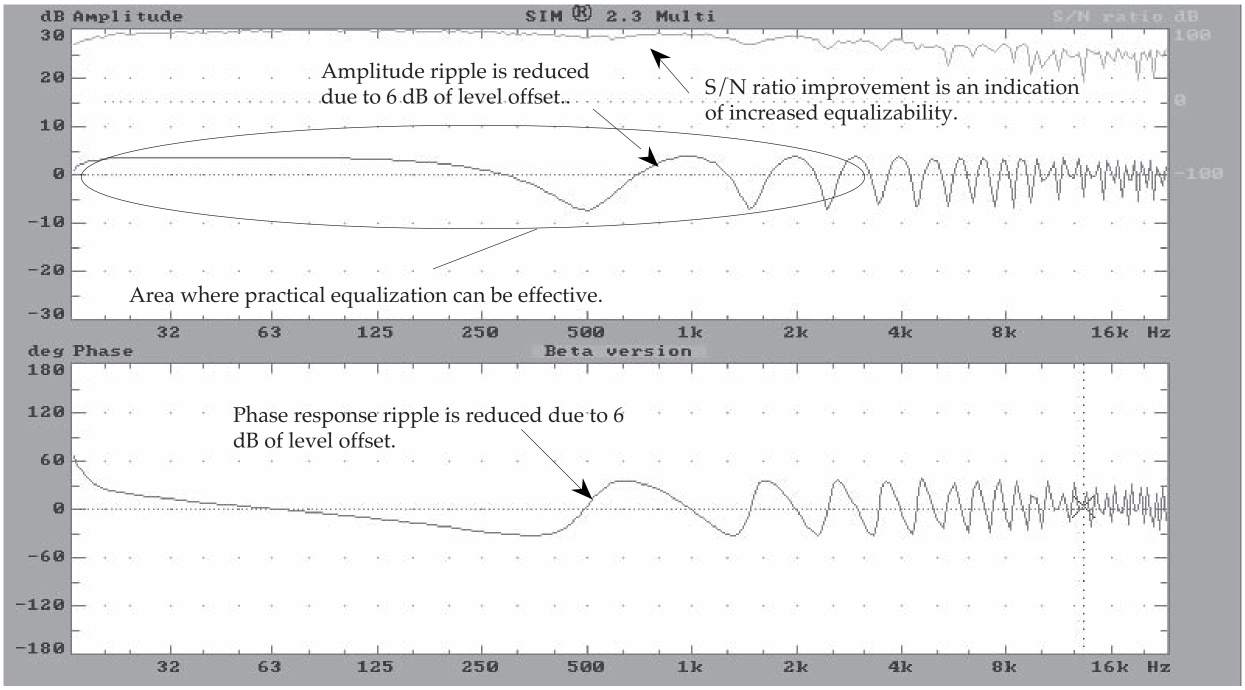


Fig 2.1i Comb filtering with 1 ms time offset between signals with 6 dB level offset.

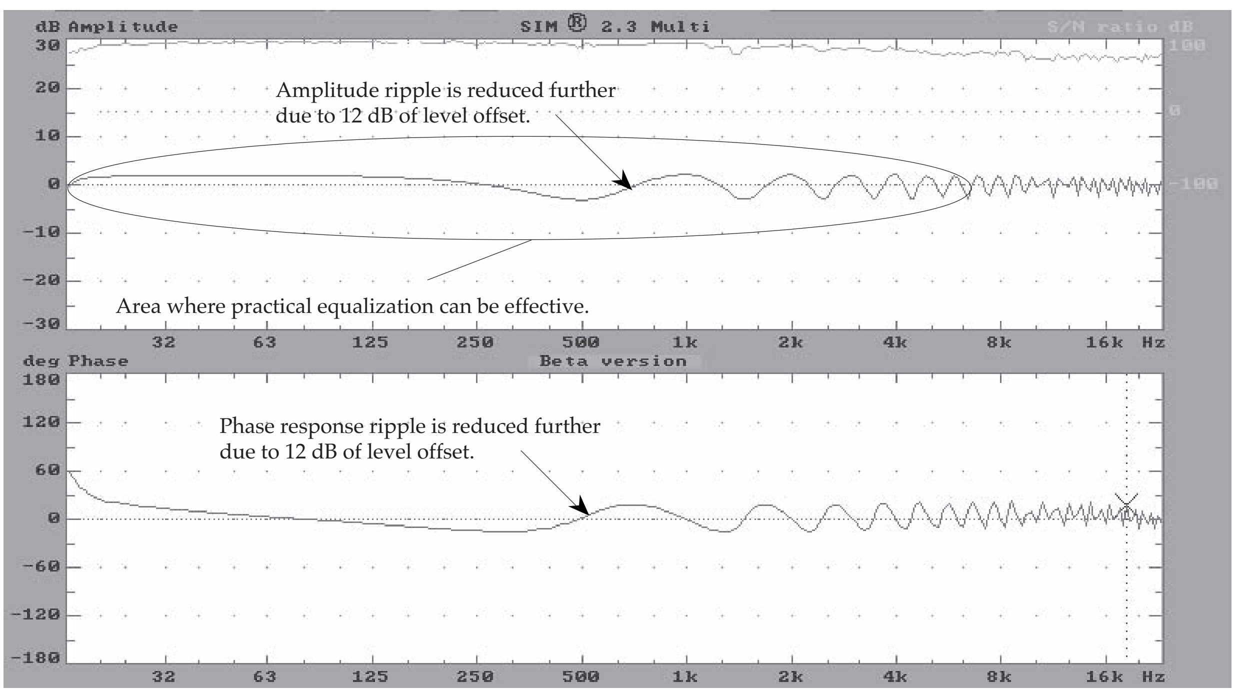
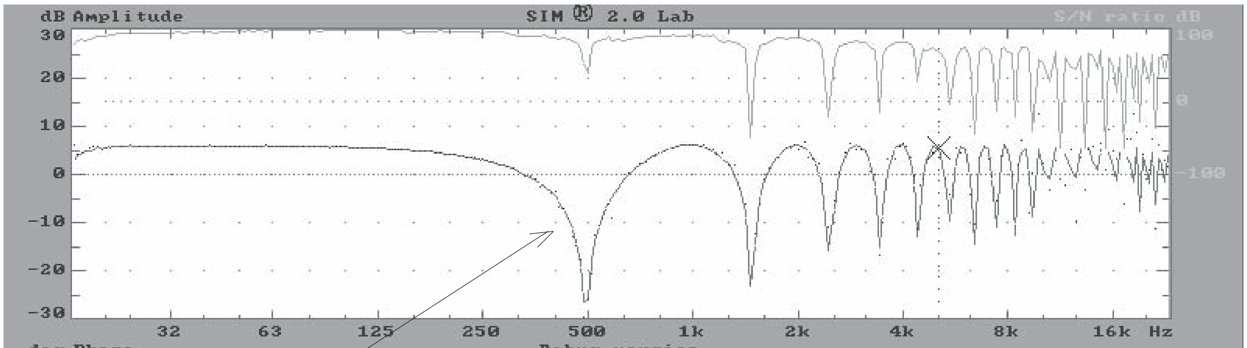


Fig 2.1j Comb filtering with 1 ms time offset between signals with 12 dB level offset.

2.1.3 Identifying Comb Filters



1st null

Spacing between nulls

First Null	Time Offset
(Hz)	(ms)
20	25.000
25	20.000
31.5	15.873
40	12.500
50	10.000
63	7.937
80	6.250
100	5.000
125	4.000
160	3.125
200	2.500
250	2.000
315	1.587
400	1.250
500	1.000
630	0.794
800	0.625
1000	0.500
1250	0.400
1600	0.313
2000	0.250
2500	0.200
3150	0.159
4000	0.125
5000	0.100
6300	0.079
8000	0.063
10000	0.050
12500	0.040
16000	0.031
20000	0.025

Spacing between peaks or nulls	Time Offset
(Hz)	(ms)
20	50.000
25	40.000
31.5	31.746
40	25.000
50	20.000
63	15.873
80	12.500
100	10.000
125	8.000
160	6.250
200	5.000
250	4.000
315	3.175
400	2.500
500	2.000
630	1.587
800	1.250
1000	1.000
1250	0.800
1600	0.625
2000	0.500
2500	0.400
3150	0.317
4000	0.250
5000	0.200
6300	0.159
8000	0.125
10000	0.100
12500	0.080
16000	0.063
20000	0.050

Table 2.1k Comb filter frequency versus delay time offset.

Measuring the frequency response of a system will show a complex series of peaks and dips. Before equalization is applied, it is helpful to identify the sources of the interactions that caused the peaks and dips. This puts us in a better position to take the most effective steps toward system optimization. The above chart can be used as a guide for identifying the causes of peaks and dips in the system response. If the center frequency of a peak or dip is known, the corresponding time offset can be found in the column to the right of the frequency (e.g., a null appears in the system response at 500 Hz). If this null is caused by comb filtering, the shortest possible time offset between the sound sources is 1 ms. However, if this is not the first null, then it may be the result of a longer offset. It could be the second null from an offset of 3x the minimum (in this case (3 ms) or the third null of 5x (5 ms), the fourth null of 7x (7 ms), etc.

The spacing between the nulls can solidify the time offset of the reflection. If the frequency spacing to the next null is 1000 Hz (or between peaks—it doesn't matter), the reflection is 1 ms. If the spacing is shorter, it is one of the longer intervals. The monitoring position could then be examined for paths that coincide with these time offsets and appropriate corrective action could be taken.

2.1.3 Identifying Comb Filters

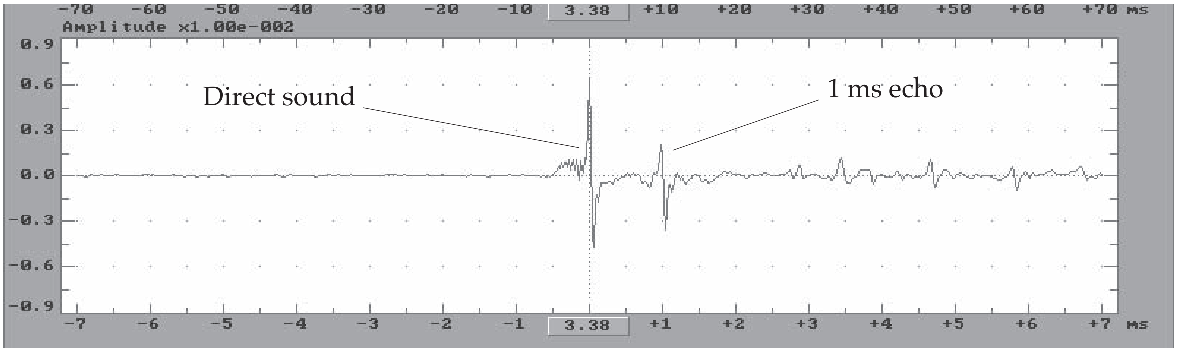


Fig 2.1l Identifying delay offset to find frequency response ripple.

Time Offset (ms)	Freq spacing (Hz)		2nd peak (Hz)	3rd peak (Hz)		1st null (Hz)	2nd null (Hz)
0.1	10,000.00		10,000.00	20,000.00		5,000.00	15,000.00
0.2	5,000.00		5,000.00	10,000.00		2,500.00	7,500.00
0.3	3,333.33		3,333.33	6,666.67		1,666.67	5,000.00
0.4	2,500.00		2,500.00	5,000.00		1,250.00	3,750.00
0.5	2,000.00		2,000.00	4,000.00		1,000.00	3,000.00
0.6	1,666.67		1,666.67	3,333.33		833.33	2,500.00
0.7	1,428.57		1,428.57	2,857.14		714.29	2,142.86
0.8	1,250.00		1,250.00	2,500.00		625.00	1,875.00
0.9	1,111.11		1,111.11	2,222.22		555.56	1,666.67
1	1,000.00		1,000.00	2,000.00		500.00	1,500.00
2	500.00		500.00	1,000.00		250.00	750.00
3	333.33		333.33	666.67		166.67	500.00
4	250.00		250.00	500.00		125.00	375.00
5	200.00		200.00	400.00		100.00	300.00
6	166.67		166.67	333.33		83.33	250.00
7	142.86		142.86	285.71		71.43	214.29
8	125.00		125.00	250.00		62.50	187.50
9	111.11		111.11	222.22		55.56	166.67
10	100.00		100.00	200.00		50.00	150.00
15	66.67		66.67	133.33		33.33	100.00
20	50.00		50.00	100.00		25.00	75.00
25	40.00		40.00	80.00		20.00	60.00
30	33.33		33.33	66.67		16.67	50.00
35	28.57		28.57	57.14		14.29	42.86
40	25.00		25.00	50.00		12.50	37.50
45	22.22		22.22	44.44		11.11	33.33
50	20.00		20.00	40.00		10.00	30.00
60	16.67		16.67	33.33		8.33	25.00
70	14.29		14.29	28.57		7.14	21.43
80	12.50		12.50	25.00		6.25	18.75
90	11.11		11.11	22.22		5.56	16.67
100	10.00		10.00	20.00		5.00	15.00

Table 2.1m Delay time offset versus comb filter frequency.

It is also possible to work in the reverse direction. The chart can be used to identify how the frequency response will be affected when the time offset between sources is known. The corresponding frequency multiplier and the frequency of the first null can be found in the columns to the right of the time offset (e.g., a time offset of 1 ms is detected between two sound sources at a monitoring position). The comb frequency spacing will be 1000 Hz with the first null at 500 Hz. There will be nulls every 1 kHz above that (1500 Hz, 2500 Hz, etc.). There will be a peak below 500 Hz and additional ones starting at 1 kHz and each 1 kHz above that (2 kHz, 3 kHz, etc.). The first null will be at 500 Hz and each 1 kHz above that.

2.2.1 Introduction

The interaction between multiple speaker systems will cause comb filtering. For this reason it is important that systems be designed for a minimum of overlap between speaker subsystems.

A typical speaker interaction is shown in Fig 2.2a. The down lobe from the upper speaker arrives onto the floor seating area. Unfortunately it is 1 ms late, and, as we discussed in the previous section, comb filtering will result in the midrange and highs.

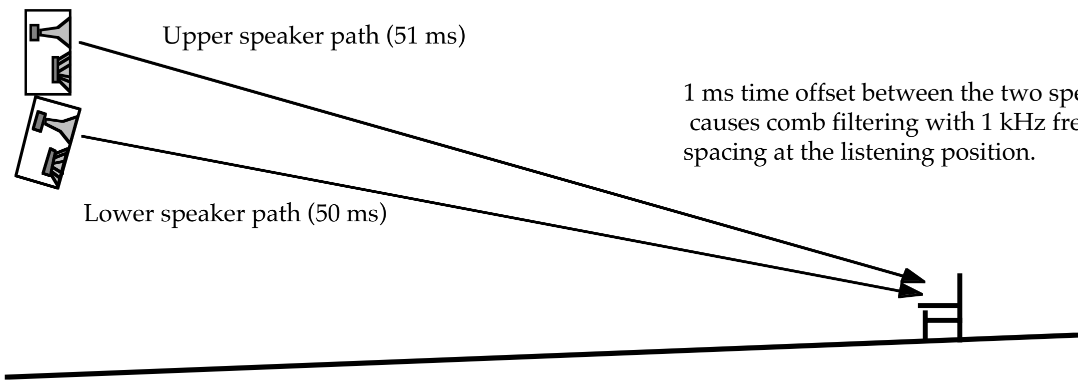


Fig 2.2a Elevation view of speaker interaction between an upper main and downfill system.

Multiple speaker arrivals can be modeled as shown in Fig 2.2b. The amount of axial attenuation will depend on the angle and coverage pattern of the contaminating speaker. The propagation time offset is a result of different arrival times between the speakers. This could be a positive or negative number depending on which speaker arrives first. The relative propagation loss and high-frequency air loss are due to the difference in path length between the speakers.

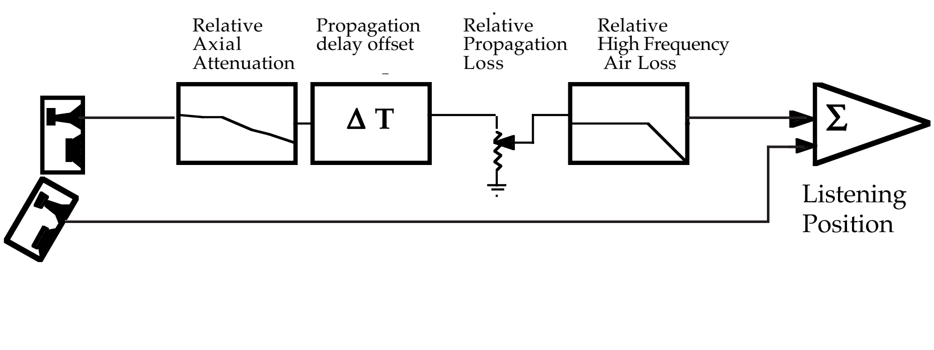


Fig 2.2b Model of speaker interaction between an upper main and downfill system.

2.2.1 Introduction

The following SIM measurements show the interaction between an upper and lower speaker system.

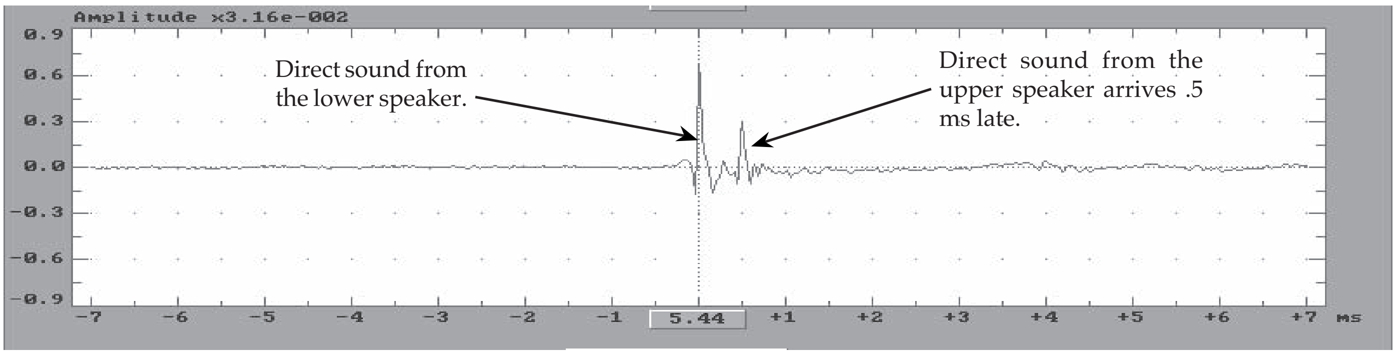


Fig 2.2c Delayfinder display of the arrivals of the two speakers.

Note that the crest factor of the impulse from the upper speaker is less than the lower. This is due to HF axial attenuation in the response from the upper system.

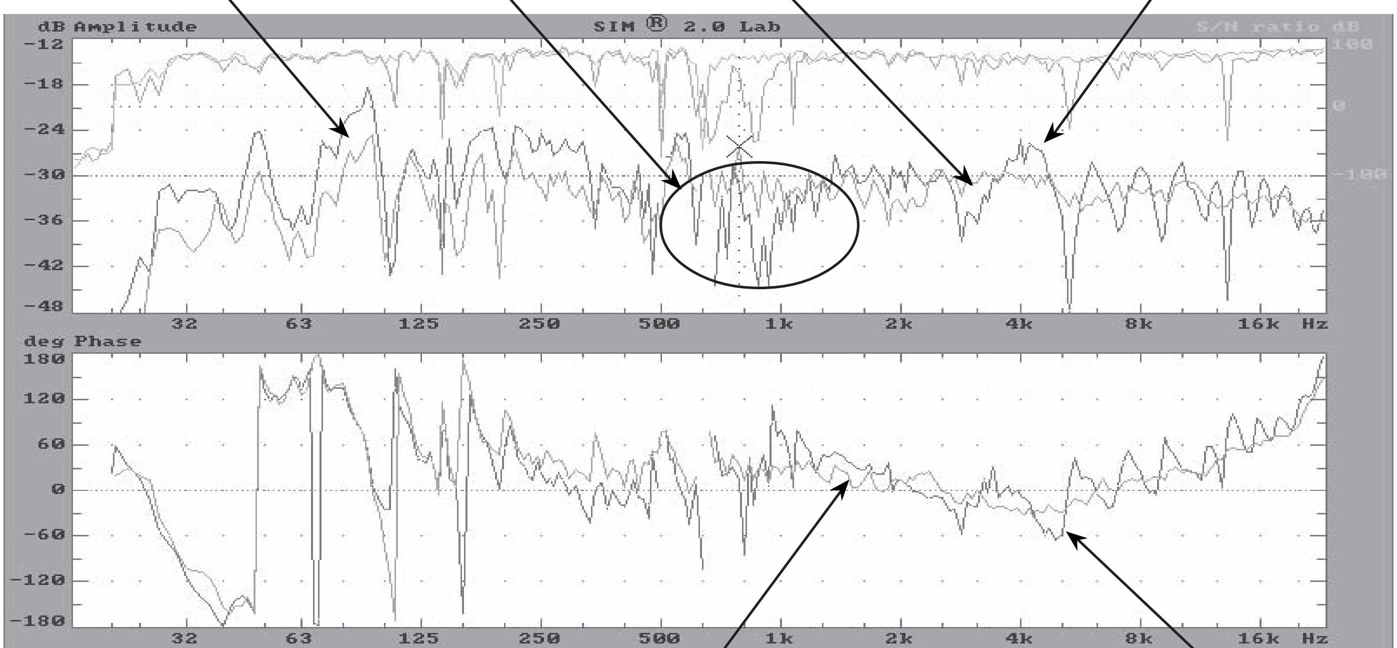


Fig 2.2d Comparison frequency response of single speaker to combination.

2.2.2 Factors Affecting Interaction

The fundamental mechanism behind speaker arrays is shown in Fig 2.2e. The behavior of speaker arrays can be simplified into the following categories.

Interaction Types

Coupling: Maximum addition and minimum cancellation.

Combining: Moderate addition and a lesser degree of cancellation.

Combing: Deep cancellation and addition.

Echo: Discrete sources are perceived.

Reverberation: Decay character perceived.

Isolation: Little or no audible effect.

Which of these occurs on any given design will depend upon two key factors.

Relative time offset: The difference in arrival time of the sound sources. This function has a frequency component, phase. For example, 1 ms is only 18° of phase shift at 50 Hz, 180° at 500 Hz.

Relative level offset: The difference in level of the sound sources. This is also frequency dependent, since the responses of the two systems may not be matched.

Coupling: Occurs when the time offset and level offset both approach zero. The signals arrive "in phase" and can add a maximum of 6 dB. This is easiest to achieve in low frequency arrays where the periods are long. Therefore the physical offset of multiple devices does not become too large and the wavefronts remain in phase.

Combining: Occurs when the time offset is low and the moderate level is offset. To achieve this the devices must be in close proximity (hence the low time offset), yet must have a method of obtaining some level offset. This can be best achieved by using directional speaker systems arrayed as a point source.

Combing: Occurs when the time offset is large but the level offset is low. This occurs when speakers are arrayed with redundant coverage patterns, such as parallel arrays. While this may give substantial addition, it is highly position dependent and causes large variations in frequency response and low intelligibility. This should be avoided if at all possible.

Echo: Occurs when the time offset is large and the isolation low, so that the systems sound like discrete sources. This also causes large variations in frequency response and low intelligibility and should be avoided if at all possible.

Reverberation: Occurs when the time offset is large but the isolation is high enough that the interaction sounds like the normal decay character of a room. If kept to a minimum this will not dramatically affect the system intelligibility. This is far preferable to combing or echo.

Isolation: Occurs when the level offset is large enough so that the second speaker has little or no audible effect on the primary speaker's response. As the time offset increases, larger amounts of level offset will be required to achieve isolation.

2.2.2 Factors Affecting Interaction

The relationship of time and level offset is shown graphically in Fig 2.2e. As level offset increases the amount of addition decreases. Notice that the preferred areas of coupling, combining and isolation are all towards the bottom of the graph, where time offsets are low. As you move from left to right there will be progressively less power addition. As the time offset increases (higher vertical positions) the coupling and combining give way to combing, and the isolation gives way to echo and reverb. As you move upward, the comb filtering moves progressively down through greater proportions of the audible range. Large time offsets with no level offset will cause the most destructive combing.

The key to speaker array design is:

- If the level offset is low, the time offset must be as low as possible. This will create coupling.
- As time offset increases, the level offset should also increase. This will create combination.
- If the time offset is large, the level offset should also be large. This will minimize combing and increase isolation.

Array Design Trade-Offs

Array design is a trade-off between the following parameters:

Coverage: As overlap increases, coverage narrows and vice-versa.

On-Axis SPL: As overlap increases, on-axis SPL increases significantly. As overlap decreases, on-axis SPL remains largely unchanged.

Level Distribution: As overlap increases, level distribution becomes uneven, most notably in the form of hot spots in the center area. As overlap decreases, level distribution becomes smoother.

Frequency Response Distribution: As overlap increases, frequency response distribution becomes uneven. As overlap decreases, frequency response distribution becomes smoother.

Equalizability: Virtually any array is equalizable at a single point. But if we can assume that the intended goal is to provide an equalization curve that is suitable for a wide part of the coverage area, arrays with even distribution patterns will respond best.

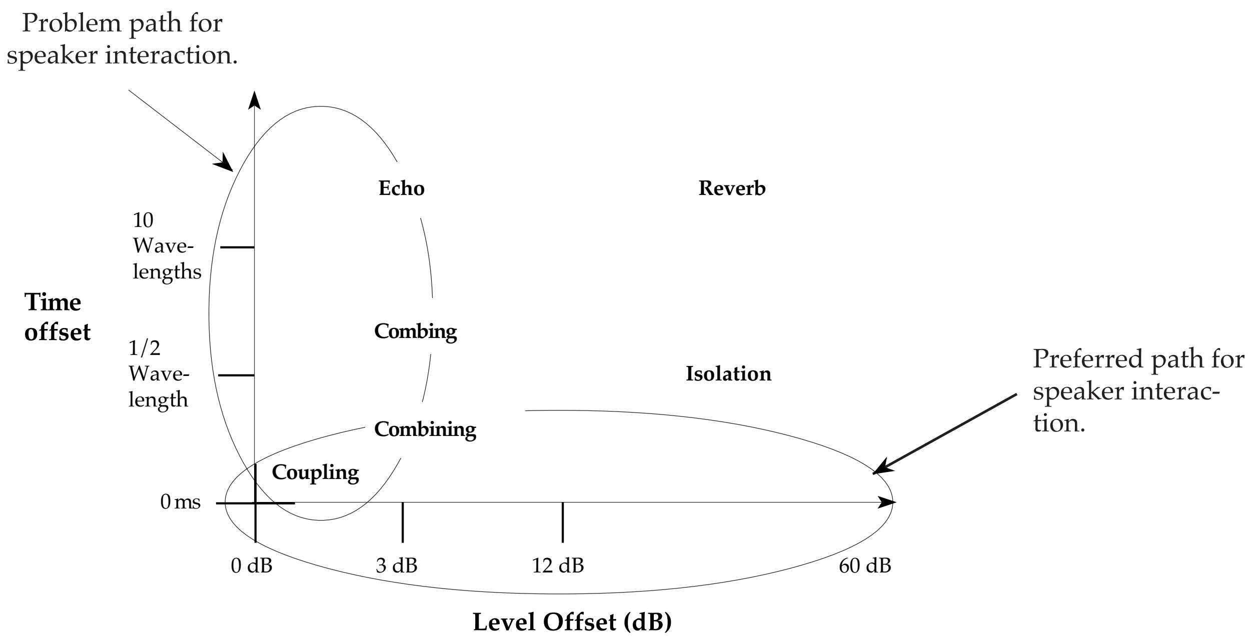
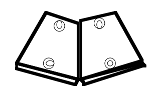


Fig 2.2e Graphic representation of speaker interaction.

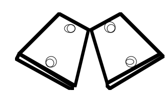
2.2.3 Array Configurations

There are seven basic types of speaker arrays, each with its own strength and weakness. The arrays can be made of adjacent or distributed elements. When the speakers are spread apart they are referred to as split speaker arrays. Split speaker arrays perform well only in a short depth of field.

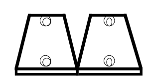
- **Point-source narrow:** Two speakers are arrayed in an arc. The individual speaker patterns are wider than the splay angle. The combined pattern narrows due to summation at the center.



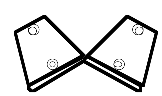
- **Point-source wide:** Two speakers are arrayed in an arc. The individual speaker patterns approximate the splay angle. The combined pattern widens with minimal summation at the center.



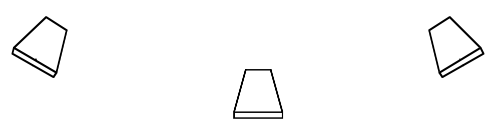
- **Parallel:** Two speakers are arrayed in a parallel plane. The patterns overlap and create an inconsistent response. *Not recommended.*



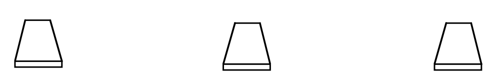
- **Crossfire:** Two speakers cross directly in front of the horn. This has significantly higher interference problems than the point-source approach and has no advantages over it. *Not recommended.*



- **Split point-source:** The speakers are arrayed in an extended arc. This type of array is relatively consistent, but lacks LF coupling.



- **Split-parallel:** When speakers are placed parallel over an extended line they will resemble a series of distinct point sources in the HF range and a single elongated source in the LF region.



- **Split crossfire (point destination):** The inverse of a point-source. The focal point is the central destination point of the speakers. This type of array is most useful when covering a central area from two sides. This array type has very inconsistent sound at the center.

2.2.3 Array Configurations

Each array configuration has unique tendencies toward coupling, combing and combining as shown in Figs 2.2f and 2.2g. The wide point-source array combines the best due to the minimal overlap zone and time offset. The narrow point-source has more overlap, hence more coupling and combing. The crossfire array has still more overlap. The parallel array is almost entirely overlapped showing only tendencies toward coupling and combing.

The split point-source array, having the least overlap, combines the best of the split arrays. The time offsets are larger than coupled point-source arrays due to the distance between cabinets. The split-parallel array will comb badly. The overlap will get very large as you get farther away, with very large time offsets. Point-destination arrays are useful to reach central areas from side locations. However, these have the highest tendency toward combing due to the high overlap and rapid time offset changes.

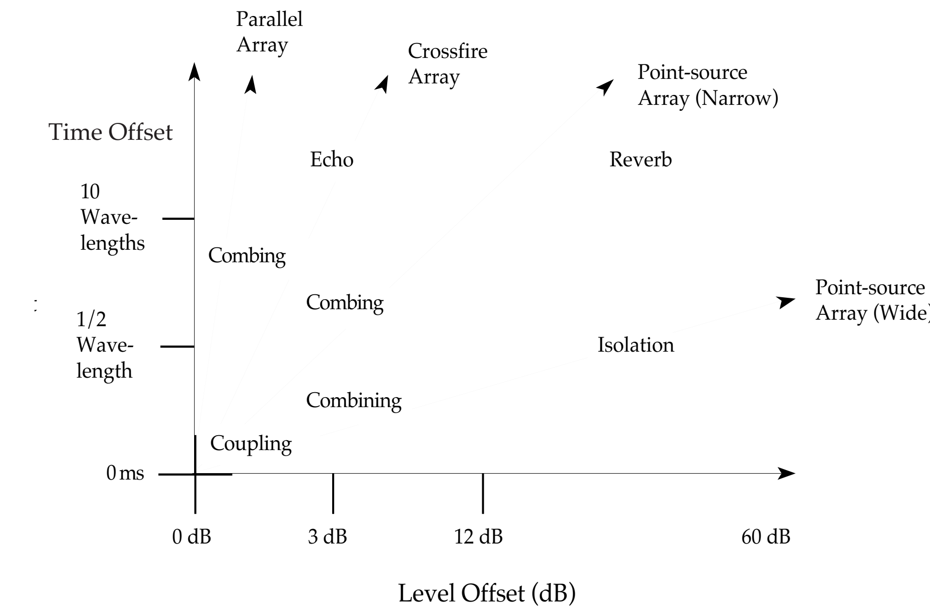


Fig 2.2f Tendencies of coupled arrays.

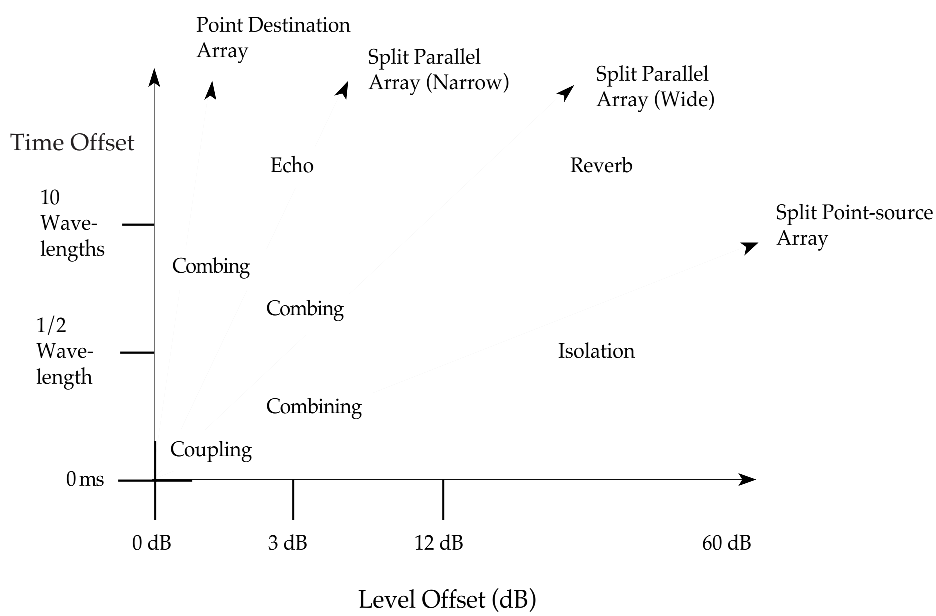


Fig 2.2g Tendencies of split speaker arrays.

2.2.3 Array Configurations

The nature of the interaction between two speakers varies with different array types. In the following pages (Figs 2.2g through 2.2aa) is a comparative study of the interaction at each of nine axial positions of an 8°speaker.

Reading the Figure

The positions represent the 10° points (from –40 to +40) on an arc at a distance of twenty-five feet from the speaker. A line is drawn from the second speaker representing its arrival into the first speaker's coverage area. The figures are shaded to represent the extent of the interference, with progressively darker shades representing deeper interference.

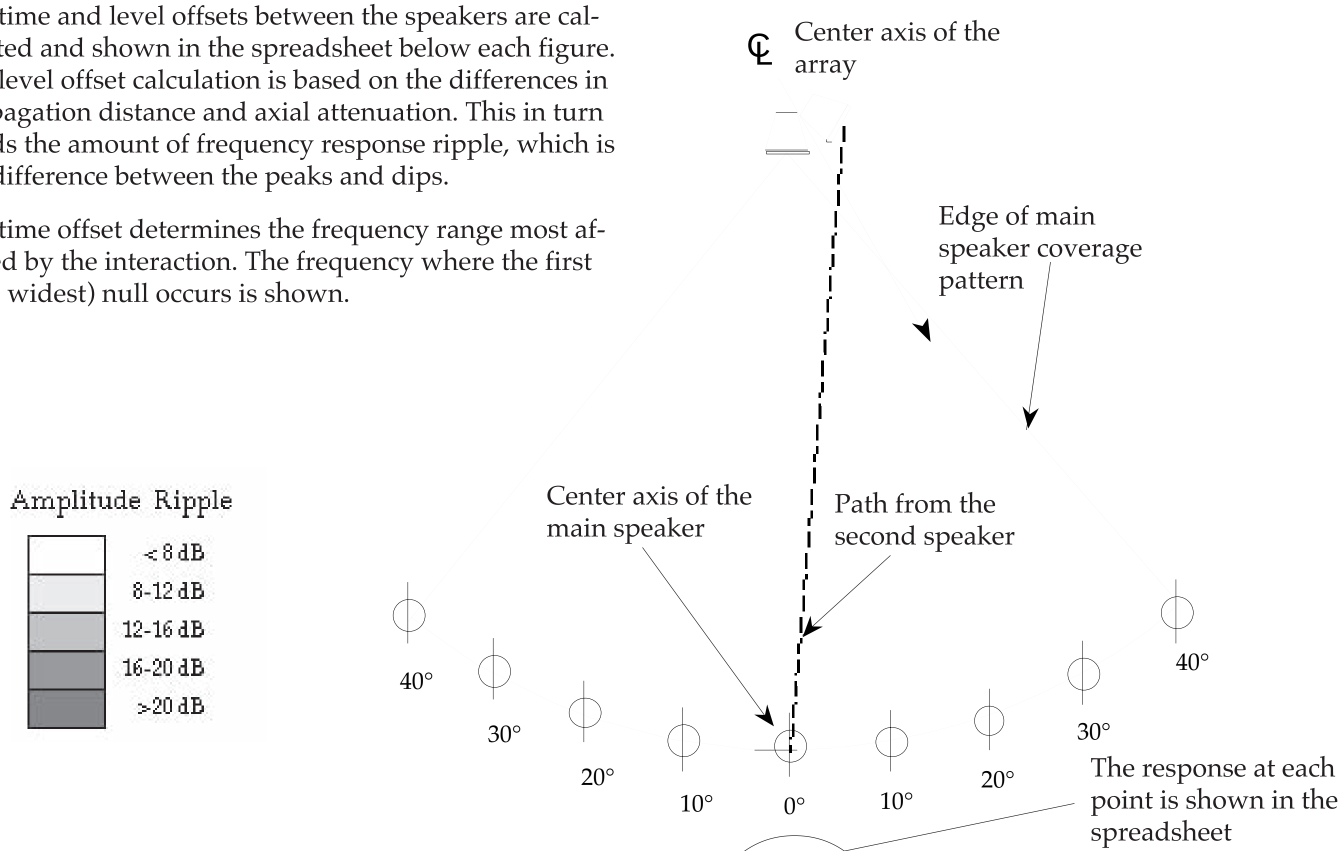
Reading the Spreadsheet

The time and level offsets between the speakers are calculated and shown in the spreadsheet below each figure. The level offset calculation is based on the differences in propagation distance and axial attenuation. This in turn yields the amount of frequency response ripple, which is the difference between the peaks and dips.

The time offset determines the frequency range most affected by the interaction. The frequency where the first (and widest) null occurs is shown.

Reading the SIM® Plots

Actual measurements were made of some of the arrays at positions near the center (–10° to +20°) and are shown in the accompanying page. The measurements were not made in an anechoic chamber and contain some room reflections. Therefore only the high-frequency range is shown where the room interaction is minimal, but the speaker interaction is easily visible.



Point Source Wide

	40°	30°	20°	10°	0°	10°	20°	30°	40°
Time offset	2.2 ms	2.1 ms	1.7 ms	1.5 ms	1.1 ms	0.7 ms	0.3 ms	0.1 ms	0.25 ms
1st null	227 Hz	238 Hz	294 Hz	333 Hz	455 Hz	714 Hz	1667 Hz	5000 Hz	2000 Hz
Level offset	12.7 dB	13.7 dB	14.6 dB	13.0 dB	11.4 dB	9.2 dB	7.1 dB	2.0 dB	5.1 dB
Ripple	4.0 dB	3.5 dB	3.5 dB	4.0 dB	4.5 dB	6.0 dB	8.0 dB	19.0 dB	10.0 dB

Fig 2.2h How to read the series of array interaction comparison figures

2.2.3 Array Configurations

Single –10°

Single 0°

Single 10°

Single 20°

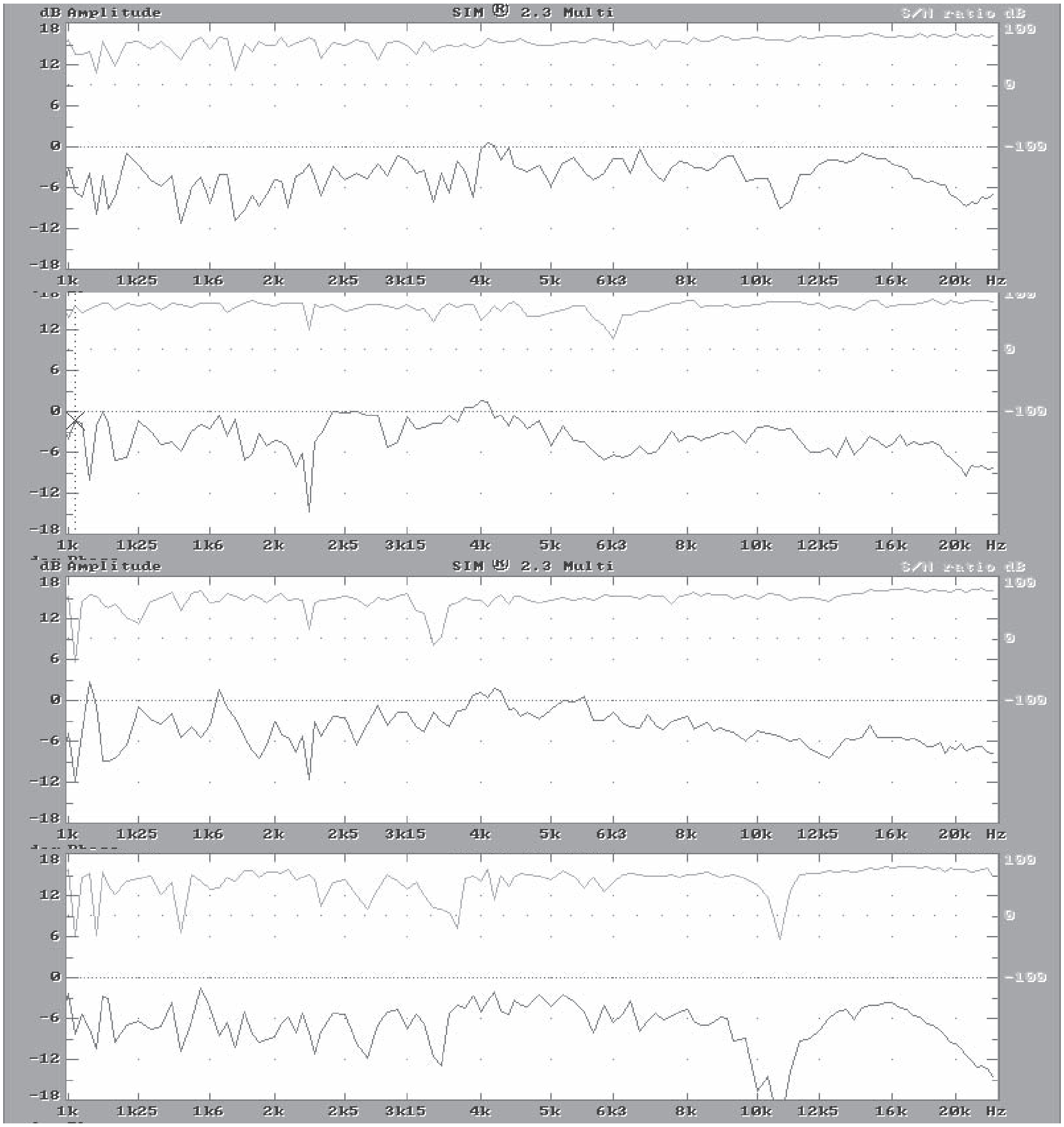


Fig 2.2i Reference for the upcoming SIM® plots of array interaction.

The four plots show the response of a single speaker (no multiple speaker interaction) measured at four points. This can be used for comparison with the plots showing the interaction of the point-source, crossfire and parallel arrays. Because the measurements were not made in an anechoic chamber, they are not free of ripple. Therefore, only the HF region is shown. Most of the ripple shown can be attributed to the room acoustics.

2.2.5 Point-Source Arrays (Narrow)

Narrow coverage arrays can be constructed by placing the cabinets directly adjacent to each other in an arc. Such arrays tend to increase the on-axis power but will have less of a widening effect on coverage than might be expected over that of a single unit, and may actually narrow it. These systems are highly interactive because the modules are arrayed at much tighter angles than the individual unit coverage angles. The on-axis point of the array contains the most overlap, causing a substantial addition in on-axis pressure, with less overlap and addition as you move to the edges. Since coverage angle is specified relative to the on-axis pressure, an addition there may cause the angle to decrease even with more cabinets.

The on-axis buildup can be reduced by the technique of amplitude tapering (see Section 3.6.3) which will widen the array's coverage.

Coverage: Center buildup causes the area between the –6 dB points to narrow.

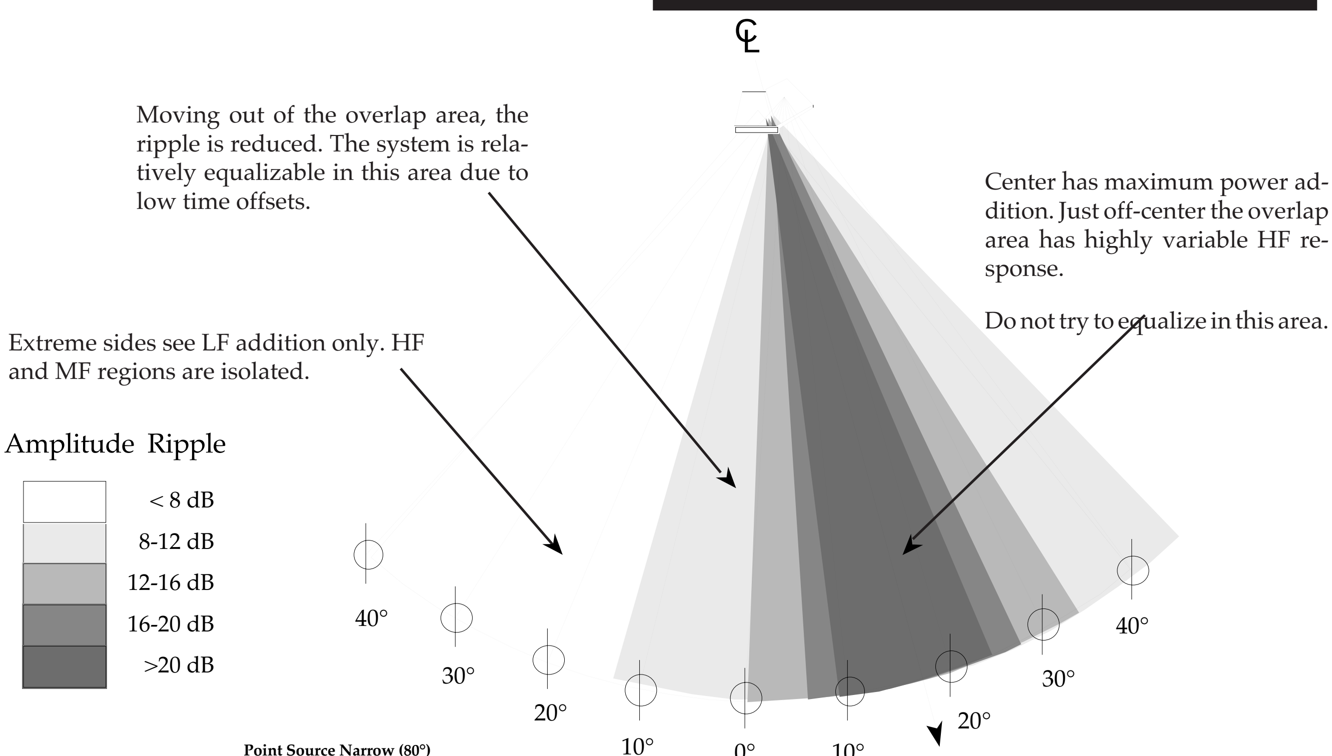
On-axis SPL: Maximum addition.

Level distribution: Large addition in center area. Less on the sides. Good LF and MF coupling.

Frequency response distribution: A large overlap area creates a wide area with a deep ripple around the center. Smoother on the sides.

Equalizability: Responds well to EQ except in the center overlap area.

Where to use: Long-throw applications or when desired coverage angle is less than that of a single enclosure.



Point Source Narrow (80°)

	40°	30°	20°	10°	0°	10°	20°	30°	40°	Mean
Time offset	1.1 ms	1 ms	0.8 ms	0.65 ms	0.45 ms	0.1 ms	0.1 ms	0.3 ms	0.5 ms	0.6 ms
1st null	455 Hz	500 Hz	625 Hz	769 Hz	1111 Hz	5000 Hz	5000 Hz	1667 Hz	1000 Hz	1791.8 Hz
Level offset	6.4 dB	8.3 dB	8.8 dB	6.7 dB	4.2 dB	1.0 dB	0.0 dB	2.1 dB	6.2 dB	4.9 dB
Ripple	9.0 dB	6.5 dB	6.0 dB	8.5 dB	11.5 dB	24.0 dB	30.0 dB	19.0 dB	9.0 dB	13.7 dB

Fig 2.2j Narrow point-source array interaction.

2.2.5 Point-source Arrays (Narrow)

Point-Source
(Narrow)

–10°

0°

10°

20°

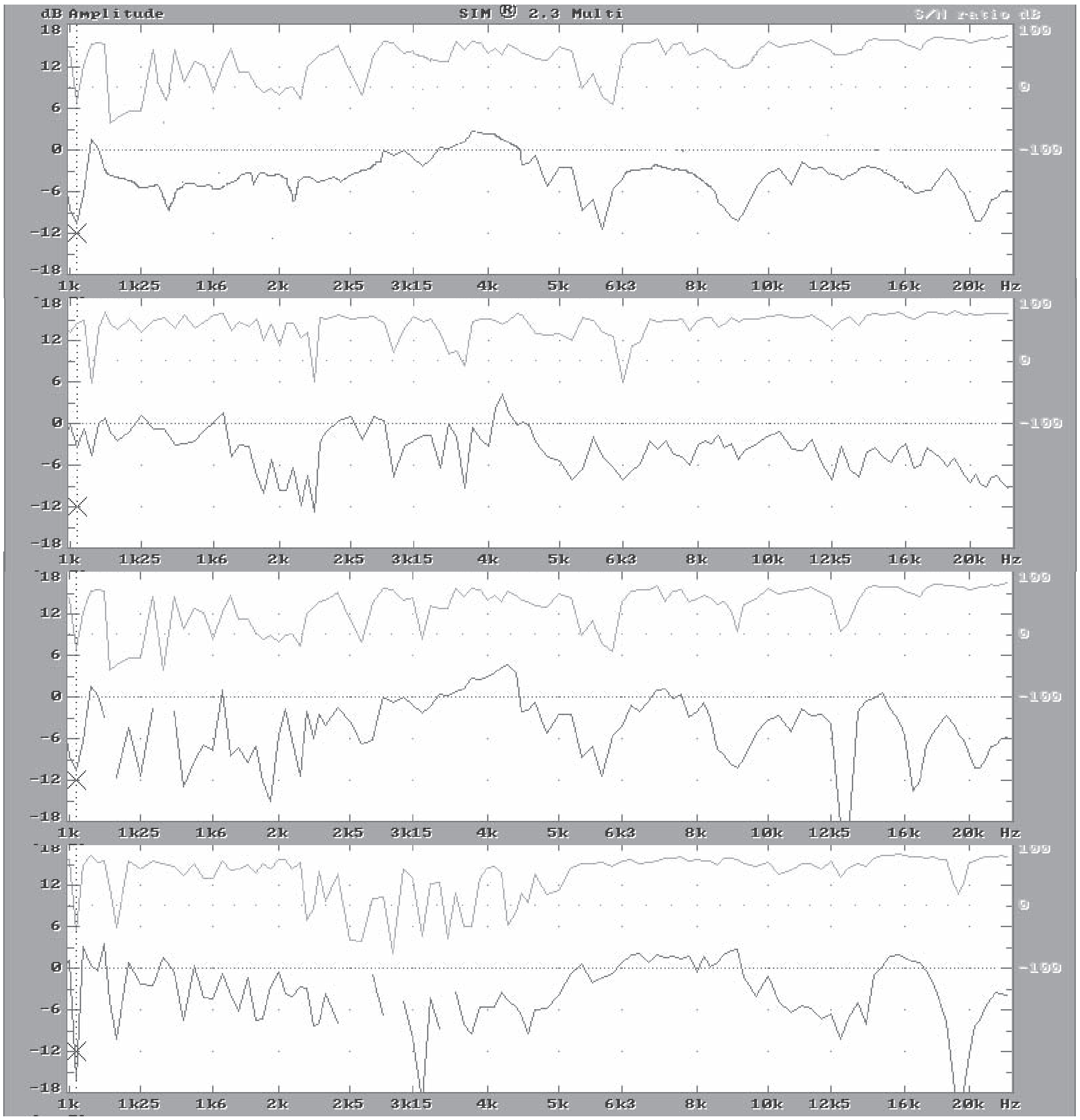


Fig 2.2k Narrow point-source array HF frequency response.

Two speakers arrayed as a narrow point source were measured using SIM System II. The response at four of the positions is shown above. Compare and contrast the above responses with those of the single speaker shown in Fig 2.2g and those that follow. There is substantially more ripple than a single speaker, but much less than the parallel or crossfire arrays.

2.2.6 Point-Source Arrays (Wide)

The narrow arrays discussed previously have substantial overlap areas as shown in Fig 2.2l.

There are two ways to reduce the overlap:

1) Use speakers with a tighter directional pattern as shown in Fig 2.2m.

2) Splay the speakers apart as shown in Fig 2.2n.

Speaker systems such as the MSL-5, MSL-6 and MSL-10A represent the first approach. Their pattern so closely matches the angle of the enclosure that they should only be arrayed adjacently. Because the coverage pattern closely matches the 30° enclosure dimension, there will be a loss in the center if the cabinets are splayed apart. Systems are extremely easy to design since each cabinet simply adds 30° to your horizontal coverage. However, because of the sharp cutoff characteristic of these systems, you must take care to have enough sections to cover the listening area fully. Otherwise, you may need to supplement the system with some additional sidefill speakers. Another important consideration is the fact that narrow arrays like this must be aimed much more precisely than the wide arrays, or the narrow arrays derived from the tight-packing of wide coverage speakers such as UPAs, MSL-2As and MSL-3s.

The second approach is achieved by splaying the system apart to various extents depending upon the coverage and enclosure angles.

Wide coverage arrays can be constructed by splaying the cabinet fronts outward, while leaving the rears touching. Wide arrays will increase the horizontal coverage but have minimal effect on increasing the on-axis power over that of a single unit.

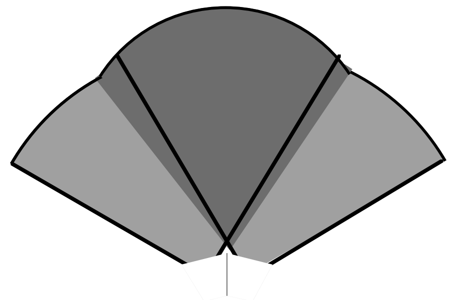


Fig. 2.2l

Narrow array (tight-pack).

On-axis addition causes polar response to elongate in the center.

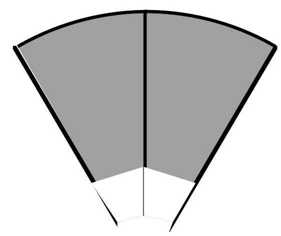


Fig. 2.2m

Wide array (tight pack).

Coverage pattern widens but on-axis power is not increased.

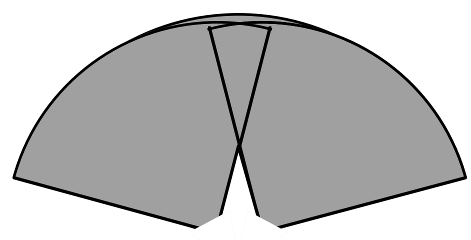


Fig. 2.2n

Wide-angle array (optimized).

Coverage pattern widens but on-axis power is only minimally increased.

2.2.6 Point-Source Arrays (Wide)

The following example shows the characteristics of a speaker with a much tighter pattern of 45°. (The previous examples were 80°.) Notice that the overlap area is a much smaller percentage of the coverage area. This allows us to achieve low ripple through a large area. Contrast the time and level offset number shown here with those of the narrow point-source array shown earlier. Notice that the time offsets remain small, but the level offset rapidly increases as you move out of the center area. This is the key to creating smooth frequency response distribution.

Coverage: Minimal center buildup. Therefore, the pattern widens.

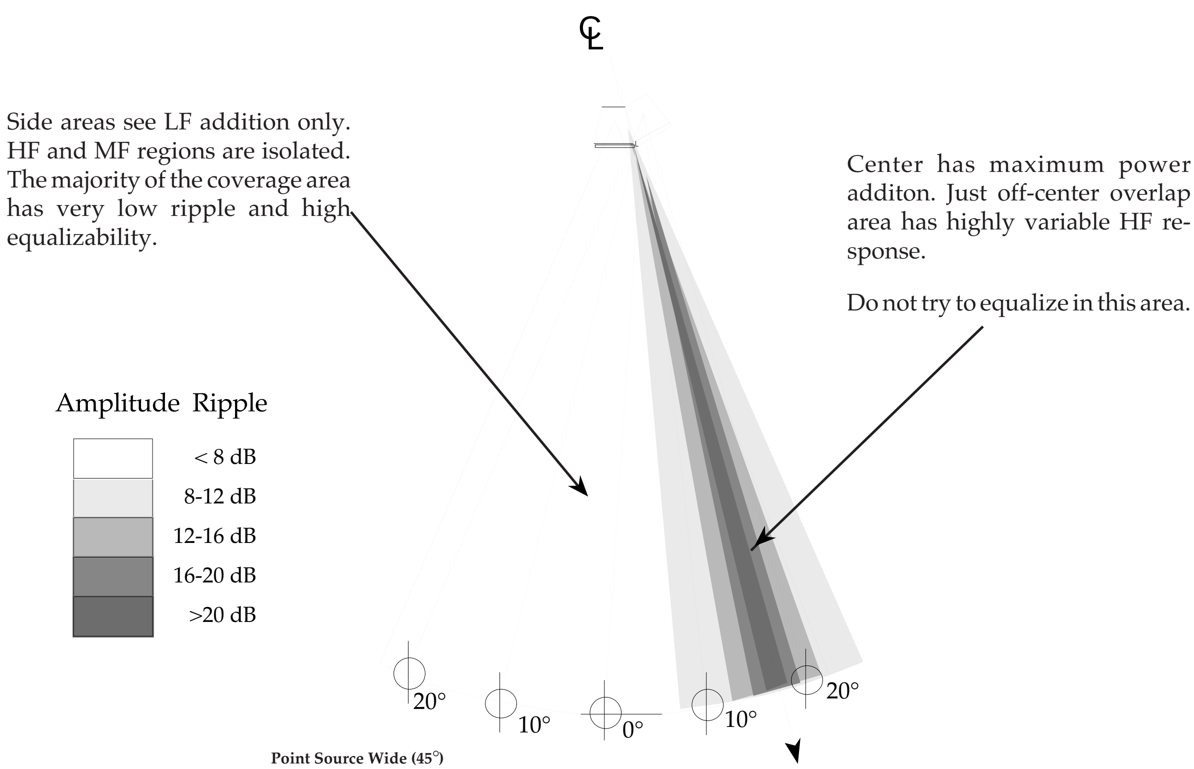
On-axis SPL: Minimal addition.

Level distribution: Smooth due to lack of overlap.

Frequency response distribution: The small overlap creates a narrow area with a deep ripple around the center. Very smooth except just off-center.

Equalizability: Responds very well to EQ except in the center overlap area.

Where to use: When the desired coverage angle is wider than that of a single enclosure



Point Source Wide (45°)

	20°	10°	0°	10°	20°	Mean
Time offset	0.8 ms	0.65 ms	0.45 ms	0.1 ms	0.1 ms	0.4 ms
1st null	625 Hz	769 Hz	1111 Hz	5000 Hz	5000 Hz	2501.1 Hz
Level offset	7.8 dB	9.2 dB	9.2 dB	4.5 dB	2.5 dB	6.6 dB
Ripple	7.0 dB	6.0 dB	6.0 dB	11.0 dB	16.0 dB	9.2 dB

Fig 2.2p Wide point-source array interaction.

2.2.6 Point-Source Arrays (Wide)

An alternative method of achieving a wide coverage point-source array is to splay the fronts of the cabinets apart. This reduces the overlap are in the center and spreads the energy out over a wider area.

Coverage: Minimal center buildup. Therefore, the pattern widens.

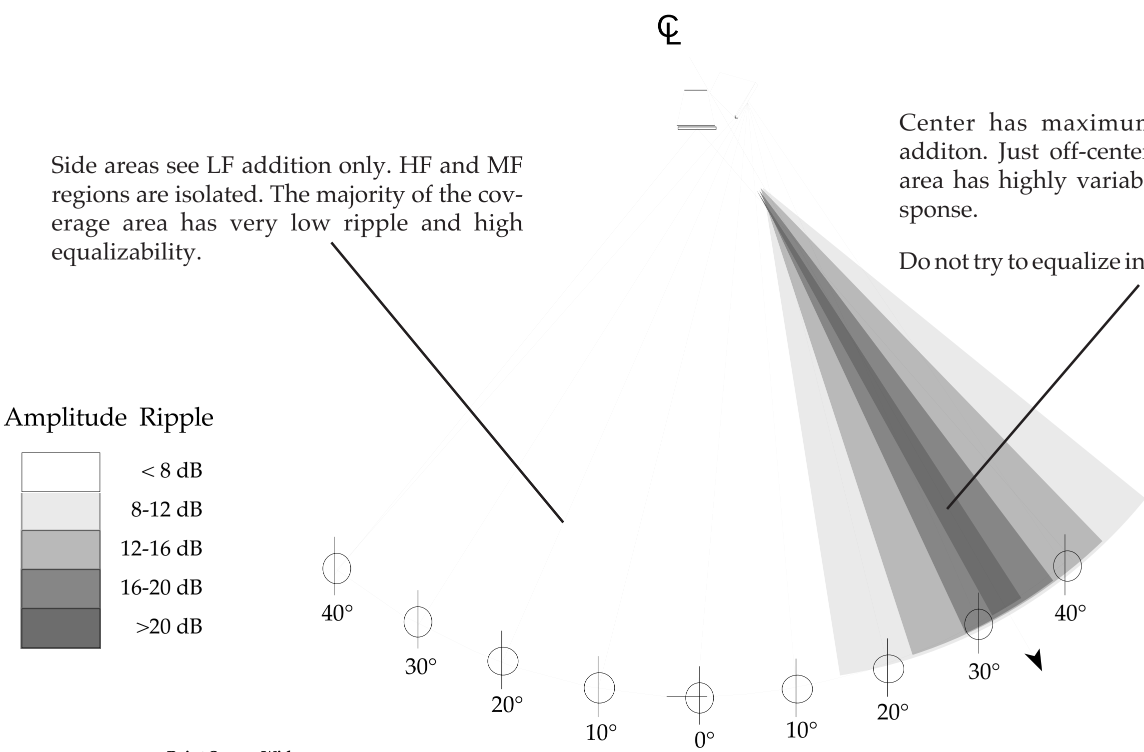
On-axis SPL: Minimal addition.

Level distribution: Smooth due to lack of overlap.

Frequency response distribution: The small overlap area creates a narrow area with a deep ripple around the center. Very smooth except just off-center.

Equalizability: Responds very well to EQ except in the center overlap area.

Where to use: When desired coverage angle is wider than that of a single enclosure



Point Source Wide

	40°	30°	20°	10°	0°	10°	20°	30°	40°	Mean
Time offset	2.2 ms	2.1 ms	1.7 ms	1.5 ms	1.1 ms	0.7 ms	0.3 ms	0.1 ms	0.25 ms	1.1 ms
1st null	227 Hz	238 Hz	294 Hz	333 Hz	455 Hz	714 Hz	1667 Hz	5000 Hz	2000 Hz	1214.3 Hz
Level offset	12.7 dB	13.7 dB	14.6 dB	13.0 dB	11.4 dB	9.2 dB	7.1 dB	2.0 dB	5.1 dB	9.9 dB
Ripple	4.0 dB	3.5 dB	3.5 dB	4.0 dB	4.5 dB	6.0 dB	8.0 dB	19.0 dB	10.0 dB	6.9 dB

Fig 2.2q Wide Point-Source Array Interaction.

2.2.6 Point-Source Arrays (Wide)

Point-Source (wide) –10°

0°

10°

20°

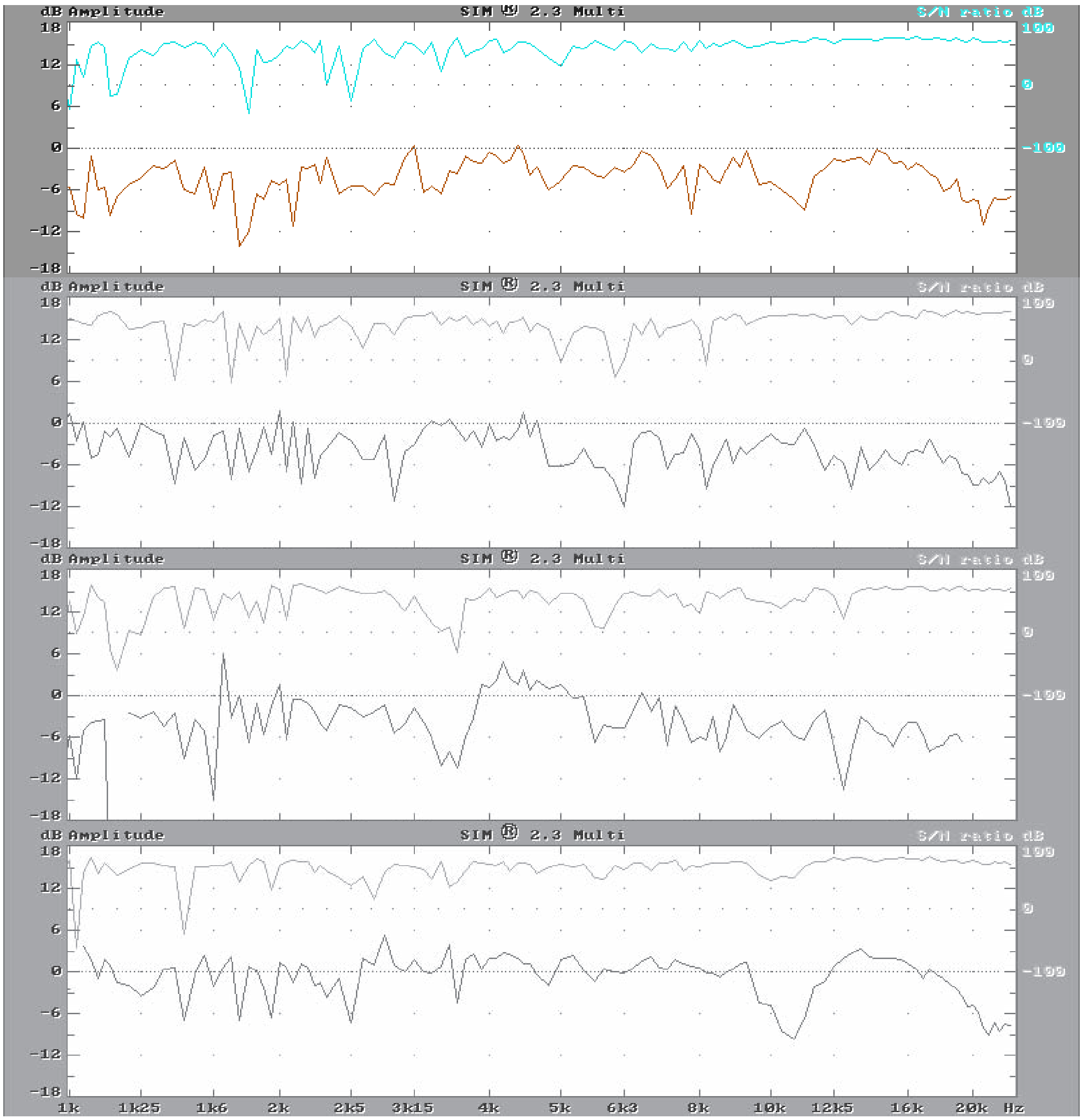


Fig. 2.2r Wide point-source array HF frequency response.

This configuration clearly has the least ripple of any of the multiple speaker array configurations. This gives it the most consistent frequency response and therefore, the most equalizable. Notice also that there is very little addition.

2.2.7 Parallel Arrays

This type of design has maximum overlap. However, as the time increases the level offset does not. This causes highly variable combing. Aligning speakers in a row with redundant horizontal orientation will cause an uneven frequency response over the listening area. While such arrays may generate lots of acoustical power, the redundant coverage will create large amounts of comb-filtering, making it respond poorly to equalization.

Notice in the chart below that the time offsets are relatively low. However, there is virtually no level offset at any position since axial orientation is the same for both speakers. This results in severe HF ripple.

The parallel configuration is only suitable for low-frequency devices in which the coupling effect can be beneficial. In such cases, the time offset is small enough so that the first null is above the HF cutoff of the subwoofers.

Coverage: Same coverage as for a single cabinet.

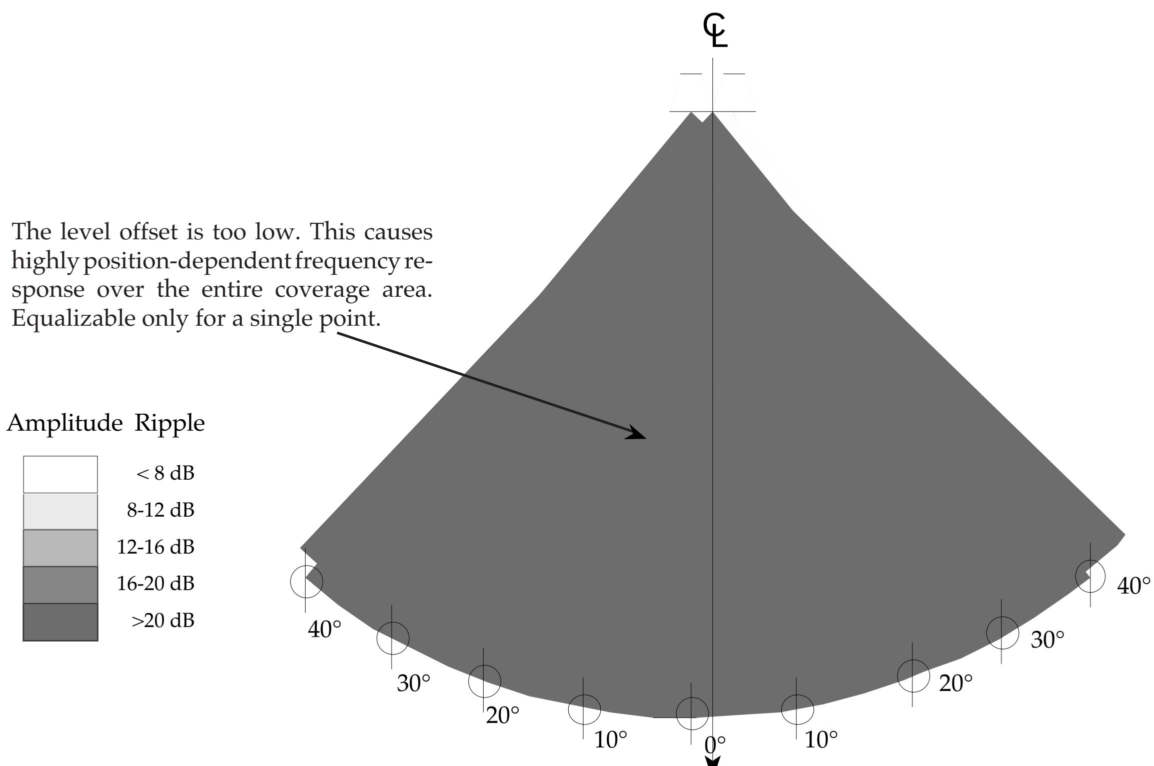
On-axis SPL: Maximum addition.

Level distribution: Same coverage as for a single cabinet.

Frequency response distribution: Every position has a unique frequency response. HF ripple is severe at all locations.

Equalizability: Can only be equalized for one position.

Where to use: Subwoofers only!!!!

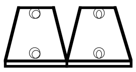


Parallel

	40°	30°	20°	10°	0°	10°	20°	30°	40°	Mean
Time offset	1.2 ms	0.8 ms	0.7 ms	0.4 ms	0.06 ms	0.04 ms	0.7 ms	1 ms	1 ms	0.7 ms
1st null	417 Hz	625 Hz	714 Hz	1250 Hz	8333 Hz	12500 Hz	714 Hz	500 Hz	500 Hz	2839.3 Hz
Level offset	0.4 dB	0.3 dB	0.2 dB	0.1 dB	0.0 dB	0.0 dB	0.2 dB	0.3 dB	0.3 dB	0.2 dB
Ripple	30.0 dB	30.0 dB	30.0 dB	30.0 dB	30.0 dB	30.0 dB	30.0 dB	30.0 dB	30.0 dB	30.0 dB

Fig 2.2s Parallel array equalizability.

2.2.7 Parallel Arrays



Parallel
–10°

Parallel 0°

Parallel 10°

Parallel 20°

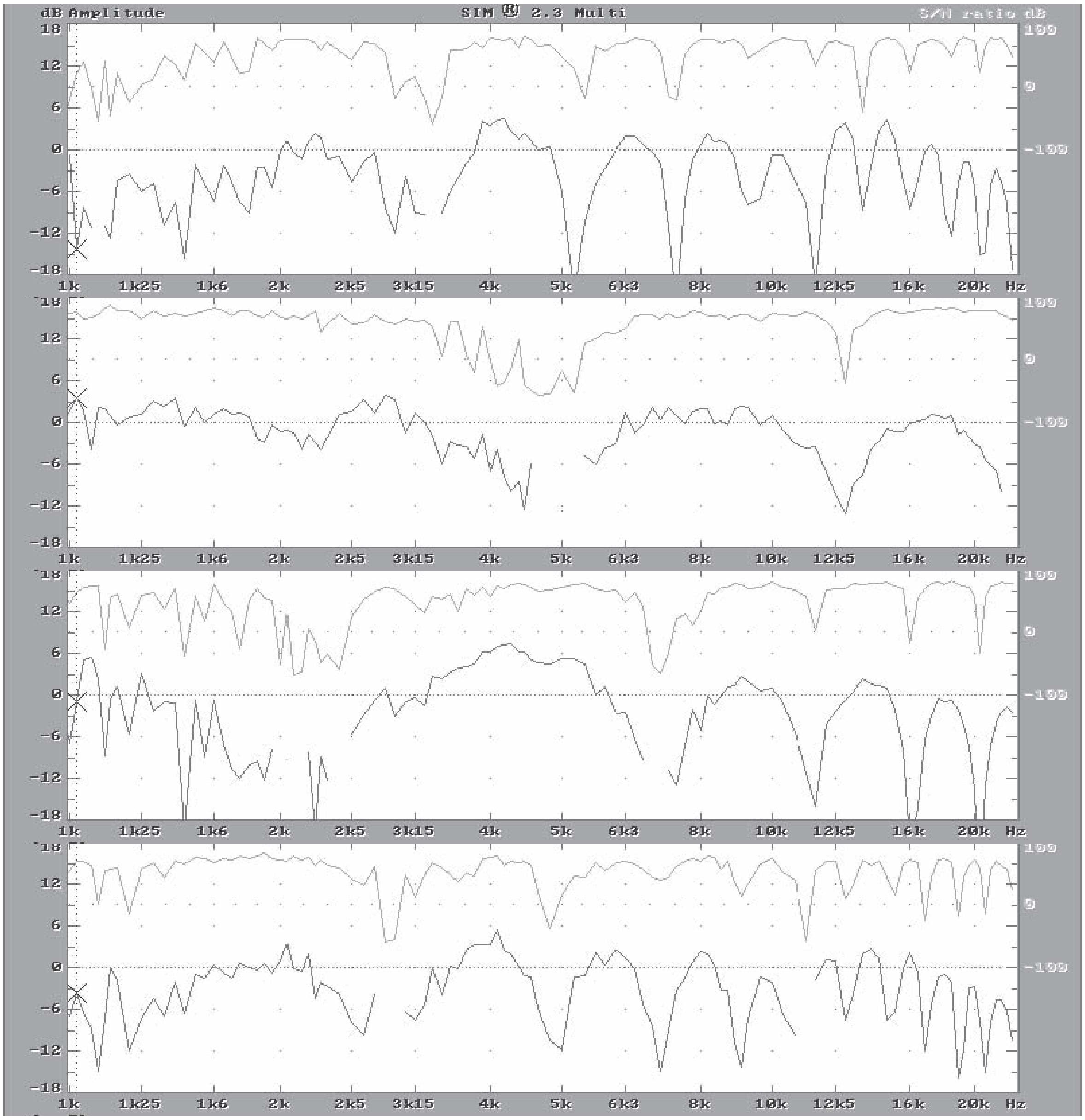


Fig 2.2t Parallel array HF frequency response.

An array of two speakers was measured using SIM System II. The response at four of the positions is shown above. Notice the large change in response over 40° of the coverage area. The cancellations are deep and highly variable making it difficult to find an equalization solution that would serve more than one position. Notice also that there is substantial addition. (Unfortunately there are even more substantial cancellations.)

2.2.8 Crossfire Arrays

Crossfire arrays (not recommended) function similarly to narrow point-source arrays but have worse combing. There is no advantage to crossfire arrays over point-source arrays. Therefore, they are not recommended.

Notice in Fig 2.2u the overlap zone covers most of the main speaker's coverage area. This causes a large amount of power addition in the center area. Contrast the time offsets here with those of the narrow point-source array described previously. You will notice that they are consistently larger than the narrow point-source array, creating a wider area of deep notches in the center.

Note: Crossfire arrays are not recommended.

Coverage: Center buildup causes narrowing of the area between –6 dB points.

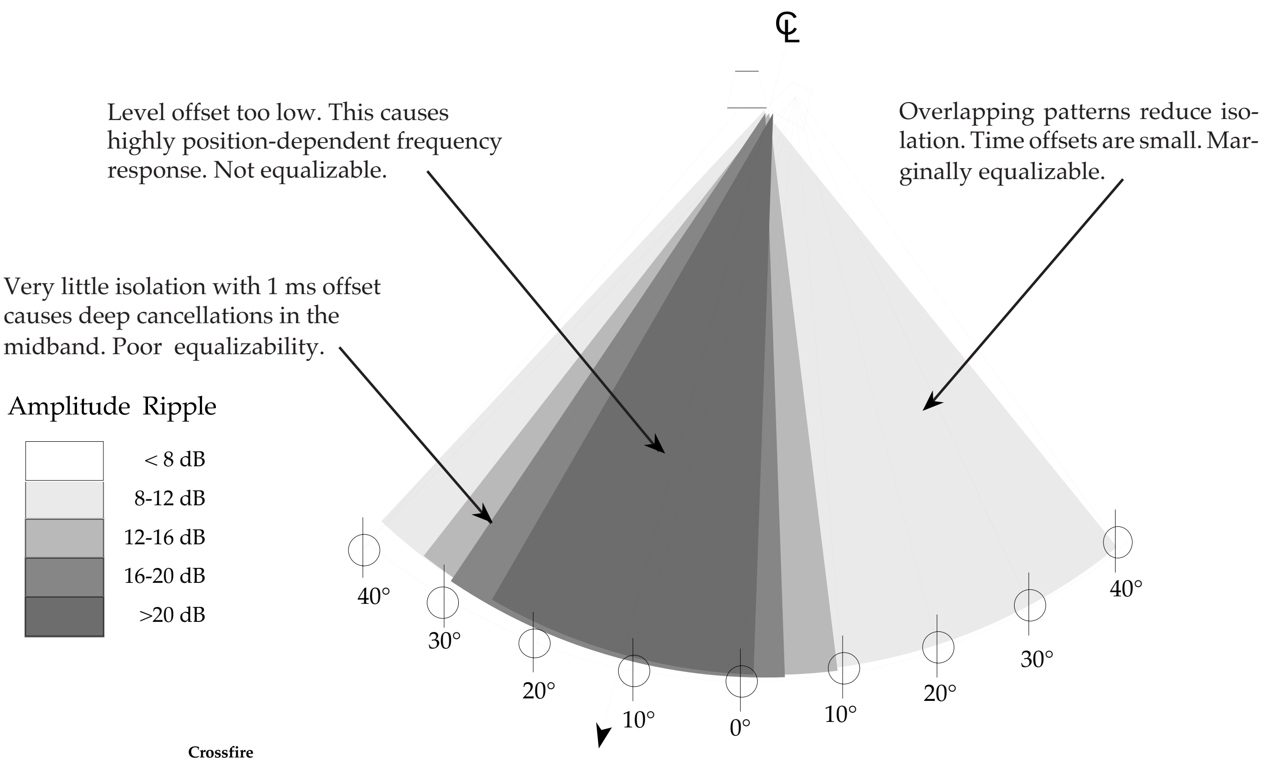
On-axis SPL: Maximum addition.

Level distribution: Hot in the center.

Frequency response distribution: Poor. Most of the center area has deep ripple.

Equalizability: Poor. High variability through the coverage area.

Where to use: Not recommended.



Crossfire

	40°	30°	20°	10°	0°	10°	20°	30°	40°	Mean
Time offset	1.4 ms	1 ms	0.6 ms	0.2 ms	0.2 ms	0.65 ms	0.9 ms	1.6 ms	1.9 ms	0.9 ms
1st null	357 Hz	500 Hz	833 Hz	2500 Hz	2500 Hz	769 Hz	556 Hz	313 Hz	263 Hz	954.5 Hz
Level offset	6.5 dB	2.3 dB	0.2 dB	0.1 dB	1.1 dB	5.2 dB	7.3 dB	7.5 dB	5.6 dB	4.0 dB
Ripple	9.0 dB	17.5 dB	30.0 dB	30.0 dB	24.0 dB	10.0 dB	8.0 dB	8.0 dB	10.0 dB	16.3 dB

Fig 2.2u Crossfire array interaction.

2.2.8 Crossfire Arrays

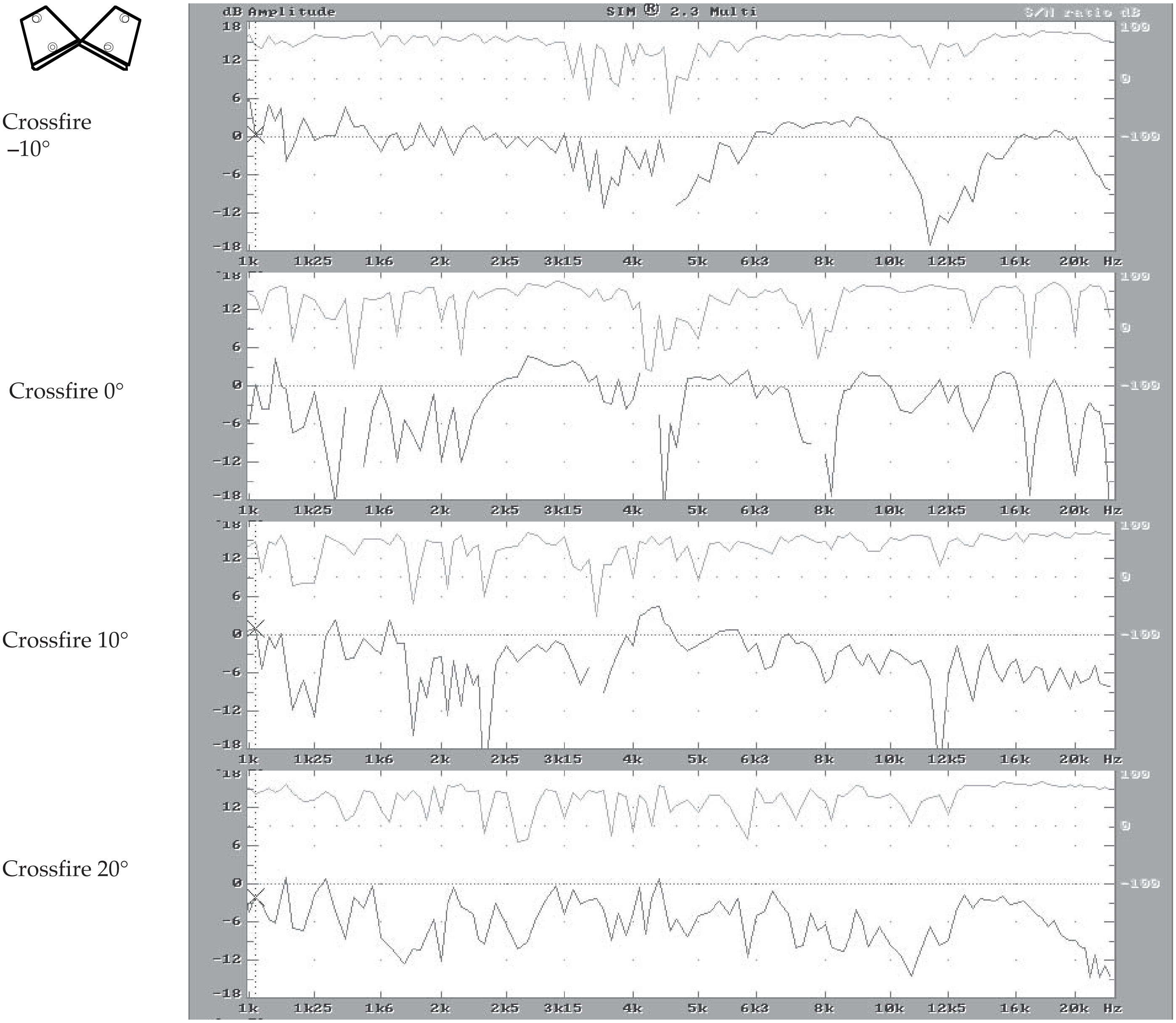


Fig 2.2v Crossfire array HF frequency response.

An array of two speakers was measured using SIM System II. The response at four of the positions is shown above. Notice the large change in response over 40° of the coverage area. The cancellations are deep and highly variable making it difficult to find an equalization solution that would serve more than one position.

2.2.9 Split-Parallel Array (Narrow)

Split-parallel arrays are often used for fill systems. This type of array will work best if the depth of the coverage is small, allowing smooth level distribution over a wide area. The key to using this type of array is to minimize the overlap zones, which are prone to severe ripple. Some tips on designing with this type of array are shown in Section 3.7.3, Frontfill Systems.

The key to equalizing the interaction of this type of array is *don't.* The response is too variable and the ripple too deep. The best method is simply to mute one of the speakers and equalize for the speaker / room interaction of the remaining speaker. Then restore the other speaker.

The following example shows what happens when the depth of coverage is too deep. The overlap areas have large time offsets and high ripple. This creates very low intelligibility and highly variable frequency response.

Coverage: Wide.

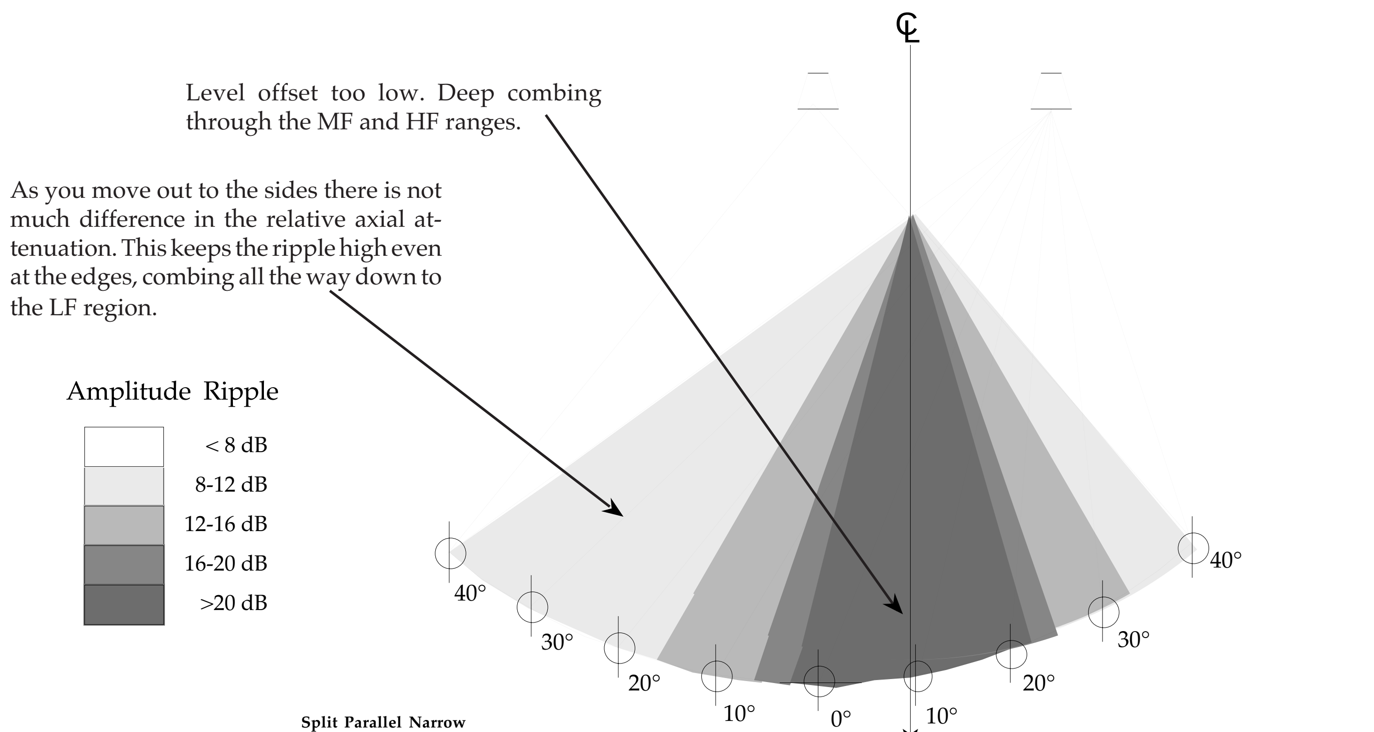
On-axis SPL: Some addition in the center area.

Level distribution: Highly variable with hot spots on the axis to the speakers and at the midpoint.

Frequency response distribution: Very poor. Large overlap area has large time offsets causing ripple deep into the LF range. The comb frequency changes very rapidly as you move off center.

Equalizability: The interaction is only equalizable at very low frequencies where it more closely resembles a single source. The MF and LF ranges should only be equalized as single sources—not as a combined system.

Where to use: Fill applications where the depth of coverage is very small and wide.



Split Parallel Narrow

	40°	30°	20°	10°	0°	10°	20°	30°	40°	Mean
Time offset	7.5 ms	6 ms	5 ms	3.5 ms	2 ms	0.37 ms	1.3 ms	3.1 ms	5 ms	3.8 ms
1st null	67 Hz	83 Hz	100 Hz	143 Hz	250 Hz	1351 Hz	385 Hz	161 Hz	100 Hz	293.3 Hz
Level offset	5.3 dB	6.9 dB	7.6 dB	3.1 dB	1.2 dB	0.1 dB	0.4 dB	3.0 dB	7.6 dB	3.9 dB
Ripple	10.0 dB	8.0 dB	7.5 dB	14.5 dB	24.0 dB	30.0 dB	30.0 dB	14.5 dB	8.0 dB	16.3 dB

Fig 2.2w Split-parallel (narrow) array interaction.

2.2.10 Split-Parallel Array (Wide)

When the proper distance is used between speakers this type of array works well, provided that the depth of the coverage is small. This creates smooth, level distribution over a wide area and minimal overlap zones. Some tips on designing with this type of array are described in Section 3.7.3, Frontfill Systems.

These types of arrays can be equalized quite effectively in the on-axis area of one the speakers. (Do not try to EQ in the overlap zones.)

The following example shows what happens when the depth of coverage is shallow. The overlap areas are small, leaving the majority of the coverage area with very low ripple.

Coverage: Wide.

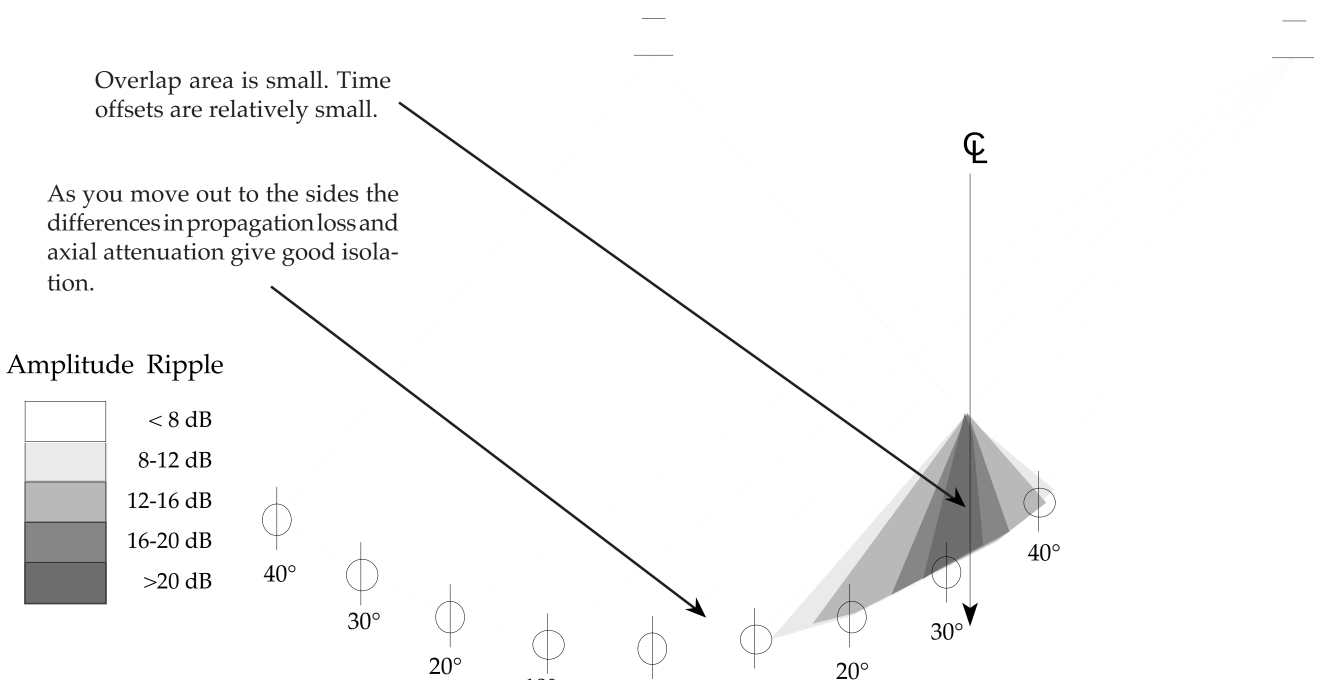
On-axis SPL: The same as a single speaker plus minimal addition in the center area. Low-frequency coupling will be minimal.

Level distribution: Good.

Frequency response distribution: The speakers act largely independently. There will be ripple in the overlap area but the on-axis area will have sufficient isolation for the ripple to be low.

Equalizability: The interaction is only equalizable at very low frequencies where it more closely resembles a single source. The MF and HF ranges should only be equalized as single sources—not as a combined system.

Where to use: Fill applications where the depth of coverage is very small and wide.



Split Parallel Wide

	40°	30°	20°	10°	0°	10°	20°	30°	40°	Mean
Time offset	22 ms	20 ms	17.5 ms	15 ms	11.5 ms	8.5 ms	5 ms	1.3 ms	3.5 ms	11.6 ms
1st null	23 Hz	25 Hz	29 Hz	33 Hz	43 Hz	59 Hz	100 Hz	385 Hz	143 Hz	93.3 Hz
Level offset	10.5 dB	13.1 dB	14.6 dB	12.1 dB	10.8 dB	9.0 dB	7.6 dB	1.4 dB	5.1 dB	9.4 dB
Ripple	5.0 dB	4.0 dB	3.5 dB	4.0 dB	4.0 dB	6.0 dB	8.0 dB	22.0 dB	10.0 dB	7.4 dB

Fig 2.2x Split-parallel (wide) array interaction.

2.2.11 Split Point-Source Arrays

An alternative array for fill systems is the split point-source. The speakers can be placed closer together and achieve minimal overlap by using the axial attenuation of the speakers. This type of array will work best if the coverage is shaped as an arc. The depth of the coverage area can be much deeper than for parallel arrays, since the angling of the speakers keeps the overlap area relatively small.

The overlap areas have larger level offsets than the comparably spaced split-parallel (narrow) array. This reduces ripple and improves isolation.

These types of arrays can be equalized quite effectively in the on-axis area of one the speakers. (Do not try to EQ in the overlap zones.)

Coverage: Wide.

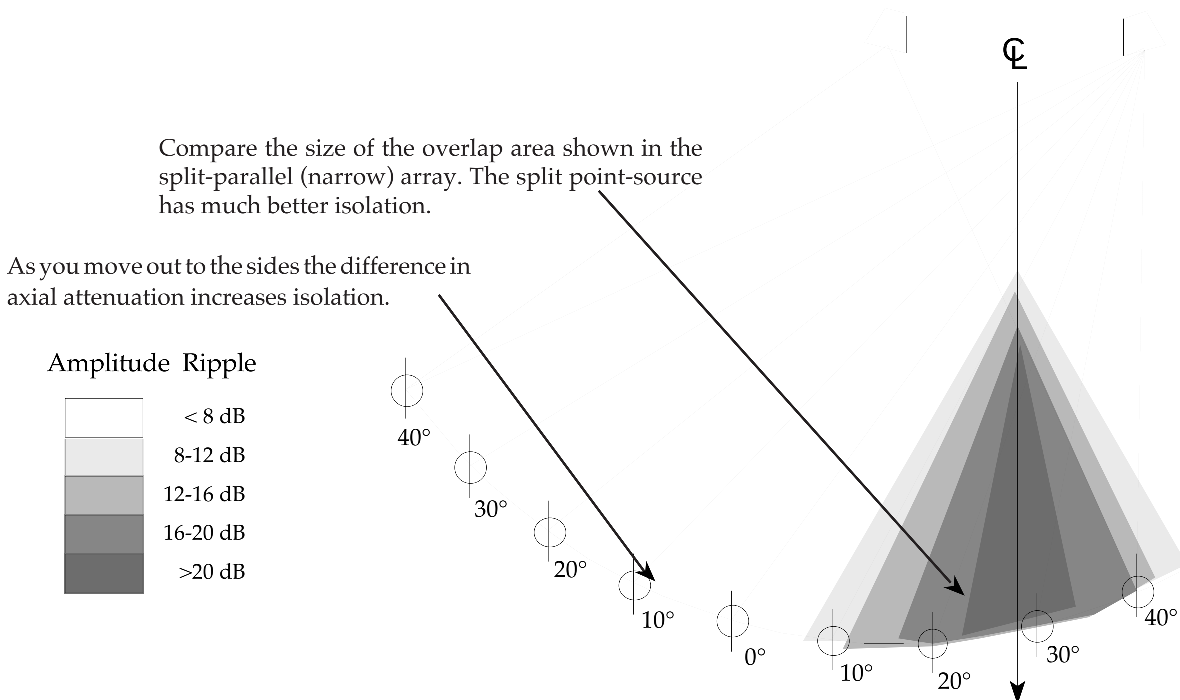
On-axis SPL: The same as a single speaker plus minimal addition in the center area. Low-frequency coupling will be minimal.

Level distribution: Good.

Frequency response distribution: The speakers act largely independently. The overlap area is much less than a split parallel array of similar dimensions.

Equalizability: The interaction is only equalizable at very low frequencies where it more closely resembles a single source. The MF and HF ranges should only be equalized as single sources.

Where to use: Fill applications where the depth of coverage is very small and wide.



Split Point Source

	40°	30°	20°	10°	0°	10°	20°	30°	40°	Mean
Time offset	9 ms	8.2 ms	7 ms	6 ms	4.5 ms	3 ms	1.4 ms	0.6 ms	2.4 ms	4.7 ms
1st null	56 Hz	61 Hz	71 Hz	83 Hz	111 Hz	167 Hz	357 Hz	833 Hz	208 Hz	216.4 Hz
Level offset	9.7 dB	12.5 dB	13.1 dB	11.9 dB	9.4 dB	7.5 dB	3.0 dB	0.7 dB	6.8 dB	8.3 dB
Ripple	5.0 dB	4.0 dB	4.0 dB	4.0 dB	5.0 dB	8.0 dB	14.5 dB	26.0 dB	8.0 dB	8.7 dB

Fig 2.2y Split point-source array interaction.

2.2.12 Point-Destination Arrays

Point-destination arrays have the most variable responses of any of the split arrays. Extreme caution should be used when designing these into your system. This type of array has extremely large overlap areas and very little axial attenuation, resulting in full range combing.

The best use for point destination arrays is for in-fill applications (i.e., where you are trying to reach a center area from the sides). The key is to minimize the level for these systems so *only* the center area is covered. The larger the area covered, the worse the combing.

You may notice the resemblance of this array type to the standard stereo configuration. Bear in mind that stereo systems (theoretically) contain different signals for the left and right channels. Therefore, the interaction is randomized by the difference in signals. This is normal for stereo. However, if the signal is panned to the center, the interaction will occur as shown here.

The key to equalizing the interaction of this type of array is *don't.* The response is too variable and the ripple too deep. The best method is simply to mute one of the speakers and equalize for the speaker/room interaction of the remaining speaker. Then restore the other speaker.

Coverage: Narrow.

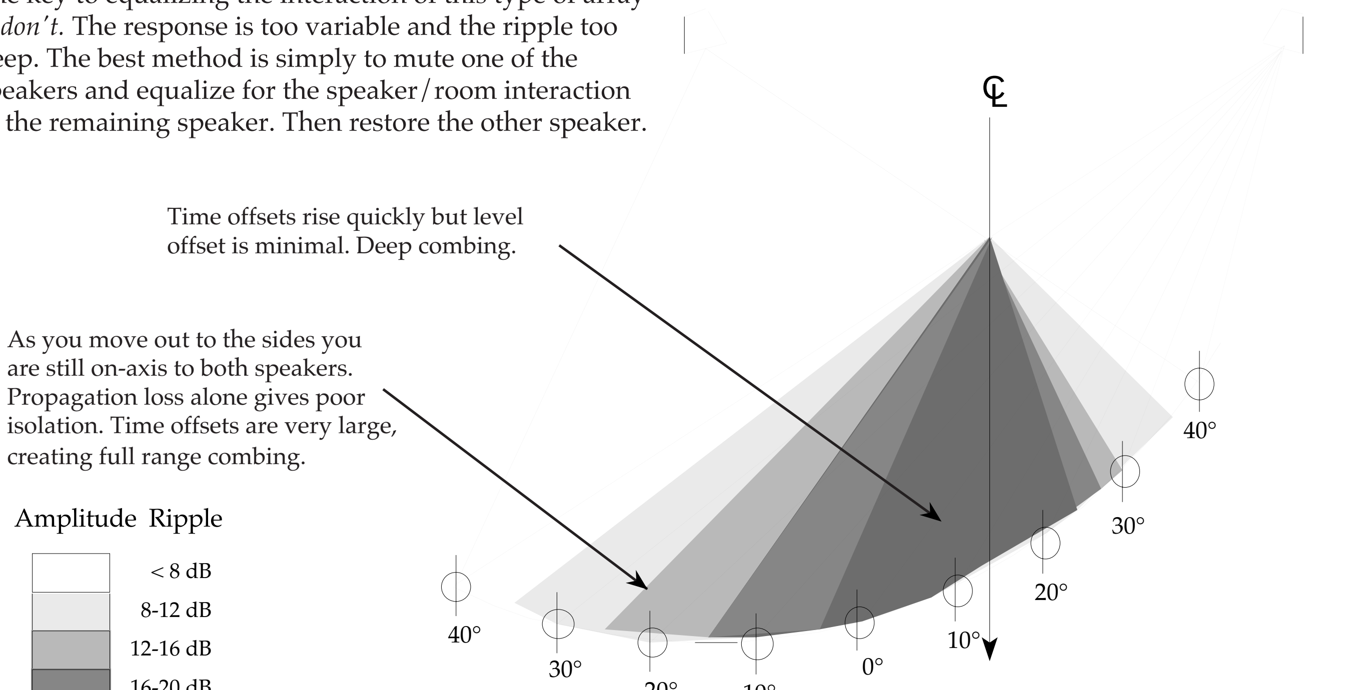
On-axis SPL: Maximum addition.

Level distribution: Large center area buildup.

Frequency response distribution: Every position has a unique frequency response. HF ripple is severe just off center, moving down in frequency as you move to the sides.

Equalizability: The interaction is not equalizable. Should only be equalized as single sources.

Where to use: In-fill speakers to cover near center area. Depth of coverage must be kept to a minimum or excess overlap will cause deep combing.



Point Destination

	40°	30°	20°	10°	0°	10°	20°	30°	40°	Mean
Time offset	16.6 ms	14 ms	11 ms	8 ms	4.5 ms	1.5 ms	2 ms	6 ms	10 ms	8.2 ms
1st null	30 Hz	36 Hz	45 Hz	63 Hz	111 Hz	333 Hz	250 Hz	83 Hz	50 Hz	111.3 Hz
Level offset	8.4 dB	5.9 dB	3.2 dB	2.4 dB	1.4 dB	0.5 dB	0.7 dB	3.9 dB	8.9 dB	3.9 dB
Ripple	7.0 dB	9.0 dB	14.5 dB	17.0 dB	22.0 dB	30.0 dB	28.0 dB	12.0 dB	6.0 dB	16.2 dB

Fig 2.2z Point-destination array interaction.

2.2.13 Monitor Sidefills

Monitor sidefills are the ultimate point destination array. In this case, both speakers face directly into each other. When approaching one speaker you move away from the other. Therefore, the time offsets change so rapidly that at 1 foot (30 centimeters) off-center, combing is as low as 275 Hz. Since you are on-axis to both speakers, there will be no axial attenuation, leaving only the relative propagation loss to isolate the systems. The result is very severe ripple through most of the coverage area.

This is not to say that monitor sidefills should be abolished. But, it is important to know what happens when they are used. In other words, don't be surprised when the response changes at every spot, and think you are doing something to cause it. And, put to rest crazy "solutions" like polarity reversing one of the speakers.

Here is a solution that will work: If the singer is off-center and the mic will not be moved, delay the nearer sidefill so that both speakers are synchronized at the mic. Maximum addition is ± 1 foot.

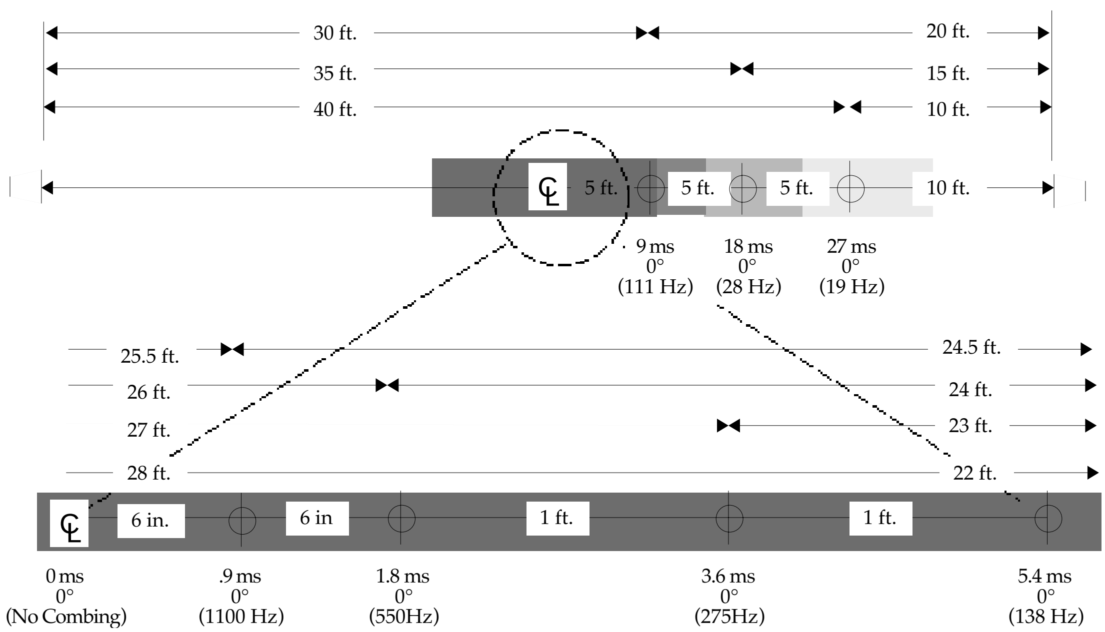
On-axis SPL: Maximum addition at the center.

Level distribution: Large center area buildup.

Frequency response distribution: Every position has a unique frequency response. HF ripple is severe just off-center, moving down in frequency as you move to the sides.

Equalizability: The interaction is not equalizable. Monitor sidefills should only be equalized as a single source.

Where to use: Monitor side fill.



Point Destination (Sidefill)

	0	.5 ft	1 ft	2 ft	3 ft	5 ft	10 ft	20 ft	Mean
Time offset	0.025 ms	0.9 ms	1.8 ms	3.6 ms	5.4 ms	9 ms	18 ms	27 ms	8.2 ms
1st null	20000 Hz	556 Hz	278 Hz	139 Hz	93 Hz	56 Hz	28 Hz	19 Hz	145.8 Hz
Level offset	0.0 dB	0.3 dB	0.6 dB	1.2 dB	1.7 dB	2.7 dB	4.7 dB	6.4 dB	2.2 dB
Ripple	0.0 dB	30.0 dB	27.0 dB	23.0 dB	21.0 dB	16.0 dB	11.0 dB	8.5 dB	17.1 dB

Fig 2.2aa Monitor sidefill array interaction.

2.2.14 Vertical Arrays

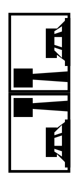
Although the trapezoidal enclosure design has no benefit for vertical arrays, the same rules of logic apply. Cabinets arrayed vertically should also form an arc, with a virtual point source behind the array. The exception to this is the long-throw array configuration, in which the horns are placed together with no angular differential. This creates an extreme narrowing of the vertical coverage pattern, doubling the distance that the speakers can throw, at some cost in frequency response smoothness.

Narrow (Tight-Pack) Arrays

Narrow coverage arrays can be constructed by placing the cabinets directly adjacent with the HF horns together on the same vertical plane. The horns couple together, creating, in effect, a single horn of half the coverage pattern with an on-axis SPL increase of up to 6 dB. The trade-off here, however, is that the frequency response distribution is noticeably poorer than that of the other recommended vertical arrays. The long-throw array will work best if the cabinets are directly coupled. Since we are dealing with high frequencies, the displacement between the horns becomes critical. If the cabinets are moved apart, the frequency response distribution will degrade. Long-throw arrays should be used only when extreme pattern narrowing is needed.

Long-throw vertical array.

Horns are directly coupled as close together as possible.



TechNotes™

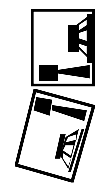
Meyer Sound has published a series of technical notes describing the array behavior of various configurations of UPAs, MSL-2As and MSL-3As. Each array configuration was measured and the coverage angle and on-axis maximum SPL determined, providing designers with an easy method to begin the process of array selection. Contact your Meyer Sound dealer to receive your copy of TechNotes™

Narrow (Optimized) Arrays

Narrow optimized arrays are a hybrid of the long-throw and wide-angle arrays. In this case, the horns are coupled together but splayed apart at the front . This configuration gives a wider response than the standard, long-throw configuration, and has improved frequency response distribution. The amplifier voltage gain of the two horns should be matched, and the rear of the cabinets should be as close as possible. There is currently research under way to best determine the optimal angles for narrow arrays. The current recommended angle between the horns is one-fourth the horns' coverage angle.

Long-throw optimized vertical array.

Horns are placed together and splayed outward at the front.

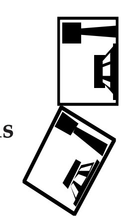


Wide-Angle Arrays

Wide coverage arrays can be constructed by splaying the cabinet fronts outward while leaving the rears touching. The vertical orientation of the cabinets is the same (both horn up or both horn down). These arrays will tend to increase the vertical coverage but have less of an effect on the on-axis power. These arrays are less sensitive than the long-throw arrays and will work well at various angles ranging from one-fourth to one-half the unit coverage angle. The smoothest frequency response and level distribution will generally occur when the splay angle is one-half the coverage angle.

Wide vertical array (adjustable).

Horns are not coupled. Cabinet fronts are splayed. Coverage widens as splay angle increases.



2.2.14 Vertical Arrays

Parallel Arrays (Not Recommended)

Aligning speakers in a row with redundant vertical orientation will cause an uneven frequency response over the listening area. While such arrays may generate large amounts of acoustical power, the redundant coverage will create extensive comb filtering, making it unequalizable except for a single point.

Crossfire Arrays (Not Recommended)

Concave array shapes will also focus large amounts of energy at the center but will suffer from similar comb-filtering problems as the parallel arrays.

Vertical Arrays

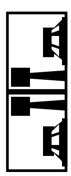
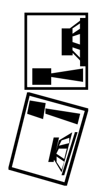
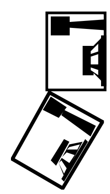
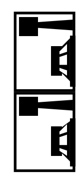
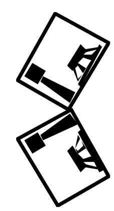
Vertical Arrays	Narrow (Long-throw)	Long-throw Optimised*	Wide Angle Optimised*	Parallel	Cross-fire
Configuration					
Vertical Coverage	Minimum (1/2 single unit coverage)	Narrow Moderately consistent	Wide Consistent	Narrow Highly variable	Narrow Highly variable
On-axis SPL Addition	Maximum 6 dB (typ.)	Moderate	Modest	Maximum	Maximum
Level Distribution	Fairly uneven	Fairly smooth	Smooth	Extremely uneven	Extremely uneven
Freq Response Distribution	Fairly rough	Fairly smooth	Good	Very poor	Very poor
Equalizability	Moderate Equalize as single block.	Good Equalize as single block	Very good Equalize as two blocks	Impossible Except for single location	Impossible Except for single location
Note:	HF Horns must be directly coupled for best result.	*UPA-1 = 15° UPA-2 = 15° MSL-2A = 15° MSL-5 = 10° MSL-10A = 10°	*UPA-1 = 30° UPA-2 = 30° MSL-2A = 30° MSL-5 = 20° MSL-10A = 20°	Not recommended under any circumstances	Not recommended under any circumstances

Fig. 2.2bb Vertical array reference chart.

2.3.1 Introduction

Reflections are virtually indistinguishable from speaker interactions. The same mechanisms that cause the additions and cancellations in speakers are at work with reflections. A reflective surface can be visualized as a phantom speaker source adding energy into the space. Reflected energy can be useful. If the reflected energy is near in time and level it will couple, as when subwoofers are placed on the floor. But why doesn't this work for HF horns? Because the time offset is too long to provide coupling. It combs instead.

The absorption coefficient of the surface, and its angle relative to the source, are primary factors in the nature of the speaker/room interaction. This, in effect, determines its equalizability. As absorption rises and the angle widens, the effect of the reflections is diminished.

Fig 2.3a contrasts the different types of reflections in terms of time and level offset. Five different types of reflections are shown with their corresponding tendencies. The ideal reflections would follow the same paths as outlined for speaker interaction (Fig 2.2e).

The following pages map out the effects of grazing, parallel, corner and back wall reflections. These scenarios are intended to provide generalized guidelines for approaching reflective surfaces, not to provide specific design criteria for a speaker or room. The example speaker has an 80° coverage pattern (–6 dB) and is measured at a "distance" of 75 micro-seconds in 10° increments. While the speaker is pictorially represented in plan view (horizontal axis), all of the information is equally relevant to its vertical axis. To simplify matters, the absorption coefficient of the surface is assumed to be 0 for all frequencies. Any of the scenarios will be aided by the addition of absorption. Therefore, these represent worst-case scenarios.

Notice that each of the reflection scenarios has a counterpart in the speaker interaction scenarios discussed in Section 2.2. If you have a clear understanding of the interaction of speakers, you will find it very easy to transfer that knowledge to these reflections. In this case, the relationship between the real and "phantom" speaker (the reflected image) is similar to that of multiple speakers. For example, the wide grazing reflection creates a split point-source array, whereas the corner reflection creates a point-destination array. Similar considerations will hold true in terms the equalizability of these reflections.

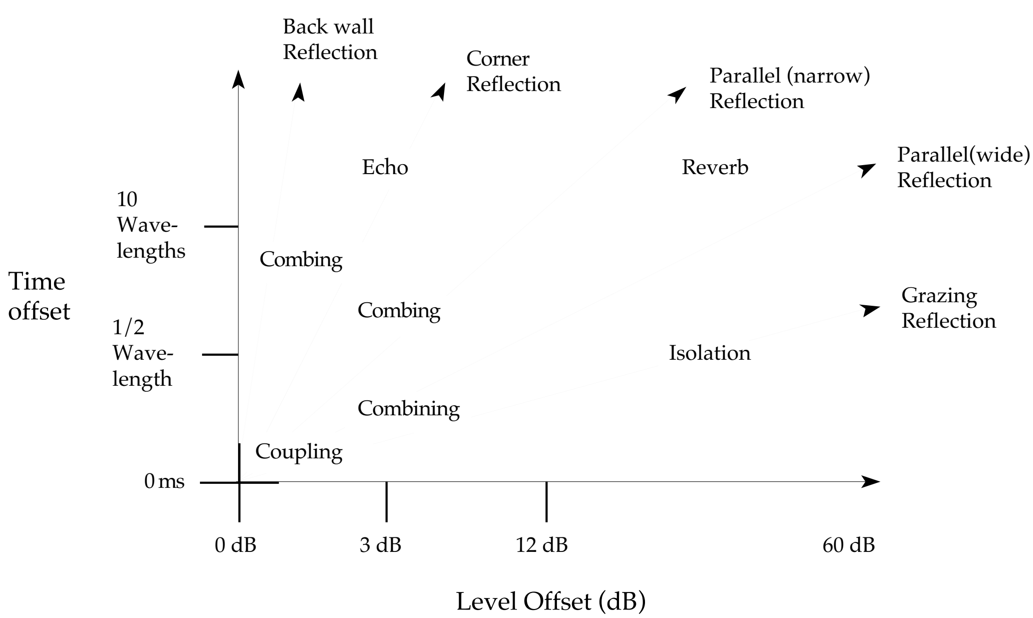


Fig 2.3a Tendencies for reflections to couple, combine and comb.

2.3.2 Grazing Wall Reflections

Grazing wall reflections occur when the speaker is pointed away from the plane of the surface. This allows only the speaker's off-axis signal to reach the wall, maximizing the axial attenuation of the reflection. Such a situation is typical of side walls in proscenium theatres and some ceilings as well.

Grazing reflections act like a second speaker in a split point-source array, and therefore are relatively well behaved.

Moving away from the wall the time offsets increase, but at the same time the isolation also increases. This creates a smooth predictable response in the center area.

Equalization can be very effective, particularly in the central area. Overall ripple is very low except at the extreme edges near the wall.

This is the most favorable of all the reflection scenarios. Careful speaker positioning—so that the speaker's pattern matches the grazing angle—will take advantage of this.

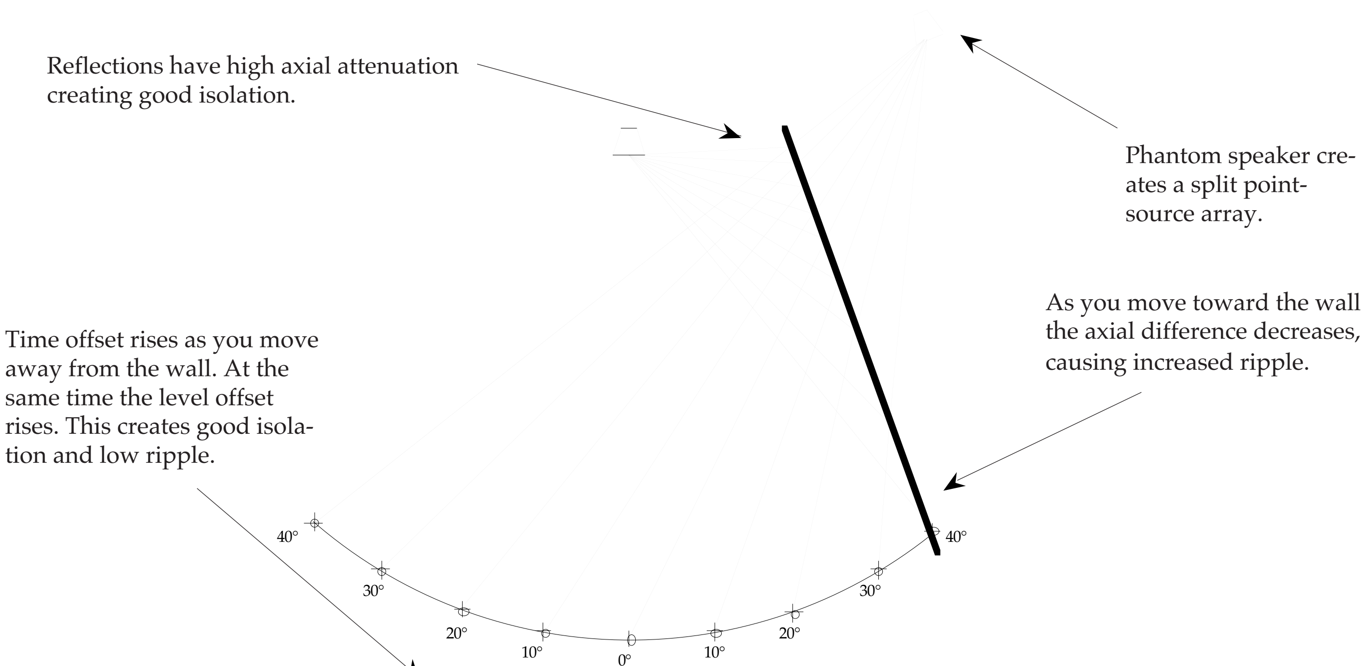
Features: The on-axis area of the speaker should have very low ripple primarily due to the large axial attenuation difference between the direct and reflected signals. As the grazing angle widens the ripple is further decreased due to the increased axial attenuation of the reflected path.

Speaker interaction counterpart: Split point-source array.

Frequency response distribution: Moving toward the side wall the comb frequency rises and the ripple deepens. Time offsets are high in the center but the ripple is very low.

Equalizability: Good equalizability in the on-axis area. Lower equalizability as approaching the side wall.

Examples: Proscenium side walls, ceilings.



Grazing Reflection (Narrow)

	40°	30°	20°	10°	0°	10°	20°	30°	40°
Time offset	44 ms	40 ms	36 ms	32 ms	26 ms	20 ms	15 ms	60 ms	0.0025 ms
1st null	11 Hz	13 Hz	14 Hz	16 Hz	19 Hz	25 Hz	33 Hz	8 Hz	200000 Hz
Level offset	14.0 dB	16.2 dB	16.4 dB	15.6 dB	14.1 dB	12.1 dB	10.1 dB	10.1 dB	0.0 dB
Ripple	3.5 dB	2.5 dB	2.5 dB	3.0 dB	3.5 dB	4.0 dB	5.0 dB	5.0 dB	30.0 dB

Fig 2.3b Grazing wall reflection.

2.3.3 Parallel Wall Reflections

Parallel side walls and floors are the most typical type of reflections. Their counterparts in speaker interaction are the parallel arrays. When directly coupled to the surface (for example, when you place a speaker on the floor or in a corner), low frequency coupling will occur. As frequency rises, the time offsets will become too large and create combing.

As the speaker moves away from the surface, it begins to act similarly to the split parallel arrays. As with those arrays, the interaction of parallel walls is best controlled if the coverage is confined to a shallow area, thereby minimizing the overlap (in this case, direct and reflected). If the coverage is deep, additional options include using narrower speakers, or repositioning the speaker away from the surface. This makes it a grazing reflection.

With parallel reflections the time offsets are large and the level offsets are not. As you approach the surface both offsets are reduced and eventually coupling resumes again.

The effectiveness of equalization will depend on the amount of isolation between the speaker and surface. Equalization is best applied in the central areas where the isolation is best.

This is one of the more favorable of the reflection scenarios in that there are practical solutions (repositioning, absorption and EQ) that will greatly reduce its effect.

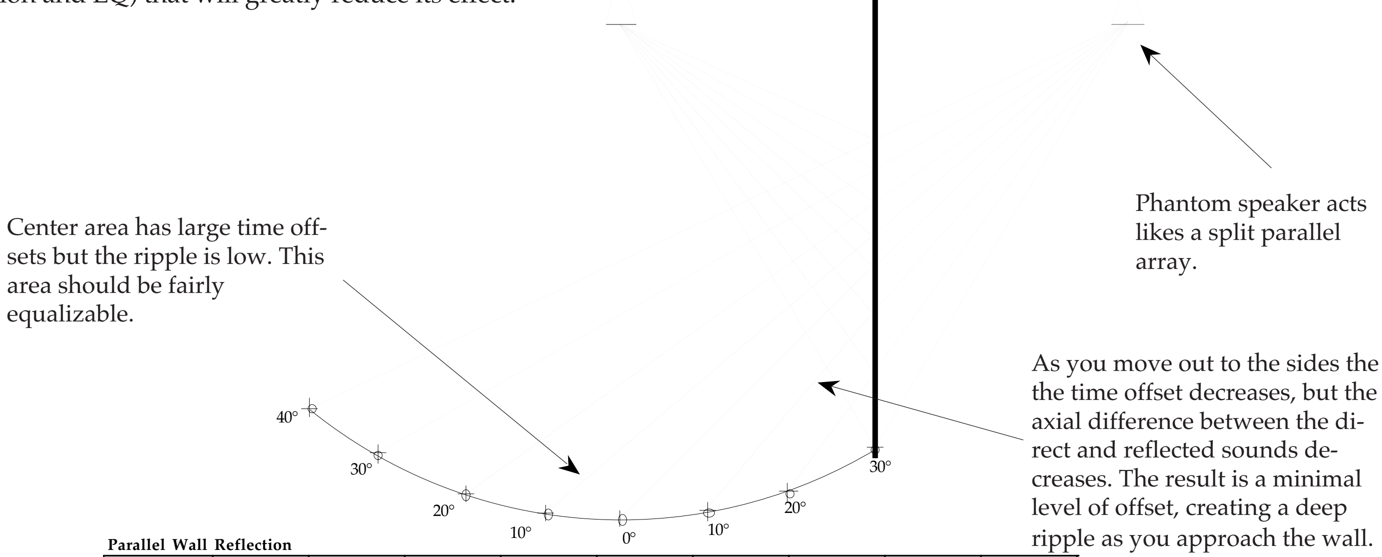
Features: The principal factors are the proximity to the wall, desired depth of coverage and speaker coverage angle. If the required coverage is deep, the speaker must be more distant from the wall, or else more directional.

Speaker interaction counterpart: Parallel (when directly coupled), split-parallel array (wide) if the coverage is shallow and wall is distant (or the speaker is narrow). Split-parallel array (narrow) if the coverage is deep and the wall is close (or the speaker is wide).

Frequency response distribution: As you move toward the side wall the time offset decreases but the ripple increases. The center area has the opportunity for good isolation if the wall is not too close or the speaker too wide.

Equalizability: Equalizability increases as you approach the speaker axis and decreases as you approach the wall.

Examples: Side walls, floors, ceilings.



Parallel Wall Reflection

	40°	30°	20°	10°	0°	10°	20°	30°	40°
Time offset	62 ms	56 ms	48 ms	40 ms	31 ms	22 ms	13 ms	0.025 ms	ms
1st null	8 Hz	9 Hz	10 Hz	13 Hz	16 Hz	23 Hz	38 Hz	20000 Hz	Hz
Level offset	10.2 dB	12.8 dB	13.3 dB	11.7 dB	10.0 dB	8.2 dB	6.4 dB	0.0 dB	dB
Ripple	5.0 dB	4.0 dB	4.0 dB	4.5 dB	5.0 dB	8.0 dB	9.0 dB	30.0 dB	dB

Fig 2.3c Parallel wall reflection.

2.3.4 Rear Wall Reflections

One of the biggest advantages to outdoor concerts is the lack of a rear wall. Indoors it is virtually impossible to cover the audience without also targeting the rear wall with on-axis energy. The strength of the rear wall reflection is proportional to the listener's distance from it. The closer you are, the more powerful the reflection. However, that does not necessarily mean that the audibility of the problem is worse there. Near the rear wall the low time and level offsets can create LF addition, while at the same time causing deep MF and HF ripples. However, as you move away from the wall the time offsets can become very large and are perceived as discrete echoes. Even though the ripple is not as deep, the intelligibility is worse and the distraction of a slap echo can be very annoying.

This is one of the least favorable of all the reflection scenarios and is the one most in need of additional absorption

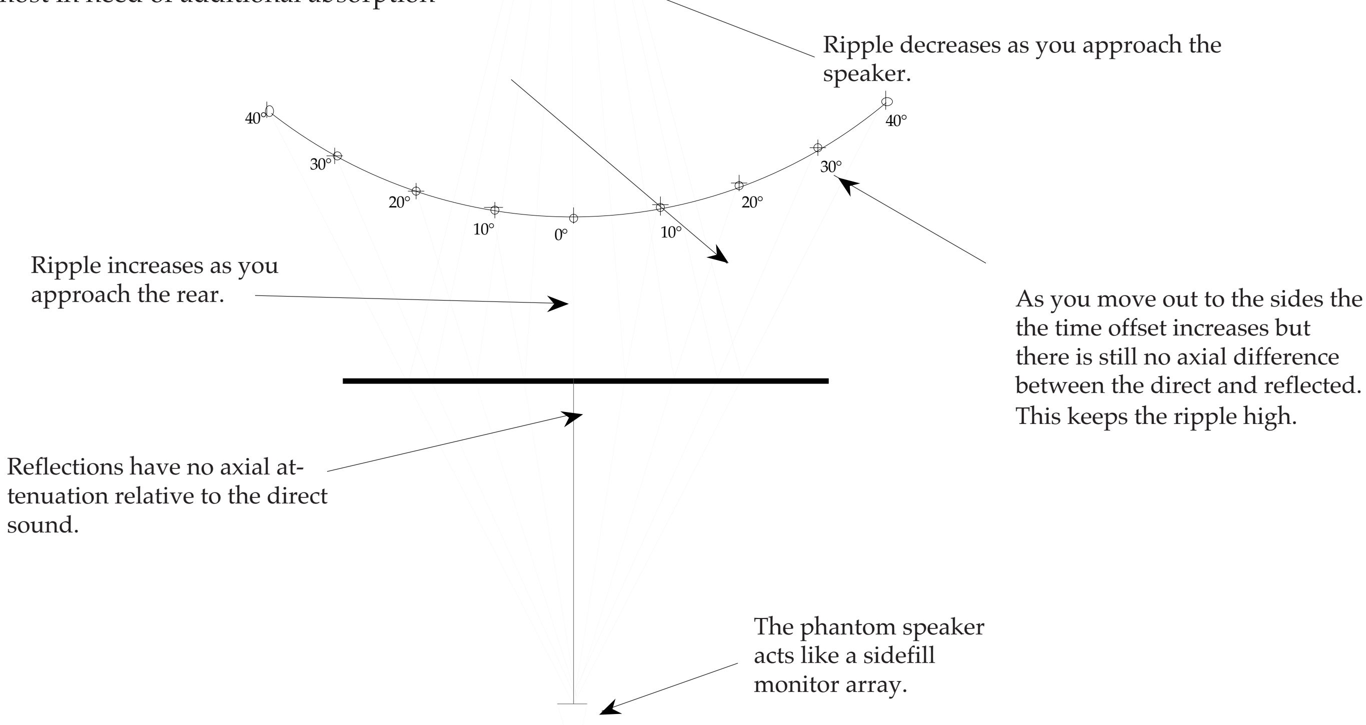
Features: The principal factor here is the proximity to the wall. The axial position is secondary since there is never any difference in axial attenuation between the direct and reflected sound.

Speaker Interaction Counterpart: Sidefill monitor array

Frequency response distribution: As you move toward the side wall the comb frequency rises but the ripple decreases slightly. It becomes highly variable near the rear.

Equalizability: Equalizability increases as you approach the speaker and decreases as you approach the wall.

Examples: Back walls.



Rear Wall

	40°	30°	20°	10°	0°	10°	20°	30°	40°
Time offset	75 ms	65 ms	57 ms	52 ms	50 ms	52 ms	57 ms	65 ms	75 ms
1st null	7 Hz	8 Hz	9 Hz	10 Hz	10 Hz	10 Hz	9 Hz	8 Hz	7 Hz
Level offset	0.0 dB	3.4 dB	4.9 dB	4.6 dB	4.4 dB	4.6 dB	4.9 dB	3.4 dB	0.0 dB
Ripple	30.0 dB	13.0 dB	10.0 dB	11.0 dB	11.0 dB	11.0 dB	10.0 dB	13.0 dB	30.0 dB

Fig 2.3d Rear wall reflection.

2.3.5 Corner Reflections

A corner reflection is the opposite of a grazing reflection. The surface is angled inward so that the energy comes back into the on-axis area. This corresponds to the point-destination array and suffers from large time offsets and low isolation. The worst case scenario of these types (not shown) is when the surface is curved, creating a parabolic mirror effect, which has "whisper gallery" focus points and virtual images.

One common corner reflection is the downward angled roof structure in the rear of halls. The top of a speaker's vertical pattern can bounce down into the rear seats and reduce intelligibility. Practical solutions for this include the use of highly directional speakers or an "eyebrow" curtain above the speaker that absorbs the unused part of the vertical pattern. The more common but less effective method is to direct the speakers downward away from the roof. While this may help the people downstairs (by reducing reverb) it will not fix things in the rear since listeners are losing both direct and reverberant sound together. See Section 3.4.3, Speaker Placement.

With corner reflections, the time offsets are highly variable and the level offsets are low. Moving away from the surface, the time offset increases greatly, but the ripple does not, creating poor equalizability.

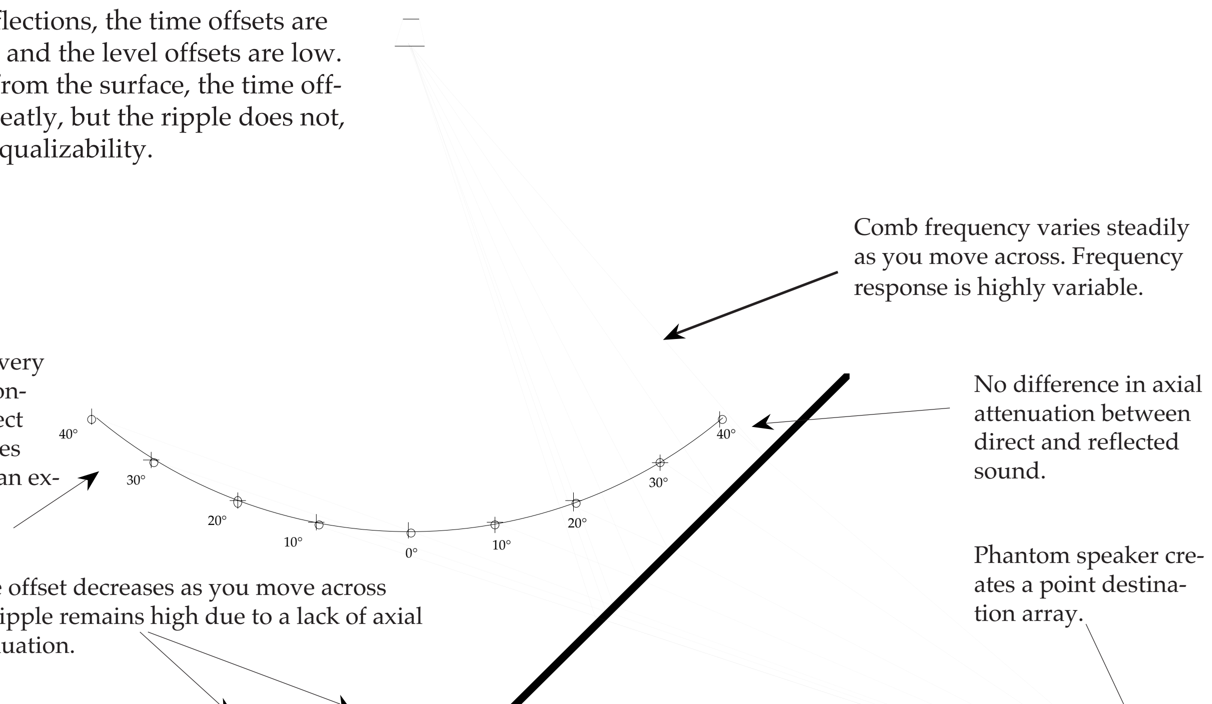
Features: The principal factors are the size of the angled area and its orientation to the speaker. The larger the angled area and the closer the angle of the surface is to the on-axis angle of the speaker, the worse it gets.

Speaker interaction counterpart: Point-destination array.

Frequency response distribution: As you move the seaker coverage from left to right the time offset decreases but the ripple does not. This creates an extremely inconsistent response.

Equalizability: Poor. Try absorption, speaker repositioning, or demolition.

Examples: Angled side walls, ceilings



Corner Reflection

	40°	30°	20°	10°	0°	10°	20°	30°	40°
Time offset	101 ms	90 ms	76 ms	64 ms	50 ms	40 ms	29 ms	22 ms	16 ms
1st null	5 Hz	6 Hz	7 Hz	8 Hz	10 Hz	13 Hz	17 Hz	23 Hz	31 Hz
Level offset	1.4 dB	4.8 dB	6.1 dB	5.4 dB	4.4 dB	4.2 dB	4.3 dB	4.2 dB	1.7 dB
Ripple	21.0 dB	10.0 dB	9.0 dB	9.5 dB	11.0 dB	11.0 dB	11.0 dB	11.0 dB	20.0 dB

Fig 2.3e Corner reflection.

2.4.1 Temperature

The formula below shows that the speed of sound in air changes with temperature. The interrelation of temperature and speed causes the wavelength to change for a given frequency. Table 2.4a illustrates the effect of different temperatures on the speed and wavelength. As the speed increases, the wavelength increases. The change in wavelength will modify the structure of standing waves and room modes. In addition, these changes will affect the timing of reflections relative to the direct sound.

The Speed Of Sound in Air (c):

$c = (1052 + 1.10T)$ feet/second,

where T is the temperature in degrees Fahrenheit.

For example, at 68° Fahrenheit the speed of sound in air is:

$c = 1052 + (1.10 \times 68)$

$c = 1052 + 74.8$

$c = 1126.8$ feet/second

Temp	Velocity	T delay	Wavelength		Temp	Velocity	T delay	Wavelength
		at 100 ft.	at 50 Hz				at 30 m	at 50 Hz
(°F)	(ft/sec)	(ms)	(ft)		(°C)	(m/sec)	(ms)	(m)
50	1107.30	90.31	22.15		10	337.47	88.90	6.75
52	1109.51	90.13	22.19		11	338.08	88.74	6.76
54	1111.72	89.95	22.23		12	338.68	88.58	6.77
56	1113.94	89.77	22.28		13	339.29	88.42	6.79
58	1116.15	89.59	22.32		14	339.90	88.26	6.80
60	1118.36	89.42	22.37		15	340.50	88.10	6.81
62	1120.57	89.24	22.41		16	341.11	87.95	6.82
64	1122.78	89.06	22.46		17	341.72	87.79	6.83
66	1125.00	88.89	22.50		18	342.33	87.64	6.85
68	1127.21	88.71	22.54		19	342.93	87.48	6.86
70	1129.42	88.54	22.59		20	343.54	87.33	6.87
72	1131.63	88.37	22.63		21	344.15	87.17	6.88
74	1133.84	88.20	22.68		22	344.75	87.02	6.90
76	1136.06	88.02	22.72		23	345.36	86.87	6.91
78	1138.27	87.85	22.77		24	345.97	86.71	6.92
80	1140.48	87.68	22.81		25	346.57	86.56	6.93
82	1142.69	87.51	22.85		26	347.18	86.41	6.94
84	1144.90	87.34	22.90		27	347.79	86.26	6.96
86	1147.12	87.18	22.94		28	348.40	86.11	6.97
88	1149.33	87.01	22.99		29	349.00	85.96	6.98
90	1151.54	86.84	23.03		30	349.61	85.81	6.99
92	1153.75	86.67	23.08		31	350.22	85.66	7.00
94	1155.96	86.51	23.12		32	350.82	85.51	7.02
96	1158.18	86.34	23.16		33	351.43	85.37	7.03
98	1160.39	86.18	23.21		34	352.04	85.22	7.04
100	1162.60	86.01	23.25		35	352.64	85.07	7.05

Table 2.4a The effect of temperature on sound transmission.

2.4.2 Humidity

Humidity will modify the manner in which air transmits sound. As humidity increases high frequency transmission improves. Lower humidity causes greater high frequency attenuation. The losses accumulate over distance causing the HF to roll off as you move away from the source. Outdoor venues have noticeable changes in HF transmission over time due to humidity changes. Fig 2.4b is a chart of transmission loss over frequency. Fig 2.4c is a field example showing the HF transmission loss over distance.

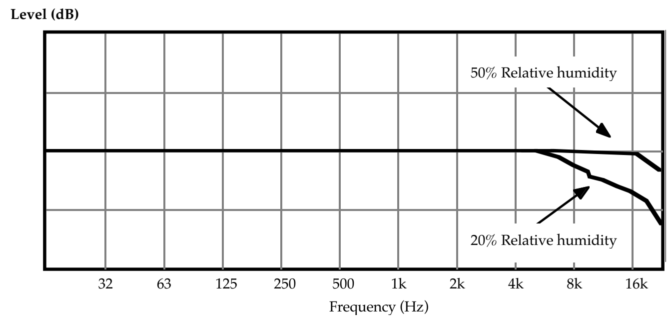


Fig 2.4b Relative humidity effect on frequency response.

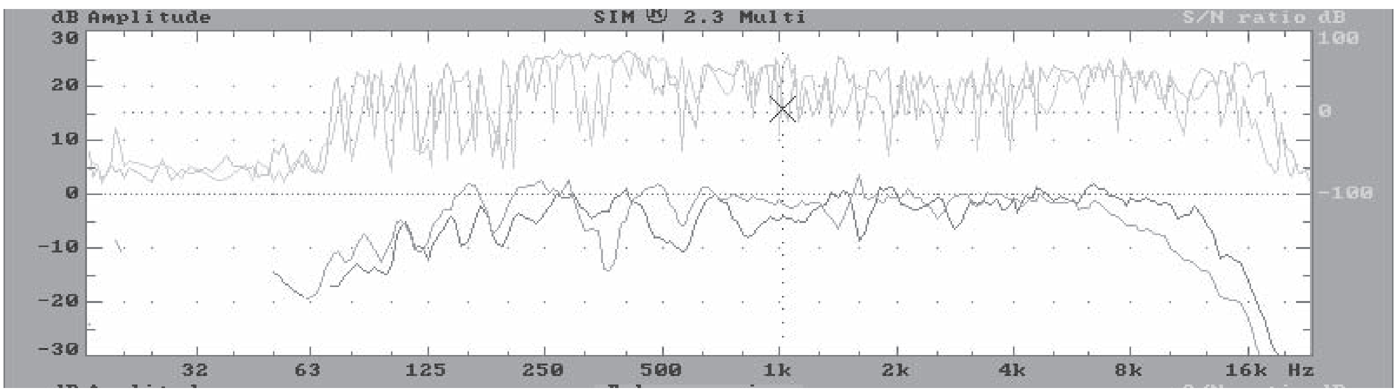


Fig 2.4c Air absorption causing HF loss over increased distance.

2.4.3 Absorption Coefficient

The absorption coefficient of materials determines the amount of energy that will be lost as a sound wave impinges upon an object. This subject is complex and well documented in other texts. An in-depth treatise is beyond the scope of this text. However, the following aspects should be noted:

1) A low absorption coefficient will cause strong echoes to be combined with the direct sound, causing deep comb filtering.

2) A change in absorption coefficient, such as when an audience comes in and covers a cement floor, will modify the frequency response by decreasing the strength of the echoes.

An application example illustrating how absorption affects a speaker system response is found in Section 5.4.

3.1.1 Requirements

If there is one word that encompasses the process of sound system design it might be *compromise*. Very few of us are given blank checks to design systems for ideal quality without regard for practical concerns or other needs. This does not mean, however, that a designer needs to capitulate immediately to a compromised sound quality. Instead, wise choices must be made between ideal, realistic, minimal and unacceptable conditions.

Before we can design a system, we need to have some idea as to what the client needs or wants. Then begins the process of turning this into reality.

At a minimum, the following questions should be answered:

- How many channels?
- Over what frequency range?
- At what maximum level?
- From what position(s)?
- Over what coverage area?
- For how much money?

Channels

The number of channels depends on the program material and the physical logistics. Possibilities include mono (for typical voice-only systems), two-channel mono or stereo (concert sound), left, center, right and surround (cinema), or multichannel (theatrical and other complex systems).

Frequency Range

The choice of program material will allow us to evaluate both frequency and power bandwidth requirements. For example, a vocal-only system will have little need for subwoofers, while a pop music system would be useless without them.

Power Requirements

The level requirements (interrelated with coverage) will guide you toward a choice of enclosure. If you need ninety-six dB SPL continuous at 500 feet (160 meters) you will probably want to use two MSL-5s rather than sixteen UPAs. On the other hand, if you need 90˚ of coverage at 100 feet (30meters) to achieve ninety-six dB, the single UPA-1C is more practical than three MSL-10s.

Speaker Positions and Orientation

Good speaker positions are critical to obtaining consistent sound quality and realistic imaging. Often, speaker positions are determined by outside factors such as staging, lighting, sightline, budget and aesthetic concerns. Poor speaker positions can make the rest of your efforts worth little, regardless of whether you have plenty of great speakers. Whenver possible, design for maximum flexibility in position and orientation. However, in most circumstances, once a speaker position is chosen users are forced to live with it. More than any other factor the choice of speaker position is a compromise involving the most interaction with other departments, many of which, unfortunately, have minimal knowledge or concern for sonic quality. Therefore, we must know what we want ideally, what we need minimally and where to draw the line on a position that will not work.

Coverage Area Requirements

The coverage requirements are largely a function of the venue shape. The arrayable speakers provide us with the building blocks for speaker arrays, which can be customized for each venue.

Budget

This all-important aspect cannot be overlooked or the design will never exist except on paper.

3.2.1 Frequency Range: Introduction

For each signal channel we need to know its required range. There are various acoustical texts that can detail the response ranges for the human voice as well as each and every musical instrument. In practical terms, however, we are rarely called upon to design a "flute only" or "tuba only" sound reinforcement system. Therefore, the frequency range requirements tend to break down into two categories: voice-only and full-range music.

Frequency range requirements:

- Voice only 80 Hz–18 kHz
- Music 40 Hz–18 kHz

While only one octave, the difference between these requirements, generally represents a large difference in system size, budget and pattern control. All of the biamplified systems (with the exception of the MSL-5) are fully capable of vocal reproduction by themselves. Music systems can be constructed with the addition of subwoofers.

3.2.2 Three-Way System Configuration

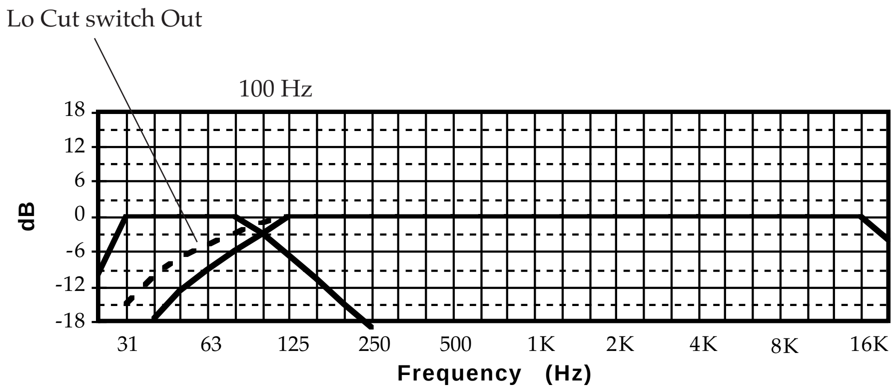
Full Range + Subwoofer

This system configuration creates a full-range triamplified system consisting of subwoofer and integral two-way full-range enclosures. The typical acoustical crossover is at 100 Hz between the systems, but it will shift down slightly if the two-way system is run full range.

CEU and LD-1A Settings

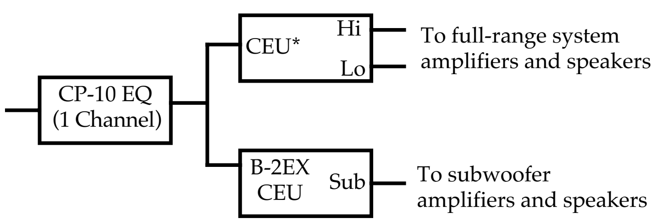
Full-range system: The Lo Cut switch is **In** when the full-range speakers are directly coupled to the subwoofers.

Full-range system: The Lo Cut switch is **Out** when the full-range speakers are flying above or separated from the subwoofers.



Three-way system frequency range chart.

Externally Powered

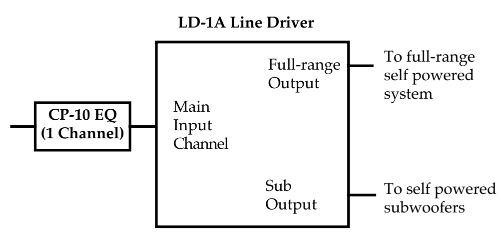


Flow block

Subwoofer System		Full-range System	
CEU	**Speaker**	**CEU**	**Speaker**
B-2EX	650-R2	M-1A	UPA-1C
	USW-1	M-1A	UPA-2
	MSW-2	M-1A	UM-1C
		S-1	MSL-2A
		S-1	USM-1
		M-3A	MSL-3A
		M-5	MSL-5
		M-10A	MSL-10A

Fig 3.2a Speaker / CEU Reference.

Self-Powered



Flow block

Subwoofer System	Full Range System
650-P	CQ-1
PSW-2	CQ-2
PSW-4	PSM-2
	MTS-4
	MSL-4
	MSL-6

Fig 3.2b Speaker / CEU Reference.

3.2.3 Three-Way DS-2 System Configuration

Full Range + Mid-bass

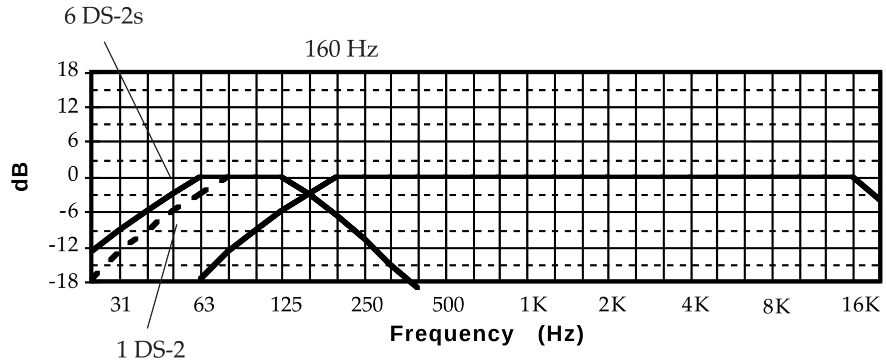
This system configuration creates a triamplified system, consisting of the DS-2 mid-bass system and integral two-way full-range enclosures. It has more superior low-frequency directional control than the standard three-way system. The typical acoustical crossover is at 160 Hz between the systems. Note that the full-range CEU is driven by the D-2 output. This system is restricted in low-frequency range down to between 60 and 50 Hz, depending upon the number of DS-2s.

CEU and LD-1A Settings

Full-range system: The Lo Cut switch is **In** at all times for the full-range speakers.

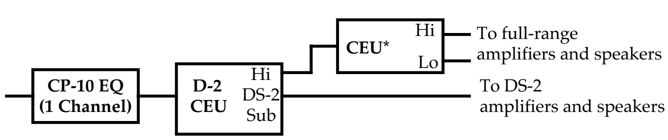
D-2 CEU: "DS-2 Only."

LD-1A: DS-2 plus Sub switch is **Out**.



Three-way DS-2 system frequency range chart.

Externally Powered

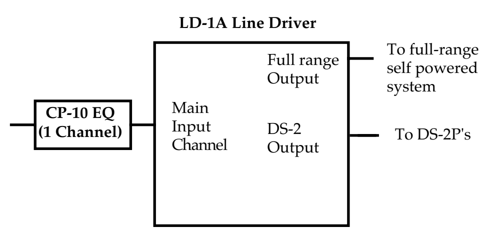


Flow block

Mid-bass System		Full-range System	
CEU	**Speaker**	**CEU**	**Speaker**
D-2	DS-2	M-1A	UPA-1C
		M-1A	UPA-2
		M-1A	UM-1C
		S-1	MSL-2A
		S-1	USM-1
		M-3A	MSL-3A
		M-5	MSL-5
		M-10A	MSL-10A

Fig 3.2c Speaker / CEU Reference.

Self-Powered



Flow block

Mid Bass System	Full Range System
DS-2P	CQ-1
	CQ-2
	PSM-2
	MTS-4
	MSL-4
	MSL-6

Fig 3.2d Speaker / CEU Reference.

3.2.4 Four-Way System Configuration

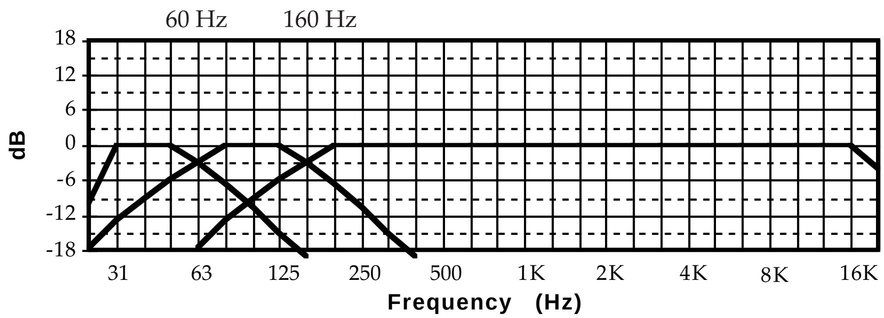
Full Range + Subwoofers + Mid-bass

This system configuration maximizes low-frequency power capability and directional control. The D-2 CEU resets the crossover to accomodate the combination of the 650-R2s and DS-2s at 60 Hz. This configuration is well suited for rock music applications, which typically require large amounts of low-frequency power. The superior directional control of the DS-2 creates a longer throw for the mid-bass. The acoustical crossovers are 60 Hz and 160 Hz between the systems.

CEU and LD-1A Settings

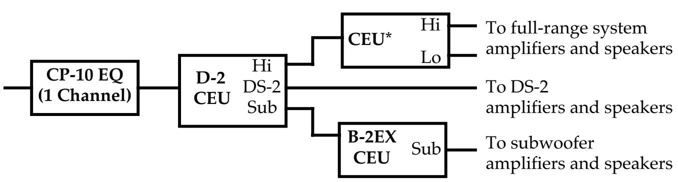
Full-range system: The Lo Cut switch is **In** at all times for the full-range speakers.

D-2 CEU and LD-1A : "DS-2 + Subwoofers."



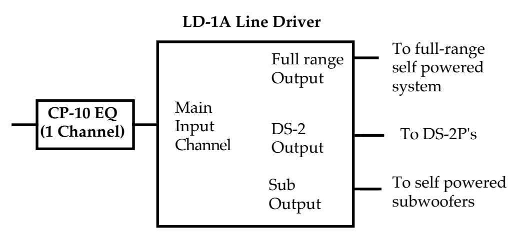
Four-way system frequency range chart.

Externally Powered



Flow block

Self-Powered



Flow block

Subwoofer System		Mid-bass System		Full-range System	
CEU	**Speaker**	**CEU**	**Speaker**	**CEU**	**Speaker**
B-2EX	650-R2	D-2	DS-2	M-1A	UPA-1C
	USW-1			M-1A	UPA-2
	MSW-2			M-1A	UM-1C
				S-1	MSL-2A
				S-1	USM-1
				M-3A	MSL-3A
				M-5	MSL-5
				M-10A	MSL-10A

Fig 3.2e Speaker / CEU Reference.

Subwoofer System	Mid Bass System	Full Range System
650-P	DS-2P	MSL-4
	PSW-2*	MSL-6
	PSW-4*	

Fig 3.2f Speaker / CEU Reference.

3.2.5 Five-Way System Configuration

Full Range + Subwoofers + Mid-bass + VHF Tweeter Array

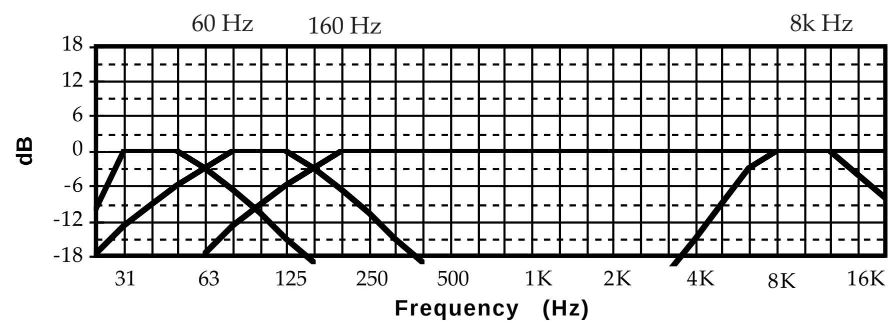
This system configuration is designed for very long-throw applications where distance related high-frequency attenuation becomes significant. Typically this configuration is reserved for high-power long-throw systems such as the MSL-3, MSL-5 and MSL-10A. The MST-1 Super Tweeter Array is a very directional VHF system which crosses in at 8 kHz. The logistics of MST-1 placement may dictate that a delay line will be needed to align the tweeters.

CEU Settings

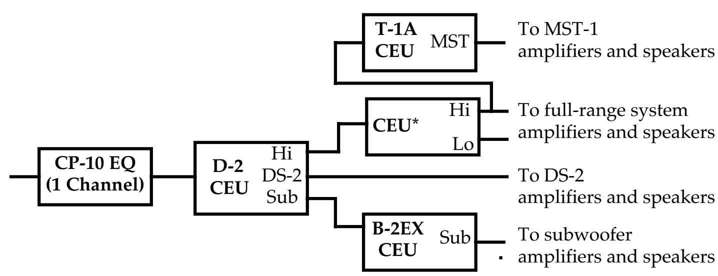
Full-range CEU: The Lo Cut switch is **In** at all times for the full-range speakers.

D-2 CEU: "DS-2 + Subwoofers."

Four-way, Five-way System Speaker / CEU Reference



Five-way system frequency range chart.



Five-way system flow block.

Subwoofer System		Mid-bass System		Full-range System		VHF System	
CEU	**Speaker**	**CEU**	**Speaker**	**CEU**	**Speaker**	**CEU**	**Speaker**
B-2EX	650-R2	D-2	DS-2	M-1A	UPA-1C	T-1A	MST-1
	USW-1			M-1A	UPA-2		
	MSW-2			M-1A	UM-1C		
				S-1	MSL-2A		
				S-1	USM-1		
				M-3A	MSL-3A		
				M-5	MSL-5		
				M-10A	MSL-10A		

Fig3.2g Five-way System Speaker / CEU Reference.

3.3.1 Power Loss Over Distance

Propagation loss of a speaker system in free-field conditions occurs at 6 dB per doubling distance from the source, as shown in Fig 3.3a. This property, known as the "inverse square law" provides a good estimate for outdoor systems, and to a lesser extent, indoors. Nevertheless, it represents the minimum SPL (sound pressure level) numbers for indoor systems since the addition of reverberant energy will cause the losses to be less than in free-field. Chart 3.3b shows the propagation loss over distance in feet and meters, respectively.

To calculate the maximum SPL at a given location:

A) Determine the on-axis maximum SPL at one meter. (This can be done by using the Meyer Sound TechNotes™ and/or various data sheets.)

B) Measure the distance from the speaker array to the listening position.

C) Find the attenuation for that distance on Chart 3.3b and subtract it from the one-meter rating.

This will give you a conservative on-axis maximum SPL estimate.

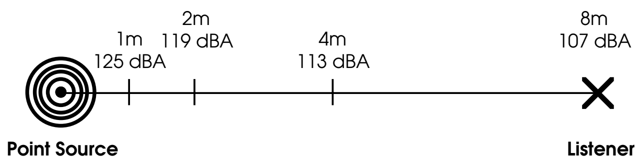
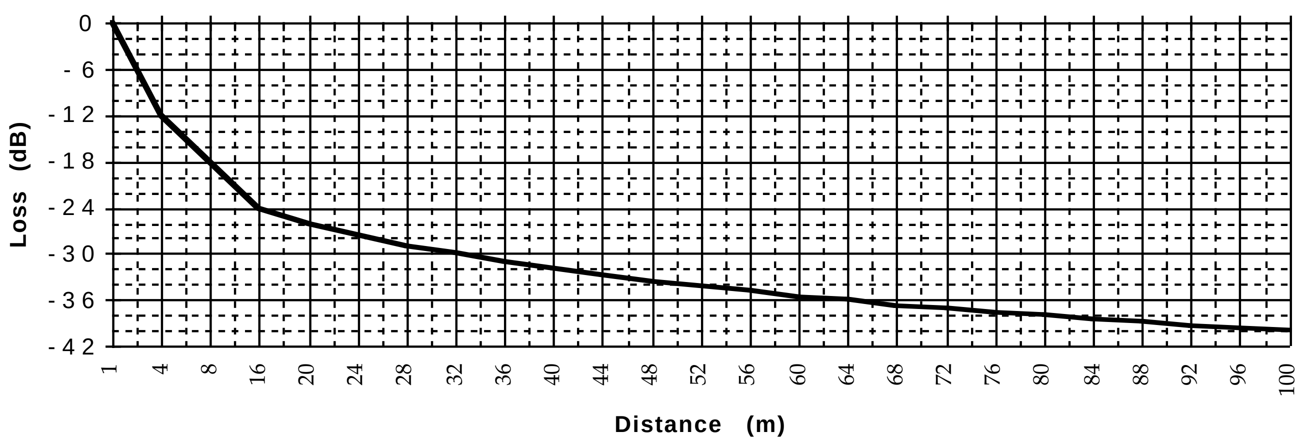


Fig 3.3a Inverse square law propagation loss.



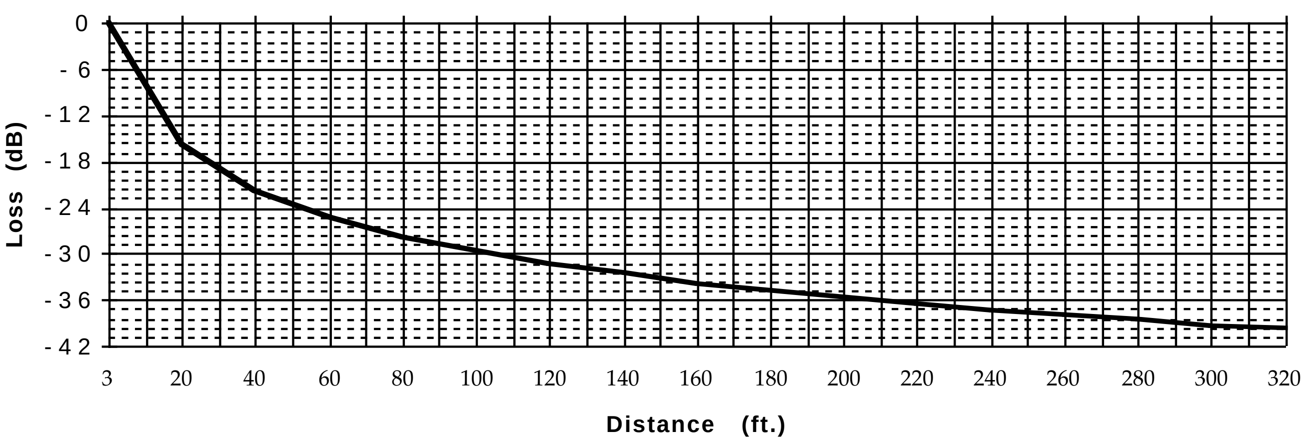
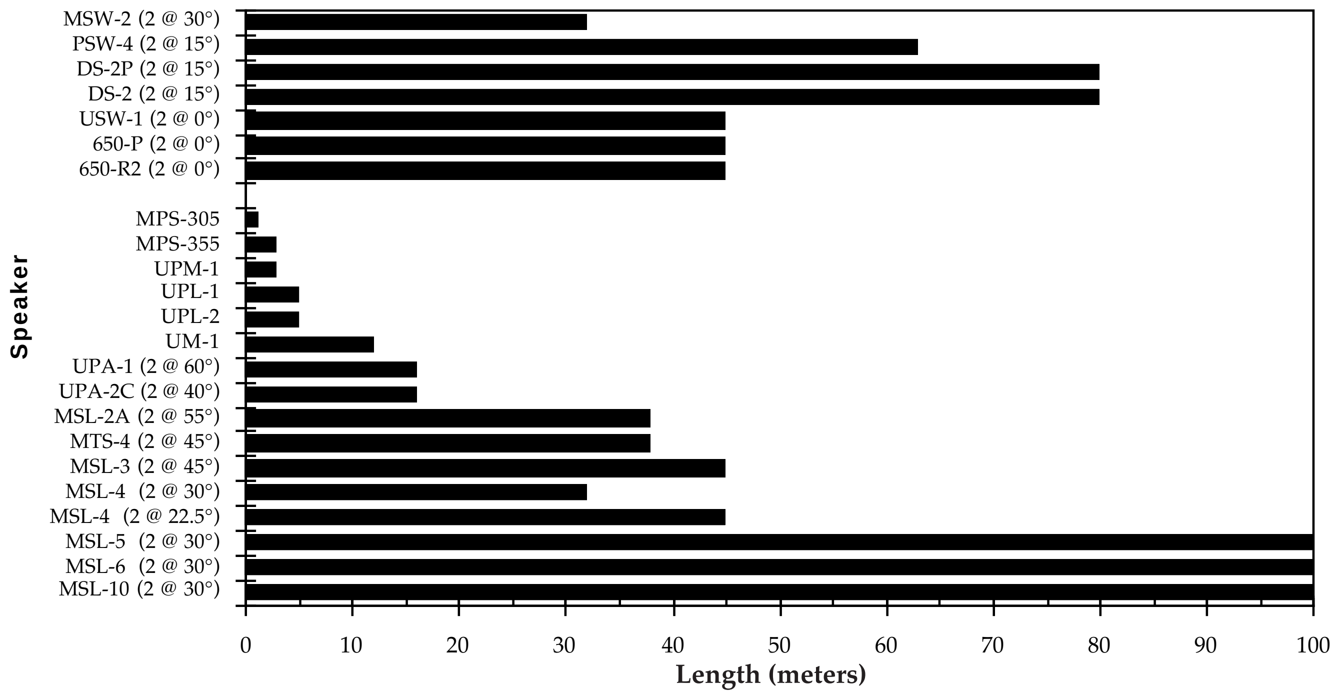


Fig 3.3b Propagation loss over distance (free-field) referenced to 1 meter SPL (in meters and feet).

3.3.2 Speaker Power Over Distance



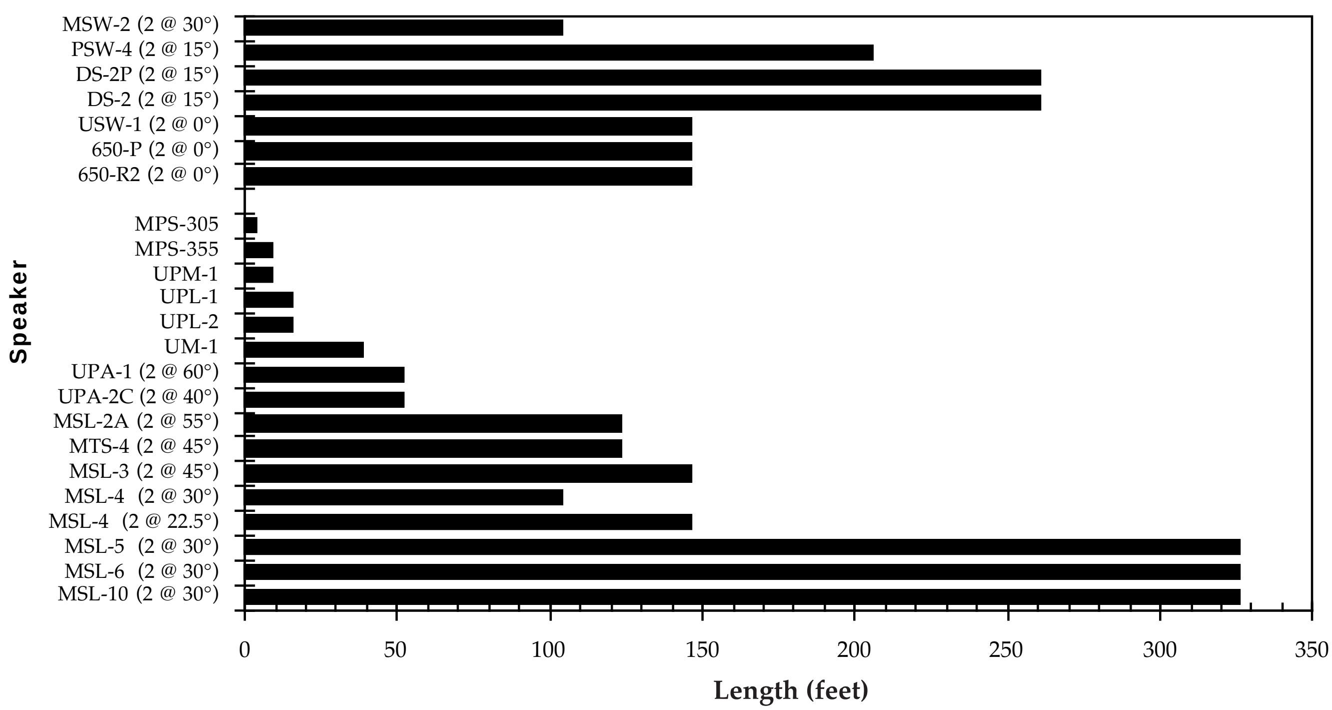


Fig 3.3c Meyer speaker SPL/distance reference (meters and feet).

Each model of speaker has a unique maximum power capability. This chart shows the *distance* at which the speaker's maximum SPL drops to 110 dB (peak) in one-half space loading. In most cases it is two speakers arrayed horizontally.

3.3.3 Power Capability Over Frequency

Maximum power capability and frequency response have a more complex relationship than first meets the ear. When referring to loudspeakers, the term "Frequency response" generally means the relative amplitude response of the system over frequency. Specifications give a range within which the response falls (± x dB) and LF and HF cutoff points (typically – 3 dB). Corrections for peaks and dips can, to some extent, be corrected with equalization.

Maximum sound pressure level (SPL), unless otherwise specified, describes the pressure a system can produce when driven simultaneously over its full passband. This, however, does not mean that it can reach that same maximum SPL at every frequency in the passband when driven individually. This is governed by its maximum power capability over frequency. Deficiencies in this response cannot be corrected by equalization and will probably require changes in gain structure or additional speakers.

Below is a simple example to contrast frequency range and maximum power capability.

A small tweeter can be equalized so that its frequency range extends to 30 Hz although it may require more than 40 dB of boost equalization in the LF range. However, this does not change the tweeter's maximum output capability at 30 Hz which is, of course, practically nothing.

The following example should help to illustrate the difference between these concepts:

The system used for stadium scale opera reinforcement shows might utilize the following cabinets:

6 MSL-5s (160 Hz–18 kHz)

12 DS-2s (60 Hz–160 Hz)

2 650-R2s (30–60 Hz)

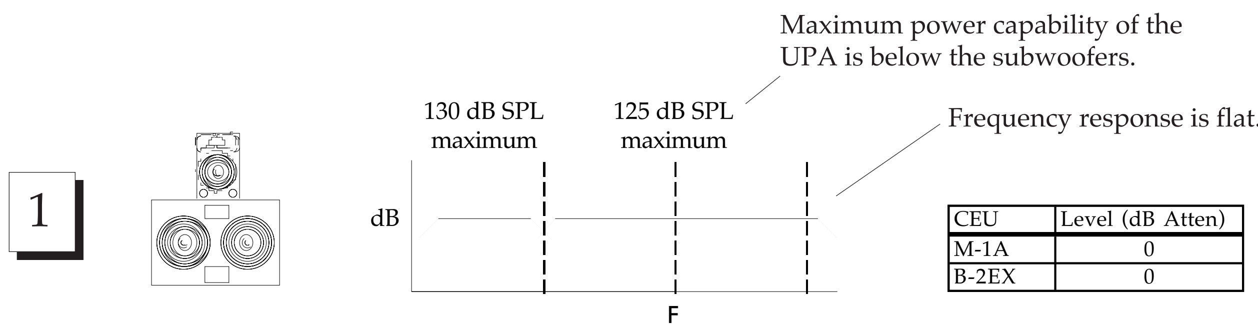
At first glance this system has a very low ratio of 650s to the other cabinets. Does this mean that the frequency response will show a dip in the area below 60 Hz? Not necessarily. In fact, the system may have a peak there, since the frequency response will be governed by the CEU and amplifier voltage gains as well as the ratio of cabinets and the room conditions. So while it is possible to set the frequency response flat, the power bandwidth is indeed very low in the area below 60 Hz. However, this system works very effectively because the low frequency power requirements of operatic music are minimal.

Rock music, on the other hand, has the majority of its power requirements below 250 Hz. This operatic system configuration would not work well for rock music because the low frequency power requirements would far exceed the capabilities of the two subwoofers.

Because of rock's extreme low frequency power requirements, systems are often aligned with an exaggerated low-frequency response. Even the operatic system could be aligned this way, but it still would not be suitable for rock because *this is not a frequency response issue*. An equalizer cannot make a system more powerful! If your system does not have the power where it needs it, it will run into distortion. When that happens, the frequency response hardly matters.

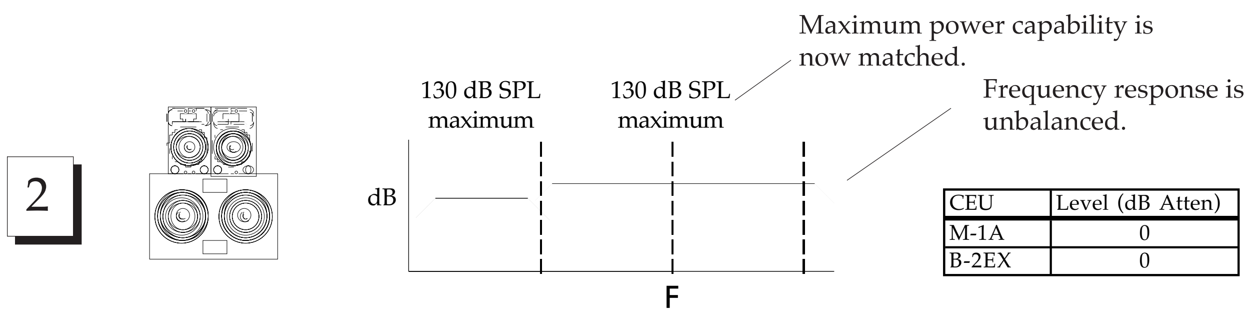
3.3.3 Power Capability Over Frequency

Here is an application example illustrating the difference between frequency response and power capability over frequency.



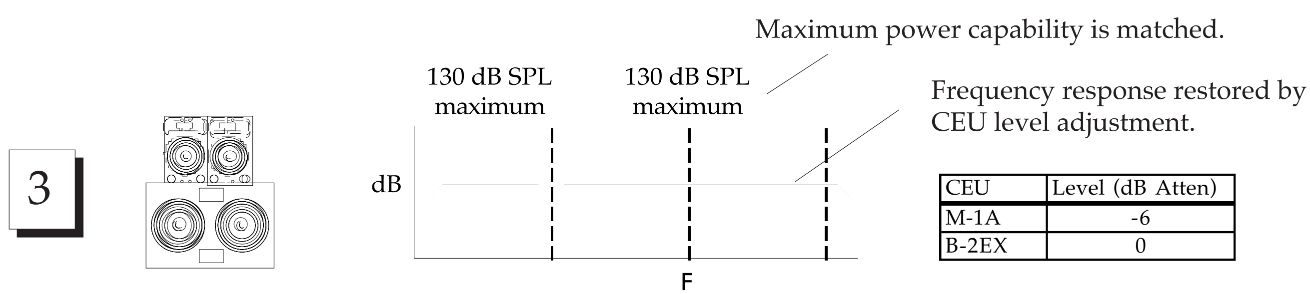
CEU	Level (dB Atten)
M-1A	0
B-2EX	0

A single UPA is placed on top of a 650-R2 subwoofer. CEU levels and amplifier voltage gains are matched, creating a combined flat frequency response. If the system is run to full power over a broad spectrum, the UPA will compress first since its maximum power capability is less than that of the 650.



CEU	Level (dB Atten)
M-1A	0
B-2EX	0

A second UPA-1C is added to the system, which increases the power capability in the MF and HF range. This creates an even maximum power capability over the full range. The coupling that produces the additional power capability also causes the frequency response to rise in the MF and HF range. This unbalancing of the frequency response will need to be remedied or the system will sound thin.



CEU	Level (dB Atten)
M-1A	-6
B-2EX	0

The most effective way to rebalance the system above is to turn down the M-1A CEU by 6 dB. This will restore the frequency response to its original response. The maximum power capability over frequency has been changed so that both the UPAs and subwoofers will reach their limits at the same time.

3.4.1 Coverage Angle and Distance

The traditional rendering of a speaker coverage pattern is a simple radial arc of the nominal angle between the speaker's –6 dB points as shown in Fig 3.4a. For simple designs and gross approximations of coverage, this is often sufficient. The simple radial arc is limiting in that it creates the appearance that all points along the arc are comparable in level and frequency response, giving the impression that all seats within the arc are well served and all areas outside of the pattern are unaffected. Alternatively, it leads to exaggerated fears of excess energy spilling onto side walls and ceiling. Both of these limitations ultimately affect the choice of speaker array, coverage angle and aim point, so it is worthwhile to investigate further.

The key concept is the relationship between propagation loss and coverage angle. For each doubling of distance from the source a 6 dB loss accrues. Movement from on-axis to the off-axis point accrues a 6 dB loss as well. In free field conditions the effect of these is quite different. The propagation loss attenuates all frequencies evenly, while the axial losses are greater in the range of frequencies that are directionally controlled by the speaker; usually the mids and highs.

Indoors, however, the propagation loss takes on a character very similar to the axial loss. As you move further into the room the rate of LF loss decreases, due to the coupling of the early reflections. The HF response continues to fall, creating an off-axis type character to the response.

This similarity can be used to maximize uniform level and frequency response.

Fig 3.4b shows the relationship between propagation loss and coverage angle. The farthest on-axis point is marked "A1". The mid point between the speaker and "A1" is the point marked "A2," which is 2x louder (+6 dB) than "A1." Points "B1" and "B2" represents a point equidistant to "A1" and "A2" respectively but at the axial edge (-6dB points) of the speaker coverage pattern. Notice that at "B2" the axial loss (-6dB) is compensated by the its closer proximity (+6 dB) resulting in the same level as at "A1."

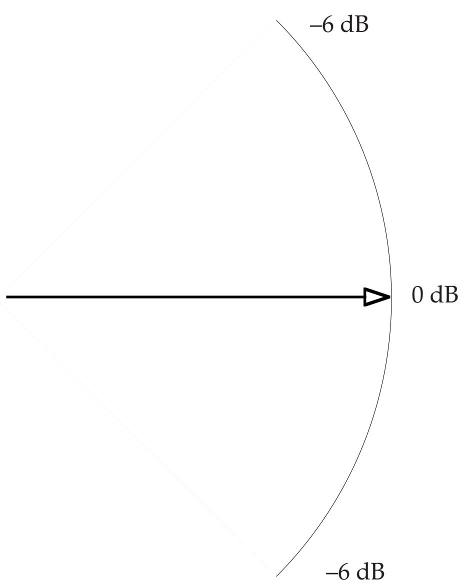


Fig 3.3a Simple 40° radial arc.

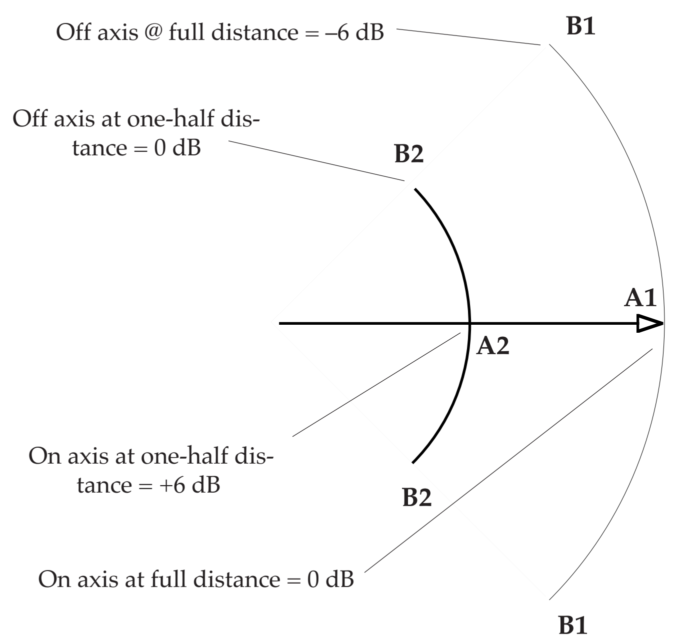


Fig 3.3b Relationship between propagation loss and coverage pattern.

3.4.1 Coverage Angle and Distance

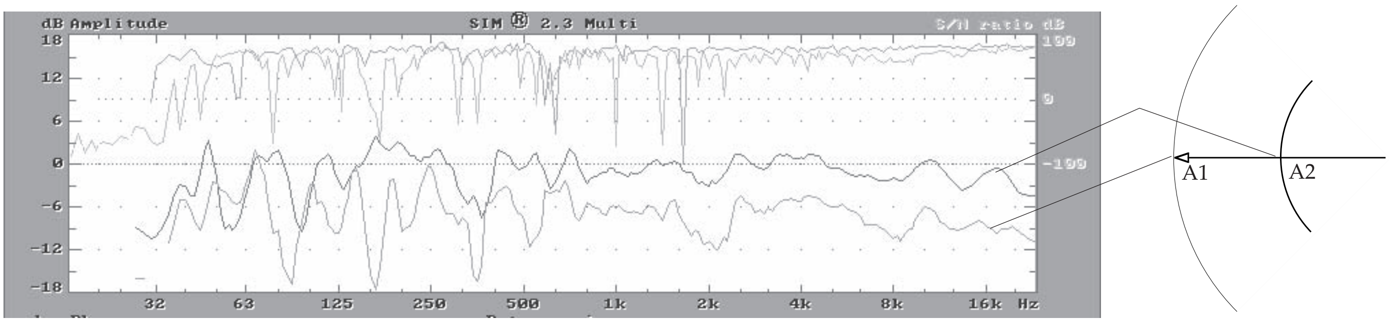


Fig 3.3c (1) Comparison of on-axis far (A1) and on-axis near (A2).

These two curves show a difference of 6 dB through the MF and HF region as would be expected from the inverse square law. The room reflections add energy in the LF range causing less than 6 dB attenuation. The extent of this effect will vary depending on the room acoustics and the directional control of the speaker.

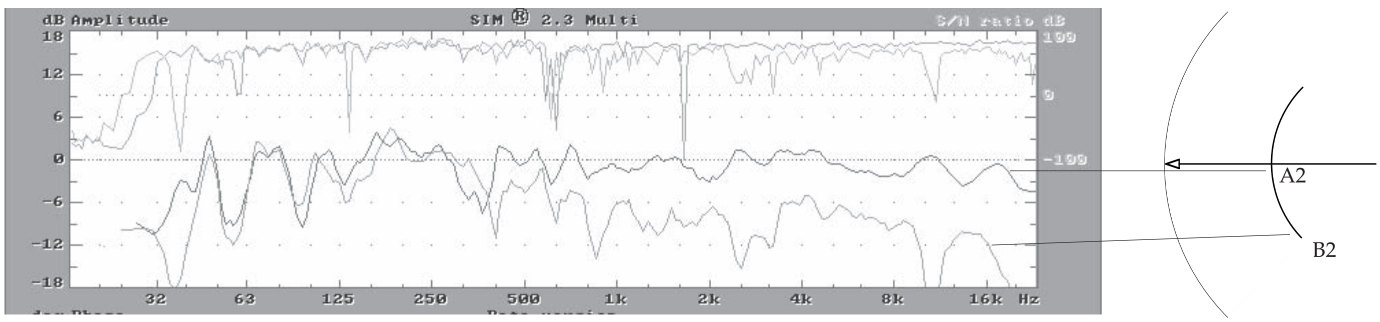


Fig 3.3d Comparison of on-axis near (A2) and off-axis near (B1).

These two positions are equidistant. The LF region has less directional control, therefore the energy is equal. The MF and HF regions are reduced in the off-axis response.

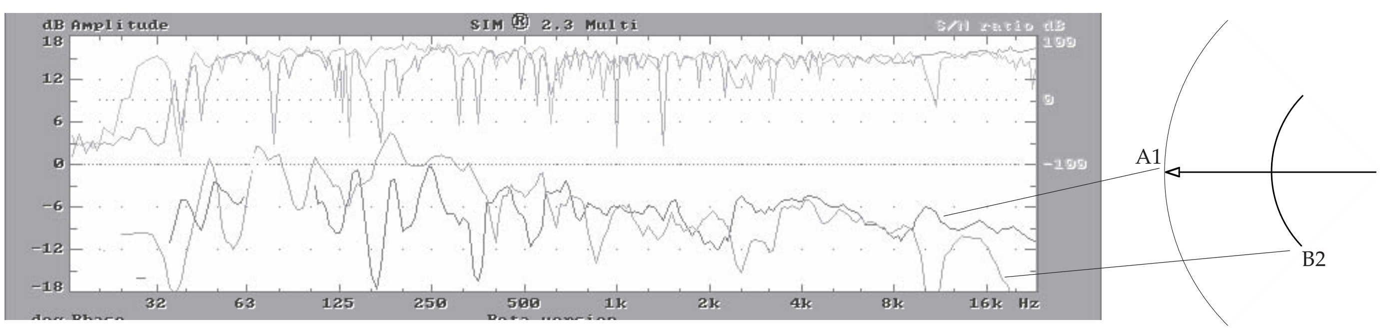


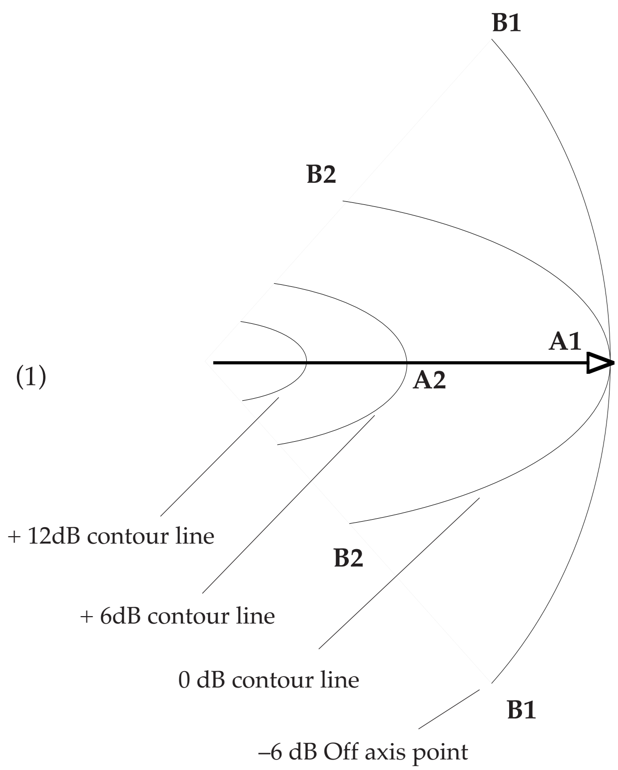
Fig 3.3e Comparison of on-axis far (A1) and off-axis near (B2).

These are the most closely matched of the responses. The HF attenuation at the off-axis position creates a similar response to the room, loading at the more distant on-axis position.

3.4.2 Equal Level Contours

We have found two points of equal level by closer examination of the coverage pattern, but we can go further. The line that connects the two points A1 and B2 (Fig 3.4f) represents the equal level contour, similar in concept to a topographical map. This "isobar" type approach can be viewed as a series of successive contours, each denoting a rise or fall in level. This is shown as a series of darkened shades in Fig 3.4f.

At the time of this writing Meyer Sound's research team is engaged in extensive research into isobar analysis. The approach utilizes the precise polar response measurements made in the Meyer Sound anechoic chamber and couples them to an algorithm that displays the equal energy contours of single speakers and arrays. Unfortunately, this data is too latebreaking to be included in this book. The isobar responses shown here are graphical approximations, rather than actual data.



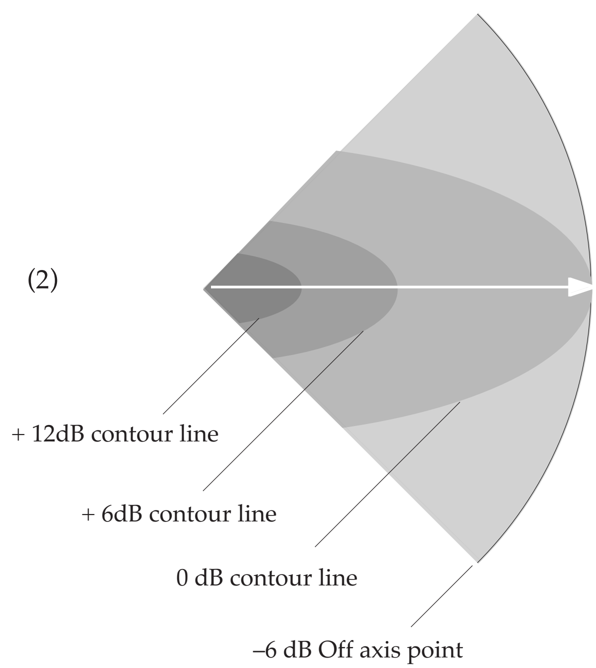


Fig 3.4f Isobar type rendering.

Equal energy contours form successive 6 dB lines inside the speaker's coverage pattern.

(1) Unshaded rendering to illustrate the relationship between distance, coverage angle and equal level contours.

(2) Shaded version. Each successive darkening represents a 6 dB rise.

3.4.2 Equal Level Contours

The isobar approach has the advantage of showing the shape of the energy coming from the speaker (or array) more clearly than a simple radial arc. As the pattern narrows the shape of the isobar "balloon" is squeezed into an elongated shape, as shown in Fig 3.4g. This illustrates the fact that highly directional speakers do not merely occupy a smaller radial arc but also have a sharper cutoff. The shape of the coverage balloon will have a deep influence on the choice of speaker model and its position. The first step is to ascertain the shape of the desired coverage area in the room. In its most basic form, the shape is taken from the "aspect ratio" of the intended coverage area. The aspect ratio is the ratio of length-to-width of the coverage area. To determine this, draw a box over the intended coverage area.

From this parameter you can begin to look for a coverage pattern match. If the aspect ratio is greater than 1:1, a point source array will be most suitable. If the ratio is less than 1:1, a split-parallel or split point-source array is best.

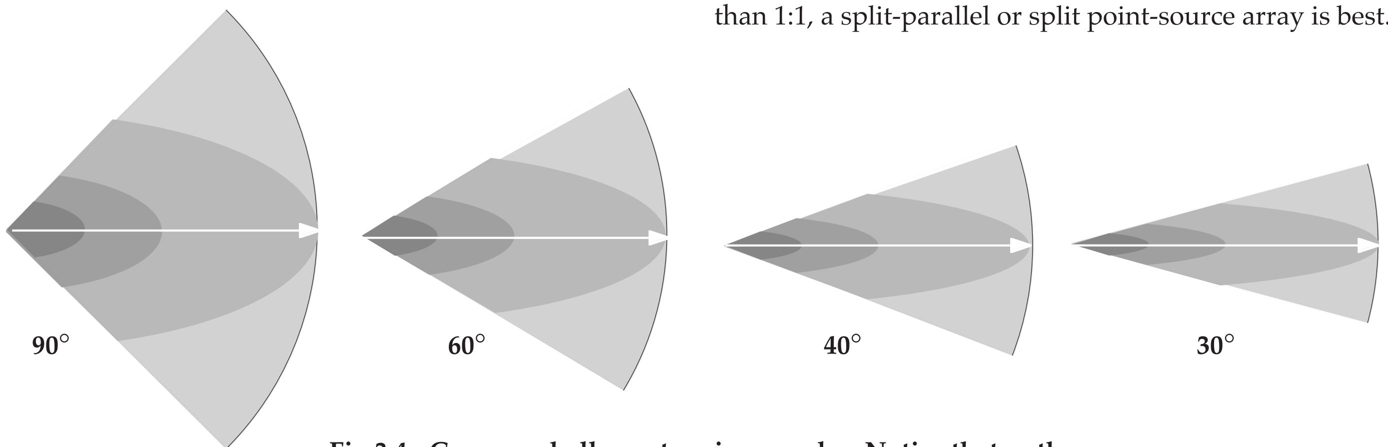


Fig 3.4g Coverage ballons at various angles. Notice that as the coverage narrows the cutoff becomes sharper.

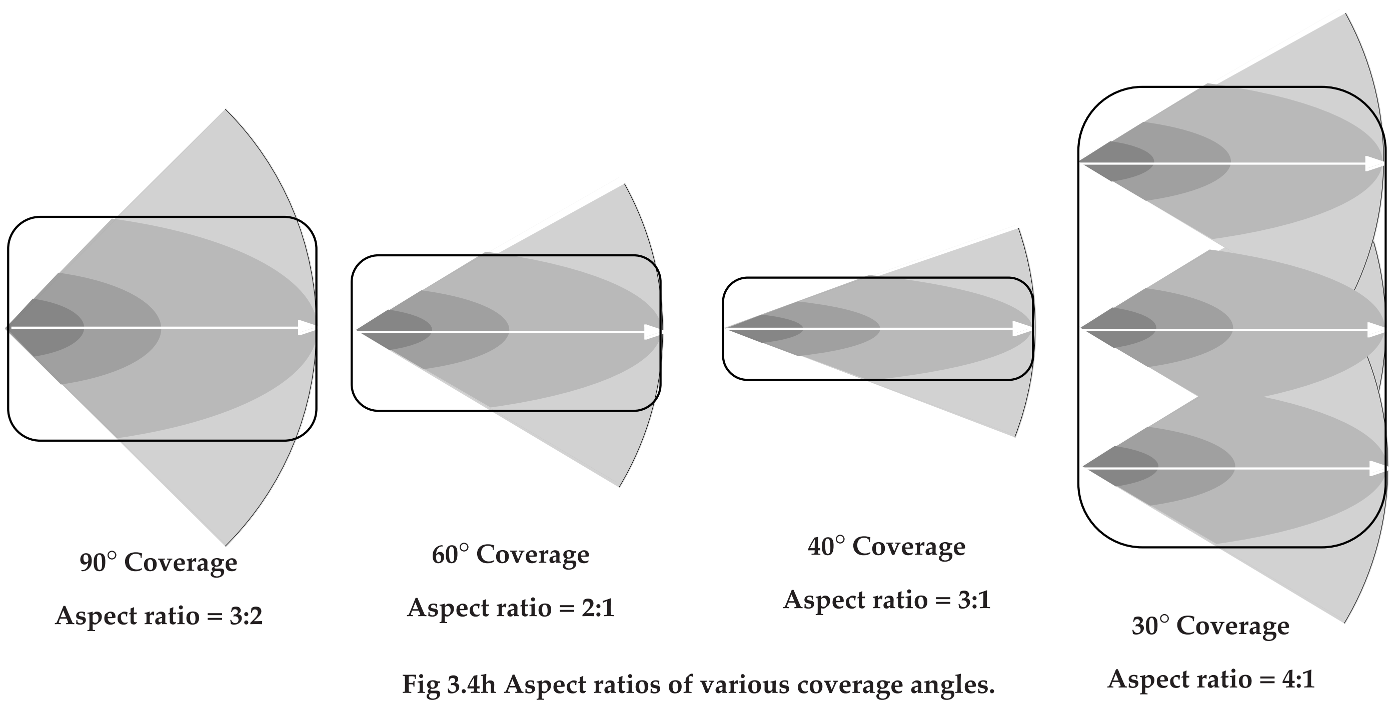


Fig 3.4h Aspect ratios of various coverage angles.

3.4.3 Speaker Placement

All speakers manufactured by Meyer Sound are designed for flat frequency response in a free-field acoustical environment. However, it is extremely unlikely that you will ever do a concert in free-field conditions. Every surface that reflects or refracts the sound waves emanating from a speaker will alter its frequency response. However, such effects can be minimized if the speaker positions are carefully chosen .

Free-field conditions are the most ideal from the standpoint of frequency response linearity, and the least ideal in terms of efficiency. By contrast, 1/8 space loading (two side walls and the floor) is exactly the inverse of the above.

Half-space loading occurs when the speaker is adjacent to a single boundary, such as the floor. The increase in efficiency gained in the low frequency range is known as "coupling." This is a common practice for subwoofer placement and works well for low frequencies. The period (1/Frequency) of low frequencies is long, and reflected energy from the boundary arrives nearly in phase with the direct signal. The reflected energy, therefore, adds with the direct, giving the system a higher efficiency in the LF region. However, as frequency increases, the period shortens and the reflected energy begins to fall behind the direct signal by more than 1/4 wavelength. The coupling then gives way to comb filtering. In practical applications the LF sections can be coupled to the floor and the main systems flown.

Some key aspects to speaker placement are:

- Position speakers to create a sonic image from the stage area.
- Avoid recessed positions where near-field HF reflections will occur.
- Keep the speaker away from near-field boundaries (particularly the HF horn).
- Try to avoid scrims or curtains in front of the speaker. If you must use scrims, get the most transparent cloth possible.
- Avoid redundant coverage by speakers. If you must, keep the time offset between such systems to a minimum. Do not create echoes by having large offsets and multiple sourcing.

Aim Points

The orientation of the speaker determines where its on-axis energy will be focused. Obviously, the intention is to focus the energy on the audience and away from reflective surfaces as much as possible. Once you have chosen the area that you want a particular speaker to cover, it becomes relatively simple to ascertain the orientation of the speaker.

As a starting point, calculate the edges of the area you intend to cover, horizontally and vertically, and the depth of field you want the system to throw. From this you can determine where the center axes are as well as the halfway point in the depth of field. The horizontal orientation is usually the most straightforward: simply aim the speaker into the center of the horizontal coverage area.

The vertical axis is complicated by the fact that the audience is usually closer to the bottom half of the vertical pattern than the top. If the vertical axis is aimed directly at the mid-point of the depth of field, the level will be noticeably louder in the front than the back. However, the vertical axial attenuation of the speaker can be used advantageously to help compensate for the vertical depth of throw differences. If the speaker is aimed above the depth of throw mid-point, the level will be more consistent. As you move closer to the speaker, the axial attenuation will lessen the effect of the SPL rise. As you move away the SPL loss will be decreased by the fact that you are receiving less axial attenuation. This is shown on the following page.

3.4.3 Speaker Placement

Here is a simple example to illustrate the speaker angle selection. Let's begin by looking at the area to be covered, as shown in Fig 3.4i. In this example, it is a given that a speaker with a 40° vertical pattern has been selected as the main system. The speaker aim point was selected by the radial arc method mentioned at the beginning of this section. The pattern reaches up to the last seat. The front area will need to be covered by a downfill speaker. Will it work?

Let's look at the coverage with the equal level contours as shown in Fig 3.4j. The middle seating area is on the +6 dB contour, while the rear area is in the –6 dB area. This 12 dB differential will be very noticeable. In addition the frequency response will be very different as the axial attenuation and room reflections will both be strong at the rear, which causes the system to have a large LF build up there. The central area would have neither of those factors, leaving its response relatively flat. For those who might be thinking that this angle is vital to preventing roof reflections, consider the strength of the floor reflections in this scenario.

Let's look at a second scenario where the on-axis point "A1" is positioned at the last seat in the hall as shown in Fig 3.4k. Notice that most of the seats in the hall are situated on the 0 dB contour line, right in the "sweet spot" of the speaker's coverage. As you move back, the distance loss is compensated by axial gain. Not only does the level remain constant the frequency response does as well. As the LF builds up in the rear, the HF comes up to meet it as you move into the axial center. You might be thinking that half of the vertical pattern is being wasted. It's true. In actual fact this will usually be the case unless the rake of the hall is extremely steep or they have seated people on the ceiling. If the ceiling is highly reflective in the high frequencies this optimal position may not be practical because the HF reflections will arrive back into the hall center. However, the people up top will suffer as a result. If the ceiling is only reflective in the lows and low mids, keep the angle up. The 10° to 20° variation in angle will make very little difference to the omnidirectional low frequencies. However, the loss of high frequencies at the rear will be extreme. One solution that I learned from Alexander Yuill-Thornton II is the placement of an eyebrow curtain above the speaker as shown in Fig 3.4l. This will keep the HF off the roof, without giving up the prime angle.

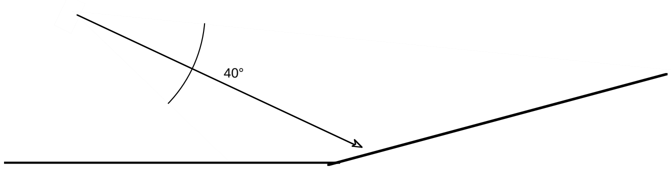


Fig 3.4i Vertical coverage by the radial arc method.

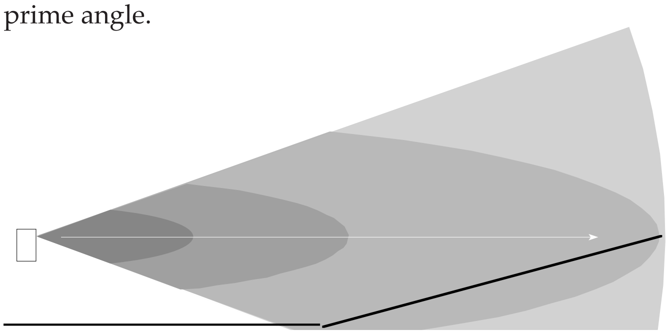


Fig 3.4k An elevation view of a speaker aimed for constant coverage. All the seats in the coverage area are near the 0 dB contour.

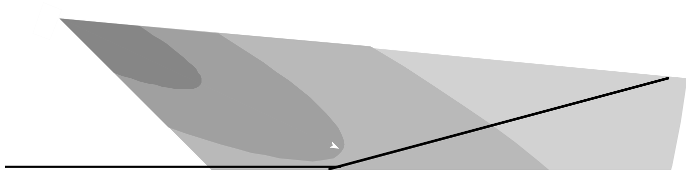


Fig 3.4j The speaker is aimed too low and will not achieve constant coverage. The level at the rear is 12 dB down from the main floor.

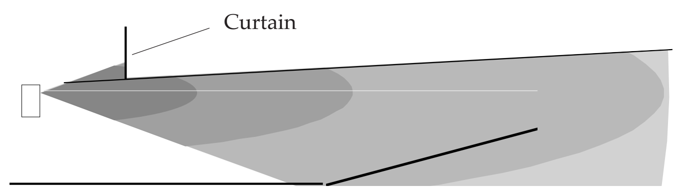


Fig 3.4l An eyebrow curtain prevents HF leakage onto the roof, while maintaining an optimal angle.

3.4.4 Theatre Coverage Example

Here is an example of typical orchestra floor coverage for a musical theatre. The system is for monaural vocal reproduction. The main system has a throw distance 110 feet. At the midpoint the coverage requirement is 45°. The aspect ratio is approximately 2.8:1, suitable for the 45° coverage UM-1C. The inner system is an inside fill, needing only to reach to the center. The throw is only 40 feet but the coverage angle requirements are similar. The UM-1C is chosen again. The frontfill and underbalcony delays have a very low aspect ratio so a split-parallel array is chosen.

The vertical coverage is typically broken into three levels: Orchestra, Mezzanine and Balcony. (In England these are referred to as Stalls, Circle and Balcony.) The coverage could be accomplished with a single central cluster, however, this causes the image to pull up too high. The proscenium-based orchestra and mezzanine systems keep the image down low. The systems are in close proximity and prone to overlap. Therefore, highly directional speakers such as the UM-1C are usually chosen.

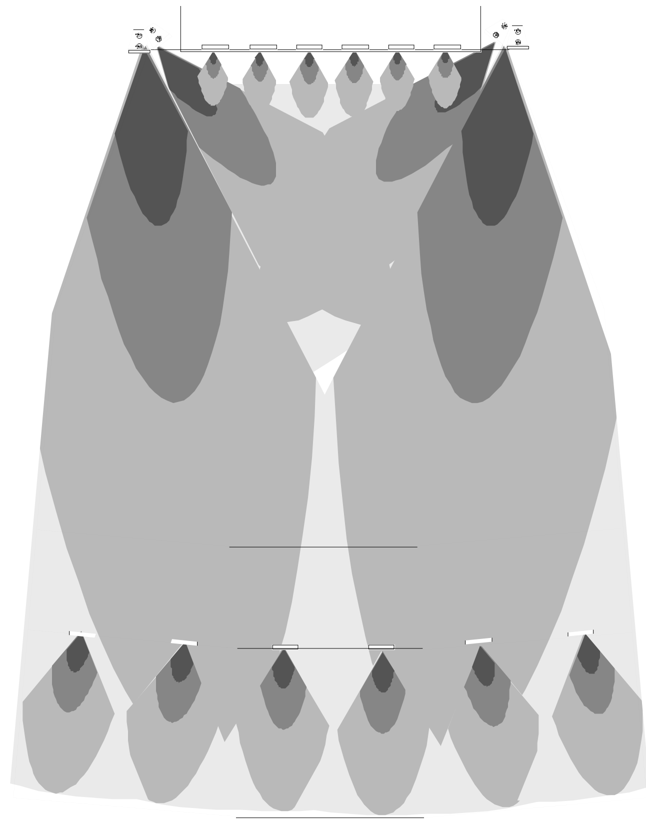


Fig 3.4m Plan view of an example of a musical theatre system.

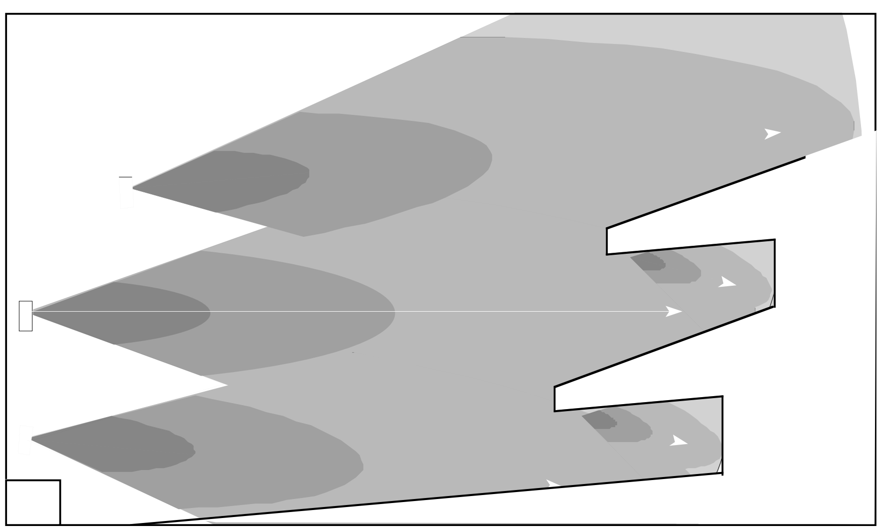


Fig 3.4n Section view of an example of a musical theatre system.

3.4.5 Arena Coverage Example

Here is an example of a typical arena setup for a pop concert. The mains are a point source stereo system (one side is shown). The main system has a throw distance of approximately 270 feet. At the midpoint the coverage requirement is 90°. The aspect ratio is approximately 3:2, suitable for three sections (90°) of MSL-5s, MSL-6s or MSL-10s. The outer system is a side fill, needing only to reach about half the distance of the mains. The coverage angle requirements are around 60°. The sidefill could be done with an additional pair of MSL-5s, MSL-6s or MSL-10s but they must have a separate level control to trim the level. The job could also be done with a lower power system such as the MSL-4 since it is half the throw distance. The frontfill and underbalcony delays have a very low aspect ratio so a split-parallel array is chosen.

The vertical requirements are best suited for a main system with 30° coverage. This makes the MSL-6 the best candidate. The downfill system could be MSL-5s or MSL-6s to cover the floor seating with CQ-2s to cover the front. The extreme front area is best covered with a split parallel array across the stage front.

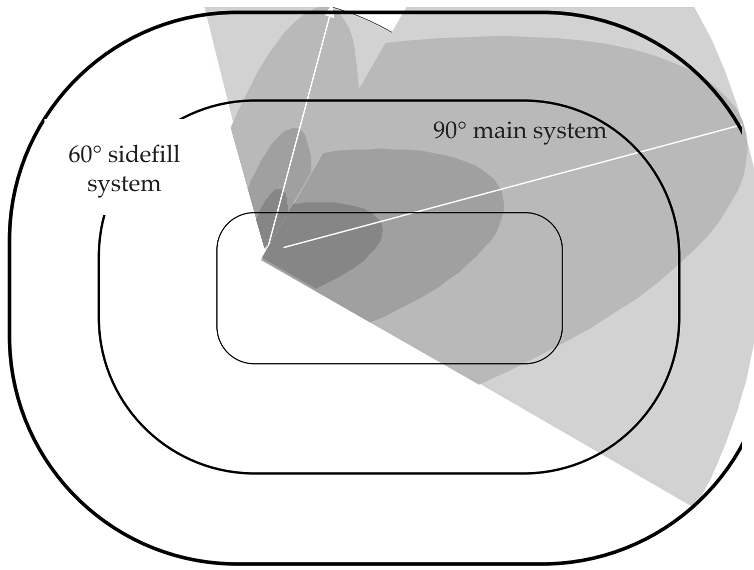


Fig 3.4p Plan view of an example of an arena concert system.

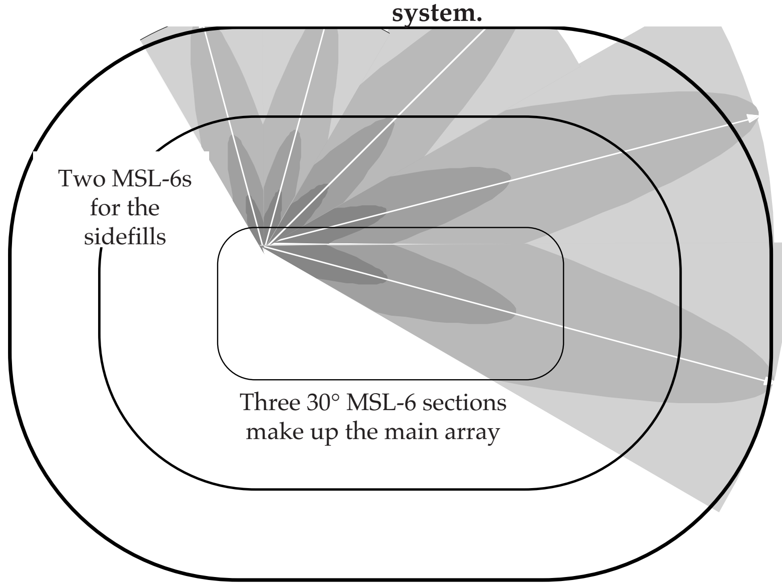


Fig 3.4q Plan view of an example of an arena concert system.

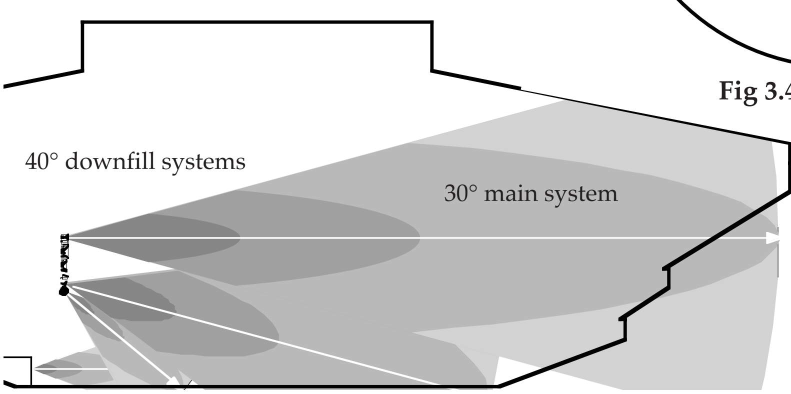


Fig 3.4r Section view of an example of an arena concert system.

3.5.1 System Subdivision

In an ideal world we might find ourselves with every speaker in our sound system having its own dedicated equalizer, delay line and CEU, and we have all of the time and tools we need to optimize each speaker's response. In the real world, however, we must be practical in our designs, working with limited resources, tools and time. Choosing the correct ratios is critical. Overly simplistic system designs may yield uneven coverage with no means of recovery, while an overly complex system may leave no time to complete the alignment in time for a show.

The basic signal flow of a speaker goes through the mixer, delay line, equalizer, CEU, amplifier and speaker. It is naive to believe that a single equalization curve aligned for one position will be beneficial over a large and varied listening area. If uniform frequency response and level are to be obtained, each distinct area will need to be adjusted separately. This approach to sound design is not new. Theatre sound designers have been implementing and expanding on these techniques for decades. Pop music professionals have been surprisingly slow to embrace this approach, citing time and budget constraints. This is changing, however, with the aid of SIM System II. The alignment of complex subdivided sound systems can now be done in a very practical time frame and the results are unsurpassed.

System subdivision is appropriate when you have:

- Separate channels (e.g., stereo).
- Distinct vertical coverage areas (e.g., downfill arrays).
- Distinct horizontal coverage areas (e.g., sidefill arrays).
- Physically separated speaker systems (e.g., frontfill or delay systems).
- Differences in depth of throw (e.g., a center cluster with a longer throw to the back than to the sides).
- Differences in the acoustical environment of the listening area (e.g., absorptive rear with reflective side walls.)

Subdivision Levels

System subdivision is an incremental process with several levels of complexity, each providing more options to optimize the combined response.

Separate Speakers

With distinct enclosures you have the option of adjusting their relative position. This allows for coverage angle adjustment by changing the splay angle, and allows for time offset adjustment by physical placement.

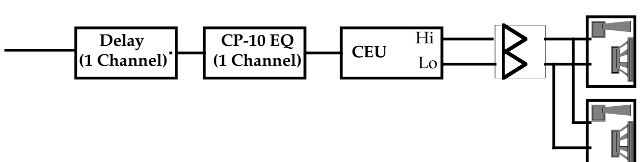


Fig 3.5a

Separate speakers give coverage angle adjustability.

Separate Amplifiers

This option opens up the possibility of adjusting the coverage angle electronically by amplitude tapering of the amplifier level controls. This can be very effective when part of an array must throw farther than another. It is also useful for reducing the interaction between speakers in larger arrays. (See Section 3.6.3, Amplitude Tapering.)

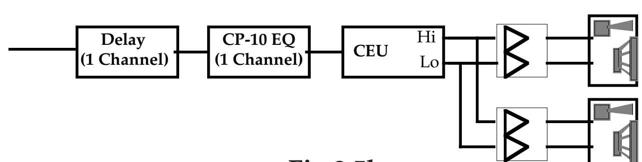


Fig 3.5b

Separate amplifiers allow for amplitude tapering.

3.5.1 System Subdivision

Separate Equalizer and CEU

This level opens up the possibility of enacting equalization separately for subsystems. This is essential when subsystems are physically separated since they will be operating in distinct acoustical environments. (The exception being when systems are symmetrical opposites or identical.) For example upper and lower proscenium side systems will be aimed toward the balcony and floor, respectively, requiring different equalization for their environments. It is usually recommended that vertical arrays be broken into separately equalizable subsystems for each row. Although horizontal seating in most applications is fairly gradual in depth, vertical seating tends to have distinct break points, creating the need for a separate adjustment.

Separate Delay Line

Delay lines are mandatory for distributed systems such as underbalcony delays. Delays are also used to synchronize systems such as frontfills to the live acoustic source being reinforced. In addition to these examples are also more subtle reasons, such as the synchronization of horizontally and vertically splayed array subsystems. See Section 3.7.1, Downfill Arrays for an example of this.

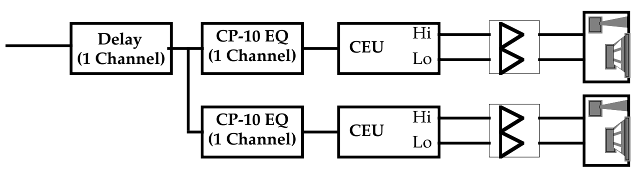


Fig 3.5c

Separate equalizers allow for independent frequency response correction.

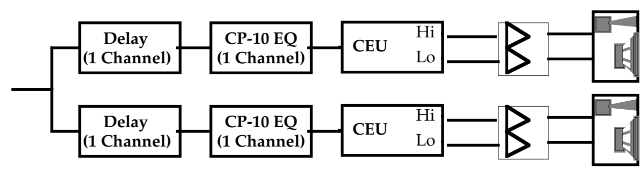


Fig 3.5d

Separate delays allow for time offset compensation.

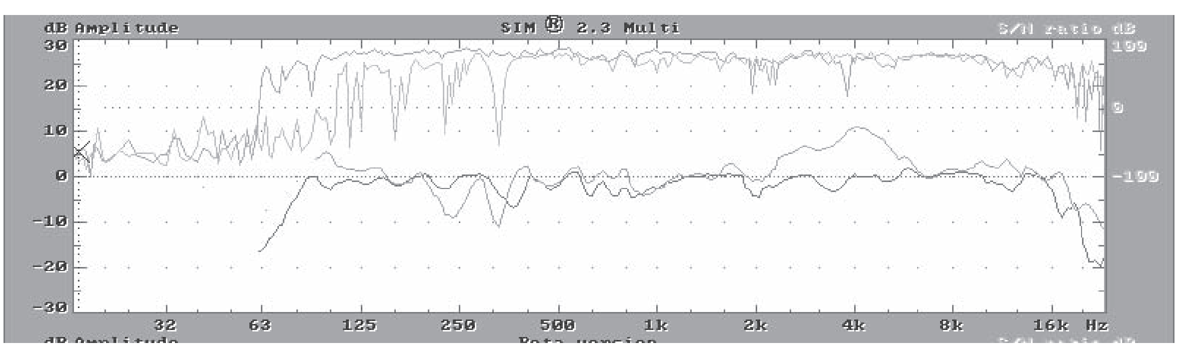


Fig 3.5e This is the amplitude response of two separate systems at their respective on-axis locations before equalization. The responses are not matched in the 4 kHz range. Therefore, separate equalization should be applied.

3.6.1 Main Arrays: Introduction*

Speakers such as the UPA, MSL-2A, MSL-3, MSL-4, MTS-4, DS-2, MSL-5, MSL-6 and MSL-10A are designed for use in multiple unit configurations known as arrays. These models are easily identified as arrayable speakers by their trapezoidal enclosure design. This design concept, introduced by Meyer Sound in 1980 with the patented UPA-1 speaker, helps with the mechanical aspects of constructing arrays. Prior to the introduction of the UPA, typical sound designs consisted of multiple speaker elements stacked in rows and columns with many of the drivers having redundant orientation. While this type of array can produce large amounts of acoustical power, it has the disadvantage of creating an uneven frequency response which is highly variable with position. The principal behind the enclosure design is that the elements of the array are aligned in an arc, combining to create an array that acts like a point source, or, to be more precise, a section of a point source. (A point source is a radiating spherical surface. Omnidirectional radiation is rarely practical in real-world sound reinforcement.) Arranging full-range cabinets into arc formations, if the cabinets are consistent in frequency and phase response, creates a "phantom" focal point some distance behind the array, thus approximating a point source.

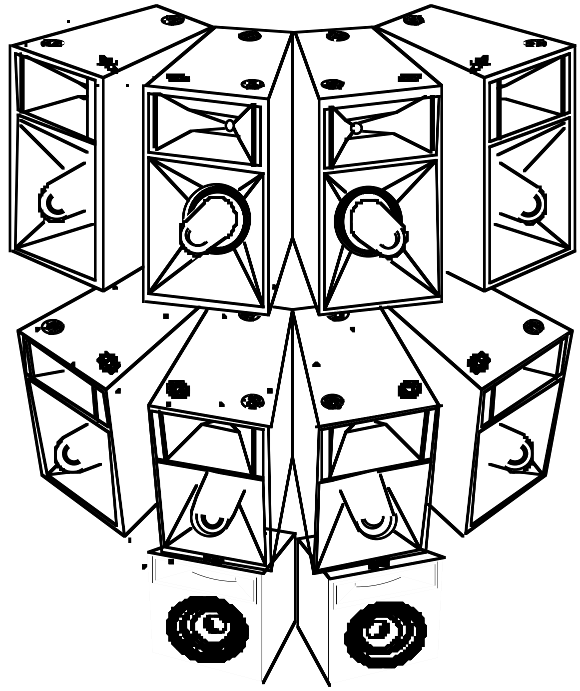


Fig 3.6b Point-source Array.

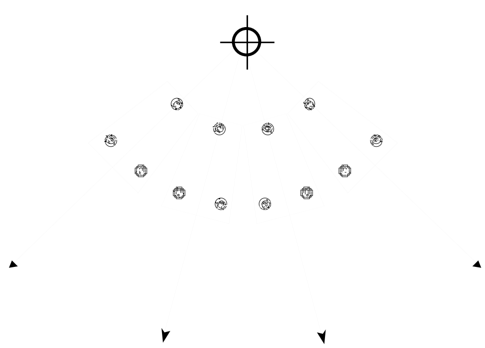


Fig 3.6a Horizontal point-source array with the focal point behind the speakers.

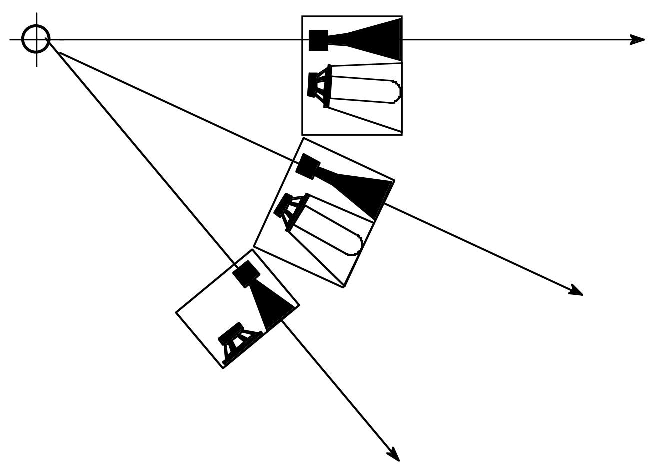


Fig 3.6c Vertical point-source array with the focal point behind the speakers.

*The text on this page was written by Ralph Jones.

3.6.1 Main Arrays: Introduction

Having the focal point behind the array has the advantage of reducing interaction between speakers, creating a smooth uniform coverage pattern over the listening area. However, creating smooth, controlled arrays is not as simple as cutting angles onto an enclosure. In fact, the trapezoidal shape has no effect on the polar pattern of the speaker, serving only as a mechanical aid to create the optimal angles for the array.

The two primary factors in array performance are *coverage angle* of the speakers and *splay angle* between the enclosures. As a general rule, as the splay angle (center to center) approaches the coverage angle, the smoothest coverage will be obtained with the least interaction. However, this is made more complex by the fact that while the splay angle is a simple fixed constant, the coverage angle varies over the frequency range of the device. The coverage angle increases as the frequency decreases, leaving the array more interactive at lower frequencies.

You might wonder why Meyer Sound would design speakers with a coverage pattern wider than the enclosure design. This was done intentionally on models such as the UPA-1, MSL-2A and MSL-3A in order to provide the maximum flexibility for use. When the pattern is significantly wider than the enclosure angle, and units are placed adjacent, the coverage pattern may expand only slightly or actually narrow while greatly increasing the on-axis power. This is particularly true of the MSL-3A, which has a horizontal pattern that is less for three cabinets in a tight-pack array than for a single unit. However, the on-axis maximum SPL is almost 10 dB louder for the three-unit array.

Meyer Sound could have chosen to make the enclosure angle equal to the coverage angle, but this would take away the option of creating high-power arrays from multiple compact enclosures. It would also make for a challenging truck pack.

Coverage Angle and Enclosure Shape

• The coverage pattern of the speaker is not necessarily the same as the angle described by the enclosure.

• The angle of the trapezoid constitutes the *minimum angle* for multiple speaker units—*not necessarily the optimum* angle in all respects.

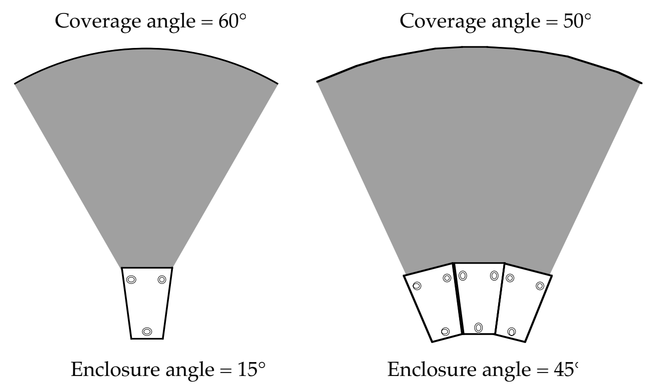


Fig 3.6d MSL-3A coverage pattern narrows while on-axis power increases as additional units are packed together. Use this configuration for long-throw applications.

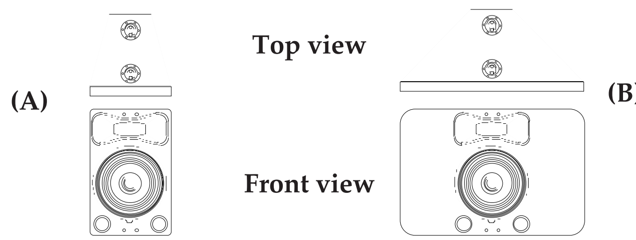


Fig 3.6e UPA-1C Enclosure design: (A) actual; (B) if the enclosure angle were equal to the coverage angle.

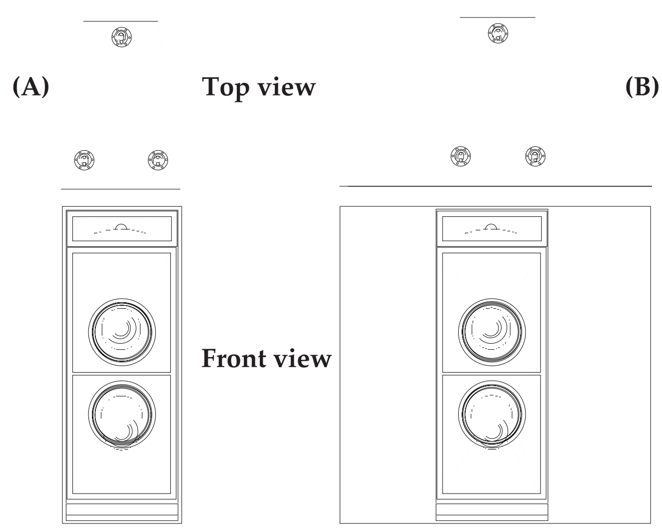


Fig 3.6f MSL-3A Enclosure design: (A) actual; (B) if the enclosure angle were equal to the coverage angle.

3.6.2 Splay Angle and Coverage

The interaction between speakers in arrays was discussed in detail previously in Section 2. This section details the tendencies of point-source arrays toward coupling, combining, and isolating, depending upon the splay angle between cabinets. Array design is a compromise between these effects and depends upon the application. For example, heavy metal rock music is much more concerned with on-axis power than smooth frequency response. The desired design for heavy metal would be a highly interactive array with lots of coupling. On the other hand, the top priorities for classical music sound reinforcement are smooth frequency response and maximum coverage uniformity. The desired design for classical music is well isolated speakers with minimal interaction.

Array Design Tradeoffs

Coverage: As overlap increases, the coverage narrows and vice-versa.

On-axis SPL: As overlap increases, on-axis SPL increases significantly. As overlap decreases the on-axis SPL does not increase much.

Level distribution: As overlap increases, level distribution becomes uneven, most notably in the form of hot spots in the center area. As overlap decreases, the level distribution becomes smoother.

Frequency response distribution: As overlap increases, the frequency reponse distribution becomes uneven primarily due to comb-filtering. This results from phase cancellation due to the multiple arrival times of the different drivers in the listening area. As overlap decreases the frequency response distribution becomes smoother due to decreased comb-filtering.

Equalizability: Virtually any array is equalizable at a single point. But if we can assume that the intended goal is to provide an equalization curve that is suitable for a wide part of the coverage area, arrays with even distribution patterns will respond best.

Coverage Angle of Horizontal Arrays

It would be very handy if the coverage angle of an array could be calculated by simply adding the splay angle of each additional unit to that of the first unit. Such a scheme is shown below. Unfortunately this is only true when the "isolating" splay angle is used. If the speakers are coupled close together the splay angle may be much narrower than even a single unit, while the on-axis power increases greatly. A compromise "combine" position falls in between, with increased on-axis power but a wider pattern.

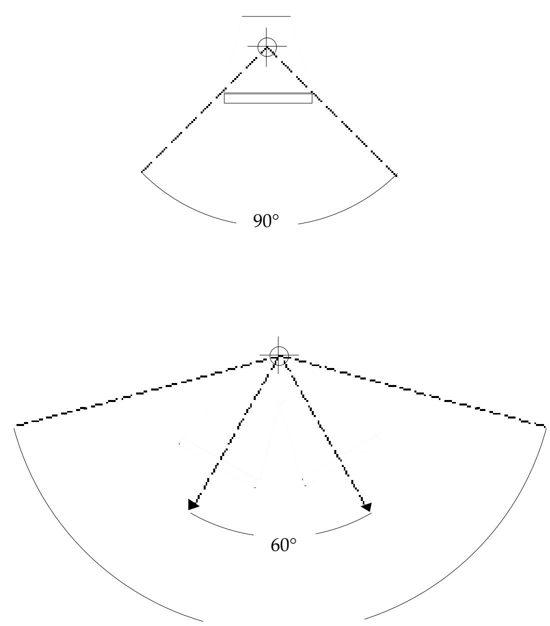


Fig 3.6g

Calculating coverage for isolated horizontal arrays.

The above example shows the coverage pattern when speakers are combined for minimal overlap. The array angle is the sum of the coverage angle of one speaker plus the splay angle of each additional unit.

Caution: This calculation is not valid for narrow (coupled) arrays.

3.6.2 Splay Angle and Coverage

Coupled, Combined and Isolated Point-Source Arrays

The following chart shows guidelines for splay angles of each speaker model to create arrays for coupling, combining, and isolating respectively. This is an aid for selecting the optimal splay angle between cabinets for your application. For applications where smoothness of coverage is the most critical parameter, the "isolate" angle will be best. Where brute force on-axis power is the overriding concern, select the "coupling" angle. The "combine" angle provides a compromise value in power and response uniformity.

Point Source Array Angle Reference				
	Enclosure	**Couple**	**Combine**	**Isolate**
On axis addition (approximate)		4-6 dB	2-4 dB	Minimal
Interaction		High	Moderate	Low
UPA-1C	30°	30°	45°	60°
UPA-2C	30°	30°	35°	40°
MSL-2A	30°	30°	40°	55°
CQ-1	20°	40°	50°	60°
CQ-2	20°	20°	30°	40°
MSL-3	15°	15°	30°	45°
MTS-4	15°	15°	40°	55°
MSL-4	15°	15°	22.5°	30°
MSL-5, MSL-6, MSL-10A	30°	-	-	30°
SB-1 (Soundbeam)		4°	6°	8°

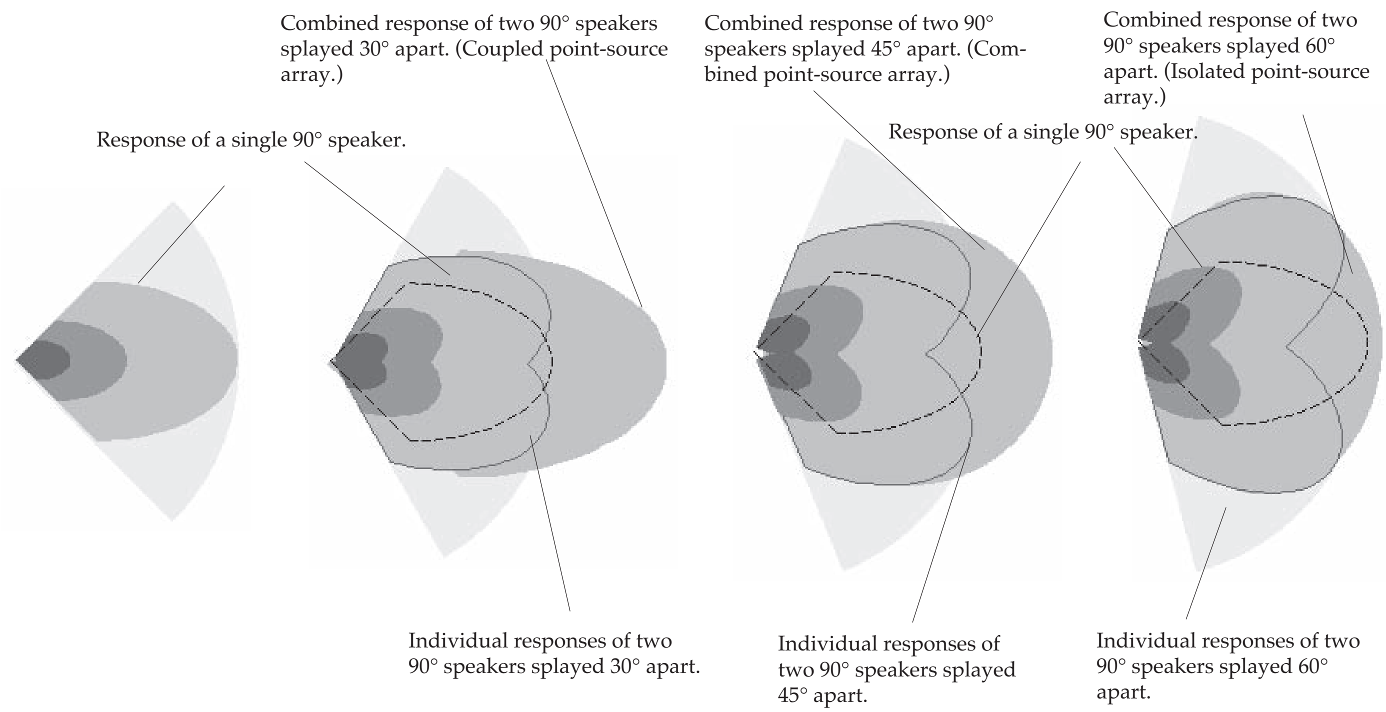


Fig 3.6h Coupled, combined and isolated point-source arrays.

3.6.3 Amplitude Tapering (Adjusting Coverage Angle Electronically)

There are two basic means of adjusting the coverage angle of an array of speakers: mechanical and electronic. The mechanical approach consists of splaying apart the cabinet fronts as described in the previous chapters. But after the array has been hung and tied off, the mechanical option is no longer available. However, the coverage angle can be adjusted electronically by modifying the relative drive levels of the array components. This process, known as *amplitude tapering* is particularly effective in larger tight-pack horizontal arrays of cabinets such as the UPA-1, MSL-2A, and especially the MSL-3A, where it expands coverage, frequency response and level distribution.

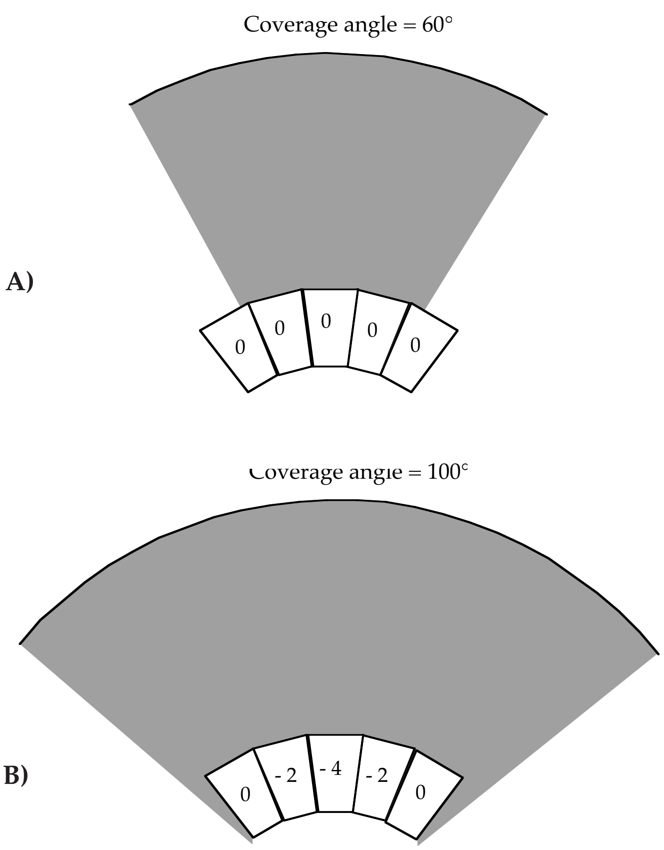


Fig 3.6i The effect of amplitude tapering on a tight-pack MSL-3A array.

A) Coverage angle narrows with all speakers at same level.

B) Coverage angle widens as inner and center speakers are turned down by 2 and 4 dB, respectively.

Example

An array of five MSL-3A loudspeakers is configured in a row with all the cabinets adjacent. With all the speakers driven at the same level, the –6dB points are 60° apart, creating a tight, relatively long-throw horizontal coverage pattern. The current venue requires coverage out to 100°. This can be accomplished by reducing the inner pair and center speakers by 2dB and 4 dB, respectively, as shown in Fig 3.6h.

To widen the coverage pattern the drive level to the center speakers are attenuated relative to the outer ones. This reduces the energy in the center overlap zone, thereby reducing the on-axis bulge and widening the angle between the 6 dB down points. This will make a tight-pack array of the wide cabinets (UPA-1, MSL-2A, MSL-3A) behave more like a narrow optimized system such as the MSL-5 or MSL-10A. The attenuation does reduce the on-axis power slightly but the improvements in the system's response make it worthwhile.

It may surprise you that while reducing the center cabinet widens the coverage, reducing the outside speakers does not necessarily do the opposite. Reducing the outside speakers also reduces the energy in the overlap zone, again reducing the bulge in the center. As the side speakers are attenuated the pattern will begin to take the shape of a single unit. It is usually unadvisable to widen the coverage by reducing the sides because the power capabilities of the center speaker may be compromised.

Amplitude tapering can be done at either the CEU level controls or at the amplifiers (provided that the Hi and Low sections are attenuated together). Note that amplitude tapering has a limited scope. Steps of 2 dB for adjacent cabinets have proven to give good results. Reduction of greater amounts effectively removes the speaker from the array, can leave coverage holes and reduce system power. The addition of greater amounts makes the speaker stand out and carry the bulk of the power requirements of the array, and can reduce overall system power and increase distortion.

To take advantage of amplitude tapering you must:

- Have a preestablished standard voltage gain from which to refer.

(See Section 1.4.2, Amplifier Voltage Gain)

3.6.3 Amplitude Tapering

Horizontal Amplitude Tapering

Arrays can be designed to take advantage of amplitude tapering. The key is the configuration of amplifier channels driving the speakers. Generally speaking, setting up a system for amplitude tapering requires only a repatch of amplifier inputs and outputs. In some cases, however, this will result in an increase in the number of amplifier channels required, but the results are worth the expense.

The following examples show strategies for driving arrays of various sizes.

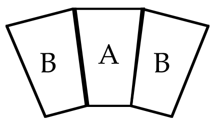


Fig 3.6j

Amplitude tapering with two speaker arrays.

Separate drive for the speakers for situations where the intended coverage area is not symmetric and/or equidistant between the two speakers.

Fig 3.6k

Amplitude tapering with three and four speaker arrays.

Separate drive for the inner (A) and outer (B) speakers allows for array amplitude tapering. For wide coverage reduce A. For narrow long throw keep all systems at the same level.

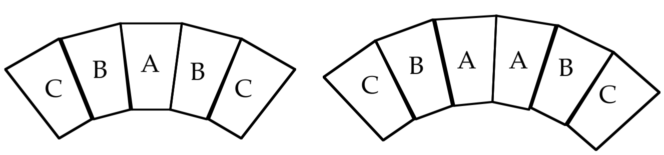


Fig 3.6l

Amplitude tapering with five and six speaker arrays.

Separate drive for the center (A), inner (B) and outer (C) speakers allows for array amplitude tapering. For wide coverage reduce B and A by 2 dB and 4 dB, respectively. For narrow long throw keep all systems at same level.

Vertical Amplitude Tapering

This works under the same principles as described above for horizontal array tapering. Amplitude tapering is particularly important for vertically arrayed systems because in most cases the audience is significantly closer to one of the systems.

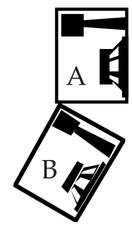


Fig 3.6m Vertical amplitude tapering.

Separate drive for the speakers for situations where the intended coverage area of the two speakers is not symmetrical or equidistant. Typical in vertical arrays because seating is usually closer to the lower systems.

Long-Throw Vertical Arrays

Note: Amplitude tapering is *not* recommended for long-throw vertical arrays. Long throw vertical arrays rely on equal level at both horns in order to create proper coupling.

> **TechNotes™**
>
> Meyer Sound has published a series of technical notes describing the array behavior of various configurations of UPAs, MSL-2As and MSL-3As. Specific figures for horizontal amplitude tapering of tight-pack arrays are included in TechNotes. Contact your Meyer Sound dealer to receive your copy of TechNotes™

3.6.4 Array Coverage Patterns

All 90° coverage patterns are not created alike as shown in Fig 3.6m. The rate of cutoff (how quickly the pattern moves from 0 dB down to 6dB and 10 dB) is a function of the horn geometry and the coupling of arrays. The following graphs contrast the cutoff rate of three 90° clusters.

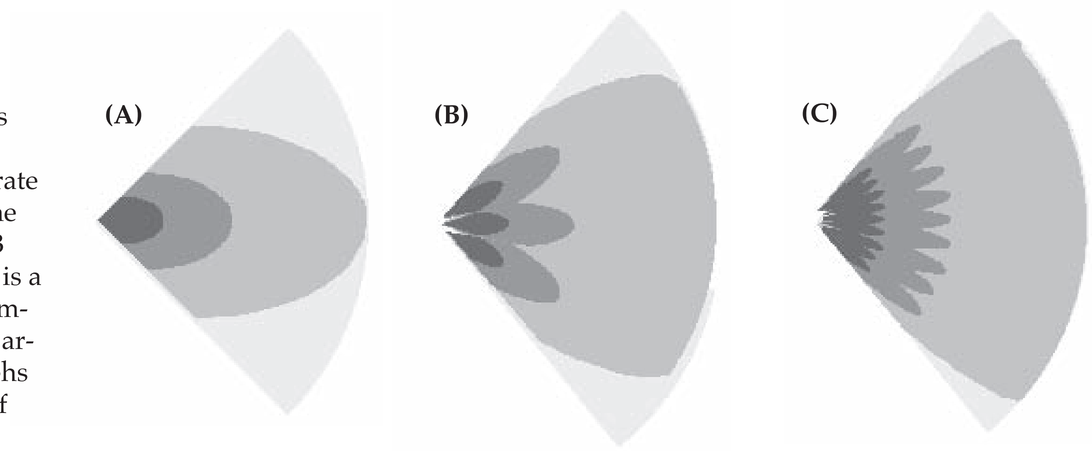


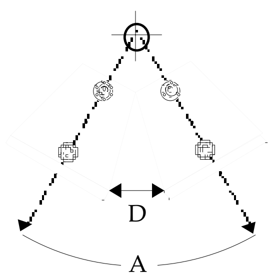
Fig 3.6n 90° coverage patterns as derived from single and multiple units.

A) 90° coverage from a single UPA-1C. The loss from on-axis to off-axis is very gradual.

B) 90° coverage from an array of three MSL-4s. The three narrow horns cause the edge of the pattern to have a steeper cutoff.

C) 90° coverage from an array of three MSL-5, MSL-6 or 10A sections. The nine extremely narrow horns cause the edge of the pattern to have an incredibly steep cutoff.

3.6.5 A Simple Way to Verify Splay Angle



In many situations it is impractical to use a protractor to verify that you have achieved the desired splay angle. Reference chart 3.6n allows you to determine the splay angle by measuring the gap between the speaker fronts. It is assumed that the rears are adjacent. The gap (D) for each angle (A) is shown in inches and centimeters for each speaker model. For cabinets with a protruding front grill (such as UPAs, MSL-2A and CQs) the gap is measured from the wood edge, not the grill edge.

	MSL-3A, MSL-4, DS-2 PSW-4,MTS-4		CQ-1, CQ-2		MSL-2A, MSW-2		UPA 1&2	
"A"	"D" (in)	"D" (cm)	"D" (in)	"D" (cm)	"D" (in)	"D" (cm)	"D" (in)	"D" (cm)
15°	0.00	0.0						
20°	3.00	7.6	0	0.0				
25°	5.00	12.7						
30°	8.00	20.3	3.5	8.9	0.00	0.0	0.00	0.0
35°	10.50	26.7			1.50	3.8	1.00	2.5
40°	13.00	33.0	7.25	18.4	3.50	8.9	2.50	6.4
45°	15.50	39.4			5.00	12.7	3.50	8.9
50°	18.25	46.4	10.75	27.3	6.50	16.5	5.00	12.7
55°	20.50	52.1			8.25	21.0	6.00	15.2
60°			14	35.6	9.75	24.8	7.00	17.8

Fig 3.6o Splay angle reference chart.

3.6.6 Array Coverage and Maximum SPL Charts

A series of outdoor tests were conducted at Meyer Sound to determine the coverage angle and on-axis maximum SPL for arrays with one and two horizontal rows of up to six elements each at numerous splay angles. The measurements were conducted at a distance of 8 meters with one-half space loading. On-axis values were interpolated from 8 meters to 1 meter. The coverage for the array is the result of averaging the –6 dB points from 125 Hz to 8 kHz.

The horizontal angles in the tables represent the optimal configurations of each model for narrow and wide coverage areas. The vertical angles represent similar data with the addition of the LT (long-throw) configuration; the two horns are coupled directly together (top speaker upside down, bottom speaker upright) to form a single narrow horn.

All splay angles refer to the angle between cabinet centers.

UPA-1 Array Coverage and Maximum SPL Chart

Horizontal Units & Angle	1			2 @ 30 °			2 @ 60 °			3 @ 30 °			3 @ 60 °			4 @ 30 °			4 @ 60 °		
	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)
Vertical Rows & Angle 1	100°	90°	135	80°	90°	139	120°	90°	136	110°	90°	140	180°	90°	137	120°	90°	141	220°	90°	138
2 LT (0°)	100°	30°	139	80°	30°	143	120°	30°	140	110°	30°	145	180°	30°	141	120°	30°	147	220°	30°	142
2 @ 15°	100°	80°	136	80°	80°	142	120°	80°	139	110°	80°	143	180°	80°	140	120°	80°	144	220°	80°	141
2 @ 30°	100°	100°	133	80°	100°	139	120°	100°	136	110°	100°	140	180°	100°	137	120°	100°	141	220°	100°	138

UPA-2C Array Coverage and Maximum SPL Chart

Horizontal Units & Angle	1			2 @ 30 °			2 @ 40 °			2 @ 45°		
	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)
Vertical Rows & Angle 1	45°	45°	132	60°	45°	136	80°	45°	134	90°	45°	134
2 LT (0°)	45°	15°	138	60°	15°	142	80°	15°	140	90°	15°	140
2 @ 15°	45°	50°	137	60°	50°	141	80°	50°	139	90°	50°	139
2 @ 30°	45°	70°	134	60°	70°	138	80°	70°	136	90°	70°	136

Horizontal Units & Angle	3 @ 30 °			3 @ 40 °			3 @ 45 °			4 @ 30 °			4 @ 40 °			4 @ 45 °		
	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)
Vertical Rows & Angle 1	90°	45°	136	120°	45°	135	130°	45°	135	130°	45°	138	160°	45°	137	180°	45°	136
2 LT (0°)	90°	15°	142	120°	15°	141	130°	15°	141	130°	15°	144	160°	15°	143	180°	15°	142
2 @ 15°	90°	50°	141	120°	50°	140	130°	50°	140	130°	50°	143	160°	50°	142	180°	50°	141
2 @ 30°	90°	70°	138	120°	70°	137	130°	70°	137	130°	70°	140	160°	70°	139	180°	70°	138

3.6.6 Array Coverage and Maximum SPL Charts

MSL-3A Array Coverage and Maximum SPL Chart

Horizontal Units & Angle	1			2 @ 15 °			2 @ 30°			3 @ 15°			3 @ 30 °		
	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)
Vertical Rows & Angle 1	60°	70°	140	60°	70°	146	70°	70°	145	50°	70°	149	100°	70°	148
2 LT (0°)	60°	35°	146	60°	35°	152	70°	35°	151	50°	35°	155	100°	35°	154
2 @ 15°	60°	70°	144	60°	70°	150	70°	70°	149	50°	70°	153	100°	70°	152
2 @ 30°	60°	85°	143	60°	85°	149	70°	85°	148	50°	85°	152	100°	85°	151

Horizontal Units & Angle	4 @ 15°			4 @ 30°			5 @ 15°			5 @ 30°			6 @ 15°			6 @ 30°		
	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)
Vertical Rows & Angle 1	60°	70°	152	120°	70°	150	60°	70°	153	160°	70°	150	80°	70°	154	180°	70°	150
2 LT (0°)	60°	35°	158	120°	35°	156	60°	35°	159	160°	35°	156	80°	35°	160	180°	35°	156
2 @ 15°	60°	70°	156	120°	70°	154	60°	70°	157	160°	70°	154	80°	70°	158	180°	70°	154
2 @ 30°	60°	85°	155	120°	85°	153	60°	85°	156	160°	85°	153	80°	85°	157	180°	85°	153

MTS-4 Array Coverage and Maximum SPL Chart

Horizontal Units & Angle	1			2 @ 15 °			2 @ 30 °			2 @ 45°			3 @ 15 °			3 @ 30 °			3 @ 45 °		
	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)
Vertical Rows & Angle 1	70°	60°	140	50°	60°	146	60°	60°	145	100°	60°	142	80°	60°	149	120°	60°	147	150°	60°	145
2 LT (0°)	70°	30°	146	50°	30°	152	60°	30°	151	100°	30°	148	80°	30°	155	120°	30°	153	150°	30°	151
2 @ 15°	70°	50°	145	50°	50°	151	60°	50°	150	100°	50°	147	80°	50°	154	120°	50°	152	150°	50°	150
2 @ 30°	70°	90°	143	50°	90°	149	60°	90°	148	100°	90°	145	80°	90°	152	120°	90°	150	150°	90°	148

MSL-2A Array Coverage and Maximum SPL Chart

Horizontal Units & Angle	1			2 @ 30 °			2 @ 55°			3 @ 30°			3 @ 55 °			4 @ 30°			4 @ 55°		
	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)	Coverage H	V	Max SPL (dB Pk)
Vertical Rows & Angle 1	90°	75°	140	95°	75°	144	120°	75°	142	110°	75°	144	160°	75°	142	140°	75°	146	225°	75°	143
2 LT (0°)	90°	45°	143	95°	45°	148	120°	45°	145	110°	45°	148	160°	45°	145	140°	45°	150	225°	45°	146
2 @ 15°	90°	75°	140	95°	75°	144	120°	75°	142	110°	75°	144	160°	75°	142	140°	75°	146	225°	75°	143
2 @ 30°	90°	90°	138	95°	90°	142	120°	90°	140	110°	90°	142	160°	90°	140	140°	90°	145	225°	90°	140

3.6.6 Array Coverage and Maximum SPL Charts

MSL-4 Array Coverage and Maximum SPL Chart

Horizontal Units & Angle	1			2 @ 15 °			2 @ 22.5°			2 @ 30°		
	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)
Vertical Rows & Angle												
1	40°	35°	140	20°	35°	145	50°	35°	143	70°	35°	141
2 LT (0°)	40°	20°	146	20°	20°	151	50°	20°	149	70°	20°	147
2 @ 10°	40°	40°	145	20°	40°	150	50°	40°	148	70°	40°	146
2 @ 20°	40°	55°	144	20°	55°	149	50°	55°	147	70°	55°	145

Horizontal Units & Angle	3 @ 15°			3 @ 22.5°			3 @ 30 °			4 @ 15°			4 @ 22.5°			4 @ 30°		
	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)
Vertical Rows & Angle																		
1	55°	35°	147	80°	35°	146	100°	35°	146	70°	35°	149	100°	35°	148	130°	35°	147
2 LT (0°)	55°	20°	153	80°	20°	152	100°	20°	152	70°	20°	155	100°	20°	154	130°	20°	153
2 @ 10°	55°	40°	152	80°	40°	151	100°	40°	151	70°	40°	154	100°	40°	153	130°	40°	152
2 @ 20°	55°	55°	151	80°	55°	150	100°	55°	150	70°	55°	153	100°	55°	152	130°	55°	151

Horizontal Units & Angle	5 @ 15°			5 @ 22.5°			5 @ 30°			6 @ 15°			6 @ 22.5°			6 @ 30°		
	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)
Vertical Rows & Angle																		
1	95°	35°	150	120°	35°	147	160°	35°	146	100°	35°	150	145°	35°	148	185°	35°	147
2 LT (0°)	95°	20°	156	120°	20°	153	160°	20°	152	100°	20°	156	145°	20°	154	185°	20°	153
2 @ 10°	95°	40°	155	120°	40°	152	160°	40°	151	100°	40°	155	145°	40°	153	185°	40°	152
2 @ 20°	95°	55°	154	120°	55°	151	160°	55°	150	100°	55°	154	145°	55°	152	185°	55°	151

CQ-1 Array Coverage and Maximum SPL Chart

Horizontal Units & Angle	1			2 @ 50 °			2 @ 70 °			3 @ 50 °			3 @ 70 °			4 @ 50 °			4 @ 70 °		
	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)
Vertical Rows & Angle																					
1	80°	40°	136	100°	40°	140	150°	40°	139	170°	40°	140	220°	40°	138	220°	40°	141	300°	40°	139
2 LT (0°)	80°	20°	142	100°	20°	146	150°	20°	145	170°	20°	146	220°	20°	144	220°	20°	147	300°	20°	145
2 @ 15°	80°	45°	140	100°	45°	144	150°	45°	143	170°	45°	144	220°	45°	142	220°	45°	145	300°	45°	143
2 @ 30°	80°	60°	139	100°	60°	143	150°	60°	142	170°	60°	143	220°	60°	141	220°	60°	144	300°	60°	142
2 @ 40°	80°	80°	138	100°	80°	142	150°	80°	141	170°	80°	142	220°	80°	140	220°	80°	143	300°	80°	141

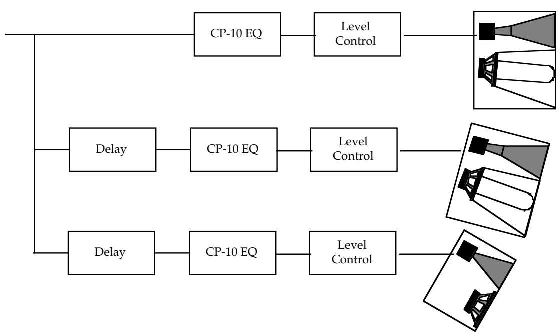
CQ-2 Array Coverage and Maximum SPL Chart

Horizontal Units & Angle	1			2 @ 30 °			2 @ 40 °			3 @ 30 °			3 @ 40 °			4 @ 30 °			4 @ 40 °		
	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)	Coverage H	Coverage V	Max SPL (dB Pk)
Vertical Rows & Angle																					
1	50°	40°	139	70°	40°	143	90°	40°	142	100°	40°	144	130°	40°	144	130°	40°	145	170°	40°	144
2 LT (0°)	50°	20°	145	70°	20°	149	90°	20°	148	100°	20°	150	130°	20°	150	130°	20°	151	170°	20°	150
2 @ 15°	50°	45°	143	70°	45°	147	90°	45°	146	100°	45°	148	130°	45°	148	130°	45°	149	170°	45°	148
2 @ 30°	50°	60°	142	70°	60°	146	90°	60°	145	100°	60°	147	130°	60°	147	130°	60°	148	170°	60°	147
2 @ 40°	50°	80°	141	70°	80°	145	90°	80°	144	100°	80°	146	130°	80°	146	130°	80°	147	170°	80°	146

3.6.7 Array Do's and Don'ts

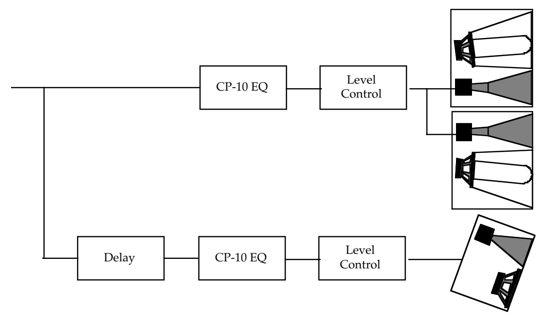
There are many different ways to construct point-source arrays, each with specific advantages and disadvantages. The following series of figures provides a general guide to point-source array design.

Note: The pictograms of the MSL-4 and CQ are shown, but the concepts are independent of the model of speaker, and apply to the full family of Meyer's arrayable speakers.



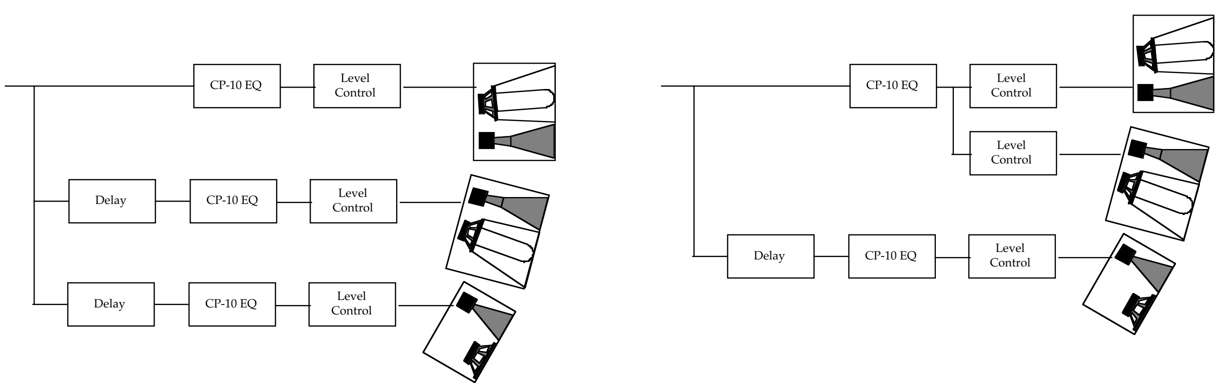
Extended Vertical Coverage with Downfill

All speakers have independent destinations. Vertical point source is achieved. Each row of speakers should be delayed with separate level and EQ.



Long-Throw Mains with Downfill

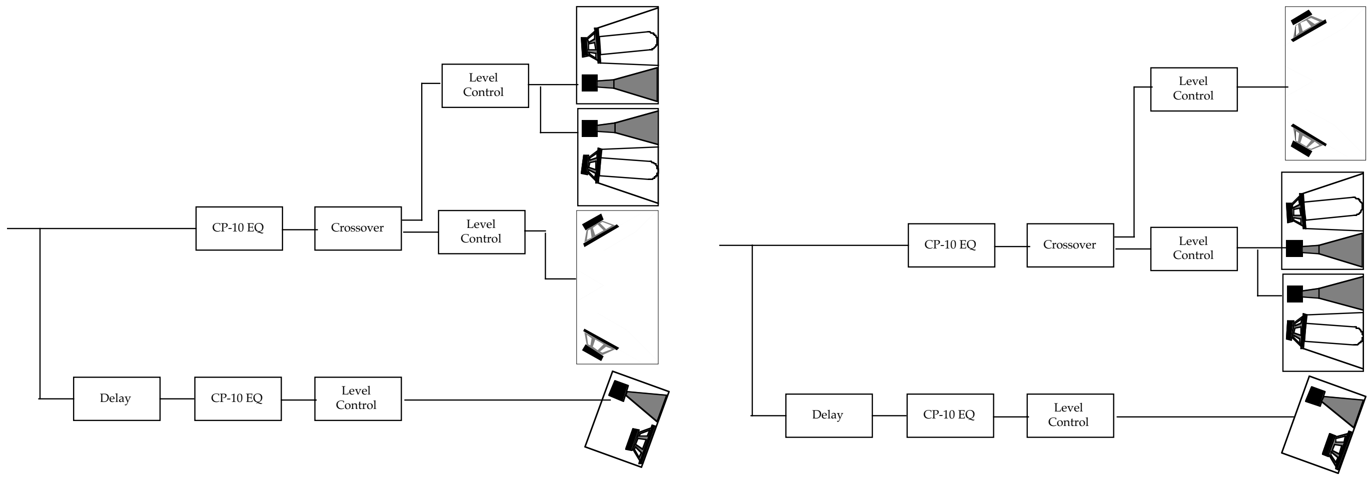
The horns of the main speakers are coupled together oriented at an identical angle. As long as the horns are coupled very closely they will add together as if they were a single unit. The result is a halving of the vertical pattern and a 6 dB increase in on-axis power. The directly coupled horns should be equalized, delayed and level set as a single system. Vertical point source is achieved. The downfill speakers should be delayed with separate level and EQ.



Narrow Vertical Coverage with Downfill

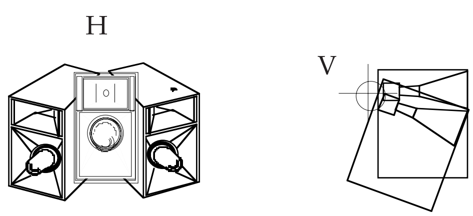
All speakers have independent destinations. Horns of the main speakers are coupled together but angled apart. The result is pattern narrowing if the angle is low. The directly coupled horns are usually equalized and delayed as a single system. If the lower horns are delayed separately the delay will be VERY small (less than 1 ms). Vertical point source is achieved. The downfill speakers should be delayed with separate level and EQ.

3.6.7 Array Do's and Don'ts



High Power Long-Throw with DS-2s

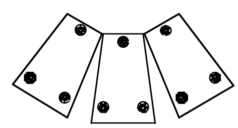
This is similar in performance to the previous long-throw configuration. The only difference is the insertion of the DS-2s. The crossover shown can be either the D-2 CEU (with the DS-2) or the LD-1A (with the DS-2P). The DS-2s can be above or below the mains. Which of these is best will depend upon the cluster trim height and the vertical requirements of the venue. A separate delay for the downfill system is vital since the time offset of the down lobe is very high.



Horizontal Array with Vertical Stagger

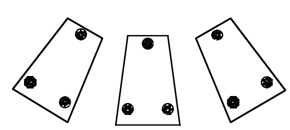
Alternating cabinets are angled downward giving vertical coverage extension. The result is similar to two vertical rows but in half the vertical space. This works well as long as the horizontal coverage of each speaker is at least twice the enclosure angle. This allows each row to achieve complete horizontal coverage. All speakers have independent destinations creating both a vertical and horizontal point source.

This technique was developed by Dave Lawler who uses this successfully with MSL-4s.



Standard Horizontal Point-Source Array

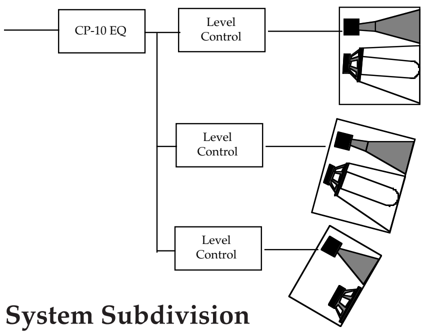
Cabinet rears are touching while the fronts are splayed apart. Each speaker has an independent destination and a horizontal point source is created. This is the most common horizontal array configuration. The close coupling of the cabinets keeps the time offsets low for maximum LF coupling and response uniformity.



Split Point-Source Array

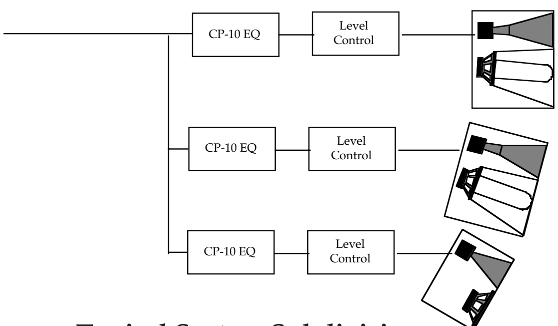
The cabinets are split apart but the splay angle is maintained. Each speaker has an independent destination, creating a horizontal point source. Small gaps between the speakers are not critical to LF coupling. Remember: Low frequency wavelengths are very long, so a few inches will not change things dramatically. In some applications the speakers are split far apart (as with delay fills). In this case LF coupling will be lost but superior level distribution can be achieved.

3.6.7 Array Do's and Don'ts



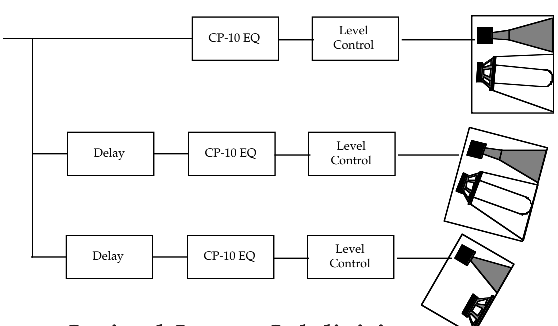
Poor System Subdivision

Level controls alone will not suffice to minimize the interaction between the vertical layers of this system. The equalization needs are totally different (different throw length and different speaker model). **Not recommended.**



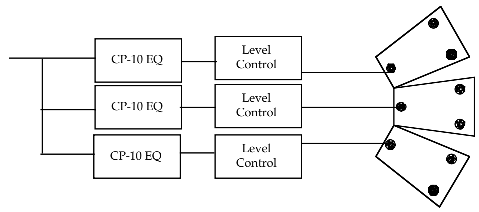
Typical System Subdivision

Vertical array sections will benefit from independent level controls, and equalization. Each system has large differences in depth of throw and speaker/room interaction, which require unique equalization. However, the lower levels will not be phase aligned with the down lobe of the upper systems. This has the potential for serious interaction problems.



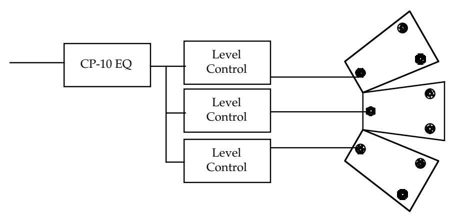
Optimal System Subdivision

Vertical array sections will benefit from independent level controls, delay and equaliztion. Each system has large differences in depth of throw and speaker/room interaction, which require unique equalization. The lower levels will be delay tapered to align with the down lobe of the upper systems.



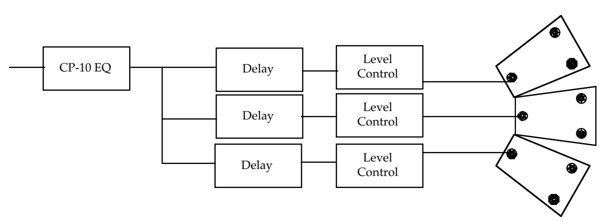
Excess System Subdivision

Directly coupled horizontal point-source arrays with identical speakers do not need separate equalizers for each speaker. (They are only needed in cases of extreme differences in throw between the center and side speakers.)



Amplitude Tapering Subdivision

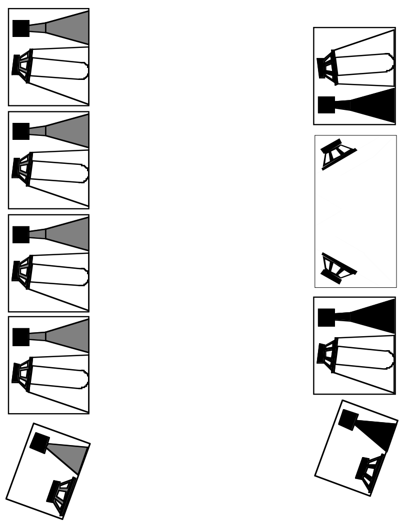
Directly coupled horizontal point-source arrays with identical speakers can benefit from independent level controls, which allows for amplitude tapering.



Delay Tapering Subdivision

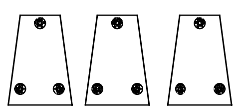
Directly coupled horizontal point source arrays with identical speakers can benefit from independent level controls and delays. This would be helpful in cases where the throw distance across the array is not the same. This allows for delay tapering.

3.6.7 Array Do's and Don'ts



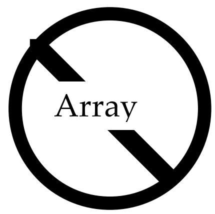
Multi-Level Split Parallel Mains with Downfill

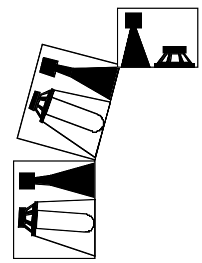
Multiple rows of speakers are oriented at the identical angle. This results in massive interaction between the upper and lower mains. Vertical point-source is *not* achieved. This type of system is very loud but will have extremely poor uniformity of coverage. **Not recommended.**



Parallel Array

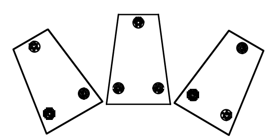
Cabinet fronts are touching and rears are splayed apart. This is a vestige of the "wall-of-sound" design strategy. Popular with the "power-at-all-costs" set. Large numbers of speakers are redundantly oriented either horizontally, vertically, or both for maximum coupling. The result is maximum combing. Frequency response uniformity in such systems is nonexistent. These "arrays" get very loud. Painfully so. And if you don't like the frequency response where you are, it's no problem. Just move a few feet and it will be totally different. **Not recommended.**





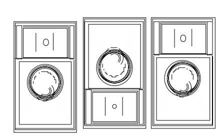
Point Destination Array (Crossfire)

The horns of all of the speakers are split apart at the rear and converge to create a pseudo point-source in front of the speakers. This is a vestige of the "flying junkyard" design concept of horn-only clusters from the 1950s. The only problem is that it doesn't work. (Go to your local sports stadium to hear for yourself.) The speakers crossfire massively for maximum interaction between the upper and lower mains and the downfill. Vertical point-source is *not* achieved. **Not recommended.**



Crossfire Array

This is another variation of the previous "wall-of-sound" type array. Cabinet fronts are touching and the rears are splayed apart with the cabinets pointed inward. This is great for creating a blast zone at the mix position. Very loud. Very narrow. Maximum combing. The frequency response uniformity is nonexistent. **Not recommended.**



Horizontal Array with Vertical Checkerboard

Alternating cabinets are turned upside-down but the vertical angle is kept the same. A vestige of the "wall-of-sound" design strategy. The main effect of this configuration is to randomize the vertical and horizontal coverage with increased combing. **Not recommended.**

3.7.1 Fill Systems: Introduction

We have now defined the main system. In addition, your system may have various supplemental subsystems, each of which has independent functions and coverage areas.

In the world of sound reinforcement there are seven different typical speaker subsystem functions.

Main System: This will cover the majority of the listening space. This system will need to have the highest power rating. If the signal requirements require more than voice only, this system will need subwoofers.

Downfill System: Supplemental vertical coverage to the main array to cover the area below. This system typically has a shorter throw than the main array.

Side/Rearfill System: These systems provide supplemental horizontal coverage to the main array. These systems typically have a shorter throw than the main array.

Frontfill System: Supplemental coverage to the stage front area. These provide localization clues to the stage and increase intelligibility in the near field.

Delay System: These systems increase intelligibility in the far field and provide some compensation for the SPL loss over distance from the main array.

Effects Systems: These systems create spatial effects. Since these systems contain separate signals their integration to the main system is not considered.

Stage Monitor System: Foldback system for the artists.

Each of these subsystems demand separate evaluation of their intended coverage area to determine the most suitable array.

The main systems are designed to cover the largest seating area, have the longest throw and therefore require the highest power handling capability of all of the systems. Since the main systems will cover the largest proportion of the audience, they will take priority over the fill subsystems in matters of alignment, equalization and level setting. (See Section 5, Alignment.)

3.7.2 Downfill/ Sidefill

Downfill and sidefill systems extend the vertical and horizontal coverage. They are treated as separate systems when the throw distances for them are significantly shorter than the mains.

Such systems are typically made up of lower power speakers with wider patterns since the audience is seated closer. In the fill listening area the sound field consists of the combined response of the main and fill speakers. Therefore the interaction between these systems will be critical to the sound quality in the downfill area. Such systems should use a delay line to synchronize them with the off-axis signal coming from the mains.

The following example details considerations in downfill systems. Sidefill systems are largely similar.

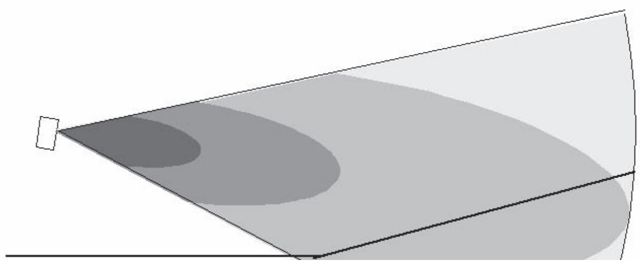


Fig 3.7a Insufficient vertical coverage.

The main system does not have sufficient vertical coverage for the front area.

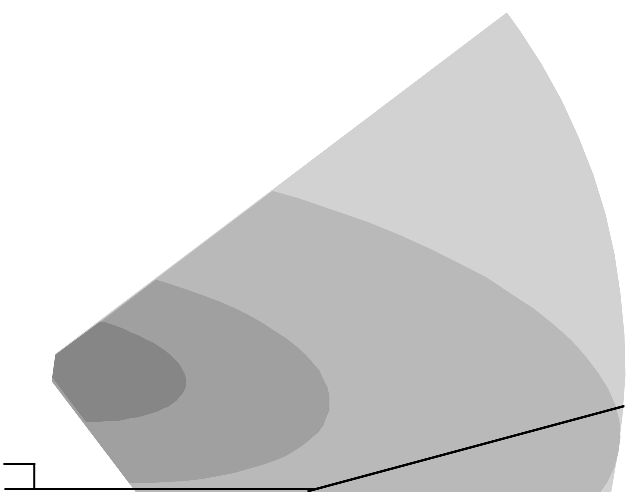


Fig 3.7b Widen the main's vertical coverage.

The 35° main system speaker could be replaced with a 90° model to widen vertical coverage without adding downfills. This has the unfortunate side effect of making it louder in the front and creating much stronger reflections off the floor and ceiling.

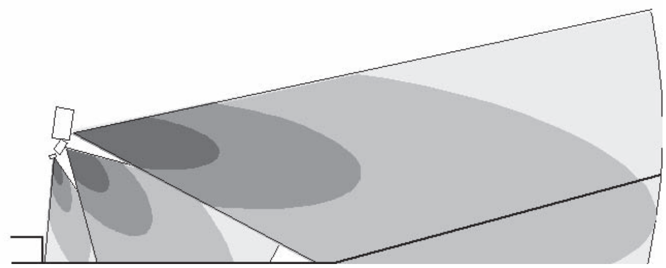


Fig 3.7c Traditional downfill.

The main system is supplemented by two levels of downfill speakers. All seats are covered but the closest seats will experience severe sonic image distortion. This approach, while effective in terms of coverage and uniformity, should be avoided if the cluster is too high unless absolutely required by architectural restrictions.

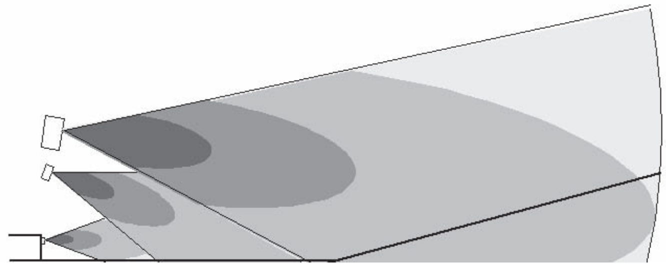


Fig 3.7d Downfill combined with frontfill.

The main system is supplemented by the downfills and a frontfill array. Level uniformity is optimized. The image is kept down by the frontfills, creating a much more natural effect.

3.7.3 Frontfill

Frontfill speakers are located at the stage front lip to cover the seats nearest the stage. They aid both vertical and horizontal sonic imaging by providing a low sound source in the direction of the performers. Frontfill speakers are delayed to synchronise them with stage sound.

The dominant design factor is that coverage is shallow and wide due to the speaker's close proximity to the listeners. They are best served by split-parallel or split point-source arrays.

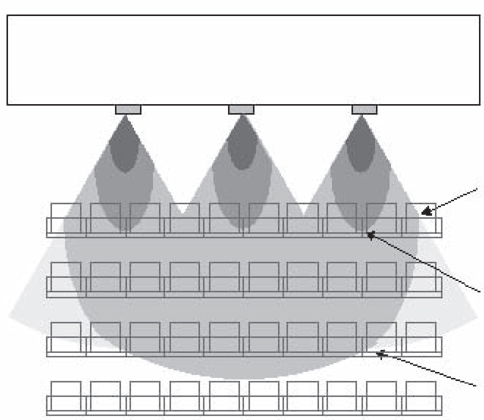


Fig 3.7e Split-parallel arrays recommended for frontfill applications.

Split-parallel arrays create the most even level distribution for frontfill applications. Three 60° speakers were placed with a 1:1 ratio between the speakers and the distance to the first seats. The level distribution across the three front rows is ± 3 dB

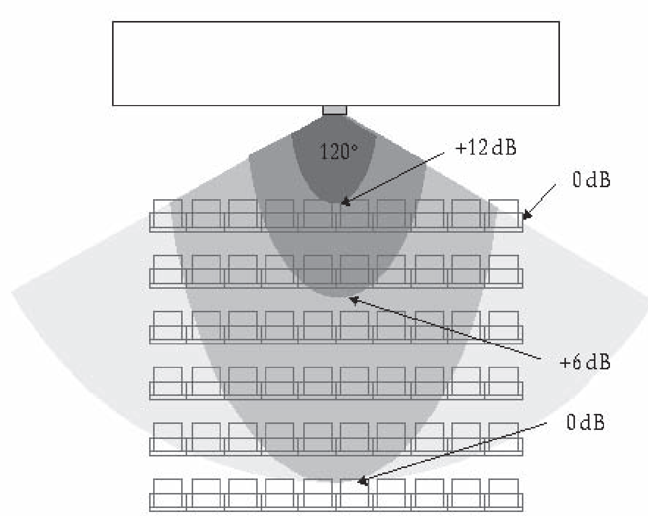


Fig 3.7f Point-source arrays or single wide coverage speakers are not recommended due to poor distribution level.

Point-source arrays create poor level distribution. Notice that the outside seat of the first row and the center seat of the 6th row have the same level. The center speaker in the first row is 12 dB louder than either of the above positions.

3.7.3 Frontfill

Frontfill Unit Spacing

When designing frontfill arrays you must consider the relationship between the speaker's coverage angle, enclosure spacing, and audience distance. It is a given that these types of arrays will have overlapping areas. The intention is to minimize the overlap, without leaving coverage gaps. The systems should overlap so that their –6 dB points converge at the first listeners to be covered.

The three factors of coverage angle, audience distance and enclosure spacing are interrelated as follows:

- As the coverage angle increases the speakers should be spread farther apart.
- As the coverage angle decreases the speakers should be moved closer together.
- As the distance to the audience increases the speakers should be spread apart.
- As the distance to the audience decreases the speakers should be spaced closer together.

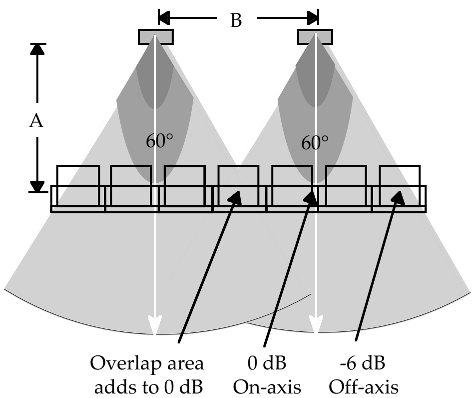


Fig 3.7g Speakers placed at proper relationship. For a 60° coverage angle A (the distance to the audience) will be equal to B (the enclosure spacing).

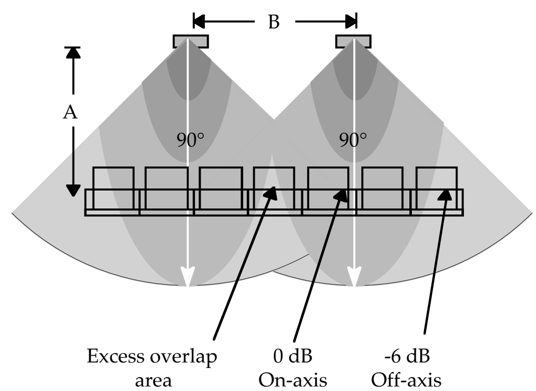


Fig 3.7h The speaker coverage angle is too wide for the current spacing. There is too much overlap, which will cause a hot spot between the two speakers.

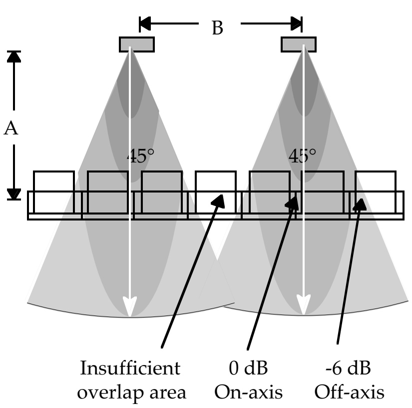
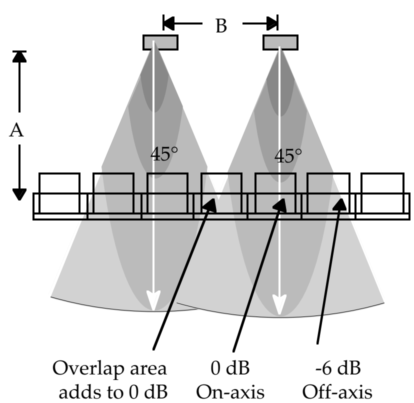


Fig 3.7i The speaker coverage angle is too narrow for the current spacing. There is a gap in coverage.



3.7j Speakers were moved closer together to optimize the response for the narrower speakers.

3.7.4 Delay Systems

Underbalcony Delay

These systems are designed to increase the direct-to-reverberant ratio in shadowed spaces such as under a balcony. They must be low profile so as not to disturb sight-lines.

Underbalcony Delays

Under balcony areas tend to suffer from:

- Strong early reflections from nearby ceiling and rear walls.
- HF loss due to distance and axial attenuation of main speakers.
- Level drop due to distance.

The loss in SPL is minimal due to the summing of the early reflections. That SPL loss is not the key factor here is borne out by the fact that turning up the main system does little to help the situation under the balcony.

The primary requirements are to increase the direct-to-reverberant ratio and restore the HF range. This can be done with a minimal increase in SPL, enabling us to improve the intelligibility without localizing to the speaker.

The old school of thought on delay speakers is to simply put an HF device to fill in the areas lost under the balcony. The result of this approach is a system that has a sudden rise in direct-to-reverberant ratio above the midband, creating a sonically unnatural characteristic. Then, in order to try to make the system less obtrusive, the delay time is intentionally offset by adding 5 to 15 ms (a distortion of the Haas effect) which decreases intelligibility and destroys the frequency response. Don't do it!

Array Types

As in the case of frontfill systems, the proximity to the audience is the dominant factor in array design. Split point-source arrays are optimal for applications with a main center cluster. This allows for the minimum number of delay channels to be used. In left/right systems the distance between the mains and delays will change substantially as you move toward the center. These systems benefit from the use of multiple delay taps.

Split-parallel arrays are also applicable. Often the decision between these two array types is more a function of available hang points.

Delay Tower

When long distances are to be covered by the main system it can be supplemented by delay towers. Delay towers are most commonly used in outdoor spaces. In contrast to the supplemental delays previously described for indoor systems, outdoor systems will have to deliver some real power as well. In outdoor spaces the losses approach 6 dB per doubling distance. The delay towers must counteract this loss without doubling back and disturbing the listeners in the front. Delay towers can also be used to increase the direct-to-reverberant ratio in large listening spaces such as stadiums. The towers must be low profile to not disturb sight-lines. They should be highly directional to prevent destructive interaction with other speaker subsystems.

Delay Towers

- If at all practical, use many small narrow towers rather than a single wide one.
- Place them as deep as possible so that they can be run at as low a level as possible. This will also decrease their size.
- Do not try to cover too wide of an area. The time offsets will be too high and the intelligibility will be lost rather than gained.
- Do not worry about stereo imaging. The dam=age to intelligibility from the overlap is substantial. People in the back are more likely to complain about a lack of intelligibility rather than a lack of stereo imaging.
- Do not try to make up all of the lost gain in the rear. The area around the delay tower will become too loud, disturbing people in the prime seating locations.

3.7.4 Delay Systems

How long can my main speaker throw before I need a delay?

Each model of Meyer Sound speaker differs in terms of its maximum throw, a combination of directivity and maximum power as shown in Figs 3.7j and 3.7k. The distances on the chart reflect the point at which it becomes advisable to supplement the main system with delays in a typical application. If the environment is highly reverberant the usable distances become shorter.

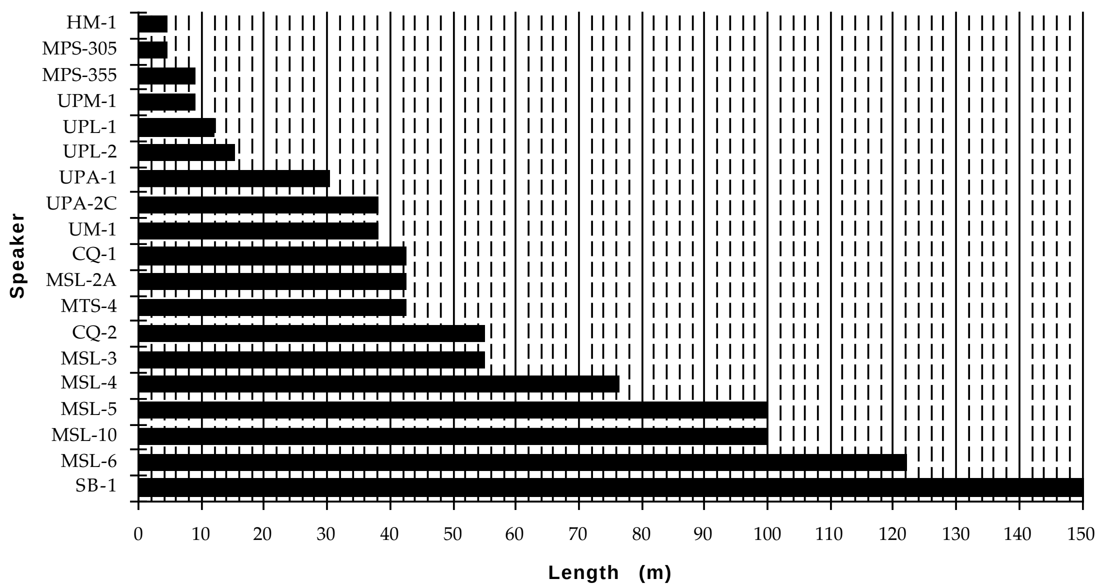


Fig 3.7k Delay speaker reference (meters).

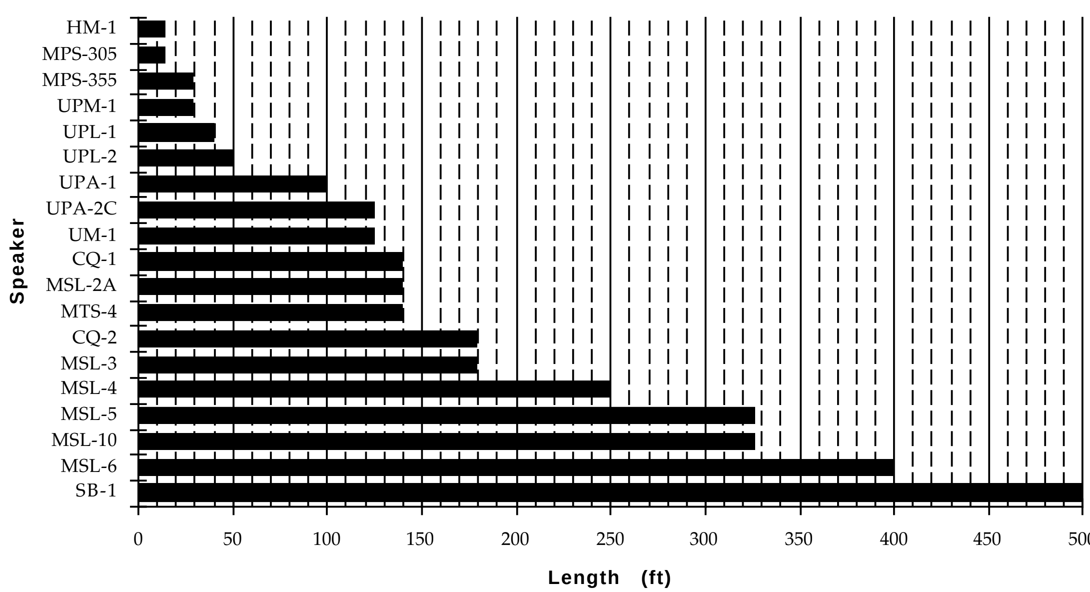


Fig 3.7l Delay speaker reference (feet).

3.7.4 Delay Systems

It is common practice to consider the usable coverage area of a speaker to be the same as its stated coverage pattern. This does not hold true for delays because of the time offset errors that accumulate between the arrivals of the main and delay speakers at different positions. It is unfortunate that the delay time setting that causes perfect synchronization of the mains and delays at one position does not do so at all positions within the delay speaker's coverage pattern. The size of the usable coverage area for a delayed speaker depends much more on the how rapidly the time offset errors accumulate. If the offset errors reach 10 ms the entire frequency response will have comb filtering and the S/N ratio of the system will be greatly compromised. The rate at which the errors accumulate is a function of the distance and angular relationship between the main and delay speakers, with the slowest rate occurring when the speakers are in a straight line. The closer the two speakers are to having the same angular orientation, the slower the accumulation of offset errors.

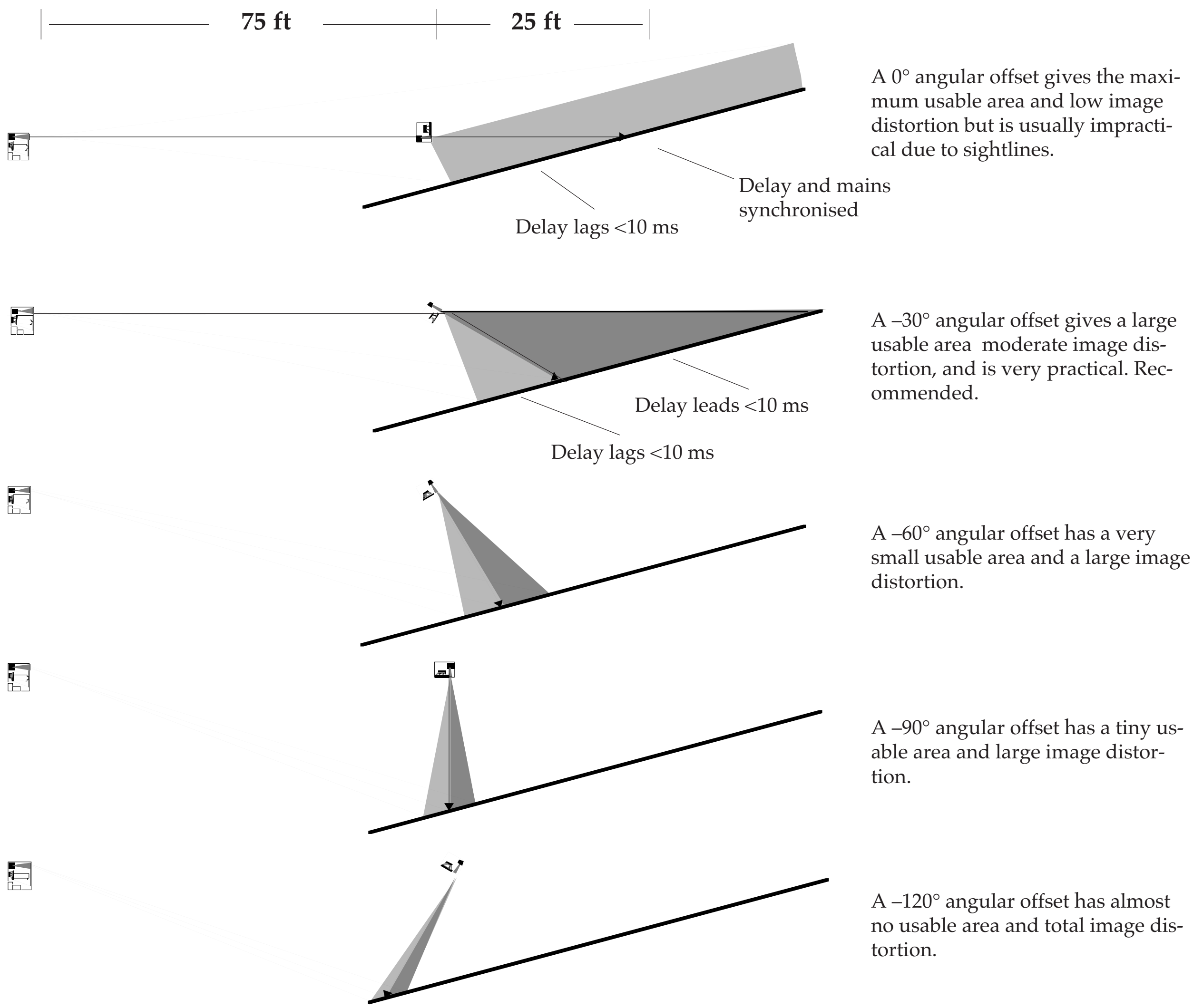


Fig 3.7m Usable vertical angle for distributed delays.

3.7.4 Delay Systems

Figs 3.7m and 3.7n show an example main and delay speaker with various vertical and horizontal angular relationships. In each case it is assumed that the speakers are synchronized in the center of the delay coverage area. The shading indicates the area where the time offset errors are less than 10 ms. The lighter shade indicates that the main speaker leads in time, the darker shade indicates that the delay speaker leads. These figures show that the usable area narrows sharply as the angular difference exceeds 30°. If steep angles are used, the delay speakers must have very tight pattern control to prevent leakage into neighboring areas.

The example shown here is for a distance of approximately 100 feet. In actual practice, if the distances are shorter the time offsets will accumulate more slowly, giving you larger usable areas. If the distances are longer, the opposite occurs.

Musical theatre designs benefit from the small distances, giving them wide usable areas to maneuver in. Such designs can not be scaled up to arena size, however, because the offsets cause huge frequency response problems. Another lesson in this is that wide coverage angle delay speakers (or arrays) are of very limited use. Even if on paper they appear to cover a large area they will not, in practice, work well due to the accumulated offsets.

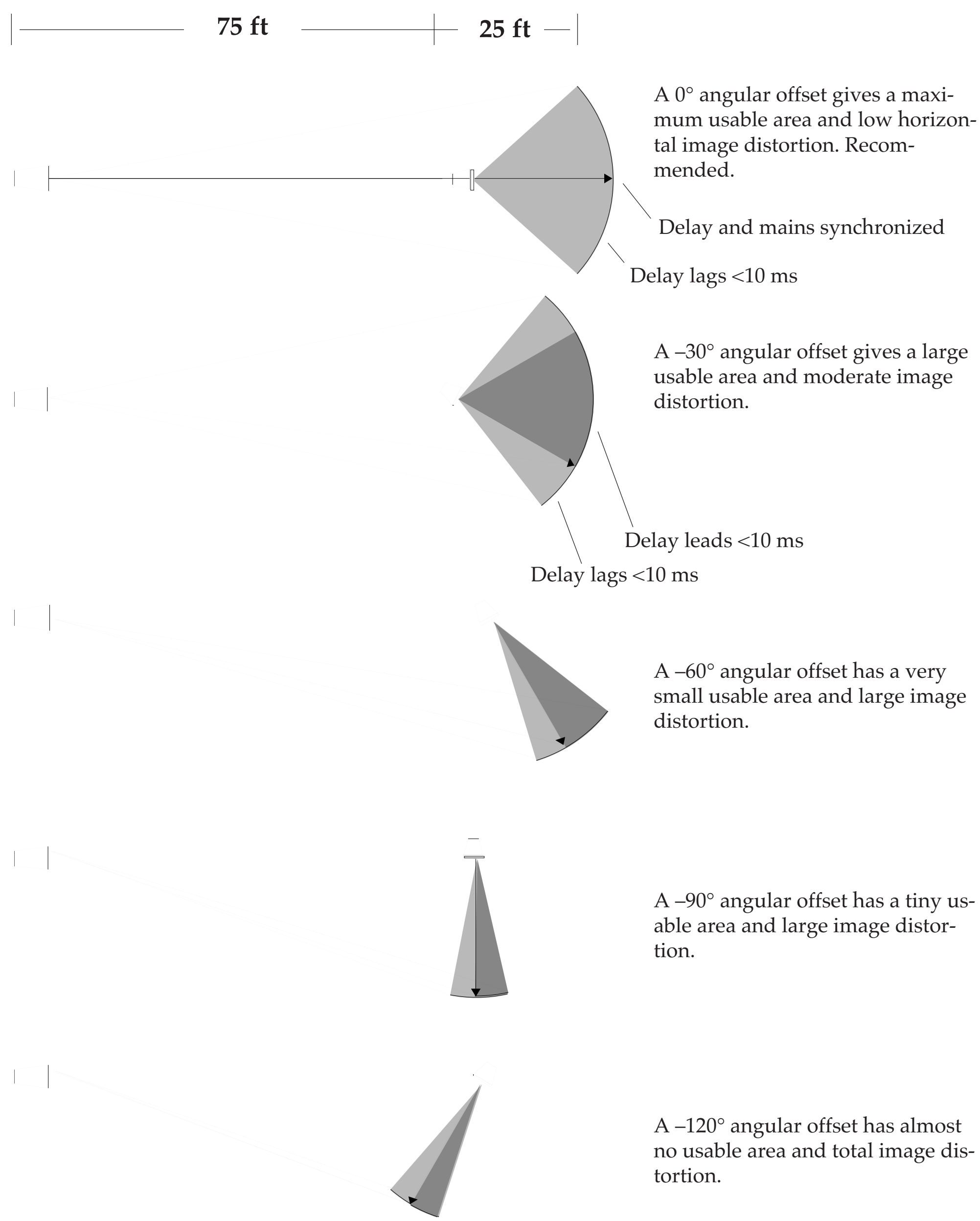


Fig 3.7n Usable horizontal angle for distributed delays.

3.8 Stage Monitors

The dominant factors in the performance of a stage monitor system are:

- The interaction of the speaker and microphone.
- The speaker response.
- Near-field boundary conditions.

The interaction of the speaker and microphone can be broken down into:

- The directional characteristics of the mic and speaker.
- The distance between the speaker and the mic.

The signals passing through a stage monitor is a mixture of direct (such as a synthesizer or a bass direct input) and regenerative (such as a vocalist). This summation of direct and delayed energy into the microphone has all of the same aspects of comb filtering as discussed in Section 2, and will have a dramatic effect on the vocal signal quality, both on stage and in the house. The comb filtering can be to some extent reduced by equalization. It could also be reduced by turning the stage monitor down, but let's be practical!

The equalization process for stage monitors can be broken into three stages:

- Equalization of the speaker system itself and the local acoustical environment. This is typically done with an outboard equalizer.
- Equalization of the regenerative path signals (vocal mics). This can be done with an inserted outboard EQ or with a channel EQ.
- Equalization of the direct signals by channel EQ.

The reason for breaking these factors apart is that when you equalize the system for the regenerative path, there are likely to be drastic measures taken. The direct path signal, which does not need this EQ will sound bad.

One solution is to double the number of stage monitors and separate them into music and vocal systems. This has the advantage allowing for separate EQ and, in addition, the musicians find it easier to localize their voice and their instruments since they come from different positions. This same psychoacoustic mechanism allows us to localize a particular conversation at a noisy party.

Another option is to route the vocal channel through an outboard equalizer, compensating for the regenerative effects there and leaving the speaker linear. Then both direct and regenerative signals will sound natural.

Monitor engineers learn to be extremely careful regarding the position of stage monitors and microphones. Slight changes in position can change the frequency response resulting in feedback. What is the mechanism that causes this?

Fig 3.8a shows the relationship of a typical stage monitor and microphone. With the mic at a height of five feet the comb filtering will result from a time offset of 5.4 ms. This relationship defines where the peaks and dips will be, and the rejection of the mic and level required to deafen the musician will determine the depth of the combing. If we move the mic up for a taller musician the time offset changes to 5.9 ms.

Now let's look at what happens with this 0.5 ms change. The deep null at 1 kHz is now a peak. The same effect will occur at 3 kHz, 5 kHz and upward. The opposite effect will occur at 2 kHz, 4 kHz etc. A change of one inch will reverse the response at 5 kHz, 10 kHz and 15 kHz. Unfortunately, however, it is a universal law of musicians that they must readjust their mic stand immediately upon coming on stage.

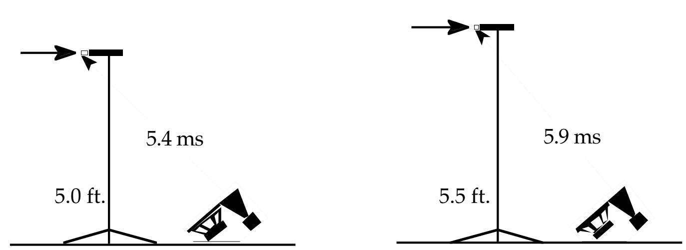


Fig 3.8a The repositioning of the mic stand creates a new time offset relationship between the mic and speaker. This 0.5 ms change can change the response at 1 kHz by over 20 dB.

3.9.1 Speaker System Selection: Introduction

Many people wonder, which speakers are the best for my application? There are many different ways to go about determining this. The method we will use is to first discern the function, followed by the power needs, and finally the directional pattern control. This guide will aid you in selecting the best building blocks for your system. Once chosen you can customize your arrays for the proper horizontal and vertical coverage by referring to the charts in section 3.6.6.

System Power Classifications

From power and pattern control standpoints the Meyer product line can be broken into five classes.

Stadium level: The MSL-5, MSL-6, MSL-10A and SoundBeam systems. These are high-power very high-Q arrayable systems. These systems have very high intelligibility over long distances. Their tightly controlled pattern gives them maximum intelligibility in the far field even in highly reverberant environments. However, the splay angle between cabinets cannot be adjusted. The typical range of these systems is 75 to 500 feet.

Arena level: The MSL-4 has excellent pattern uniformity and arrayability. The horizontal splay angle is adjustable over a short range. Typical range of these systems is 50 to 350 feet.

Concert level: The MSL-3A, MSL-2A, CQ-1, and CQ-2 and the MTS-4 systems. These are high power, medium-Q, arrayable systems. These systems are highly flexible in terms of arraying angles. However, the overlap between speakers increases the variability of the response and decreases intelligibility. These types of systems work well in moderate quantities. If your main array has more than eight to twelve of these speakers, you should consider moving up to the arena level systems. Typical range of these systems is 35 to150 feet.

Theatre level: The UPA series systems. These are medium power (compared to above), medium-Q, arrayable systems. These systems are highly flexible in terms of arraying angles. However, the overlap between speakers increases the variability of the response and decreases intelligibility. These types of systems work well in moderate quantities. If your main array has more than eight to twelve of these speakers, you should consider moving up to the concert level systems. Typical range of these systems is 20 to 100 feet.

Near field: The UPM, HM-1 and UPL series systems. These systems are for short-throw applications only (less than 40 feet).

Stadium	Arena	Concert	Theatre	Near Field
SB-1	MSL-6	MSL-5	MSL-4	UPL-2
MSL-6	MSL-10	MSL-4	MTS-4	UPL-1
MSL-10	MSL-5	MSL-3	MSL-2A	UPM-1
MSL-5	MSL-4	CQ-2	CQ-1 & 2	MPS-355
		MTS-4	UM-1	MPS-305
		MSL-2A	UPA-2C	HM-1
		CQ-1	UPA-1	
			UPL-2	
			UPL-1	

Table 3.9a The Meyer speaker line divided into basic application classes.

The presence of a speaker in a particular class indicates that is a typical choice as a main system in that type of venue.

3.9.2 Main Systems

MSL-10A: Long-throw, high-power system. Very sharp cutoff in horizontal coverage. Ideal for controlling coverage in reverberant environments. Fixed array splay angle (30°). Each additional horizontal unit adds 30° to the coverage. Vertical coverage is a more gradual rolloff so that downfill requirements are minimized. MSL-10As may be permanently installed outdoors. The enclosure is weatherproofed and contains an internal environmental system with thermostatically controlled heaters and fans. This system weighs 700 pounds, which limits its movement to fork lift only.

MSL-5: Long-throw high-power system. The MSL-5 is a smaller, lighter road version of the MSL-10 with virtually identical performance and arrayability. The LF cutoff for the MSL-5 is 100 Hz. Therefore DS-2s are used to supplement the low-mid region. The typical ratio is one to two DS-2s per MSL-5 section. The weight is 480 pounds. This system is usually moved by fork lift and hung.

MSL-6: Long-throw, high-power, high-Q system. The MSL-6 is a smaller, self-powered enclosure from the same family but with some important differences. The MSL-6 has a much tighter vertical pattern, giving it a longer throw and superior performance when used in multiple rows. Must be tight-packed.

MSL-4: This is a self-powered high-power high-Q system. This system is lighter in weight, comparable in power and much more arrayable than the MSL-3A. The ideal building block for modular arena and concert systems. At 180 pounds it can be easily flown or ground-stacked. MSL-4s have 40° of horizontal coverage. Maximum splay angle between centers is 30°.

MSL-3A: MSL-3s are a high-power, modular, arrayable system. The MSL-3s and MSL-4s have the narrowest enclosure design (15°) of any of the Meyer line, allowing them to be densely packed for high power arrays. Large arrays of tight-packed MSL-3s will generate comparable on-axis power to that of MSL-5, MSL-6 and MSL-10 arrays but with considerably less uniformity of horizontal coverage. Coverage uniformity is greatly improved by horizontally splaying the MSL-3 at 30° centers, with a corresponding reduction in on-axis maximum SPL. Maximum splay angle is 45° (center to center). The low-frequency beamwidth is tighter for the MSL-3 than for the MSL-2A and UPA-1s.

MTS-4: This is a four-way fully active self-powered system in a single enclosure. This system has comparable power and coverage to the MSL-2A with a subwoofer. It is ideal where quick setup and easy maintenance are a priority. It does not require subwoofers, although it can be supplemented with the PSW-4. It is best when maximum power is desired in a short throw. It is excellent for the relatively non-technical world of clubs, discos, etc. The MTS-4 has integral casters for transport by single person on flat ground. The weight is 300 pounds.

CQ-1: Wide pattern constant coverage system. Perfect as a single speaker or in a column. It can make very wide arrays from only a few units.

CQ-2: Also has the constant Q horn. The pattern is narrower, which makes it a great building block for small arrays.

MSL-2A: The MSL-2A answered the market need for a higher power UPA-1. It behaves very similarly to the UPA, except that it has more power and greater low-frequency range. The MSL-2A can reproduce full-range musical program with or without an additional subwoofer since it reaches 40 Hz by itself. It focuses quickly, making it ideally suited for relatively near applications (less than 125 ft.) The MSL-2A is easily hand-carried (82 pounds).

UPA-1C: The UPA-1C is a medium-power, low-Q system. The UPA-1 is the industry standard for ultra-compact speaker enclosures. The UPA-1 is best used in small arrays or singly, since its pattern is wide. Large tightly- packed UPA arrays will have a highly variable response due to excess overlap and are not recommended for this. In such cases, fewer numbers of higher-power systems should be used. The UPA-1C focuses quickly, making it ideally suited for relatively near applications (less than 100 feet). The UPA is light weight (65 pounds), the ideal small system building block.

UPA-2C: This is the narrow coverage companion to the UPA-1C. The UPA-2C was derived from the original UltraMonitor™ design. Below 1200 Hz the UPA-1C and UPA-2C are identical. Above 1200 Hz the UPA-2C has one-half the pattern of the UPA-1C (45° versus 90°). This make the UPA-2C best suited for requirements for high intelligibility, especially in reverberant environments. The UltraMonitor™ and the UPA-2C are the standard vocal reproduction speakers of Broadway and West End musical theatre.

3.9.3 Subwoofers

UPL-1 and UPL-2: These are self-powered moderate power systems for near-field to mid-field applications. These systems should be used at a range of less than 40 feet. The UPL-1 is wide like the UPA-1, and the UPL-2 is narrow like the UPA-2. The UPL series work well for conference room PAs, small theatre reinforcement and many other applications where a single compact enclosure is desired to cover the full audio range (30 Hz to 20 kHz).

650-P: This self-powered version of the classic 650 is optimized for maximum efficiency and reliability. This is the ideal subwoofer for any of the above systems unless you want to fly the subwoofers or you are very tight on space.

PSW-4: This is a two-way self-powered trapezoidal subwoofer. It is best suited for arraying with MTS-4s.

PSW-2: This is a two-way self-powered trapezoidal subwoofer. It is best suited for arraying with MSL-4s.

650-R2: This system is well suited for usage with all of the above systems. It cannot be flown.

USW-1: Compact 2 15-inch subwoofers. This system is limited in range to less than 40 Hz. It is best suited for use with UPAs.

MSW-2: The smallest of the Meyer subwoofers. It is also the lowest maximum power capability. This unit is the same size as the MSL-2A. This is best used when space is at a premium.

DS-2: The DS-2 is a directional mid-bass system designed to supplement the MSL-10A, MSL-5, MSL-4 and MSL-3A. The DS-2 provides increased power in the range of 60 to 160 Hz, the region where popular music concentrates an enormous amount of its power. In addition the DS-2 has superior directional control over the front-loaded subwoofers used with the above systems. If your application is high-power, popular music and you are at the stadium, arena or concert power level, you should seriously consider supplementing the system with DS-2s.

DS-2P: Identical performance specifications to the DS-2 with integral power amplifier and CEU.

3.9.4 Fills

Downfill Systems

The downfill systems is built of the same building blocks as above. If the length of throw for the downfills is comparable to the mains (within 70%), the same speaker should be chosen. If the throw of the downfill system is less than 70% of the mains you can select downfill speakers from the next power category down; e.g., MSL-5s can be supplemented with an MSL-4 as downfill. If the throw is less than 50% of the main system you can select downfill speakers from two power categories down; e.g., MSL-4s could be supplemented with a UPA-2C.

Side/Rearfill System

The same rules apply for the selection of these systems as the downfill.

Frontfill System

Frontfill systems should be covered with split-parallel arrays rather than point-source, with the speakers separated along the stage front. Frontfills are typically two to three power levels down from the main system; e.g. MSL-5's can be supplemented with an MSL-2A or UPA-1C as frontfill. MSL-2A's could be supplemented with the UPM-1.

Under-Balcony Delay System

These systems should be low profile to be as visually unobtrusive as possible.

UPM-1: This system is primarily designed for low profile support applications such as frontfills and underbalcony delays. Although the UPM-1 has a trapezoidal enclosure design it is rarely used in point-source arrays. The purpose of the enclosure shape, in this case, is to allow the speaker to tuck up under the balcony areas with as low a profile as possible.

UPM-2: Can also be used as the UPM-1 above but differs in that its HF pattern is much more directional. The UPM-2 would be a superior choice in applications where the speakers must be placed farther away from the listeners than ideally desired.

3.9.4 Fills

MPS-355: Provides a low-cost alternative to the UPM-1 when the trapezoidal shape is not required. The units are box shaped and have a lower-grade contractor paint finish.

MPS-305: This is a reduced power and size version when space is at a premium. The units are box shaped and have the contractor-grade paint finish.

Delay Tower

The delay tower should be a point-source array of no more than 70° of horizontal coverage. The narrower the better. The power capability of the delay tower is typically one to two levels below the main system; e.g., MSL-5s can be supplemented with an MSL-4 delay tower. MSL-4s could be supplemented with CQ-2s.

MSL-10A: These work well in pairs for large stadium and outdoor festivals. Two units can be placed directly onto a fork lift and hoisted into position since the fork-lift slots are built into the enclosure. This high power, directional system requires no scaffolding and has minimal sight line intrusion.

MSL-6: Similar to the MSL-10A above, but self-powered and with tighter vertical control. Does not have forklift tabs.

MSL-5: Similar to the MSL-10A above. It is smaller, and does not have forklift tabs.

MSL-4: This self-powered system is a great choice when small high-powered delays are required. The tight coverage pattern reduces leakage and the self-powered aspect reduces setup time and cost.

3.9.5 Stage Monitors

Floor Monitor

UM-1C: Best for high-power applications with performers in fixed positions. The HF pattern is highly directional (45°). This works well when it is desired to maximize isolation between areas of the stage.

USM-1: Higher maximum power capability than the UM-1. The wider pattern (70° by 60°) works well for applications where the musicians are moving around the stage. The extended frequency range (down to 40 Hz) makes this the best choice when lots of kick drum or bass guitar are desired in the mix.

PSM-2: Self-powered, the PSM-2 has easy setup and repatch. It has directional control similar to the UM-1 but uses the MS-2001A HF driver, giving it more HF power. The PSM-2 has two tilt angles for near and far.

UPM-1: Is commonly used for musical theatre, churches and other low-to moderate-power applications. The pattern is wide and they can be placed unobtrusively.

UPM-2: Can also be used as the UPM-1 above, but the UPM-2 differs in that its HF pattern is much more directional. The UPM-2 would be a superior choice in applications where there is a fixed mic location (such as podiums) or where the speakers must be placed farther away from the listeners than ideally desired.

Drum Monitor

USM-1 or MSL-2A: This is the most compact and unobtrusive of the high-power full range options.

CQ-2: This keeps the energy on the drummer and off the others. If they need more, add a 650-P.

UM-1 plus Subwoofer: This option takes more space and budget but retains the tight HF directional control.

PSM-2 plus PSW-2: Similar to the above but without the amp rack.

MTS-4: This complete four-way self-powered speaker system is economical, powerful and easy to set up. The superior transient response of the MTS-4 gives tremendous impulse reproduction. This is critical for drum reproduction. However, in tight spaces it may prove difficult to position so that the HF pattern is aimed precisely at the drummer.

MSL-4 with 650-P: This system offers superior directional control and high power. This allows the drummer to obtain ear-splitting levels with minimal leakage. The transient response of the MSL-4 is comparable to the MTS-4 above.

3.10.1 Self Powered Speaker Systems

	Acoustical						
Speaker	Max SPL	Coverage	Frequency	HF Driver	LF Driver	Acoustic	Impedance
	Peak (dB)	Hor / Vert	Range			Crossover	HF/LF
HM-1	118	100° x100°	40 Hz-18 kHz	1" dome tweeter			
	@ 1m			(Coaxial with			
				7" LF driver)			
UPL-2	124	45° x45°	25 Hz-22 kHz	1" dome tweeter,	MS-10		12Ω / 8Ω
	@ 1m			horn loaded			
UPL-1	124	90° x 40°	25 Hz-22 kHz	1" dome tweeter,	MS-10		12Ω / 8Ω
	@ 1m			horn loaded			
MTS-4	140	70° x 60°	32 Hz-16 kHz	MS-2001A	MS-18,	40 Hz	12Ω,8Ω
	@ 1m				MS-15	100 Hz	8Ω,8Ω
					MS-12	1 kHz	
CQ-1	136	80° X 40°	40 Hz-18 kHz	MS-2001A	MS-15	900 Hz	12Ω / 8Ω
	@ 1m						
CQ-2	139	50° X 40°	40 Hz-18 kHz	MS-2001A	MS-15	700 Hz	12Ω / 8Ω
	@ 1m						
PSM-2	140	50° X 50°	65 Hz-18 kHz	MS-2001A	MS-12	900 Hz	12Ω / 8Ω
	@ 1m						
MSL-4	140	40° X 35°	65 Hz-18 kHz	MS-2001A	MS-12	800 Hz	12Ω / 8Ω
	@ 1m						
PSW-2	136	N/A	40-120 Hz		MS-15 (x2)		8Ω
	@ 1m						
PSW-4	136	N/A	40-160 Hz		MS-18	45 Hz	8Ω
	@ 1m				and		
					MS-15		
DS-2P	148	120° x 120°	50-160 Hz		MS-15 (x2)		8Ω
	(2 units)	(2 units)					
	@ 1m						
650-P	136	N/A	30-120 Hz		MS-18 (x2)		8Ω
	@ 1m						
MSL-6	120	30° X 25°	65 Hz-18 kHz	MS-2001A (x3)	MS-12 (x2)	800 Hz	12Ω / 8Ω
	@ 30m						
SB-1		8° X 8°	500 Hz-14 kHz	MS-2001A	MS-12	600 Hz	12Ω / 8Ω

3.10.1 Self Powered Speaker Systems

	Amplifier and CEU							
Speaker	Input	Connector	Nominal	Amplifier	Driver Protection	Limit	Rear Panel	RMS™
	Type		Input Level	# Channels	Circuitry	Indicator	Selectable	Option?
HM-1	10kΩ	XLR (A-3) female	+4 dBu	Internal Amp		Red/Grn LED		No
	balanced	Term Block option		Needs external				
				48vdc supply				
UPL-2	10kΩ	XLR (A-3) female	+4 dBu	Self-	Thermo-predictive	Red/Grn LED	+4dBu/-10dBV	No
	balanced		or	Powered	limiters with		Switch	
			-10 dBV		soft peak clamps		Voltage Selector	
UPL-1	10kΩ	XLR (A-3) female	+4 dBu	Self-	Thermo-predictive	Red/Grn LED	+4dBu/-10dBV	No
	balanced		or	Powered	limiters with		Switch	
			-10 dBV		soft peak clamps		Voltage Selector	
MTS-4	5kΩ	XLR (A-3) female	+4 dBu	MP-4	Hi, Mid,	4 Red LED's	Pin 2 / Pin 3 Hot	Yes
	balanced	w/ Male Loop Out		4 Channels	Low, and Sub RMS		Selector	
					TruPower™ Limiting			
CQ-1	10kΩ	XLR (A-3) female	+4 dBu	MP-2	Hi and Low RMS	2 Red LED's	Pin 2 / Pin 3 Hot	Yes
	balanced	w/ Male Loop Out		2 Channels	TruPower™ Limiting		Selector	
CQ-2	10kΩ	XLR (A-3) female	+4 dBu	MP-2	Hi and Low RMS	2 Red LED's	Pin 2 / Pin 3 Hot	Yes
	balanced	w/ Male Loop Out		2 Channels	TruPower™ Limiting		Selector	
PSM-2	10kΩ	XLR (A-3) female	+4 dBu	MP-2	Hi and Low RMS	2 Red LED's	Pin 2 / Pin 3 Hot	Yes
	balanced	w/ Male Loop Out		2 Channels	TruPower™ Limiting		Selector	
MSL-4	10kΩ	XLR (A-3) female	+4 dBu	MP-2	Hi and Low RMS	2 Red LED's	Pin 2 / Pin 3 Hot	Yes
	balanced	w/ Male Loop Out		2 Channels	TruPower™ Limiting		Selector	
PSW-2	10kΩ	XLR (A-3) female	+4 dBu	MP-2	Hi and Low RMS	2 Red LED's	Pin 2 / Pin 3 Hot	Yes
	balanced	w/ Male Loop Out		2 Channels	TruPower™ Limiting		Selector	
					Low Excursion			
PSW-4	10kΩ	XLR (A-3) female	+4 dBu	MP-2	Hi and Low RMS	2 Red LED's	Pin 2 / Pin 3 Hot	Yes
	balanced	w/ Male Loop Out		2 Channels	TruPower™ Limiting		Selector	
					Low Excursion			
DS-2P	10kΩ	XLR (A-3) female	+4 dBu	MP-2	Hi and Low RMS	2 Red LED's	Pin 2 / Pin 3 Hot	Yes
	balanced	w/ Male Loop Out		2 Channels	Low Excursion		Selector	
650-P	10kΩ	XLR (A-3) female	+4 dBu	MP-2	Hi and Low RMS	2 Red LED's	Pin 2 / Pin 3 Hot	Yes
	balanced	w/ Male Loop Out		2 Channels	Low Excursion		Selector	
MSL-6	10kΩ	XLR (A-3) female	+4 dBu	MP-4	Hi and Low RMS	3 Red LED's	Pin 2 / Pin 3 Hot	Yes
	balanced	w/ Male Loop Out		4 Channels	VHF Peak		Selector	
SB-1	10kΩ	XLR (A-3) female	+4 dBu	MP-2	Hi and Low RMS	3 Red LED's	Pin 2 / Pin 3 Hot	Yes
	balanced	w/ Male Loop Out		2 Channels	VHF Peak		Selector	

3.10.1 Self Powered Speaker Systems

	Physical				
Speaker	Finish	Dimensions	Weight	Rigging	Transport
		Per Unit	Per Unit		
HM-1	Black textured	8.50 " W	11 lbs.	N/A	N/A
		11.00" H	(5 kg.)		
		8.40 " D			
UPL-2	Black textured	14.00" W	70 lbs.	3/8 "-16 nut plates	N/A
		20.75" H	(32 kg.)		
		14.00" D			
UPL-1	Black textured	14.00" W	70 lbs.	3/8 "-16 nut plates	N/A
		20.75" H	(32 kg.)		
		14.00" D			
MTS-4	Textured,carpet	21.25" W	280 lbs.	Aircraft pan fittings	Two 3" rubber tread
	or weatherproof	56.75" H	(127 kg)	or Blank Plates	Self-lube casters
		30.00" D			Optional Cover
CQ-1	Textured,carpet	21.25" W	130 lbs.	Aircraft pan fittings,	Optional Casterboard
	or weatherproof	30.00" H	(59 kg)	3/8"-16 or M-10 nut plates,	Optional Cover
		20.00" D		or Blank Plates	
CQ-2	Textured,carpet	21.25" W	130 lbs.	Aircraft pan fittings,	Optional Casterboard
	or weatherproof	30.00" H	(59 kg)	3/8"-16 or M-10 nut plates,	Optional Cover
		20.00" D		or Blank Plates	
PSM-2	Textured,carpet	18.00"W	90 lbs.		
	or weatherproof	24.00" H	(41 kg.)	N/A	
		14.75" D			
MSL-4	Textured,carpet	21.25" W	180 lbs.	Aircraft pan fittings,	Optional Casterboard
	or weatherproof	36.00" H	(82 kg)	3/8"-16 or M-10 nut plates,	Optional Cover
		30.00" D		or Blank Plates	
PSW-2	Textured,carpet	21.25" W	162 lbs.	Aircraft pan fittings,	Optional Casterboard
	or weatherproof	36.00" H	(74 kg.)	3/8"-16 or M-10 nut plates,	Optional Cover
		30.00" D		or Blank Plates	
PSW-4	Textured,carpet	21.25" W	205 lbs.	Aircraft pan fittings,	Optional Casterboard
	or weatherproof	39.00" H	(93 kg)	3/8"-16 or M-10 nut plates,	Optional Cover
		30.00" D		or Blank Plates	
DS-2P	Textured,carpet	21.25" W	243 lbs.	Aircraft pan fittings	Optional Casterboard
	or weatherproof	56.00" H	(110 kg)	or Blank Plates	Optional Cover
		30.00" D			
650-P	Textured,carpet	30.00" W	201 lbs.	N/A	Two 3" rubber tread
	or weatherproof	45.00" H	(91.3kg)		Self-lube casters
		22.50" D			Optional Cover
MSL-6	Textured,carpet	42.25" W	475 lbs.	12 points, pivoting lift rings,	Optional Casterboard
	or weatherproof	42.75" H	(216 kg.)	1500 lb. safe load capacity	Optional Cover
		32.00" D			
SB-1	Textured	56.00" W	293 lbs.	Aircraft pan fittings,	Optional Yoke
	or weatherproof	56.00" H	(133	3/8"-16 or M-10 nut plates,	Available
		21.00" D	kg.)	or Blank Plates	

3.10.1 Self Powered Speaker Systems

	Mains Power			
Speaker	**Power Connector**	**Voltage Selection**	**Power Requirement**	**Power Indicator**
HM-1	Terminal Strip	N/A	48 vdc, 200 watts	Red/Grn LED
	or		Can be driven by Meyer PS-1 AC Adaptor	
	XLR		or other 48 vdc power supply	
UPL-2	3-pin IEC 320	Switchable	6A @ 120 VAC, 3A @ 240 VAC	Green LED
	Male	100,120,220,240 VAC		
UPL-1	3-pin IEC 320	Switchable	6A @ 120 VAC, 3A @ 240 VAC	Green LED
	Male	100,120,220,240 VAC		
MTS-4	L6-20 Male	Intelligent AC™	Cont: 16A @ 120 VAC, 8A @ 240 VAC	Green LED
	or	95-125/208-235 VAC	Burst: 24A @ 120 VAC, 12A @ 240 VAC	
	IEC309 Male		Peak: 40A @ 120 VAC, 20A @ 240 VAC	
CQ-1	L6-20 Male	Intelligent AC™	Cont: 8A @ 120 VAC, 4A @ 240 VAC	Green LED
	or	95-125/208-235 VAC	Burst: 12A @ 120 VAC, 6A @ 240 VAC	
	IEC309 Male		Peak: 20A @ 120 VAC, 10A @ 240 VAC	
CQ-2	L6-20 Male	Intelligent AC™	Cont: 8A @ 120 VAC, 4A @ 240 VAC	Green LED
	or	95-125/208-235 VAC	Burst: 12A @ 120 VAC, 6A @ 240 VAC	
	IEC309 Male		Peak: 20A @ 120 VAC, 10A @ 240 VAC	
PSM-2	L6-20 Male	Intelligent AC™	Cont: 8A @ 120 VAC, 4A @ 240 VAC	Green LED
	or	95-125/208-235 VAC	Burst: 12A @ 120 VAC, 6A @ 240 VAC	
	IEC309 Male		Peak: 20A @ 120 VAC, 10A @ 240 VAC	
MSL-4	L6-20 Male	Intelligent AC™	Cont: 8A @ 120 VAC, 4A @ 240 VAC	Green LED
	or	95-125/208-235 VAC	Burst: 12A @ 120 VAC, 6A @ 240 VAC	
	IEC309 Male		Peak: 20A @ 120 VAC, 10A @ 240 VAC	
PSW-2	L6-20 Male	Intelligent AC™	Cont: 8A @ 120 VAC, 4A @ 240 VAC	Green LED
	or	95-125/208-235 VAC	Burst: 12A @ 120 VAC, 6A @ 240 VAC	
	IEC309 Male		Peak: 20A @ 120 VAC, 10A @ 240 VAC	
PSW-4	L6-20 Male	Intelligent AC™	Cont: 8A @ 120 VAC, 4A @ 240 VAC	Green LED
	or	95-125/208-235 VAC	Burst: 12A @ 120 VAC, 6A @ 240 VAC	
	IEC309 Male		Peak: 20A @ 120 VAC, 10A @ 240 VAC	
DS-2P	L6-20 Male	Intelligent AC™	Cont: 8A @ 120 VAC, 4A @ 240 VAC	Green LED
	or	95-125/208-235 VAC	Burst: 12A @ 120 VAC, 6A @ 240 VAC	
	IEC309 Male		Peak: 20A @ 120 VAC, 10A @ 240 VAC	
650-P	L6-20 Male	Intelligent AC™	Cont: 8A @ 120 VAC, 4A @ 240 VAC	Green LED
	or	95-125/208-235 VAC	Burst: 12A @ 120 VAC, 6A @ 240 VAC	
	IEC309 Male		Peak: 20A @ 120 VAC, 10A @ 240 VAC	
MSL-6	L6-20 Male	Intelligent AC™	Cont: 16A @ 120 VAC, 8A @ 240 VAC	Green LED
	or	95-125/208-235 VAC	Burst: 24A @ 120 VAC, 12A @ 240 VAC	
	IEC309 Male		Peak: 40A @ 120 VAC, 20A @ 240 VAC	
SB-1	L6-20 Male	Intelligent AC™	Cont: 8A @ 120 VAC, 4A @ 240 VAC	Green LED
	or	95-125/208-235 VAC	Burst: 12A @ 120 VAC, 6A @ 240 VAC	
	IEC309 Male		Peak: 20A @ 120 VAC, 10A @ 240 VAC	

3.10.2 Externally Powered Speaker Systems

	Electrical		Acoustical						
Speaker	**CEU**	**Amp Type***	**Max SPL**	**Coverage**	**Freq.**	**HF Driver**	**LF Driver**	**Acoustic**	**Impedance**
		# Channels	**Peak/Cont**	**Hor / Vert**	**Range**			**Crossover**	**HF/LF**
MSL-10A	M-10A	Type 3	120/110	60° x 40°	70-16K	MS-2001A (X3)	MS-12 (x4)	900 Hz	4Ω/4Ω
		3 Channels	(2 units/100 ft)	(2 units)					
MSL-5	M-5	Type 3	120/110	60° x 40°	100-16K	MS-2001A (X3)	MS-12 (x2)	900 Hz	4Ω/4Ω
		2 Channels	(2 units/100 ft)	(2 units)					
MSL-3A	M-3A	Type 1	135/130	70° x 60°	75-18K	MS-2001A	MS-12 (x2)	700 Hz	12Ω/4Ω
		2 Channels							
MSL-2A	S-1	Type 2	139/130	70° x 60°	40-18K	MS-2001A	MS-15	900 Hz	12Ω/8Ω
		2 Channels							
USM-1	S-1	Type 2	139/130	70° x 60°	40-18K	MS-2001A	MS-15	900 Hz	12Ω/8Ω
		2 Channels							
UM-1C	M-1A	Type 1	130/125	45° x 45°	70-18K	MS-1401B	MS-12	1200 Hz	12Ω/8Ω
		2 Channels							
UPA-1C	M-1A	Type 1	130/125	80° x 60°	60-18K	MS-1401B	MS-12	1200 Hz	12Ω/8Ω
		2 Channels							
UPA-2C	M-1A	Type 1	130/125	45° x 45°	60-18K	MS-1401B	MS-12	1200 Hz	12Ω/8Ω
		2 Channels							
UPM-1	P-1A	Type 1	118/108	80° x 60°	70-20K	2" x 5"horn loaded	MS-5 (x2)		16Ω
	MPS-3	1 Channel				piezoelectric			
UPM-2	P-2	Type 1	118/108	45° x 45°	70-20K	2" x 5"horn loaded	MS-5 (x2)		16Ω
		1 Channel				piezoelectric			
MPS-355	P-1A	Type 1	118/108	80° x 60°	70-20K	2" x 5"horn loaded	MS-5 (x2)		16Ω
	MPS-3	1 Channel				piezoelectric			
MPS-305	MPS-3	Type 1	115/105	80° x 60°	75-20K	2" x 5"horn loaded	MS-5		8Ω
		1 Channel				piezoelectric			
DS-2	D-2	Type 2	148/136	120° x 120°	50-160	N/A	MS-15 (x2)	160 Hz	4Ω
		1 Channel	(2 units)	(2 units)					
USW-1	B-2EX	Type 1 (1 Ch.)	135/130		40-100	N/A	MS-15 (x2)	100 Hz	4Ω
		Type 2 (1 Ch.)							
MSW-2	B-2EX	Type 1 (1 Ch.)	130/124		35-110	N/A	MS-18	100 Hz	4Ω
		Type 2 (1 Ch.)							
650-R2	B-2EX	Type 1 (1 Ch.)	135/130		30-100	N/A	MS-18 (x2)	110 Hz	8Ω
		Type 2 (1 Ch.)							
HF-3	M-3A	Type 1 (1 Ch.)	135/130	70° x 60°	700-18K	MS-2001A	N/A	700 Hz	12Ω
	S-1	Type 2 (1 Ch.)						900 Hz	
MST-1	T-1A	Type 1	/130	30° x 20°	8K-20K	Piezo Horn (x30)	N/A	8 kHz	8Ω
		1 Channel							

* See Amplifier Output Classifications, Page 28.

3.10.2 Externally Powered Speaker Systems

	Physical					
Speaker	**Connector**	**Finish**	**Dimensions**	**Weight**	**Rigging**	**Transport**
			Per Unit	Per Unit		
MSL-10A	Pyle w/weather	Weatherproof black	41"W x 85"H x 35"D	700 lbs.	8 points, 3/4" rigging holes	N/A
	cap, heavy AC in	coating		(318 kg.)	in steel cradle	
MSL-5	EP-4 (5), Pyle	Textured,carpet	42 1/2"W x 56 3/4"H	500 lbs.	12 points, pivoting lift rings,	Caster Board
		or weatherproof	x 32"D	(227 kg.)	1500 lb. safe load capacity	Optional
MSL-3A	EP-4(5) male	Textured,carpet	21 1/4"W x 56 3/4"H	241 lbs.	Aircraft pan fittings	Caster Board
	Pyle	or weatherproof	x 30"D	(109.3 kg.)		Optional
MSL-2A	EP-4(5) male	Textured	21 1/4"W x 24 1/4"H	82 lbs.	Aircraft pan fittings, 3/8"-16	Caster Board
		or weatherproof	x 18 1/4"D	(37 kg.)	or M10 x 1.5 nut plates	Optional
USM-1	EP-4(5) male	Textured	21"W x 24 1/4"H x	82 lbs.	Aircraft pan fittings or 3/8"-16	N/A
		or weatherproof	18"D	(37.3 kg.)	or M10 x 1.5 nut plates	
UM-1C	EP-4(5) male	Textured	14"W x 14"H x	61 lbs.	Aircraft pan fittings or 3/8"-	N/A
		or weatherproof	22 1/2"D	(27.7 kg.)	16 nut plates	
UPA-1C	EP-4(5) male	Textured	14 1/2"W x 22 3/8"H	67 lbs.	Aircraft pan fittings or 3/8"-	N/A
		or weatherproof	x 14 1/2"D	(30.4 kg.)	16 nut plates	
UPA-2C	EP-4(5) male	Textured	14 1/2"W x 22 3/8"H	67 lbs.	Aircraft pan fittings or 3/8"-	N/A
		or weatherproof	x 14 1/2"D	(30.4 kg.)	16 nut plates	
UPM-1	EP-4 or 3-pin XLR	Black textured	6 3/4"W x 18 1/8"H	16 lbs.	3/8 "-16 nut plates	N/A
	male and female		x 7 1/8"D	(7.3 kg.)		
UPM-2	EP-4 or 3-pin XLR	Black textured	6 3/4"W x 18 1/8"H	16 lbs.	3/8 "-16 nut plates	N/A
	male and female		x 7 1/8"D	(7.3 kg.)		
MPS-355	XLR (A-3) or	Black textured	6 3/4"W x 10 1/2"H	11 lbs.	3/8 "-16 nut plates	N/A
	Speak-On™		x 7"D	(5 kg.)		
MPS-305	XLR (A-3) or	Black textured	6 3/4"W x 18"H x	6.6 lbs.	3/8 "-16 nut plates	N/A
	Speak-On™		7"D	(3 kg.)		
DS-2	EP-4(5) male	Textured,carpet	21 1/4"W x 56 3/4"H	218 lbs.	Aircraft pan fittings	Caster Board
		or weatherproof	x 30"D	(98.9 kg.))		Optional
USW-1	EP-4(5) male	Textured	31"W x 21 1/2"H x	115 lbs.	Aircraft pan fittings,3/8"-16 or	N/A
		or weatherproof	21 1/2"D	(52.2 kg.)	M-10 nut plates	
MSW-2	EP-4(5) male	Textured	21 1/4"W x 24 1/4"H	66 lbs.	Aircraft pan fittings or 3/8"-16	N/A
		or weatherproof	x 20 1/4"D	(30 kg.)	or M-10 nut plates	
650-R2	EP-4(5) male	Textured,carpet	30 "W x 45"H x	176 lbs.	N/A	3" diameter
		or weatherproof	22 1/2"D	(79.8 kg.)		rubber casters
HF-3	EP-4(5) male	Black textured	21 1/4"W x 9"H	50 lbs.	Aircraft pan fittings or 3/8"-16	N/A
		over steel	x 18"D	(22.7 kg.)	or M-10 nut plates	
MST-1	EP-4(5) male	Black textured	19 3/4"W x 7 1/2"H	17 lbs.	N/A	N/A
		over steel	x 5 1/16"D	(7.7 kg.)		

3.10.3 Shipping Weights and Dimensions (Metric)

Meyer Sound Product Weights and Dimensions						
Specification	Unit Weight	Ship Weight	Shipping Dimensions (cm)			
Product			W	H	D	Vol (cu. meters)
MSL-10A	317.8 kg	317.8 kg	104	216	84	1896
MSL-6	215.7 kg	238.4 kg	104	122	84	1064
MSL-5	227.0 kg	249.7 kg	104	157	84	1375
MSL-4	81.7 kg	94.9 kg	84	107	61	545
MSL-3A painted	109.4 kg	123.9 kg	84	152	61	779
MSL-3A carpeted	116.2 kg	130.8 kg	84	152	61	779
MSL-2A	37.2 kg	46.3 kg	66	76	64	320
CQ-1	59.0 kg	68.1 kg	64	89	67	380
CQ-2	59.0 kg	68.1 kg	64	89	67	380
USM-1	37.2 kg	40.9 kg	76	69	66	345
PSM-2	40.9 kg	47.7 kg	71	28	51	101
MTS-4	127.1 kg	147.6 kg	84	152	61	779
DS-2 painted	99.0 kg	113.5 kg	84	152	61	779
DS-2 carpeted	105.8 kg	120.3 kg	84	152	61	779
DS-2P painted	110.3 kg	131.2 kg	84	152	61	779
650-R2 painted	79.9 kg	92.2 kg	86	127	69	752
650-R2 carpeted	87.2 kg	99.4 kg	86	127	69	752
650-P	91.3 kg	104.4 kg	86	127	69	752
PSW-2	73.5 kg	91.7 kg	84	107	61	545
PSW-4	93.1 kg	106.2 kg	84	107	61	545
USW-1	52.2 kg	59.9 kg	89	66	69	403
MSW-2	30.0 kg	32.2 kg	66	76	64	320
UPA-1C	30.4 kg	33.1 kg	48	69	48	160
UM-1C	30.4 kg	33.1 kg	48	69	48	160
UPM-1	7.3 kg	9.5 kg	58	30	30	54
MPS-355	5.0 kg	5.9 kg	58	30	30	54
MPS-305	3.2 kg	4.1 kg	41	30	30	38
HM-1	5.0 kg	5.9 kg	31	38	34	40
HD-1	23.2 kg	30.0 kg	58	64	48	179
HD-2	31.8 kg	34.1 kg	71	64	53	241
UPL-2	31.8 kg	34.1 kg	71	64	53	241
UPL-1	31.8 kg	34.1 kg	71	64	53	241
MST-1 (for 2)	6.4 kg	7.3 kg	56	36	36	71
SIM system II (2201)	19.5 kg	22.2 kg	76	61	30	142
SIM 2403	17.7 kg	20.0 kg	61	28	56	95
M-1A	3.6 kg	4.1 kg	8	53	30	12
B-2EX	3.6 kg	4.1 kg	8	53	30	12
M-3A	3.6 kg	4.1 kg	8	53	30	12
D-2	3.6 kg	4.1 kg	8	53	30	12
S-1	3.6 kg	4.1 kg	8	53	30	12
P-1	3.2 kg	3.6 kg	8	53	30	12
M-5	3.6 kg	4.1 kg	8	53	30	12
M-10A	4.1 kg	4.5 kg	8	53	30	12
MPS-3	3.6 kg	4.1 kg	8	53	30	12
VX-1	4.5 kg	5.0 kg	8	53	30	12
CP-10, CP-10S	5.9 kg	7.3 kg	18	51	36	32
LD-1, LD-1A	5.9 kg	7.3 kg	18	51	36	32
MP-2 Module	12.7 kg	14.5 kg	43	43	38	71
MP-4 Module	14.5 kg	16.3 kg	43	43	38	71
MS-5	1.4 kg	1.8 kg	15	10	15	2
MS-12	7.9 kg	10.0 kg	36	18	36	22
MS-15	8.8 kg	10.9 kg	41	20	41	34
MS-18	9.7 kg	12.7 kg	48	20	48	47
MS-18	5.9 kg	6.4 kg	20	20	20	8
MS-1401A	5.4 kg	5.9 kg	20	20	20	8

3.10.3 Shipping Weights and Dimensions (English)

Meyer Sound Product Weights and Dimensions						
Specification	**Unit Weight**	**Ship Weight**	**Shipping dimensions**			
Product			W	H	D	Vol (cu. feet)
MSL-10A	700 lb	700 lb	41 in	85 in	33 in	67.0
MSL-6	475 lb	525 lb	41 in	48 in	33 in	37.6
MSL-5	500 lb	550 lb	41 in	62 in	33 in	48.5
MSL-4	180 lb	209 lb	33 in	42 in	24 in	19.3
MSL-3A painted	241 lb	273 lb	33 in	60 in	24 in	27.5
MSL-3A carpeted	256 lb	288 lb	33 in	60 in	24 in	27.5
MSL-2A	82 lb	102 lb	26 in	30 in	25 in	11.3
CQ-1	130 lb	150 lb	25 in	35 in	27 in	13.4
CQ-2	130 lb	150 lb	25 in	35 in	27 in	13.4
USM-1	82 lb	90 lb	30 in	27 in	26 in	12.2
PSM-2	90 lb	105 lb	28 in	11 in	20 in	3.6
MTS-4	280 lb	325 lb	33 in	60 in	24 in	27.5
DS-2 painted	218 lb	250 lb	33 in	60 in	24 in	27.5
DS-2 carpeted	233 lb	265 lb	33 in	60 in	24 in	27.5
DS-2P painted	243 lb	289 lb	33 in	60 in	24 in	27.5
650-R2 painted	176 lb	203 lb	34 in	50 in	27 in	26.6
650-R2 carpeted	192 lb	219 lb	34 in	50 in	27 in	26.6
650-P	201 lb	230 lb	34 in	50 in	27 in	26.6
PSW-2	162 lb	202 lb	33 in	42 in	24 in	19.3
PSW-4	205 lb	234 lb	33 in	42 in	24 in	19.3
USW-1	115 lb	132 lb	35 in	26 in	27 in	14.2
MSW-2	66 lb	71 lb	26 in	30 in	25 in	11.3
UPA-1C/2C	67 lb	73 lb	19 in	27 in	19 in	5.6
UM-1C	67 lb	73 lb	19 in	27 in	19 in	5.6
UPM-1/UPM-2	16 lb	21 lb	23 in	12 in	12 in	1.9
MPS-355	11 lb	13 lb	23 in	12 in	12 in	1.9
MPS-305	7 lb	9 lb	16 in	12 in	12 in	1.3
HM-1	11 lb	13 lb	12 in	15 in	13 in	1.4
HD-1	51 lb	66 lb	23 in	25 in	19 in	6.3
HD-2	70 lb	75 lb	28 in	25 in	21 in	8.5
UPL-2	70 lb	75 lb	28 in	25 in	21 in	8.5
UPL-1	70 lb	75 lb	28 in	25 in	21 in	8.5
MST-1 (for 2)	14 lb	16 lb	22 in	14 in	14 in	2.5
SIM system II (2201)	43 lb	49 lb	30 in	24 in	12 in	5.0
SIM 2403	39 lb	44 lb	24 in	11 in	22 in	3.4
M-1A	8 lb	9 lb	3 in	21 in	12 in	0.4
B-2EX	8 lb	9 lb	3 in	21 in	12 in	0.4
M-3A	8 lb	9 lb	3 in	21 in	12 in	0.4
D-2	8 lb	9 lb	3 in	21 in	12 in	0.4
S-1	8 lb	9 lb	3 in	21 in	12 in	0.4
P-1	7 lb	8 lb	3 in	21 in	12 in	0.4
M-5	8 lb	9 lb	3 in	21 in	12 in	0.4
M-10A	9 lb	10 lb	3 in	21 in	12 in	0.4
MPS-3	8 lb	9 lb	3 in	21 in	12 in	0.4
VX-1	10 lb	11 lb	3 in	21 in	12 in	0.4
CP-10, CP-10S	13 lb	16 lb	7 in	20 in	14 in	1.1
LD-1, LD-1A	13 lb	16 lb	7 in	20 in	14 in	1.1
MP-2 Module	28 lb	32 lb	17 in	17 in	15 in	2.5
MP-4 Module	32 lb	36 lb	17 in	17 in	15 in	2.5
MS-5	3 lb	4 lb	6 in	4 in	6 in	0.1
MS-12	18 lb	22 lb	14 in	7 in	14 in	0.8
MS-15	19 lb	24 lb	16 in	8 in	16 in	1.2
MS-18	21 lb	28 lb	19 in	8 in	19 in	1.7
MS-2001A	13 lb	14 lb	8 in	8 in	8 in	0.3
MS-1401A	12 lb	13 lb	8 in	8 in	8 in	0.3

4.1 Introduction

Now that the system is designed and installed, it is time to begin the tedious task of verification. All of the careful design work amounts to little if the installation is not performed correctly. The signal path—from its original source through the sound reinforcement system to the listener—has a multitude of opportunities for inadvertent polarity reversals, unplanned signal loss (and gain), mispatching, and component compatibility problems. Each of these components will need to be verified both individually and as a whole system.

Let's use an example in order to illustrate the verification process. Fig 4.1a is a flow diagram of a fairly simple sound reinforcement system. The system includes various input channels, inserted effects and multiple mixer outputs routed to a four-way main system and downfill speakers. Also shown are the necessary tests to ensure the system is operating properly.

Sections 4.1 through 4.7 detail the types of problems one may encounter, with the remainder focusing on techniques to solve them.

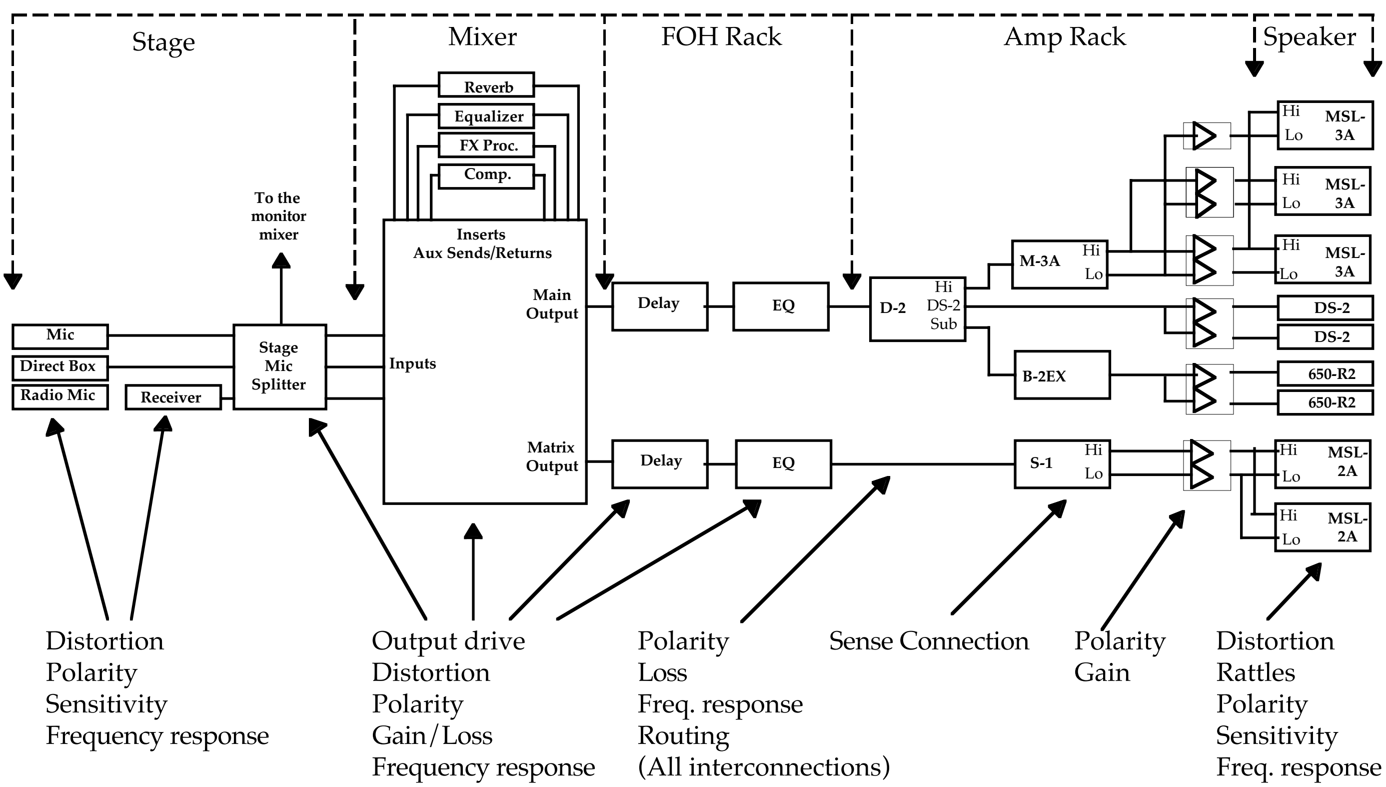
From the standpoint of verification we can subdivide the system into six sections.

- **Stage:** Microphones, direct boxes, microphone splitters.
- **Mixer:** Mixing console (desk), outboard effects, compressors, gates, equalizers, patch bays.
- **Front of house (FOH) rack**System equalizers, delay lines.
- **Amplifier racks:** CEUs, amplifiers.
- **Speakers:** LF, MF and HF drivers, networks.

And finally the one that puts it all together;

- **Connections:** Cable, connectors, wireless transmitters and receivers.

Fig 4.1a verification flow block.



4.2 Stage Components

Most of the testing on stage is basic continuity and balanced line verification. In addition, splitter networks should be verified for polarity, frequency response, headroom, distortion and signal loss. Fig 4.2a is a verification chart for stage components.

Stage component Verification	What to look for	Possible result	How to verify w/SIM®	Verify w/o SIM®
1) Microphones	Check linearity of the mic for consistency, and suitability.	If multiple units of the same mic model are inconsistent, compatibility and interchangeability problems arise.	SIM® System II: Microphone compare	Ear
	Check microphone directional pattern control.	Poor directional control will decrease gain before feedback and reduce isolation.	SIM® System II: Microphone compare	Ear, ring for feedback
	Check microphone polarity.	Polarity reversals between microphones will create compatibility and interchangeability problems.	SIM® System II: Microphone compare	
2) Direct Inputs	Check frequency response, polarity, distortion and dynamic range.	Direct input boxes usually convert high impedance unbalanced inputs to low impedance balanced outputs. These can have frequency response anomalies or polarity reversals.	SIM®: Frequency response -Amplitude and Phase. SIM distortion and dynamic range tests	
3) Stage Mic Splitter	Check frequency response, polarity, distortion and dynamic range.	Splitters may be active or transformer type. Either can have frequency response anomalies or polarity reversals.	SIM®: Frequency response -Amplitude and Phase. SIM distortion and dynamic range tests	

Fig 4.2a Stage component verification.

4.3 Microphones

Most methods of testing microphones are highly subjective. RTAs, and voltmeters are useless for anything more than a "Go" or "No-go" level test. SIM System II can be used to compare microphones directly, giving high-resolution frequency and phase response information. The mic to be tested is compared to a reference microphone such as the Bruel & Kjaer 4007, placed in front of a linear, phase-aligned full-range loudspeaker such as the HD-1. The setup is shown in Fig 4.3a. Examples of data from this technique are shown in Figs 4.3b–4.3f.

Mic comparison measurements can be used to:

- Verify frequency and phase response.
- Analyze the directional pattern of the mic over frequency.
- Ensure compatibility of mics.
- Check for aging or damage.
- Match pairs (or sets) for maximum interchangeability.

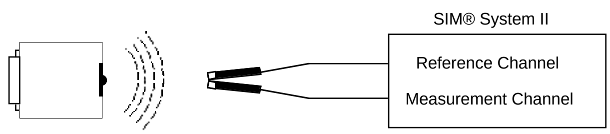


Fig 4.3a Mic comparison measurements using SIM System II.

The mic to be measured is compared to a reference microphone. The mics are placed close together in a symmetrical sound field.

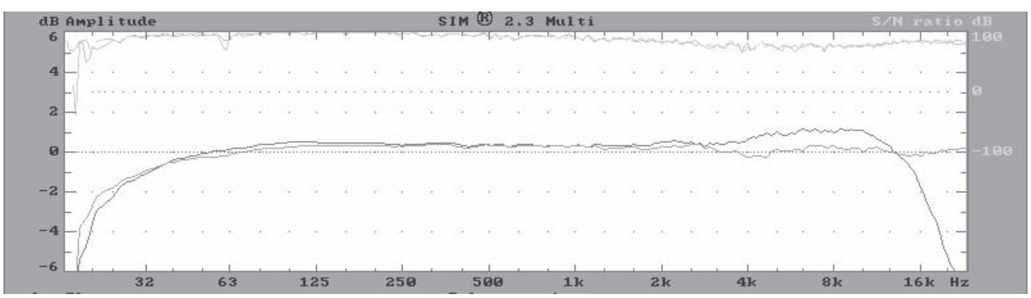


Fig 4.3b Comparison of two high-quality omnidirectional condensor mics. One of them costs three times as much as the other—Can you guess which one?

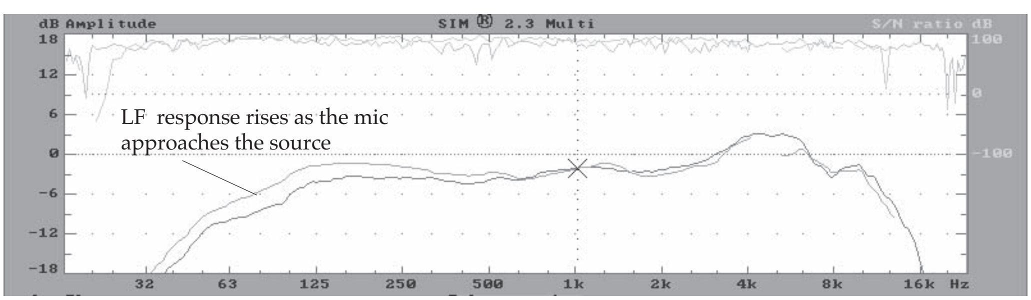


Fig 4.3c Cardioid mic exhibiting the "proximity effect" as it nears the source.

4.3 Microphones

Comparing Two of the Same Model Mics

The frequency response of two mics can be compared so that compatibility can be assured. The mic's responses are similar but differ in overall sensitivity by 2 dB. This same type of test can be utilized to compare the response of the same mic's response over time to detect aging or damage to the mic by comparing it against its previously stored response.

Comparison of Omnidirectional and Cardioid Versions of a Mic

The on-axis response is shown illustrating the differences caused by cardioid pattern control. The diaphragm is the same in both cases. This technique could also be used to preview microphone responses to aid in the placement and selection process.

Directional Pattern Control of a Hypercardioid Mic

The on- and off-axis responses of this mic are compared to best evaluate its suitability for a highly directional applications. The rejection can be read directly at any frequency. Notice that the phase response at the rear of the mic is 180° out of phase from that at the front.

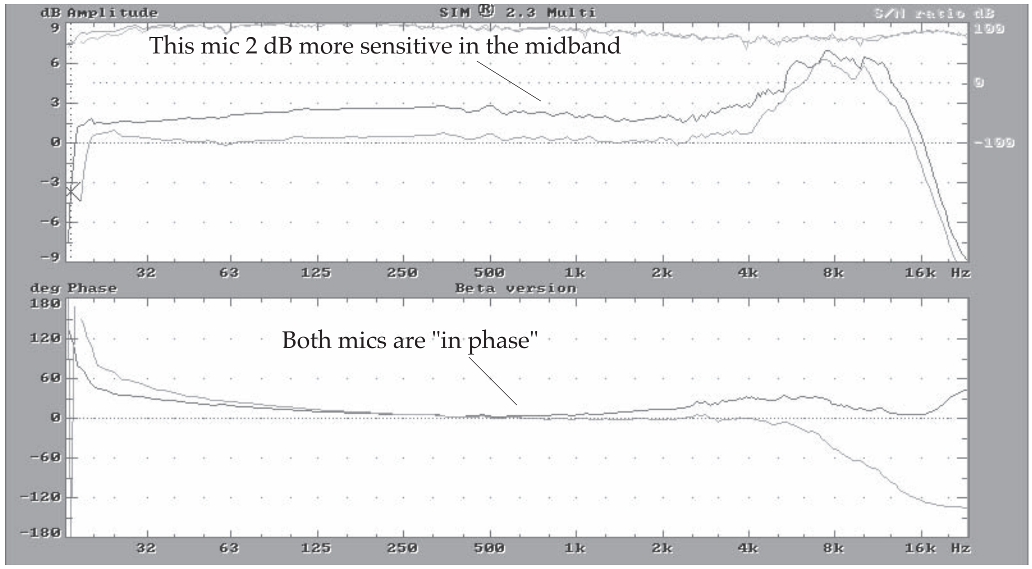


Fig 4.3d Microphone matching.

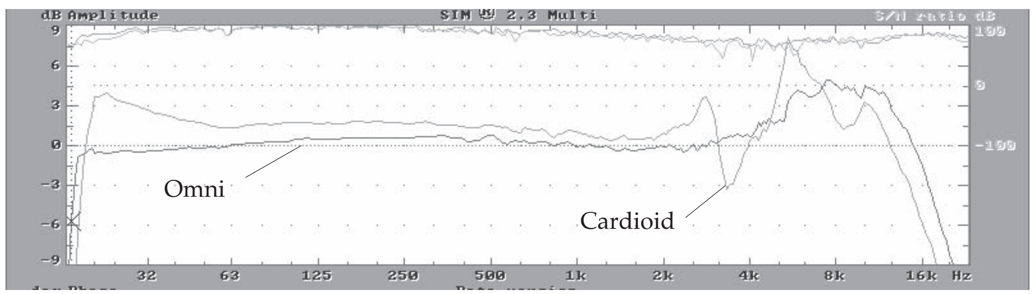


Fig 4.3e Frequency response changes with pattern switch setting.

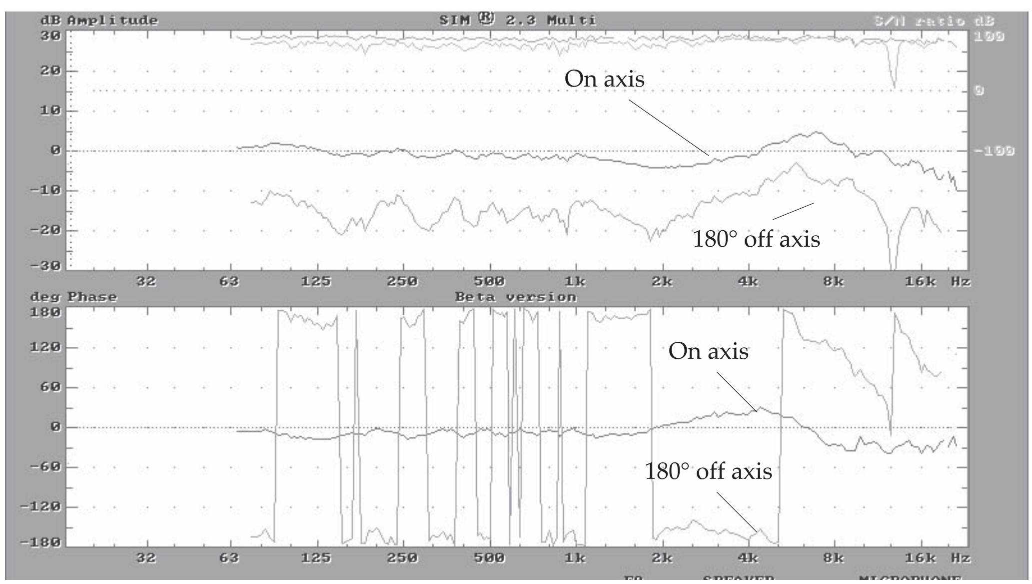


Fig 4.3f Microphone pattern verifiction.

4.4 Mixer Verification

A flow block diagram of an example mixer is shown in Fig 4.4a and 4.4b. Each numbered test point in the diagram refers to the tests included in the verification chart at right (Fig 4.4b).

Input Section

The input channel must be capable of conditioning a mic or line level signal to drive the output section. The insertion points and direct outputs should also be verified as these are sometimes overlooked by designers of low-cost mixers. Check the insertion points since they are often R-T-S phone jacks in which case anything goes. Meter accuracy and clip indication must also be verified.

Output Section

The mixing console (desk) will be tested to verify that it is capable of supplying sufficient output signal to drive the system to full power. Any professional console should be capable of +24 dBu output, *provided it is operated within its normal gain structure*. Models differ in the topology of input, subgroup, master and matrix gain structure. The goal is to get the signal from input to output without internal clipping. Meter accuracy and clip indication must also be verified.

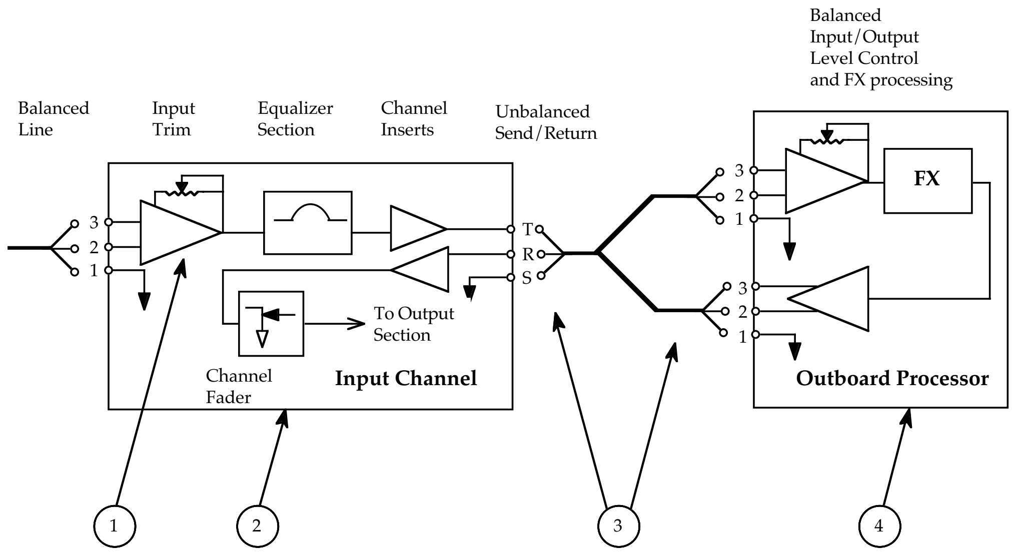


Fig 4.4a Input section verification.

Each numbered test point corresponds to the tests included in the verification chart at right.

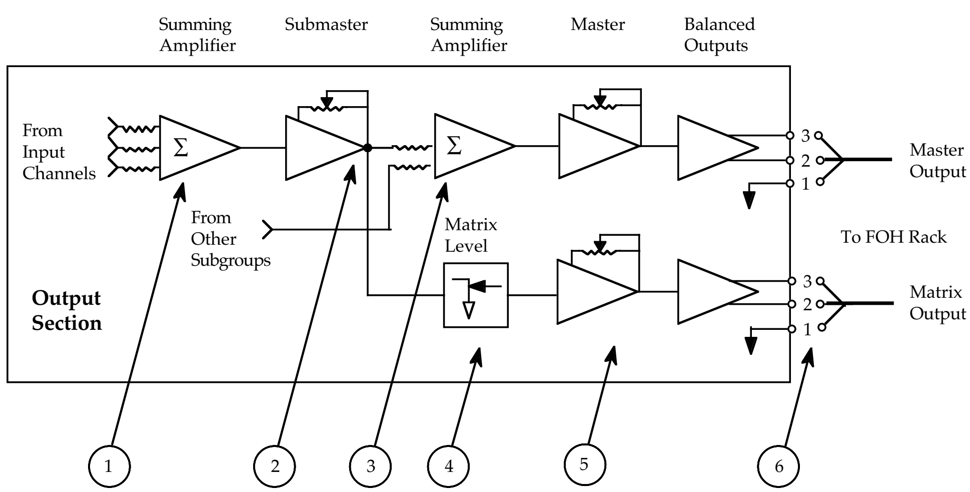


Fig 4.4b Output section verification.

4.4 Mixer Verification

Mixer Verification	What to look for	Possible result	How to verify w/SIM®	Verify w/o SIM®
Input Section				
1) Input gain trim	Check gain trim to verify THD, meters & overload indicator	We need to find a true indication of when O/L has occurred in order to optimize gain.	SIM®: Spectrum response: Maximum output capability test. Verify point at which the O/L indicator comes on and compare to actual clip point. Check status of meter vs. clip point.	
2) Channel fader	Is the fader accurate?	dB markings on channel faders can be misleading. May actually be 10 dB off.	SIM®: Frequency response -Amplitude: Find unity gain setting on the fader.	Oscillator and oscilloscope
3) Direct out and Insert	Check input channel direct output polarity	The direct output of a channel is sometimes polarity inverting. Others may be unbalanced.	SIM®: Frequency response -Amplitude and Phase: Compare the response of the input vs. the direct output.	
4) Inserted Outboard device	Check insert point gain and polarity	Insertion points are usually unbalanced R-T-S phone jacks. This can result in signal loss and polarity reversals on inserted devices.	SIM®: Frequency response -Amplitude and Phase: Balanced and unbalanced line tests. Compare the mixer output frequency response with the insert points engaged and bypassed.	
Output Section				
1) Channel Summing stage	Interstage clipping	Variations in internal gain structure can cause clipping without indication or excessive noise.	SIM®: Console check: Spectrum response: Maximum output capability test. Use various settings of channel, subgroup and master faders to find the variations in system output vs O/L point.	Oscillator and oscilloscope
2) Submaster	Check level	Submasters vary with level markings. the +10 dB marking may actually be the unity gain point. If the master has actual gain it may be capable of overloading the summing stage.	SIM®: Console check: Frequency response - Amplitude: Find unity gain setting on the fader.	
3) Submaster Summing stage	Interstage clipping	Variations in internal gain structure can cause clipping without indication or excessive noise.	SIM®: Console check: Spectrum response: Maximum output capability test. Use various settings of channel, subgroup and master faders to find the variations in system output vs. O/L point.	Oscillator and oscilloscope
4) Matrix Level	Check Level	Mismatches of Matrix levels can cause imbalances.	SIM®: Console check: Frequency response - Amplitude: Calibrate to a standard. Mark unity gain settings on mixer.	
5) Matrix output master	Check Level	Mismatches of Matrix and Master output levels can cause imbalances.	SIM®: Console check: Frequency response - Amplitude: Calibrate outputs to a standard. Mark unity gain settings on mixer.	
6) Output Drive	Gain structure	Variations in internal gain structure can cause clipping with indication or excessive noise.	SIM® Console check: Frequency response. Watch the gain structure to verify which stage have gain and which are passive. This will help to find the smoothest path from input to output with the least amount of gain changing.	Oscillator and oscilloscope
	Polarity	Most professional mixers utilize balanced inputs and outputs. Therefore, they are neither pin 2 or 3 hot. However, you may still find mixers with unbalanced outputs. These will need to be verified if you are planning to interface with additional mixers.	SIM®: Console check: Frequency response - Amplitude and Phase: Balanced line check. Compare the response of the input vs. the output.	
	Check output drive to verify THD, meters & overload indicator	We need to find a true indication of when O/L has occurred in order to optimize gain.	SIM®: Console check: Spectrum response: Maximum output capability test	Oscillator and oscilloscope
6) Matrix output master	Check Level	Mismatches of Matrix and Master output levels can cause imbalances.		Oscillator and oscilloscope

Fig 4.4c Mixer verification reference chart.

4.5 FOH Rack Verification

The front of house (FOH) rack consists of system alignment devices such as equalizers and delay lines and a variety of effects devices. A flow block diagram is shown in Fig 4.5a, along with its verification chart (Fig 4.5c).

The FOH rack will be tested to verify that it is capable of supplying sufficient output signal to drive the system to full power. Any professional equalizer or delay line should be capable of +24 dBu output, *provided it is operated within its normal gain structure*. Models differ in the input and output level controls and gain structure. The goal is to get the signal from input to output without internal clipping. In addition, the accuracy of the front panel indicators, such as filter frequency, delay time and metering, must be verified.

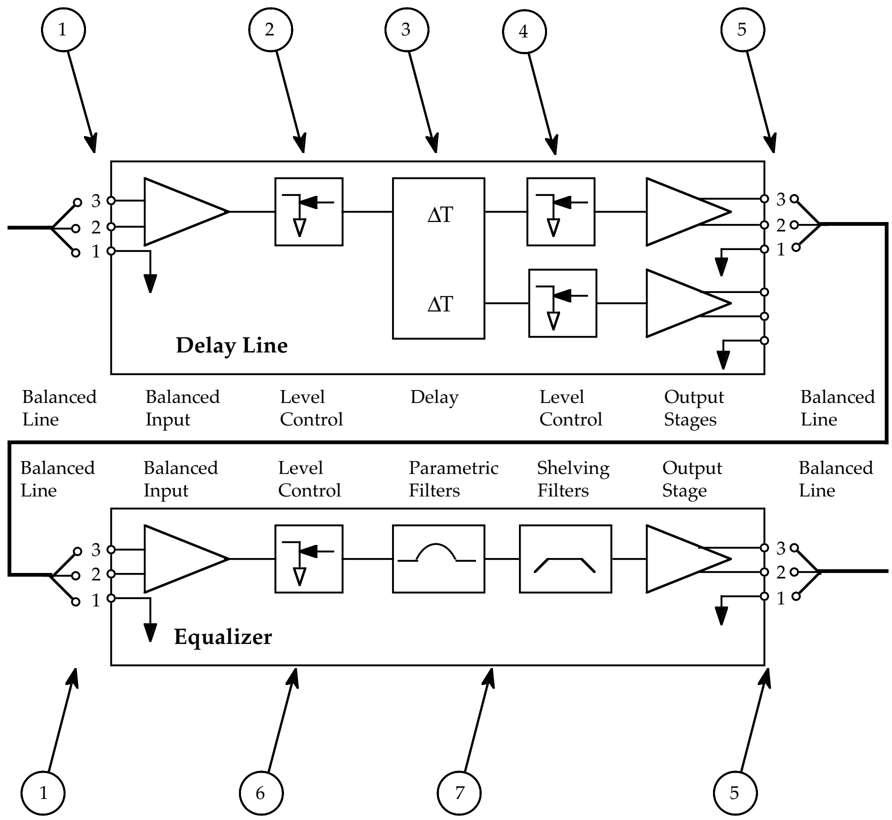


Fig 4.5a FOH rack verification.

Each numbered test point corresponds to the tests included in the verification chart at right.

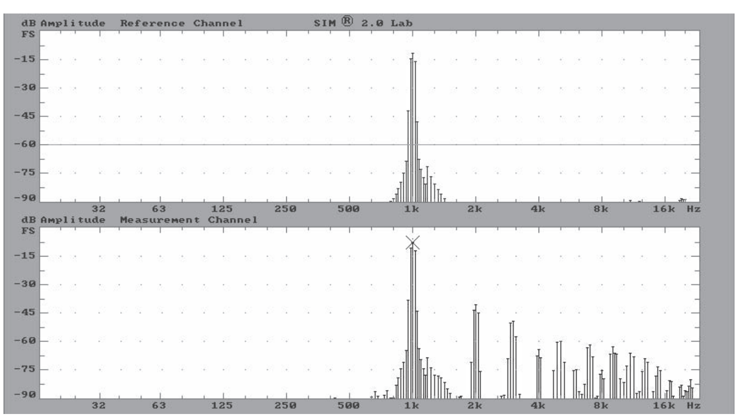


Fig 4.5b Distortion and maximum output testing with SIM.

A pure sine wave reference drives the system input (top) and is measured at the system output (bottom). The total harmonic distortion can be read directly and the overload point found, as seen by the large harmonics.

4.5 FOH Rack Verification

FOH rack Verification	What to look for	Possible result	How to verify w/SIM®	Verify w/o SIM®
1) Balanced Cable	Pins 2 & 3 reversed	Polarity reversal	SIM®: Frequency response: Amplitude and Phase: Balanced Line Check	
	Pins 2 or 3 open or shorted to pin 1	6 dB loss due to unbalancing of the line	SIM®: Frequency response: Amplitude and Phase: Balanced Line Check	
	Pins 2 & 3 shorted	Signal cancellation due to common mode input section	SIM®: Frequency response: Amplitude and Phase: Balanced Line Check	
2) Delay Line Input gain	Check for unity gain.	Equalizers and delays are best suited for unity gain operation. This is preferable from a standpoint of distortion and noise.	SIM®: Frequency response: Amplitude and Phase: Balanced Line Check	Oscillator and oscilloscope or VOM
	Check Input/output levels to verify THD, meters & overload indicator.	Find a true indication of when O/L has occurred to optimize gain. Delay lines often have separate level controls for input and output to drive the front end harder and attenuate the output (where the noise from the D/A converters is seen.)	SIM®: Spectrum response: Maximum output capability test. Use various settings of input and output gains to find the variations in system output vs O/L point.	Oscillator and oscilloscope
	Input level too low?	Possible overdrive of preceding stages. Excess noise in system due to extra gain in following stages.	SIM®: Spectrum response: Maximum output capability test. As above.	Oscillator and oscilloscope
	Input level too high?	Possible overdrive of input and following stages.	SIM®: Spectrum response: Maximum output capability test. As above.	Oscillator and oscilloscope
3) Delay Time	Does the unit have residual delay?	Every delay line has some some residual delay from the A/D conversion process. The front panel may not have factored this into its display calculation.	SIM® Delayfinder: Delay measurement test	
	Is the displayed delay time accurate?	The display may be inaccurate by design (usually a small amount) or because of a malfunction. The displayed delay time comes from the the user interface - not a measurement. If the unit is broken it may not tell you.	SIM® Delayfinder: Delay measurement test	
4) Delay Line Output gain	Output level too low?	Possible overdrive of preceding input stages.	SIM®: Spectrum response: Maximum output capability test. As above.	Oscillator and oscilloscope
	Output level too high?	Excess noise	SIM®: Spectrum response: Maximum output capability test. As above.	Oscillator and oscilloscope
5) Output Section	Is it balanced?	Some professional equalizers and delay lines offer balanced outputs as an add-on option. The unbalanced option is sometimes purchased to save money. Since they will probably drive the long line to the amp racks it is the worst place to go unbalanced.	SIM®: Frequency response: Amplitude and Phase: Balanced Line Check	
6) Equalizer Input gain	Same as for delay line above	Same as for delay line above	Same as for delay line above	Same as for delay line above
7) Equalizer section	Is the displayed equalization accurate?	The displayed eq response may not factor in the interaction between filters. Digital eq's or digital controlled analog eq's response display comes from the the user interface - not a measurement. If the unit is broken it may not tell you.	SIM®: Frequency response: Amplitude and Phase: Frequency response test	

Fig 4.5b FOH rack verification reference chart.

4.6 Amplifier Rack Verification

The amplifier rack is the last stop for line-level electronics. The connections between the CEU and the amplifier must be verified as it is vital for system protection. The amplifier voltage gain and polarity will be critical factors if systems from different sources are integrated. There are more opportunities for miswiring, incompatibility and improper operation in the amplifier rack than anywhere else in the system. This is a step that should never be bypassed.

An example amplifier rack flow block diagram is shown in Fig 4.6a along with its corresponding verification chart (Fig 4.6b).

If you are using the Meyer self-powered series speakers you don't have to deal with this!

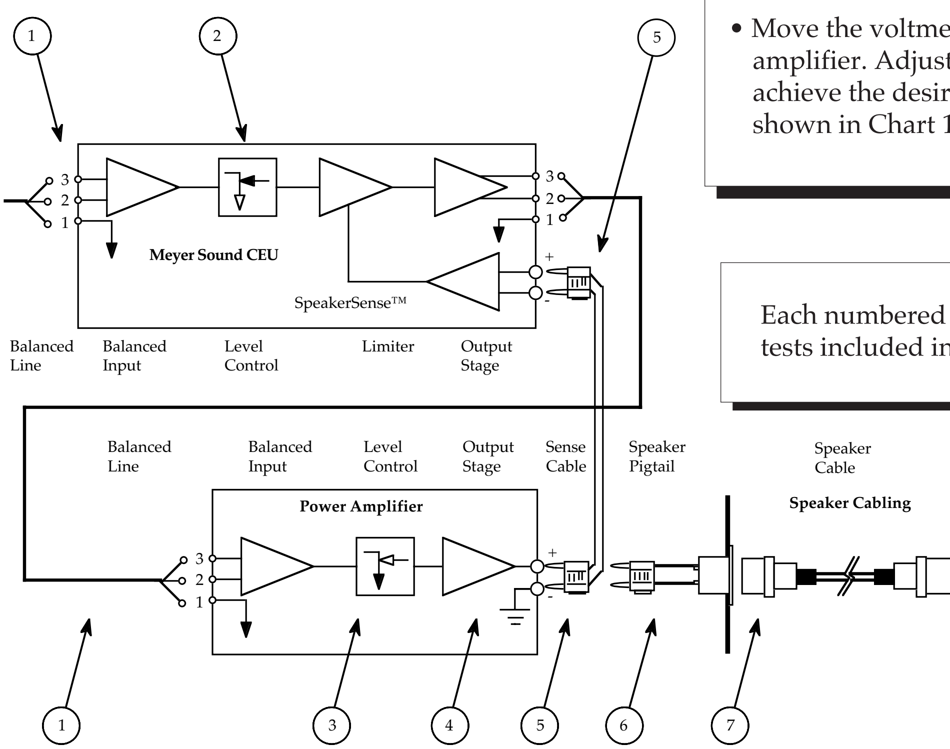


Fig 4.6a Amplifier rack verification.

Verifying Amplifier Voltage Gain

Amplifier voltage gain is a critical parameter to the operation of the system, as described in Section 1.4. It can be easily measured with a simple AC voltmeter and a sine wave generator.

To measure amplifier voltage gain:

- Disconnect the amplifier from the CEU and speakers.
- Input the sine wave generator to the amplifier. If the VOM is not frequency independent, set the sine wave frequency to 60 Hz.
- Measure the voltage across pins 2 and 3 at the output of the generator. Adjust the generator output to 1 VAC. The generator should have a low impedance output so that the input impedance of the amplifier does not load it down. To verify that the amplifier is not loading the generator down you can monitor the level while it is plugged into the amplifier input.
- Move the voltmeter to read the output of the amplifier. Adjust the amplifier level controls to achieve the desired standard voltage gain as shown in Chart 1.4a.

Each numbered test point corresponds to the tests included in the verification chart at right.

4.6 Amplifier Rack Verification

Amp Rack Verification	What to look for	Possible result	How to verify w/SIM®	Verify w/o SIM®
1) Balanced Cable	As in previous section	As in previous section	As in previous section	As in previous section
2) Ceu Level	Is the CEU level set lower than -15 dB?	Loss of headroom in FOH system. It will be difficult to drive the amplifier to full power without clipping the FOH or mixer stages. If CEU level is set too low then amplifiers tend to be set too high.		
3) Amplifier Voltage Gain	Excess voltage gain: (>30 dB)	Decreased Speakersense™ protection, excess hum & noise.		
	Insufficient voltage gain (<10 dB)	Amplifier cannot be driven to full power with overloading the CEU and preceding components.		
	Nonstandardized voltage gain	If multiple amplifier sections without the same gain, acoustical crossover will shift causing possible phase cancellations, peaks and dips.	Measure frequency response and verify whether the crossover is at the correct frequency range.	
4) Amplifier Polarity	Is the amplifier pin 2 or 3 "hot"?	Possible polarity reversals when interfacing your system with amplifiers from different manufacturers.	See following section on polarity verification	
5) Banana Plug mispatch:	Is the banana plug wired or inserted incorrectly?	Polarity reversal. Ground wire (-) should be connected to the ribbed side of the banana connector and inserted into the black terminal of the power amplifier.	See following section on polarity verification	
	Is the banana plug patched to the incorrect channel?	Channel reversal of HF and LF drive to speaker. Caution: This may damage the loudspeakers.		
5) SpeakerSense™ mispatch:	Is the sense line hooked up?	No driver protection.		Sense LED will not light when signal is applied (M-1, M-1A, M3, P-1A, P-2, MPS-3 or B-2(all). Sense LED will turn red when signal is applied (M-1E, S-1, M-3A, D-2, M-5, M-10A)
	Is the sense line connected to the incorrect channel?	Channel reversal of HF and LF sense line. Limiters will not properly protect the drivers.		
	Is the banana plug inserted incorrectly?	Polarity is not important on sense connections except for those CEU's with Multisense™. Then, if multiple sense inputs per channel are used the effectivity of the sense circuit will be defeated.		
6) Speaker pigtail miswired:	Does the pigtail wiring conform to the standard?	Polarity reversal, channel reversal or no sound.		
	Is the correct pigtail being used?	If a subwoofer pigtail is used to drive biamplified full-range products (such as the UPA) there will be no sound. The inverse of this is also true.		
				VOM: Speaker cable continuity test.

Fig 4.6b Amplifier rack verification reference chart.

4.7 Speaker Enclosure Verification

Because speakers are the last point in the signal chain they are often blamed for problems that occurred upstream. Inevitably the problem may be in the speaker itself. Blown, damaged or fatigued drivers, internal miswiring, loose screws or damage to the enclosure are the usual failure modes. Fig 4.7a is a verification chart for speakers

Counterfeit Enclosures

If you bought used speakers or are renting them from a disreputable company you may find yourself with counterfeit drivers or enclosures. There are various companies that have manufactured copies of Meyer speakers and marketed them as authentic. Fortunately, the poor workmanship of such systems makes them easy to detect for any experienced Meyer user.

Counterfeit Drivers

A more subtle form of counterfeiting is the substitution of other manufacturer's drivers or HF diaphragms. Meyer drivers can only be repaired at the factory in Berkeley, California, or at the Meyer Sound European Service Center. If someone tells you they reconed a speaker or replaced a diaphragm, you know that it is not a real Meyer part. If you suspect that you have such parts please call Meyer Sound to help determine if it is authentic.

Internal Wiring

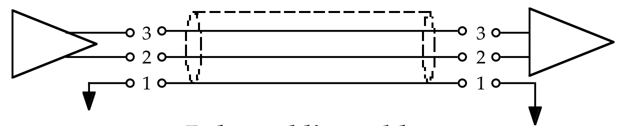
If the speakers have been field serviced it is possible that the speakers have been internally miswired. This is relatively rare, however. Opening the cabinet should be a measure of last resort in polarity verification.

Speaker Verification	What to look for	Possible result	How to verify w/SIM®	Verify w/o SIM®
1) Internal Miswiring	Red wire should go to red terminal. Black to Black. For HF Drivers White goes to red and Green to Black. This should only be considered as a cause if the drivers have been field serviced at some time in the past.	Polarity cancellation between units or at crossover.	See following section on polarity verification	Visual check
2) Open or shorted driver	No sound (open) or very low level (shorted)	Shorted driver can damage the power amplifier		
3) Rub, buzz, rattle	Loose screws. Damaged enclosure. Partially deformed driver former. Exhausted surround.	Mechanical noises and distortion increasing dramatically as output level rises.	Sine wave sweep, Distortion test	Physically inspect cabinet and driver. Sweep with sinewave and listen for mechanical noise.
4) Alien Driver Substitution	All Meyer Sound driver components have a Meyer serial # sticker on them. This should only be considered as a cause if the drivers have been field serviced at some time in the past.	Anything goes	Measure frequency response and compare to known Meyer driver	
5) Alien speaker enclosure	If the cabinet does not have a Meyer S/N in the handle cups or on the cabinet rear (smaller speakers) it may not be authentic.	Anything goes	Measure frequency response and compare to known Meyer speaker	

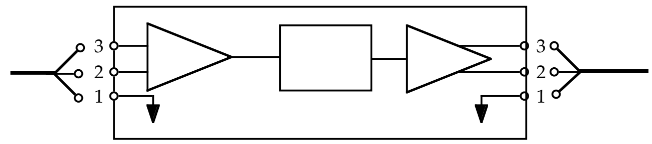
Fig 4.7a Speaker verification reference chart.

4.8.1 Normal

The continuity of cables can be easily and reliably verified by a wide range of devices. These include ohmmeters, dedicated cable testers, and frequency analyzers. Testing the balanced throughput on a line level device or transformer is best performed by a frequency analyzer, such as SIM System II. The parameters to be tested include continuity, polarity, frequency and phase response, S/N ratio and distortion.



Balanced line cable.



Balanced line device.

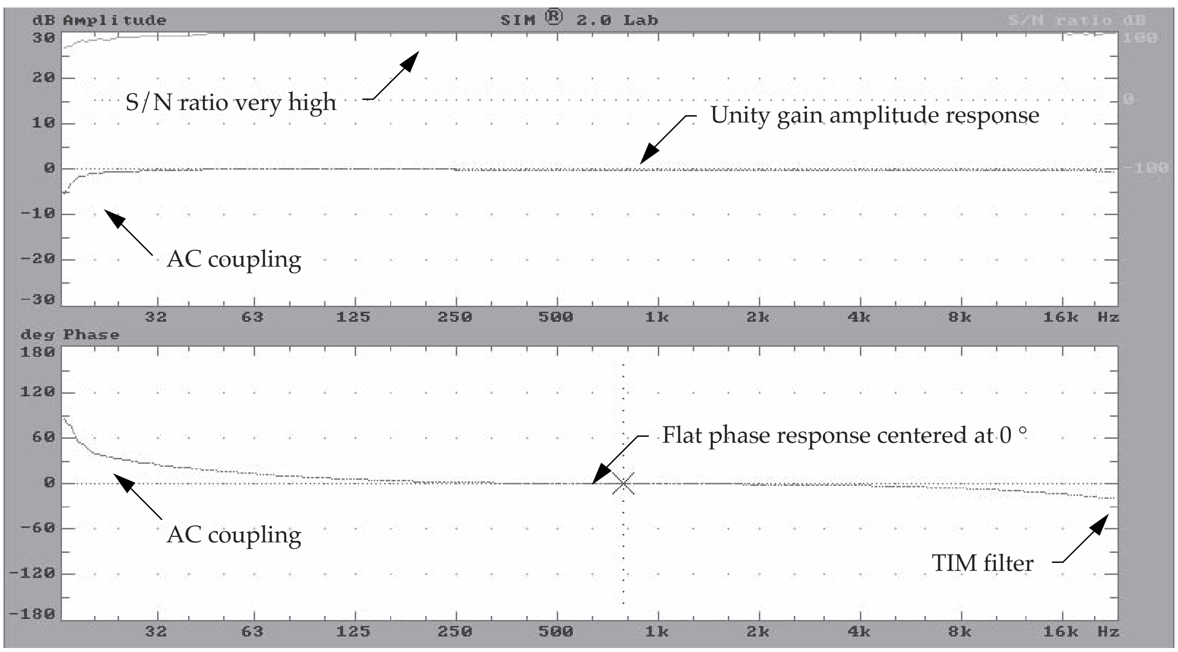


Fig 4.8a Normal balanced line. The amplitude trace shows unity gain. The phase trace is centered at 0°. This indicates a non-inverting polarity. The AC coupling and transient intermodulation (TIM) filters will only be found on active devices and transformers (not cables).

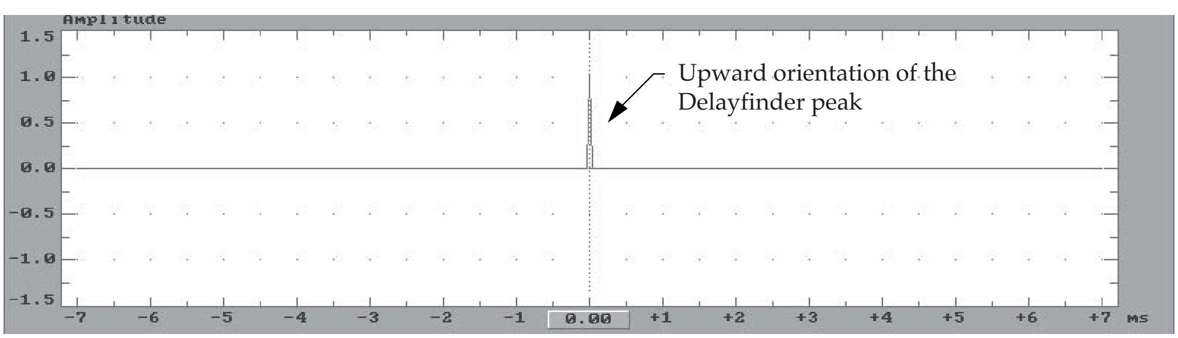
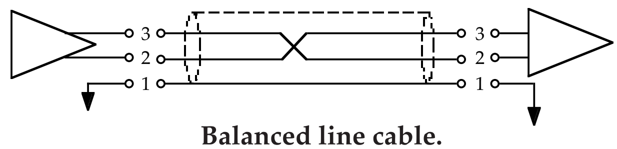


Fig 4.8b Normal balanced line. The peak of the delayfinder trace is pointing upward. This indicates a non-inverting line.

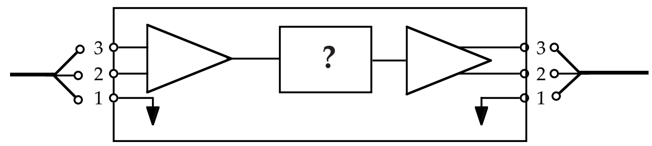
4.8.2 Polarity Reversal

There are only three wires in a typical balanced mic cable and yet there seem to be dozens of creative methods of connecting them. The most typical error is the reversal of pins 2 and 3. The result is a reversal of polarity.

Balanced line devices are *not necessarily non-inverting.* Check it!



Balanced line cable.



Device with balanced inputs and outputs.

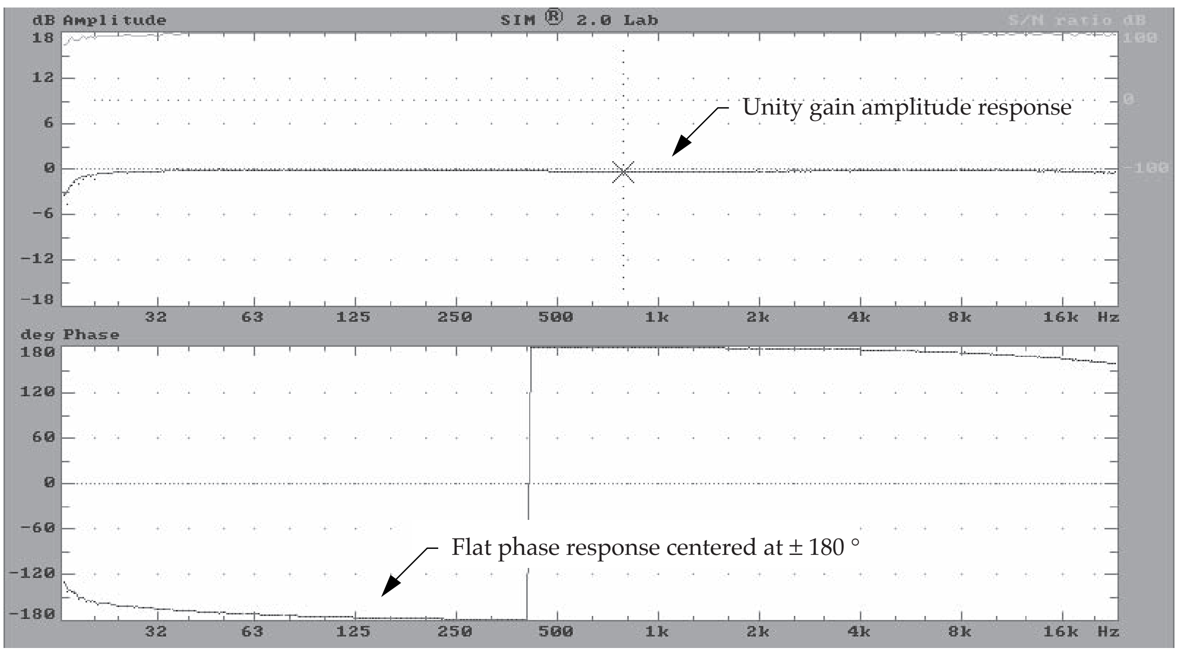


Fig 4.8c Polarity reversal of a balanced line. The amplitude trace is unaffected, however, the phase trace is centered at 180°. This indicates a polarity inversion.

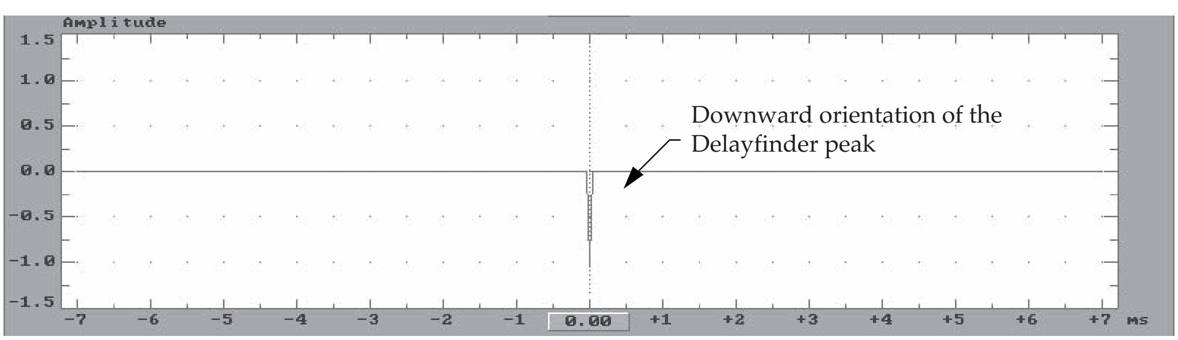


Fig 4.8d Polarity reversal of a balanced line. The peak of the delayfinder trace is pointing downward. This indicates that the system is "net inverting."

4.8.3 Unbalanced Lines

The great majority of interconnections between devices are made with balanced lines, preferred for their superior noise rejection and maximum level. Balanced lines utilize two polarity reversed signals creating a differential voltage between them. It is possible to lose one of the two signal lines yet continue to pass signal with a 6 dB level reduction. Such cables can be difficult to detect, often causing level mismatches.

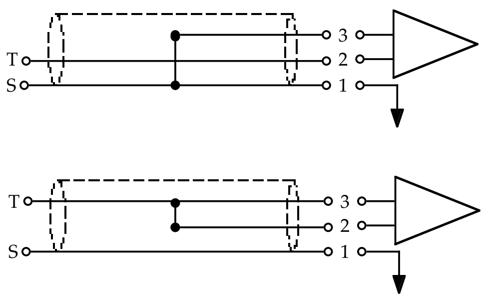


Fig 4.8e Unbalanced cable with mono phone jack.

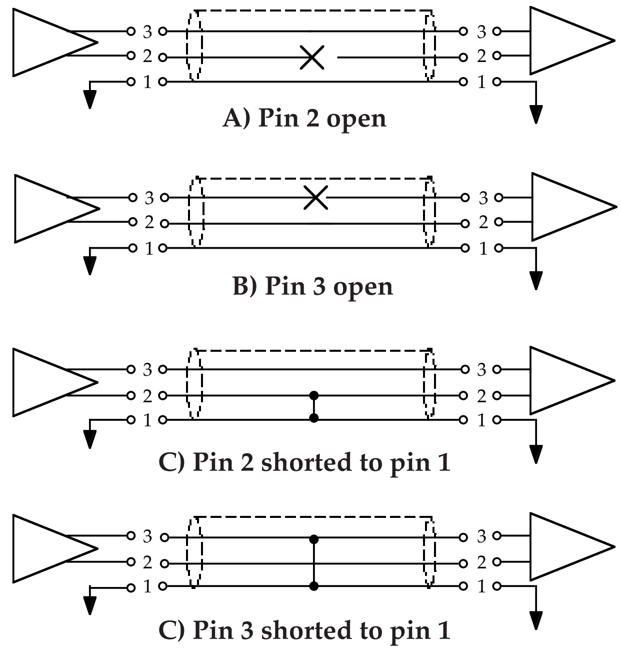


Fig 4.8f Unbalancing of a balanced line. Each of the above scenarios causes a 6 dB signal loss.

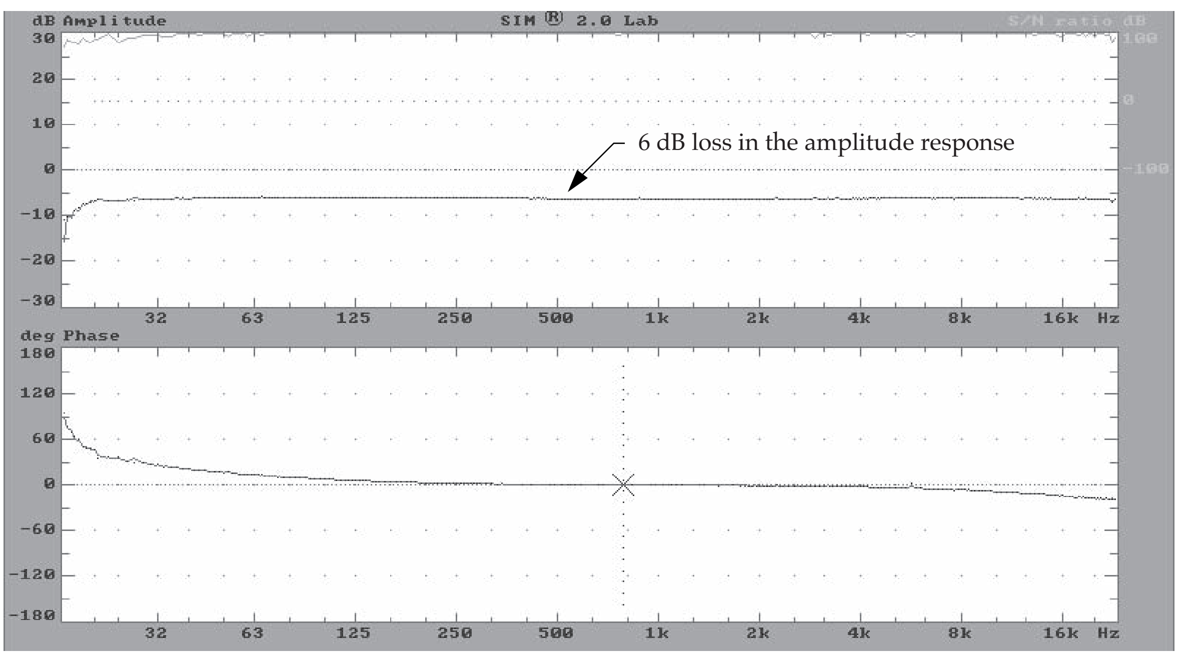


Fig 4.8g Frequency response of a balanced line that has become unbalanced. The amplitude response shows a broadband 6 dB loss due to the loss of one-half of the voltage drive. The phase response is unchanged from the balanced scenario

4.8.4 Unbalanced Cable Field Example

An unbalanced line can cause subtle problems that are difficult to detect. Here is an example of a concert hall system where the left and right channels do not match, leaving the mix engineers with poor imaging and inconsistent coverage. Various theories were espoused about the cause, such as the slight asymmetry of the room, or the length of the cable runs. Equalization and level offsetting had been implemented unsuccessfully to compensate. Using SIM System II the problem was revealed to be an unbalanced line to one speaker's LF amplifier.

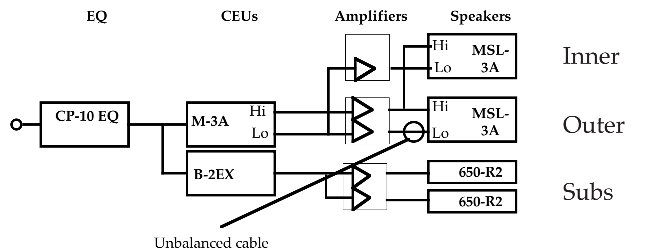
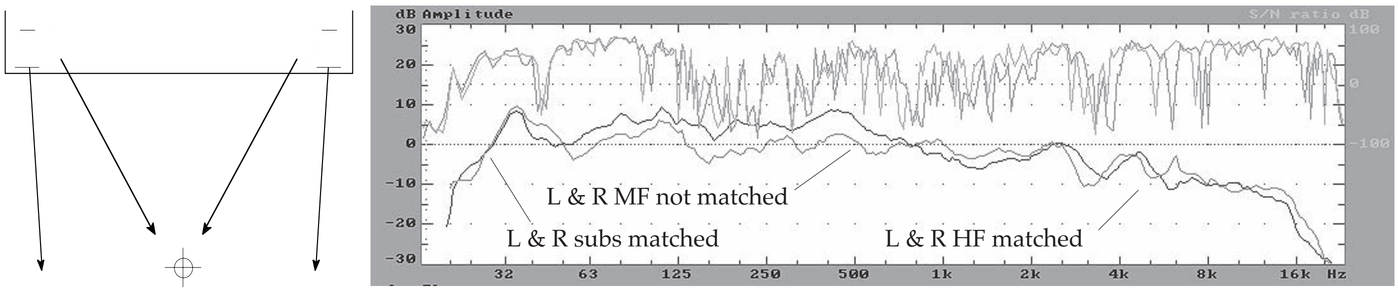
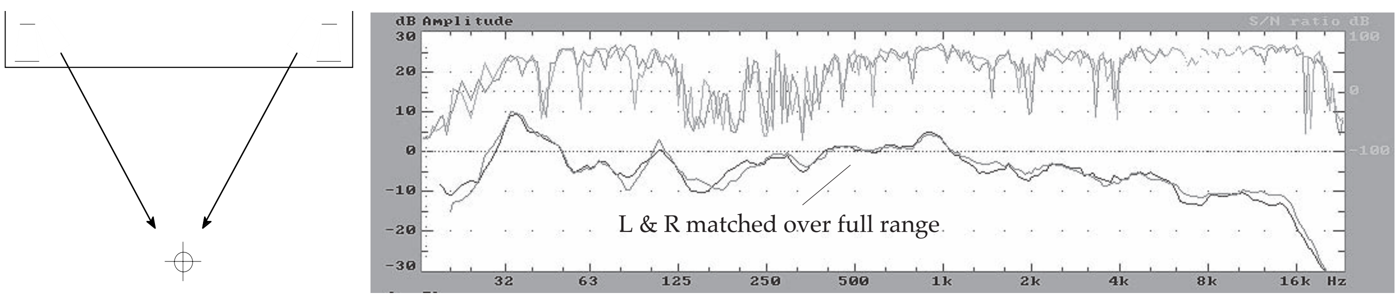


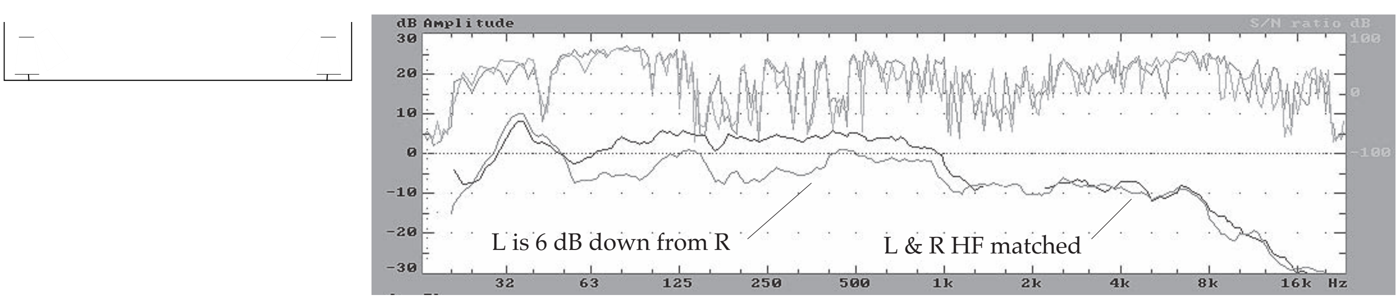
Fig 4.8i Flow block diagram of the system. The cable driving the left outer LF channel was unbalanced, creating the frequency response problems shown in Fig 4.8h below.



A) Inner and outer speakers driven. Data shows a mismatch in the midrange response of the left and right systems.



B) Inner speakers only. The responses are matched indicating no problem here.



C. Outer speakers only. The midrange area shows a 6 dB difference in level. This was caused by an unbalanced cable driving the amplifier.

Fig 4.8h The effect of a single unbalanced cable on a complex sound reinforcement system. Each screen shows a comparison of the left and right sides as measured from the center.

4.9.1 Introduction

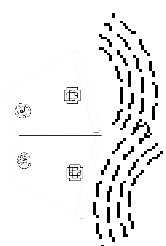
One of the most critical aspects of verification is polarity. The tables in the previous section detailed the many opportunities for polarity reversal in a system. Although reversals can happen anywhere in the signal chain, it is not until it comes out of the speakers that we get a sense that something is wrong. This section shows how to detect polarity reversals. Once detected, the previous sections will help to find *where* the miswiring has occurred.

Like Drivers Polarity Reversal

When two or more transducers cover the same frequency range, they must have the same polarity to add most efficiently and maximize system power.

Crossover Polarity Reversal

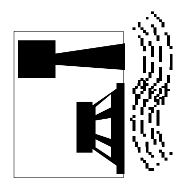
When two or more transducers cover different frequency ranges, they should have matched phase responses at crossover to add efficiently and maximize system power. This does not necessarily mean that they have the same DC polarity, as explained in the following section.



Like Drivers Polarity Reversal

If some drivers are reversed polarity from others in the same frequency range:

- The speakers will not add together properly, compromising the power response.
- Abnormal coverage patterns will be created.
- Frequency response nonlinearity will result.
- The system reliability will be reduced due to excessive excursion.



Crossover Polarity Reversal

If the polarity of two drivers is reversed in the crossover area:

- The speakers will not add together properly, compromising the power response at crossover.
- The vertical coverage pattern will be degraded.
- Frequency response nonlinearity will result.
- The high driver reliability will be reduced due to excess excursion.

4.9.2 LF Driver Polarity Verification

A positive DC voltage on the red terminal creates a forward movement on all Meyer Sound transducers.

The polarity of low-frequency drivers can be verified with one of the most inexpensive pieces of test gear on the market: A 9-volt battery.

In the case of horn loaded drivers, such as in the MSL-3A, MSL-5, MSL-6 and MSL-10A, a flashlight is required to view the driver motion.

Battery Polarity Test

1. Disconnect the speaker cable from the power amplifier.

2. Connect a 9-volt battery to the end of the speaker cable as shown in Fig 4.9a. Use the appropriate pins for each speaker model as shown in Table 4.9b.

3. The driver should move forward.

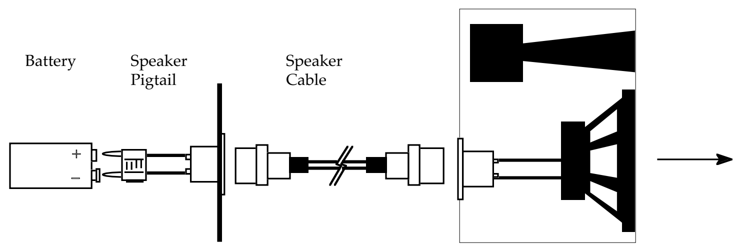


Fig 4.9a Battery polarity check.

! CAUTION: Never perform this test by applying the battery to the input of the power amplifier or other electronic components. Do not perform this test on Meyer Sound self-powered speakers.

Speaker Polarity Battery Check Table	Driver	Terminal (+)	Terminal (-)
MSL-10A	MS-12	Pin 1	Pin 2
	MS-12	Pin 3	Pin 4
MSL-5 (EP)	MS-12 (x2)	Pin 1	Pin 2
MSL-5 (PYLE)	MS-12 (x2)	Pin 2	Pin 1
MSL-3 (EP)	MS-12 (x2)	Pin 1	Pin 2
MSL-3 (PYLE)	MS-12 (x2)	Pin 2	Pin 1
MSL-2A, USM-1	MS-15	Pin 1	Pin 2
UM-1, UPA-1, UPA-2	MS-12	Pin 1	Pin 2
UPM-1, MPS-355 (XLR)	MS-5 (x2)	Pin 2	Pin 1
MPS-305 (XLR)	MS-5	Pin 2	Pin 1
UPM-2 (XLR)	MS-5 (x2)	Pin 2	Pin 1
MPS-355 (Speak-on)	MS-5 (x2)	Pin 3	Pin 1
MPS-305 (Speak-on)	MS-5	Pin 3	Pin 1
UPM-1J, MPS-355J (XLR)	MS-5 (x2)	Pin 2	Pin 3
MPS-305J (XLR)	MS-5	Pin 2	Pin 3
650-R2	MS-18	Pin 4	Pin 1
	MS-18	Pin 3	Pin 2
USW-1	MS-15	Pin 4	Pin 1
	MS-15	Pin 3	Pin 2
MSW-2	MS-18	Pin 4	Pin 1

Table 4.9b Battery polarity test reference.

4.9.3 Multiple Speaker Enclosures

When used alone, the absolute polarity of a speaker has very little, if any, perceptible effect. When combined with others the polarity issue becomes critical.

Fig 4.9c shows the amplitude and phase responses of two speakers with opposite polarities as measured in the manner described at right. The amplitude responses are a perfect match, but the phase responses show a difference of 180° over the full frequency range. This indicates a polarity reversal of both the HF and LF drivers. Singularly these speakers would be fine. If used together, there will be broadband cancellation.

How to verify that two speakers have matched polarity:

Place two speakers adjacent with a measuring microphone placed at the center line between the cabinets. Connect one speaker and input an appropriate test signal such as pink noise or music. Observe the frequency response and level. Add the second speaker. The entire response should rise approximately 6 dB. Polarity reversals between cabinets will cause severe broadband cancellation. The cancellation is most easily detected when the cabinets are adjacent.

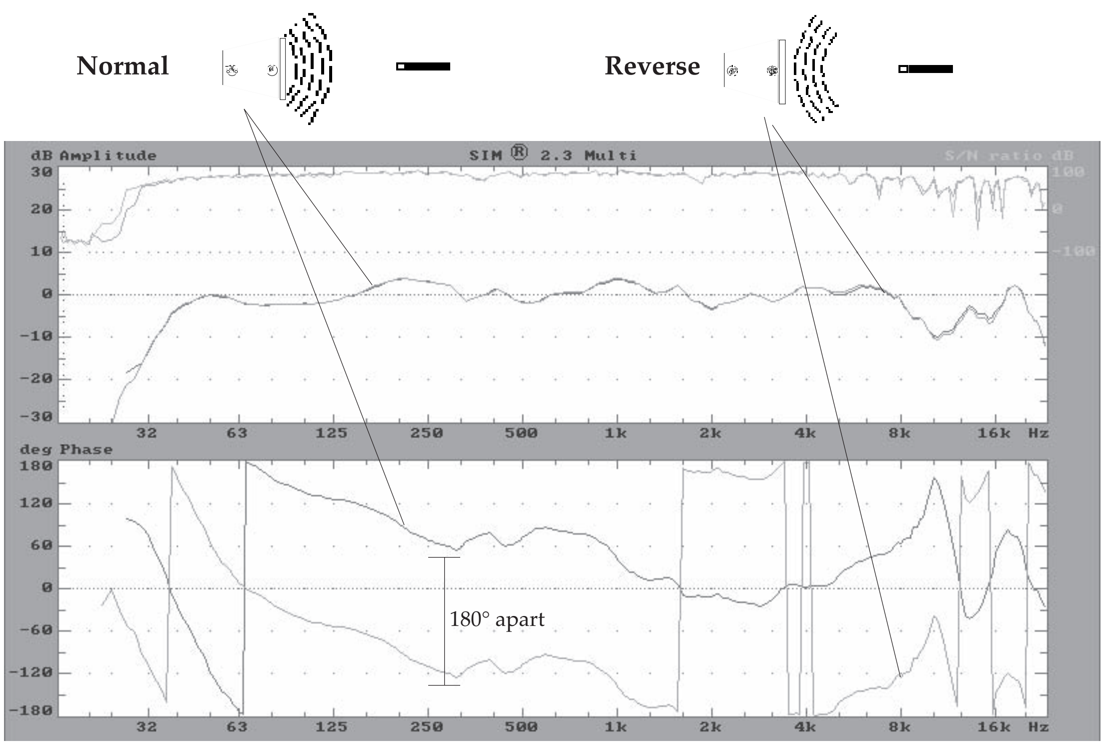


Fig 4.9c Polarity reversal of both the HF and LF drivers compared to that of a normal system. The amplitude response is identical. The phase response shows a difference of 180° at all frequencies. The sound of these two speakers will be the same. The only difference is how they combine with other speaker systems.

4.9.3 Multiple Speaker Enclosures

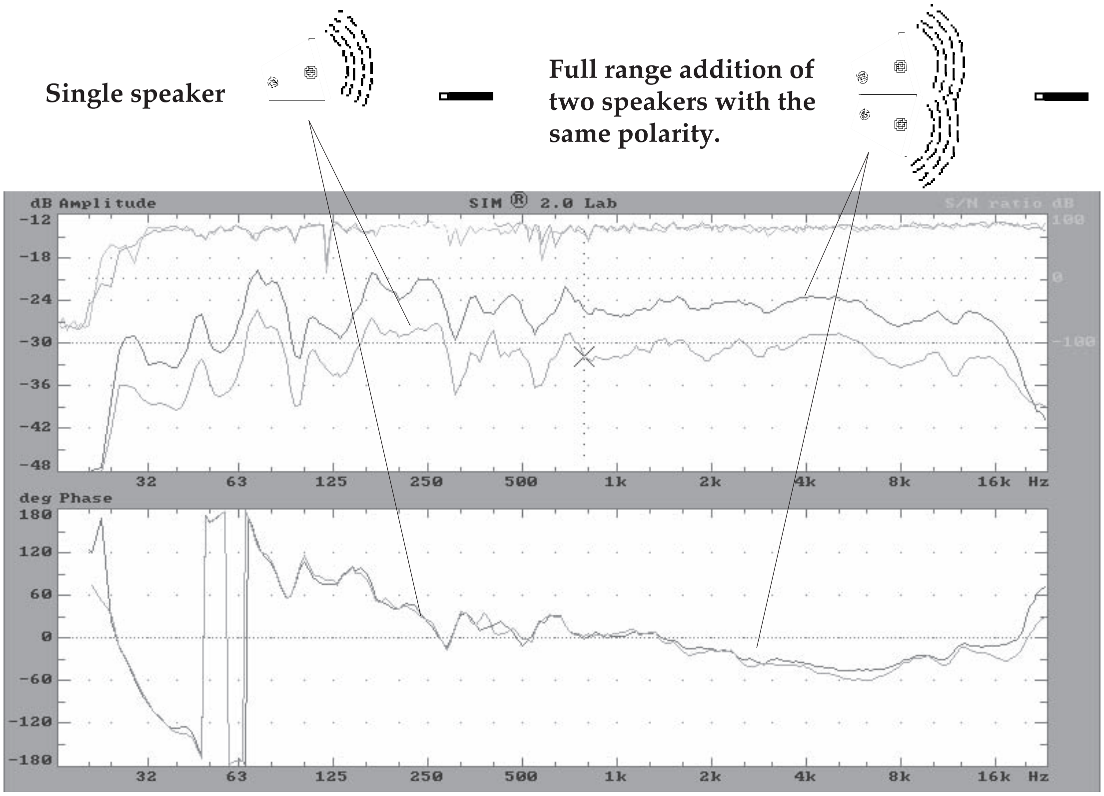


Fig 4.9d

Comparison of a single speaker versus two speakers that are matched in amplitude and phase. There is 6 dB of addition when the second speaker is added.

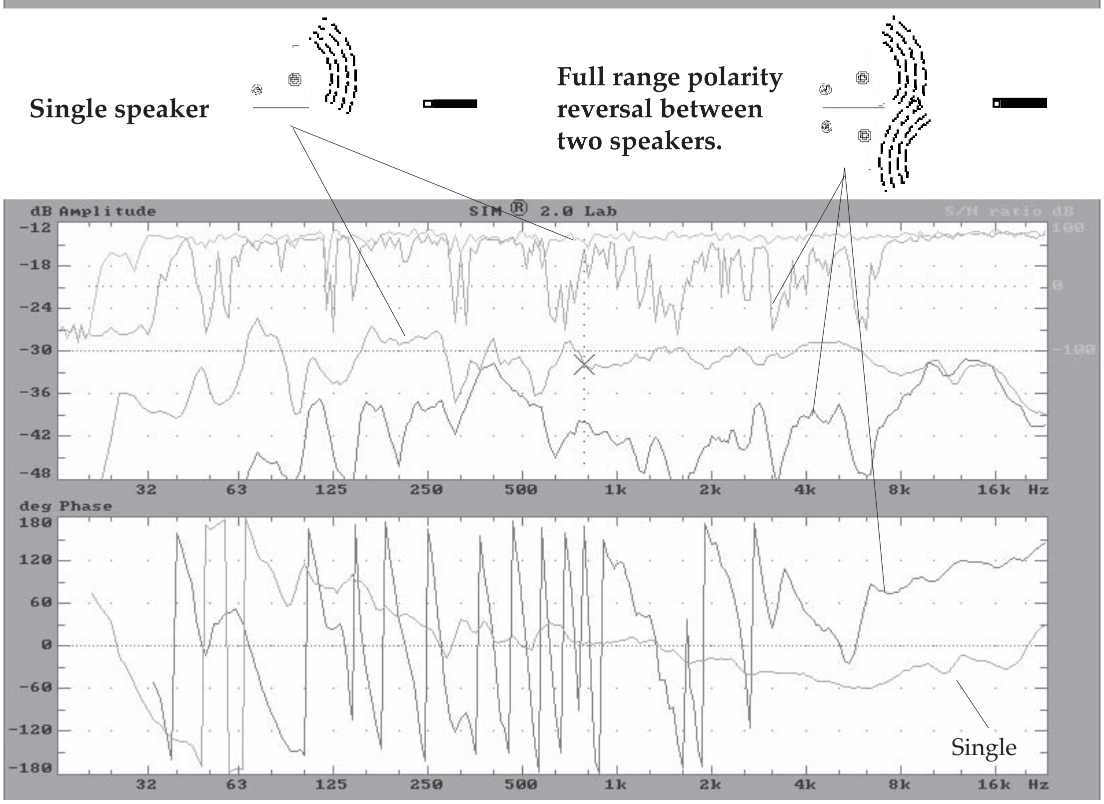


Fig 4.9e

Comparison of a single speaker versus two speakers that are matched in amplitude but reversed in polarity. The amplitude and phase responses become very irregular. The S/N ratio is greatly decreased.

4.9.4 Polarity of Multi-way Systems

An example of multi-way system polarity verification is shown in Figs 4.9f and 4.9g. In both figures the amplitude and phase responses are compared with normal and polarity reversed configurations. Notice the loss in amplitude at the acoustical crossover of 900 Hz. The amplitude trace alone cannot tell you *which* driver is reversed. In Fig 4.9f the phase responses are 180° apart above crossover, indicating a HF polarity reversal.

In Fig 4.9g the phase responses are 180° apart below crossover, indicating a LF polarity reversal.

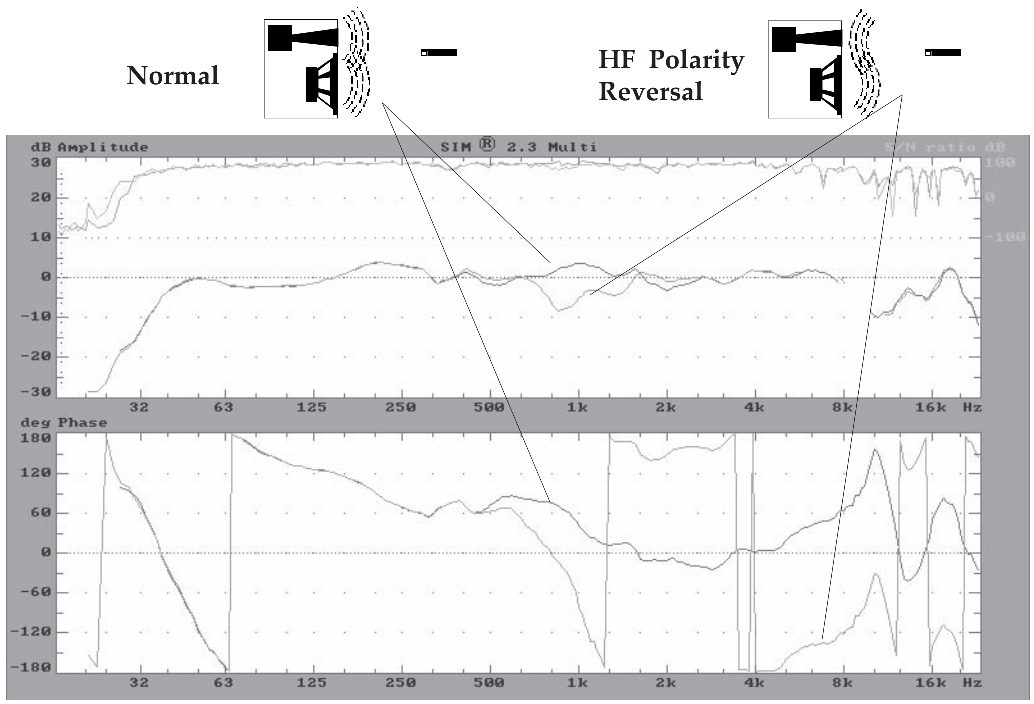


Fig 4.9f Polarity reversal of HF driver compared to a normal system.

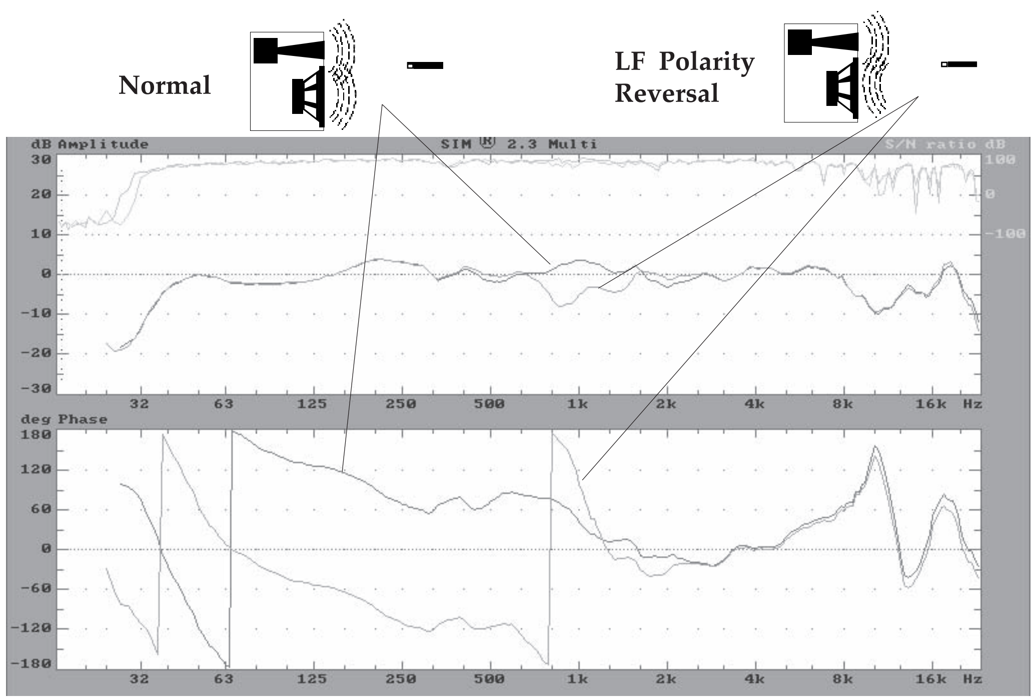


Fig 4.9g Polarity reversal of LF driver compared to a normal system.

4.9.5 Polarity or Phase?

The term "phase reversal" is commonly applied in reference to a polarity reversal. *Polarity* is a directional parameter (voltage or pressure) independent of frequency. *Phase* is a frequency dependent time delay parameter.

When identical speakers are opposite in polarity, the signals arrive in time, but reversed in pressure. Contrast this with the comparison below (Fig 4.9h) of the phase responses of two different speakers, the UPA-1C and MSL-2A. These models have the same polarity and are "in phase" through most of their mutual coverage range. However, the phase responses diverge in the low end due to their different cutoff frequencies. At 60 Hz the two speakers are 180° apart.

This difference in phase response affects:

- The LF coupling between the MSL-2A and UPA-1.
- How the two models combine with subwoofers.

The B-2 series of CEUs was designed to optimize the crossover between these subwoofers and the UPA-1 and MSL-3 full-range speakers. The MSL-2As extended low-frequency response created a different phase scenario. Therefore, it is most common to reverse the polarity of the subwoofers (or the MSL-2A) to make them "in phase." The example on the following page (Figs 4.9i-4.9k) illustrates the combination of an MSL-2A and an MSW-2 subwoofer. The scenario where the cancellation occurs is when both speakers are "normal." The best addition occurs when the MSW-2s are reversed from the MSL-2A.

The phase relationship between the systems will change if the subwoofers are separated from the mains. A polarity reversal or delay line may be required to optimize the crossover.

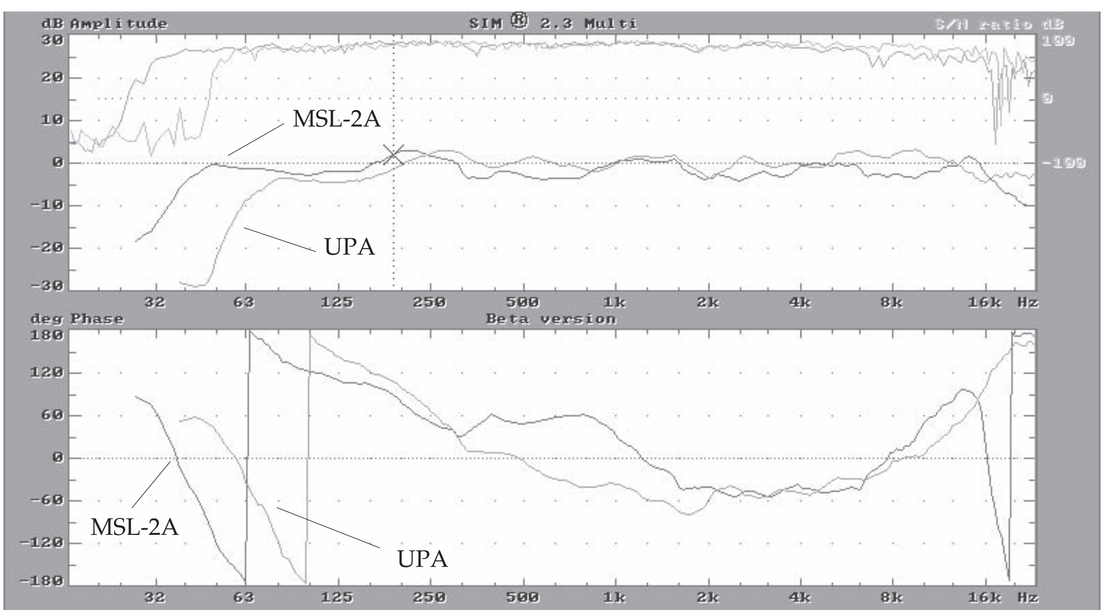


Fig 4.9h Comparison of the UPA and MSL-2A phase responses.

4.9.6 Subwoofer Polarity Optimization

Subwoofer polarity optimization is illustrated in Figs 4.9i-4.9k. Each speaker was measured individually before being combined. In Fig 4.9i the phase responses are well matched through crossover and will combine efficiently. Compare this to Fig 4.9j where the phase responses are 180° apart. The combined response, and resulting power loss, is shown in Fig 4.9k.

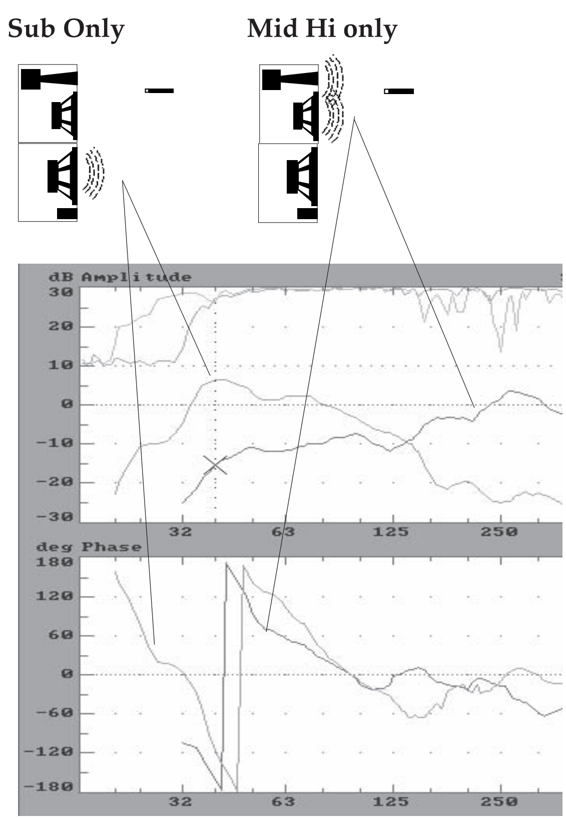


Fig 4.9i Subwoofer and mid-ranges with matched phase responses.

These speakers will add together with maximum efficiency at crossover.

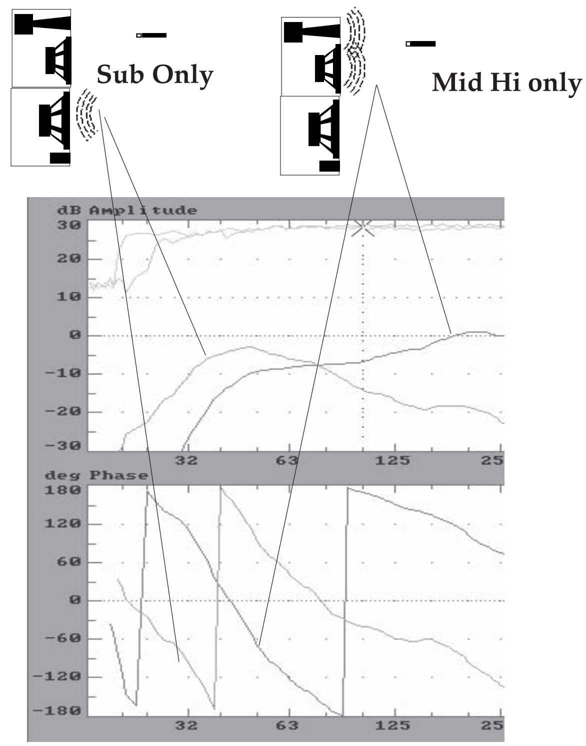


Fig 4.9j Subwoofers and mid-ranges with divergent phase responses.

These speakers will combine poorly with cancellation at crossover and minimal power addition.

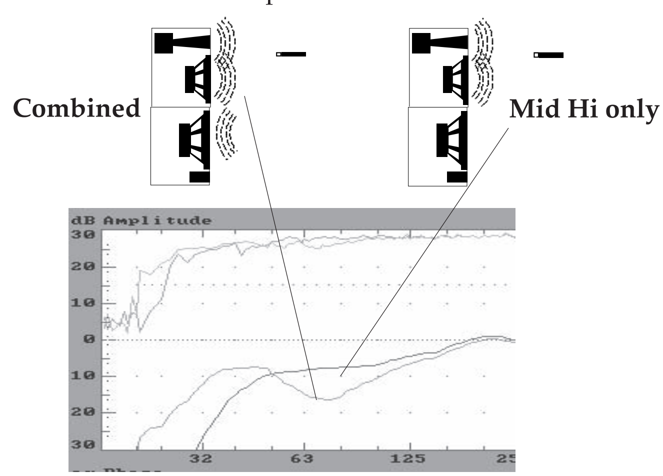


Fig 4.9k The combined response of the MSL-2A and MSW-2 subwoofer compared to the MSL-2A alone.

The combined response in the 50 to 100 Hz range is reduced from that of the MSL-2A alone. This can be remedied by reversing the polarity of the subs.

4.9.6 Subwoofer Polarity Optimization

Subwoofers represent a special case in regard to polarity optimization. Section 4.9.5 described the need for phase addition through crossover in integral multi-way systems such as UPAs and MSL-2As, etc. The same principals hold true for subwoofers. In contrast to the integral systems, subwoofers have several open variables.

Subwoofer system polarity must be evaluated on a case-by-case basis because the phase response at crossover is subject to:

- The physical relationship between the subwoofer and midrange enclosures at crossover.
- The boundary loading of the subwoofers in the space.
- The number of subwoofers and array configuration.
- The actual acoustical crossover (Section 4.10).

Sub Polarity Optimization Without Tools

Here is a simple way to verify subwoofer addition. Put on a CD or pink noise. Insert a filter in the crossover range (90Hz to 120 Hz) and give it a large boost. Listen. Reverse the polarity of the subwoofers and listen again. Use the setting that has the most energy at crossover.

Subwoofer Phase Optimization with SIM®

The subwoofer crossover is optimized when the phase responses are matched. Simple polarity-based techniques have two choices: 0° or 180°. But what if the speakers are 90° apart or 270° or 540°? A delay line can be used to bring the phase responses together. An analyzer with a phase versus frequency display is required.

Subwoofer phase alignment when the mid-ranges lead the subwoofers in time:

- Measure the subwoofer alone. Set the internal delay so that a flat phase response is created in the crossover region. Store and recall the response onto the screen.
- Measure the full range speaker alone *without* adjusting the SIM internal delay.
- Adjust the delay line for the full range speaker so that the phase responses overlay on the screen. Reverse polarity if necessary.

When the subwoofers lead in time, the role of the delay line is reversed.

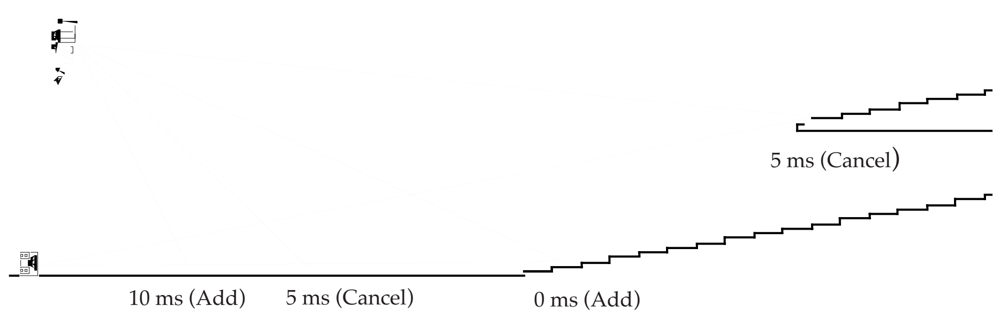


Fig 4.9l Changing time offest between mains and subwoofers.

An example of a concert application with a flying main system and subwoofers on the deck is shown in Fig 4.9l. On the floor the subwoofers arrive first. In the balcony the mains are first. Near the middle they are equal. The crossover is 100 Hz. Therefore, a time offset of 5 ms (1/2 wavelength) will cause cancelation. An offset of 0 ms or 10 ms (a full wavelength) will cause maximum addition. It is impossible for all seats to have addition when the systems are separated like this. Choose carefully where the maximum addition occurs. Hint: the mix position.

4.10.1 Acoustical Crossover

The amplifier gain should be the same for both the HF and LF drivers to obtain proper addition through crossover. (A complete explanation of this can be found in section 1.4.6, Matching Amplifier Voltage Gain.) This can be confirmed by testing the amplifier directly (Section 4.6) but inevitably it must be verified acoustically, since the wiring could also be faulty. Figs 4.10a and 4.10b show the response of a UPA-1C system with correct and improper acoustical crossover settings. The LF channel is reduced 6 dB from the HF, as could result from turning down the LF amplifier, or an unbalanced line to the LF amplifier. Notice in Fig 4.10a that the crossover region occupies a wide area centered around 1200 Hz where both drivers are close in level and matched in phase, creating maximum efficiency through crossover. In Fig 4.10b the crossover is lowered to 800 Hz, where the phase traces have diverged greatly, providing minimal addition (and perhaps cancelation). The HF driver will have to provide all of the power above 800 Hz (more than an octave lower than before). Danger!

! **The design of all Meyer two-way speaker systems is predicated on a fixed physical and electronic relationship between the LF and HF drivers.**

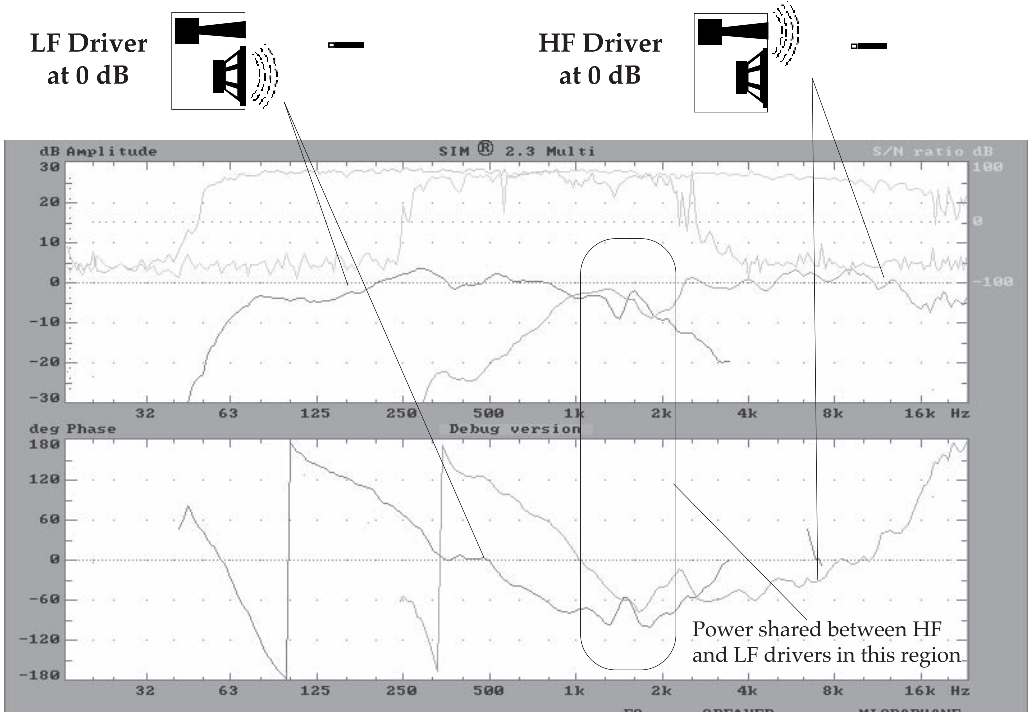


Fig 4.10a Normal crossover of the UPA-1C. Both amplifiers have the same gain.

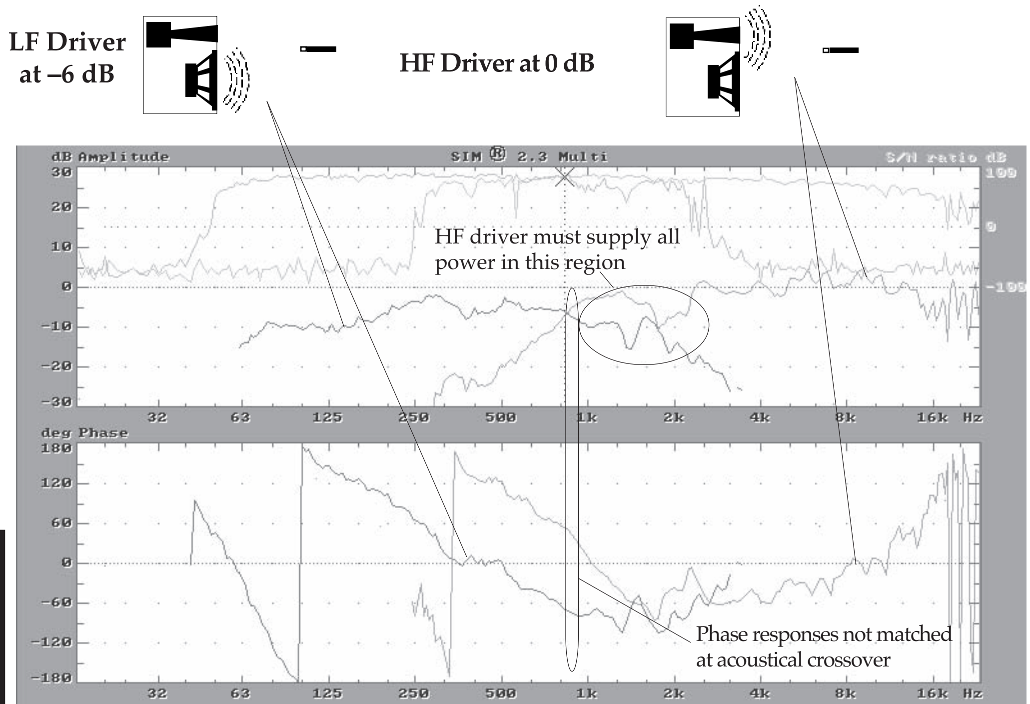


Fig 4.10b Misaligned crossover of the UPA-1C. The LF amplifier was set to –6 dB.

4.10.2 Crossover Alignment Considerations

The integral systems have a fixed ratio of LF and HF drivers. Adding another enclosure does not alter the acoustic crossover. The quantity of subwoofers is primarily a function of power requirements. The quantity of full range enclosures is a combination of coverage and power requirements. As these quantities change, acoustical crossover between them shifts.

The relative levels of the B-2 series, D-2 and full-range CEUs will depend upon:

- The *relative quantities* of subwoofers, DS-2s and full-range cabinets.
- Their *relative power amplifier voltage gain*. These *should be identical*, but because the amplifiers used for subwoofers may be different models than the others, it cannot be assumed. *Check it!*
- *Speaker coupling and room acoustics.* Subwoofer efficiency is particularly dependent on boundary loading. Half-space loading will cause a subwoofer to be more efficient than it would be in free field. Reverberant rooms are less lossy than absorptive rooms or outdoors.

For example: The recommended crossover for a 650-R2 with a full-range system is 100 Hz. A single MSL-3 and 650-R2 with matched CEU levels and amplifier voltage gains creates an *acoustical crossover* at 100 Hz. This will shift, however, if any of these variables are changed.

The primary purpose of changing the ratio of enclosures should be to tailor the power and coverage requirements, not to change the frequency response. If the ratio of subwoofers is *increased*, then the subwoofer CEU level should be *decreased* accordingly to maintain a *constant* crossover and frequency response, while *increasing* the maximum power capability in the low frequencies.

The acoustical crossover frequency rises if the:

- Ratio of subwoofer speakers to full-range speakers is increased.
- Subwoofer CEU level is increased.
- Subwoofer power amplifier voltage gain is increased.
- Subwoofers are 1/2 space or 1/4 space loaded or in a reverberant environment.

The crossover frequency falls if the:

- Ratio of subwoofer speakers to full-range speakers is decreased.
- Subwoofer CEU level is decreased.
- Subwoofer power amplifier voltage gain is decreased.
- Subwoofers are flown.
- Full-range CEU Lo cut switch is out.

The crossover frequency will remain optimized if the above factors can be compensated by a relative level adjustment of the CEU level controls.

4.10.3 Crossover Alignment Procedures

To verify the acoustical crossover of integrated full-range speakers is primarily a matter of verifying your wiring and amplifier voltage gain. The final check is an acoustic measurement.

Speaker	Acoustic Crossover
MSL-10A	900 Hz
MSL-5	900 Hz
MSL-3A	700 Hz
MSL-2A	900 Hz
USM-1	900 Hz
UM-1C	1200 Hz
UPA-1C	1200 Hz
UPA-2C	1200 Hz

To verify an integrated speaker's acoustical crossover:

- Measure the HF driver alone and store the response.
- Measure the LF driver alone and compare its level at crossover.

Crossing in Subwoofers

The factors affecting subwoofer acoustical crossovers are complex enough to prevent me from offering particular settings for each scenario. Fortunately it is quite simple to verify.

To verify subwoofer acoustical crossover:

- Measure the full-range system alone and store the response.
- Measure the subwoofers alone.
- Adjust the level of the subwoofers until they match the level of the full-range system in the crossover range (90 to 120 Hz).
- Store the subwoofer response.
- Add the systems together and check for addition in the crossover region. If there is no addition, try reversing the subwoofer polarity.

Crossing in the DS-2

It is particularly critical to verify the acoustical crossovers of the DS-2 mid-bass cabinet. The DS-2's frequency range is sandwiched between speakers capable of operating in the same range. The DS-2, however, is much more efficient in this range. The required CEU levels of the subwoofer, DS-2, and mid-hi systems depend on the quantities of the speakers. The CEU relative levels must be set to achieve acoustical crossovers of approximately 60 and 160 Hz. If the DS-2's CEU level is set too high, the crossover will move up above 160 Hz, a region where the cabinet performs poorly. When customers have come away with poor impressions of the DS-2 this has almost always been the case. If the level is set too low the DS-2 will be ineffective, as its response will be overrun by the other speakers.

The DS-2 is a tremendously powerful addition to your system *if you set its acoustic crossovers correctly.*

To verify DS-2 acoustical crossovers:

- Measure the full-range system alone and store the response. The Lo cut switch must be in.
- Measure the DS-2s alone.
- Adjust the DS-2 level until it matches the level of the full-range system in the crossover range (160 Hz to 180 Hz).
- Store the DS-2 response.
- Add the systems together and check for addition in the crossover region. If there is no addition, try reversing the DS-2 polarity. Store the combined response.
- Measure the subwoofers alone.
- Adjust the level of the subwoofers until they match the level of the DS-2s in the crossover range (60 Hz to 80 Hz).
- Store the subwoofer response.
- Add the systems together and check for addition in the crossover region. If there is no addition, try reversing the subwoofer polarity.

5.1.1 Alignment Goals

At this point we have a fully verified sound system. All of the previous steps have put us in a position to align the system to its highest potential. Let's take a moment to review the goals of the alignment process.

The goals of the alignment process:

- Provide the most accurate reproduction of the input signal's amplitude and phase response.
- Maximize intelligibility.
- Provide a consistent sound pressure level and frequency response over the listening area.
- Create realistic sonic imaging.
- Minimize the effects of poor acoustics.

We are now ready to face the remaining obstacles to system optimization discussed in Section 2 and to overcome them as much as possible. Let's take a moment to review them:

The obstacles to system optimization:

- Speaker interaction.
- Reflections.
- Dynamic conditions.

These are all acoustical interactions with the speakers. Potential electronic obstacles such as our own miswiring have already been dealt with in the verification stage.

There are several techniques available for us to overcome the problems posed above.

Techniques for system optimization:

- Architectural modification.
- Speaker repositioning.
- Electronic time delay.
- Gain structure adjustment.
- Complementary equalization.

Each of these options have their own fiscal impacts and relative effectiveness and practicality depending upon your application.

The alignment process will require dedicated high quality acoustic measurement instrumentation able to analyze all of the stated obstacles and verify the effect of the corrective actions.

The tools for system optimization:

- Your ears (don't leave home without them).
- SIM SystemII (surprise!).
- High-quality measurement microphones such the Bruel & Kjaer 4007.

5.1.2 Dividing Lines

The "Art/Science" Line

The system will be subdivided in order to allow the measurements required for alignment to proceed without constant repatching.

From the viewpoint of alignment, the sound system is composed of three sections as shown in Fig 5.2a.

Source: This includes all pre-equalizer system components. This is the original input into the sound system. Our primary goal is to reproduce this signal accurately. The alignment process makes no effort to control the spectral content of the source, which is totally under the control of the artist and mix engineer. This is considered the artistic side of the system. Usually included in this category is a "house" equalizer under the control of the mix engineer for overall response shaping. The only items to be aligned in the source category are the system delay lines and relative matrix output levels. Specific channels, such as body mics, may also be equalized when desired.

Equalizer: We have now crossed the "Art/Science" line. The equalizer will be used to align the response of the system to compensate as much as possible for the effects of speaker interactions, reflections and dynamic acoustical conditions. The system will have separate equalizers for each subsystem. These will be used to maximize the consistency of the response throughout the room.

Room+Speaker: This includes all post-equalizer system components, the room acoustics and dynamic conditions, and the measurement microphone. The response of the speaker will be modified by these interactions causing variations in level, intelligibility and frequency response. We will attempt to minimize these effects by damping reflections and steering the speakers away from the reflecting surfaces. Delay lines will compensate for time offsets. Level setting will minimize the speaker overlap and smooth out the SPL distribution. The remainder will be compensated by equalization.

The concept of the "art/science" line is critical to understanding the alignment process. The *overall* sound is a matter of artistic expression. While science and technology may be used to aid artistic expression, they should not dictate or control it. On the other hand, the difference in sound between the mix position and the audience in the balcony is not "an artistic decision." It is a challenge posed by acoustical physics and can only be solved within the realities of the laws of physics.

If the mix engineer does not understand this they may feel that the SIM® System Operator has intentions of taking away their artistic control. The opposite is true. Artistic control is enhanced for the mixer by a simple subdivision of responsibility: The mix engineer will make it sound good where his or her ears are; The SIM® system operator will make it sound the same where the audience is.

The modern mix engineer has little time to walk around the hall and analyze the response throughout the hall. Their most pressing concerns are the 100 channels of audio, thirty-two outboard devices, and the forty-eight sound cues during the show. The increasing sophistication and expectation of modern audiences, along with the skyrocketing costs of concert tickets, means that making it sound good at the mix position is simply not good enough. It must sound the same in the audience area, and the only way to ensure that is to measure all over the hall and compensate for the differences. *The only system in the world designed to do this is SIM® System II.*

The Art/Science line is illustrated by this real world example.

In the middle of a concert a patron from the balcony approaches me (the SIM Engineer) in the mix area.

"I don't like the way it sounds," he says.

"How does it sound here?" I ask.

If he answers that it sounds bad here too, we have a difference in artistic opinion. I send the listener to the mixer, and let them work it out. *ART.*

If he answers that its sounds great at the mix position, I will need to solve the problem of why it sounds different in the balcony. *SCIENCE.*

5.2 Interfacing the Measurement System

Fig. 5.2a illustrates the dividing lines between the source, EQ, and room+speaker systems. Measurement zones are divided into *branches* which are defined simply as an equalizer, speaker(s), and a measurement mic.

Each branch will have three measurement access points:

- EQ input.
- EQ output.
- Microphone.

The data from these three points gives us all we need to determine the effectiveness of architectural modification, the need for and effects of speaker repositioning, and to set level, delays and equalization.

The spectrum at each of these points in the signal chain is analyzed. The difference between spectrum pairs is found by analyzing their transfer function, which gives the amplitude, phase and S/N ratio of these systems (as discussed in Section 1.9). Three transfer functions are derived from these inputs by comparing the following pairs of points.

Room+Speaker: EQ Output versus Mic

This is the unequalized response of the system. The room plus speaker response will be affected by changes in speaker or microphone position, dynamic acoustic conditions, and the interaction of other speakers. Changes in amplifier or CEU level would also be detected here. The response of the equalizer does not affect this transfer function since it does not create a *difference between* the equalizer output and the mic.

The effect of architectural modifications and speaker position can be assessed with this measurement.

Equalizer: EQ Input versus EQ Output

The equalizer system is a purely electrical system that must be capable of creating a response that is complementary in *amplitude* and *phase* to the response of the room plus speaker system. This process is termed *complementary equalization* (Section 5.8). The EQ transfer function shows the actual response of the equalizer (In contrast to the front panel display of an EQ). The EQ response can be viewed together with the room plus speaker response to ensure that precise complement is created.

Result: EQ Input versus Mic.

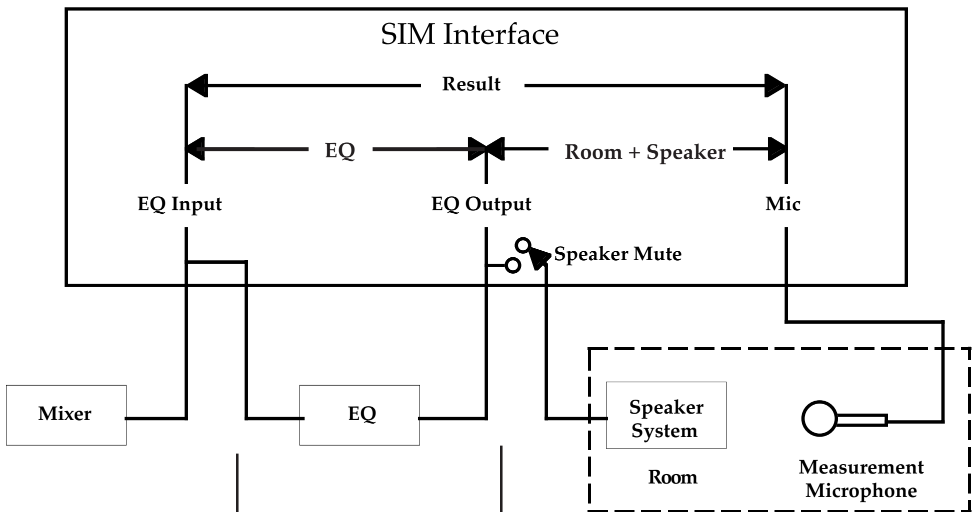
This response shows the effect of the correction. If the room plus speaker response and the EQ response are complementary the result response will be flat.

The time difference between the electrical signal at the equalizer input and the microphone (due to the propagation delay) can be analyzed. This measurement is used to set delay lines.

The level relationship between the EQ input and mic will show the relative level at a given location. This will be used to set relative levels for subsystems.

Fig 5.2a also shows how SIM System II is patched into the sound system. SIM System II contains all the cabling required to interface into the system, provided that the equalizer inputs and outputs are XLR type connectors. Notice that the signals at the EQ inputs and outputs are routed through the SIM system and returned, rather than simply paralleled. This enables the SIM System to control speaker muting. The measurement procedures call for each subsystem to be individually measured and then later combined. The muting capability enables the SIM Operator to move quickly through these procedures for each of the speaker subsystems without having to go down to the amplifier racks to mute a speaker.

5.2 Interfacing the Measurement System



Source	Equalizer	Room + Speaker
Stage Mics	Equalizer	Speaker
Direct Inputs		CEUs
Mixing Console		Amplifiers
Channel EQ	The "Art/Science" Line	Speaker/Speaker Interaction
Outboard Gear		Room/Speaker Interaction
Delay Lines		Dynamic Acoustical Conditions
House EQ		Delay Lines
		Measurement Mic

Fig 5.2a SIM interface patched into a sound system.

EQ input and output signals are routed into the SIM interface and returned.

5.3.1 Primary Microphone Positions

Measurement microphone placement plays a critical part in the alignment process. While it is important to maintain good techniques for mic placement, do not give too much thought to finding the “perfect” position. (Unless you are in a recording studio, in which case you are reading the wrong book.) Our goal is to provide ourselves with *global* information so that the system can be aligned for maximum consistency. Each microphone provides a *local* view of the system's response with the global characteristics more or less hidden. *The key to mic placement is to use positions that provide a global representation with a minimum of unique local conditions.*

Guidelines for Measurement Mic Placement:

- Avoid positions that have obvious unique local conditions, e.g., a pillar a few feet away.
- Avoid placing mics in aisles. They tend to have strong reflections from the open floor in front of them and are not representative of the area at large.
- Always point the mic toward the speaker source. Even “omnidirectional” microphones become directional above 5 kHz at angles of 90° or greater.
- Beware of exact center points in rooms since they will have unique reflection patterns.
- If at all possible avoid the offer to run your mic through the house patch bay to save running the cable. It almost never works.
- Do not place the mic at sitting head height. This position will have a strong local reflection off of the next row. This is not representative of the response with audience members seated. Standing head height usually works better. When working in extreme proximity, however, such as frontfills, the sitting height may work better because the standing height will be out of the vertical pattern.

Where are the Best Measurement Positions?

In the ideal world, everywhere your audience is. In the practical world, however, we must prioritize the importance of positions. First priority are positions that are most representative of each system's coverage area. In SIM jargon this position is termed a "primary position."

Primary Mic Position

- On-axis (± 10%) in the horizontal plane. If it is an array of speakers the mic should be placed on-axis to a central speaker in the array. (See Section 2.)
- At the midpoint, in terms of depth of coverage. (This may be below the vertical axis, depending upon the rake and depth of the seating area.)

To determine a primary position we must have a clear idea of what areas our speakers are covering, as discussed in the Section 3.

This position represents the "average" seat in the speaker system's coverage area. This position is the reference point in terms of speaker positioning and will take first priority in terms of delay, level and EQ setting.

Speaker Positioning

Horizontal: The coverage pattern can be verified by comparing side coverage areas to this position. When you have reached the –6 dB point you are at the edge of the pattern. If the pattern edges do not occur as intended, it is time to reposition the speaker.

Vertical: This position should be average in level for the seating area. As you move closer, expect to see a rise in level. As you move farther expect to see a loss. This is a complex function of the axial attenuation and propagation loss, which hopefully will combine to create a minimal difference in level over depth. If this does not occur, consider repositioning.

Level Setting

Once the speaker position is determined, the primary mic position serves as the relative level marker for this subsystem with respect to others. If all subsystems are set to create this same relative level, then maximum consistency will be achieved throughout the hall. This will be complicated, however, by the interaction of the subsystems, which tend to cause some additions.

5.3.1 Primary Microphone Positions

Delay Setting

The primary position offers a good representation for delay alignment. For most situations the best result will be achieved when the primary mic position is used as the synchronous point of alignment.

EQ Setting

The primary position provides a first look. However, it is highly recommended that additional positions be looked at and factored in to the final decision on EQ setting.

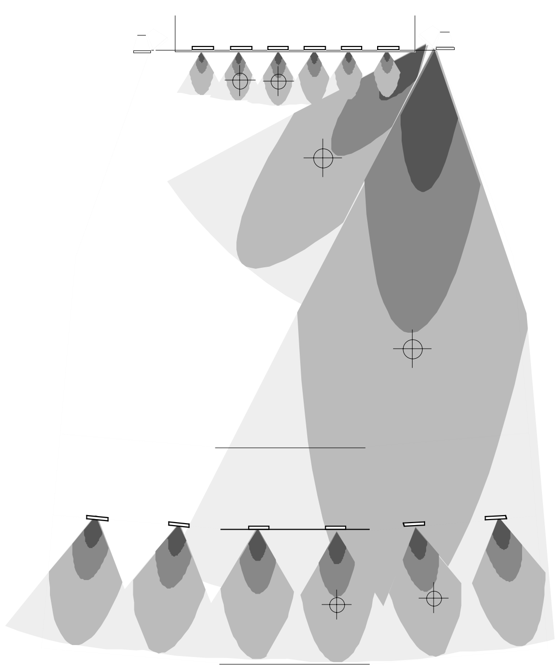
The Mix Position

You might notice that no mention has been made of placing a mic at the mix position. This is because the mix position is, after all, just another seat. It is best if the mix position is a primary location. However, if it is not, it will not help to pretend otherwise. In fact, aligning a system to a poor position will create a worse effect for audience and mixer alike. There are many reasons why a mix position may not be suitable for alignment (or mixing for that matter), such as being off-axis, at the back wall, under the balcony, or all of the above. While it is true that the mixer's reference point is critical, it is futile and destructive to align the system for a bad mix position.

Top Ten Excuses

For Not Measuring the Audience Area

1. Those are the cheap seats anyway.
2. Neither the mixer nor band manager ever leave the booth so it doesn't matter.
3. If we align it perfectly at the mix position then it's as good as it can get.
4. It takes too long. Leaves us less time in catering.
5. It's impossible to make it the same everywhere, so why get yourself depressed?
6. They should have come early enough to get a seat near the mix position.
7. The audience will steal my mics.
8. Lighting doesn't have to check how it looks in every seat so we don't either.
9. By the time the newspaper review comes out we will be in the next city.
10. The security guards keep people away from the booth so we won't get any complaints.



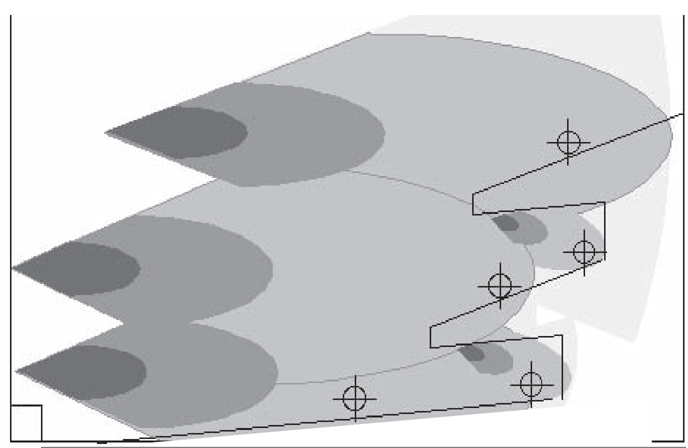


Fig 5.3a Primary mic positions.

Primary mic is on the horizontal axis, at one-half the depth of field of coverage as shown in plan and elevation views.

5.3.2 Secondary Microphone Positions

No matter how well placed the primary mic is, it is still only a single point in the room. I cannot say enough about the benefits of analyzing additional mic positions—or about the potential dangers of basing your entire alignment on a single position. Every position has unique local response characteristics in addition to more global ones. Extreme peaks and dips can be found at one position and disappear a few seats later. Moving the mic or using multiple mics is a form of insurance against making decisions that will not create global solutions. I can attest to many instances where a problem appeared to be solved at one position only to be revealed later that the "solution" had merely repositioned the problem a few seats away.

Secondary mic positions provide secondary opinions on the data. The global aspects of the system become readily recognizable when multiple placements are compared. These are the major tendencies of the system, the ones that will be the keys to getting the system under control.

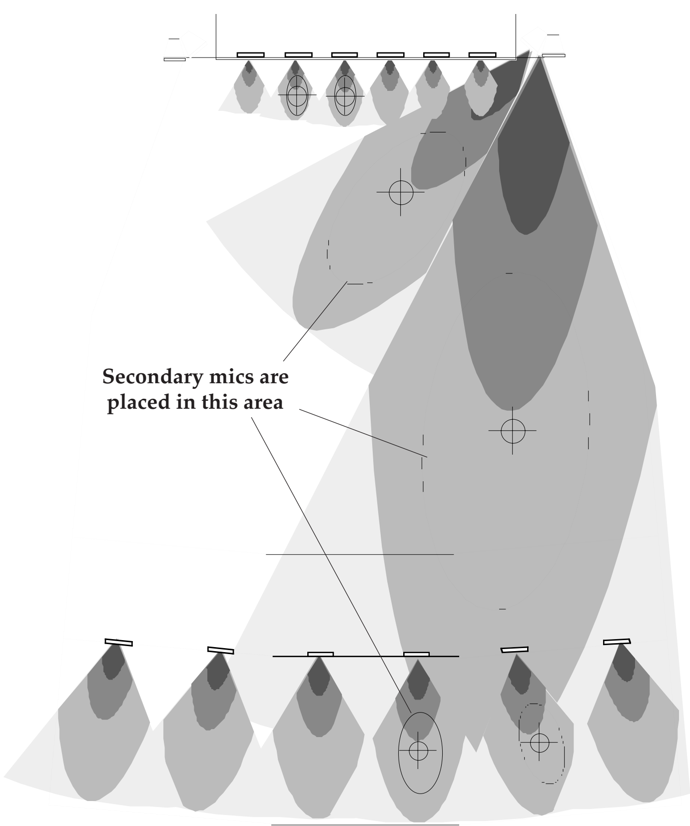


Fig 5.3c Plan view of secondary mic positions.

Secondary positions are found within the coverage area of the speaker but away from primary position. SIM engineers such as Roger Gans employ a star pattern of positions around the primary mic. This gives a map of the horizontal and vertical variations.

Secondary Mics:

- Can be placed over a wide range within the speaker's intended coverage area.
- Provide additional local information to help ascertain the global parameters of the speaker system response.

The unfortunate side effect of taking secondary measurements is the presence of blatantly contradictory data. The simple technique of overlaying the 1/EQ curve over the room+speaker response becomes complicated in the face of these discrepancies. Typically, the equalizer settings are appropriate in some frequency ranges and contradictory in others, indicating that the corrections for one measurement position hinders another. A few high resolution looks at what goes on with position changes will very quickly sober one's fantasies about automating the equalization process.

In most cases the new positions will show normal variations in frequency response due to the interaction of the speakers and the room, as shown in Figs 5.3d and 5.3e. They may also turn up unexpected results indicating coverage gaps or excess overlap.

The secondary data can be compared to the primary. Major trends will emerge where the responses match. Decisions must be made about those areas where they differ.

At first glance it might appear that taking the two different samples and creating an average would make a suitable "average" response. The inverse filter could be generated (manually or automatically) and applied and you are finished. But unfortunately, it doesn't work that way.

5.3.2 Secondary Microphone Positions

Why Automatic Equalization is not Around the Corner:

Automatic equalization is quite appealing as a concept but very risky in practice. An automated system must have access to more information than the simple amplitude response, such as the speaker's position, distance, the type of hall and the complexity of speaker interaction. If not, the automated system will make the same mistakes that a person would make if they had to work blindly without knowing whether the speaker was even pointing in the right direction or wired correctly.

A 20 dB peak and a 20 dB dip of a one-tenth octave bandwidth average out to 0 dB. But we hear the 20 dB peak as a massive coloration, while the 20 dB dip is only marginally perceived. If you leave the 20 dB peak in the system you will soon be looking for employment.

All data should not automatically be considered as equally valid. The S/N ratio response will indicate which position has more reliable data and, therefore, is more likely to benefit from equalization. Areas with a low S/N ratio should not be considered as relevant as those areas with a high S/N ratio. The S/N ratio trace may in fact be warning you that the 40 Hz peak you are reading is coming from the air handling system. Try equalizing that!

If the positions yield extreme differences, the system should be subdivided and equalized, or delayed separately. It could also be indicative of a speaker positioning problem or something that should have been caught in the verification stage, such as a blown driver or polarity reversal. An automated equalizer is unlikely to notice that the lighting crew has now placed a 1K lamp in front of your high horn.

In some cases a similar peak in the response may appear in both positions but differ in amplitude or bandwidth. In such cases, an average between them is applicable.

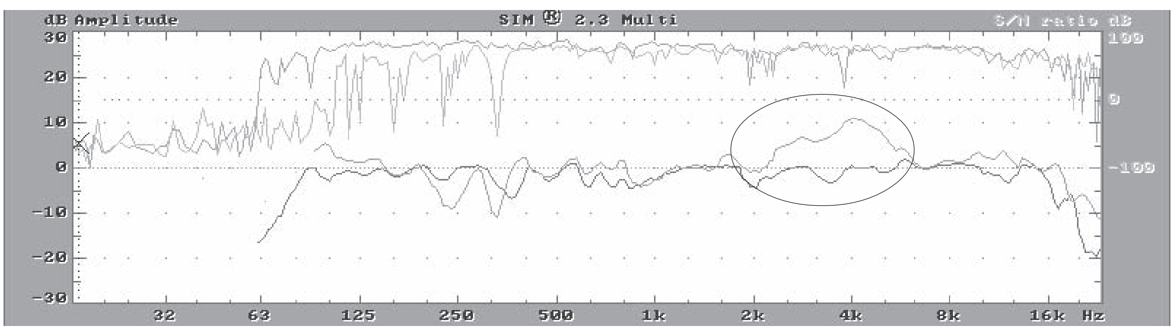


Fig 5.3d Primary and secondary positions.
The main trends are clearly seen as similar, but a large difference is seen in the 4 kHz area.

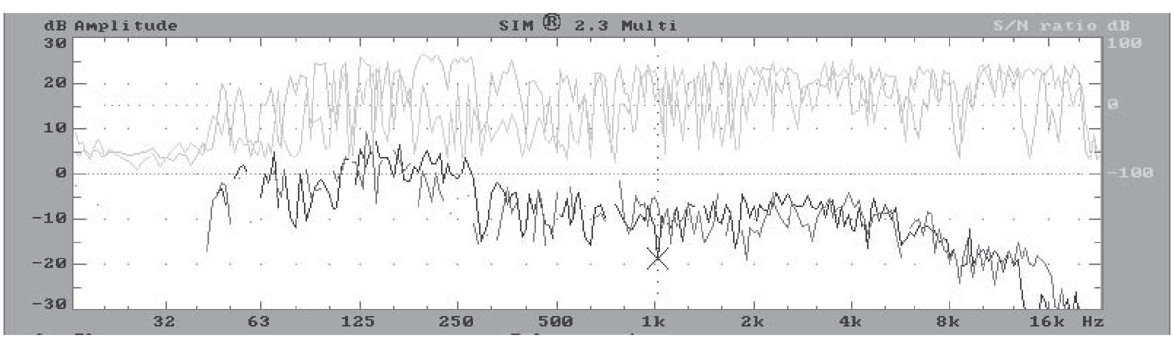


Fig 5.3e Primary and secondary positions.

This screen shows two mic positions taken seventy-five feet apart in an arena with a MSL-10A system. The unparalleled consistency of the MSL-10A's horizontal response can be clearly seen here, which makes for easy EQ decisions.

5.3.3 Tertiary Microphone Positions

Tertiary positions are a third class of placements used to verify various aspects of the speaker, such as proper wiring, gain structure and position. The data from these positions is not used to make level, delay or equalization decisions.

Tertiary Mic Position Sample Applications

- Near-field verification of crossover and polarity as described in Section 4.
- Coverage angle verification: The mic is placed at the expected axial edge.
- Seam analysis: The mic is placed at the transition zone between systems to verify that the coverage has no gaps.
- Sound leakage onto the stage: The mic is placed on stage after the system is aligned to observe the nature of the leakage.
- Analysis of a particular seat that your client is very concerned about, such as for a critic.

Common Misconceptions About Multiple Microphones

When multiple measurement microphones are mentioned there are a few misconceptions that tend to arise. The most common is that we will sum the microphones electrically to produce an average response. This has no validity whatsoever due to the comb filtering resulting from the summation of the signals with their different propagation delays. Such an idea could only work in a world without phase. Another misconception is that we multiplex the mics, switching from one to another in rapid succession. This is a vestige of real-time analysis and again totally neglects phase. A third misconception is the idea of "spatial averaging," where the mic is moved around while measurements are in progress. This may be useful for noise analysis but not for the alignment of a speaker system.

A final point regarding multiple mic positions is that they are not a random sampling as might be performed to check the chlorine content of a swimming pool. Each mic position is carefully chosen to give data about a *particular* speaker system, so that decisions can be made about *that speaker's position, level, delay and equalization.* A random sampling may provide interesting data but this is an alignment process, not a survey.

An application example of a teriary mic position is shown in Fig 5.3f.

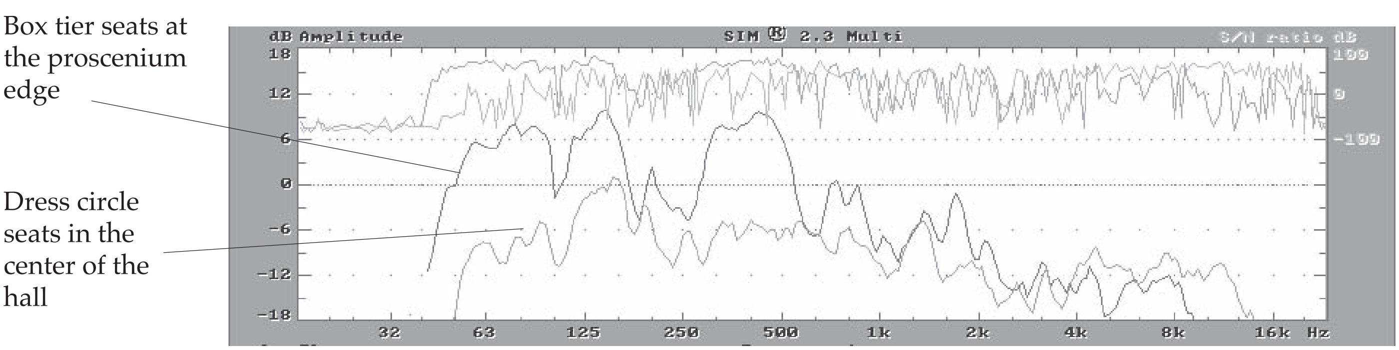


Fig 5.3f Tertiary mic position reading of off-axis box tier seating.

Many older theatres and opera houses have upper box tier seating wrapping around to the proscenium edge. These seats must be for people to be seen, since you can neither see nor hear the show from most of them. Not surprisingly, they can be a source of complaints to house management. In this example the client was hoping that the proscenium-based speakers would sneak harmlessly past the box patrons without blowing off their toupees. The response was measured and is shown here in its relative level relationship to the dress circle on-axis position. How did we do?

5.3.4 Multiple Microphone Positions Example

Figure 5.3g shows an application example of vertical coverage mapping of a speaker designed to cover the mezzanine and lower balcony levels. The areas above and below the speaker were covered by delays and downfills, respectively. An elevation view (Fig 5.3h) shows the positions of each mic. Notice the uniformity of response in the intended coverage area (positions B, C and D). Note also the HF rolloff and loss of S/N ratio at the positions outside the intended coverage area (mics A and E).

This was a case where the actual coverage area worked as designed.

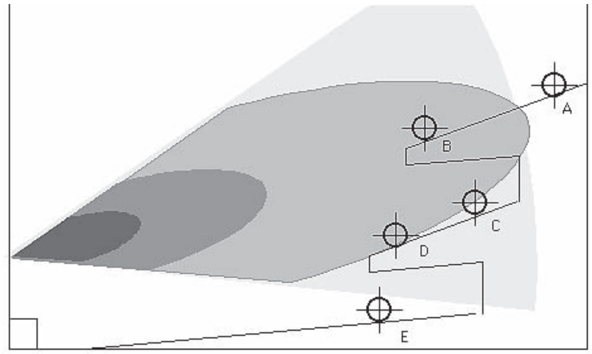


Fig 5.3h Mic positions and intended coverage.

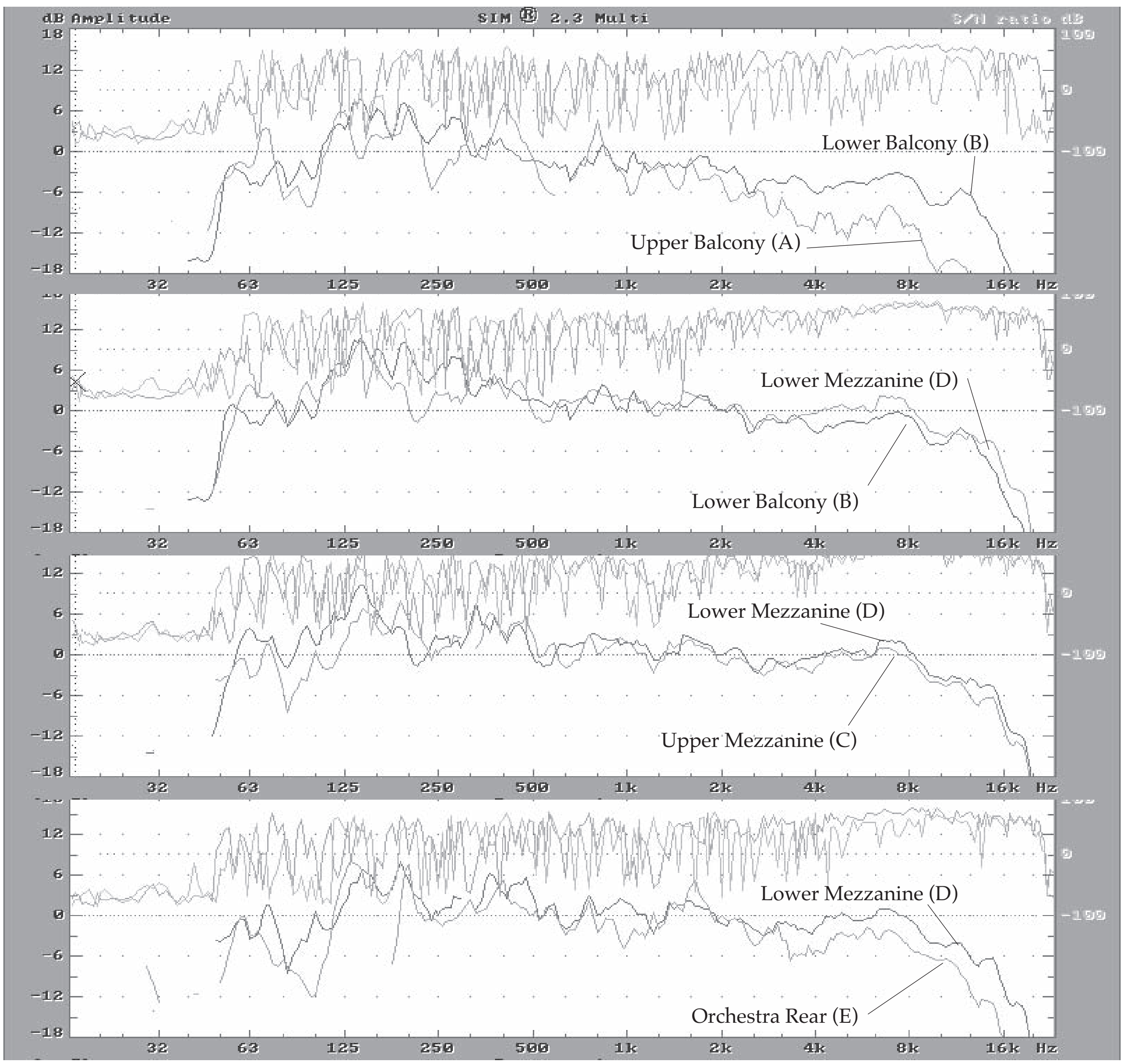


Fig 5.3g Vertical coverage map using multiple microphones.

5.4 Architectural Modification

Architectural modification is the most effective of the available options, but it is also the most expensive and slowest to enact. The costs of purely acoustical solutions tend to be inversely proportional to frequency. For example, something as simple as a curtain can be very effective for high frequencies, while the cost of giant bass traps for low frequencies is staggering, and the results are questionable. To the touring professional, the options in this category may be nonexistent due to budget, time and the availability of materials.

The advantage that absorption has over other techniques is its global benefit throughout the hall.

Figs 5.4a and 5.4b show the effect of adding absorption to a wall. The delayfinder trace of Fig 5.4a shows a sharp reduction of the power of the reflection. The frequency response is shown in expanded detail in Fig 5.4b. Notice that the ripple in the HF response is reduced sharply, decreasing the need for equalization. Notice also that this curtain (although quite thick) was totally ineffective in the low frequencies.

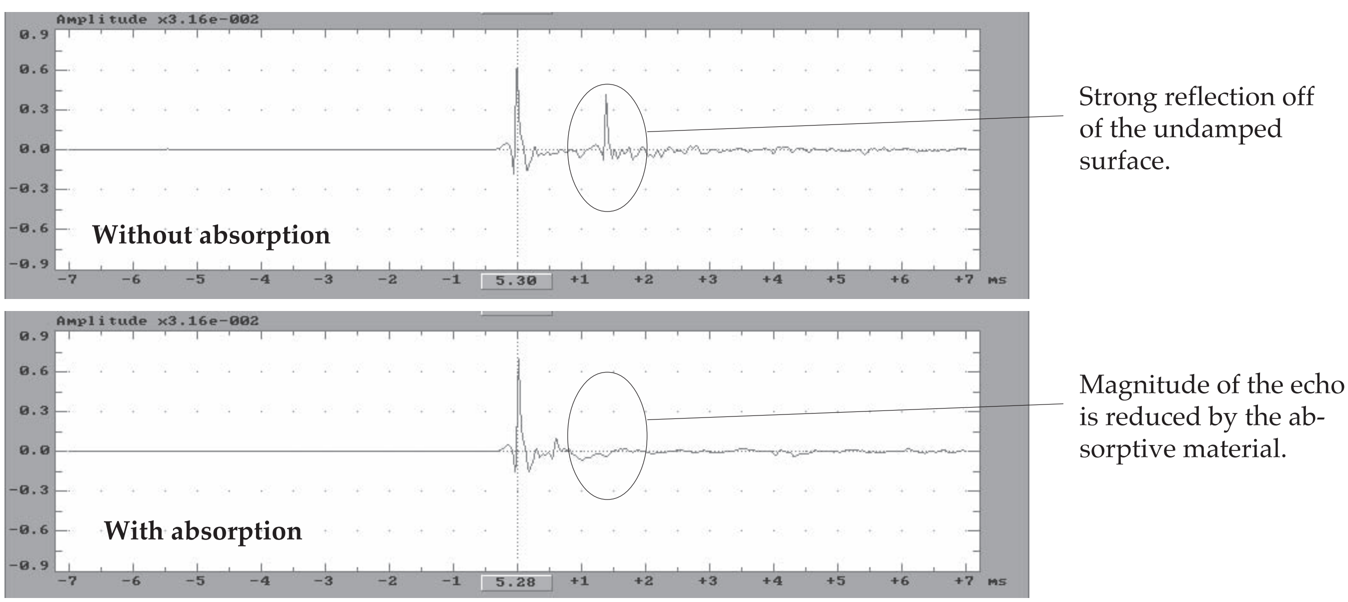


Fig 5.4a Effect of absorptive materials on the impulse response.

5.4 Architectural Modification

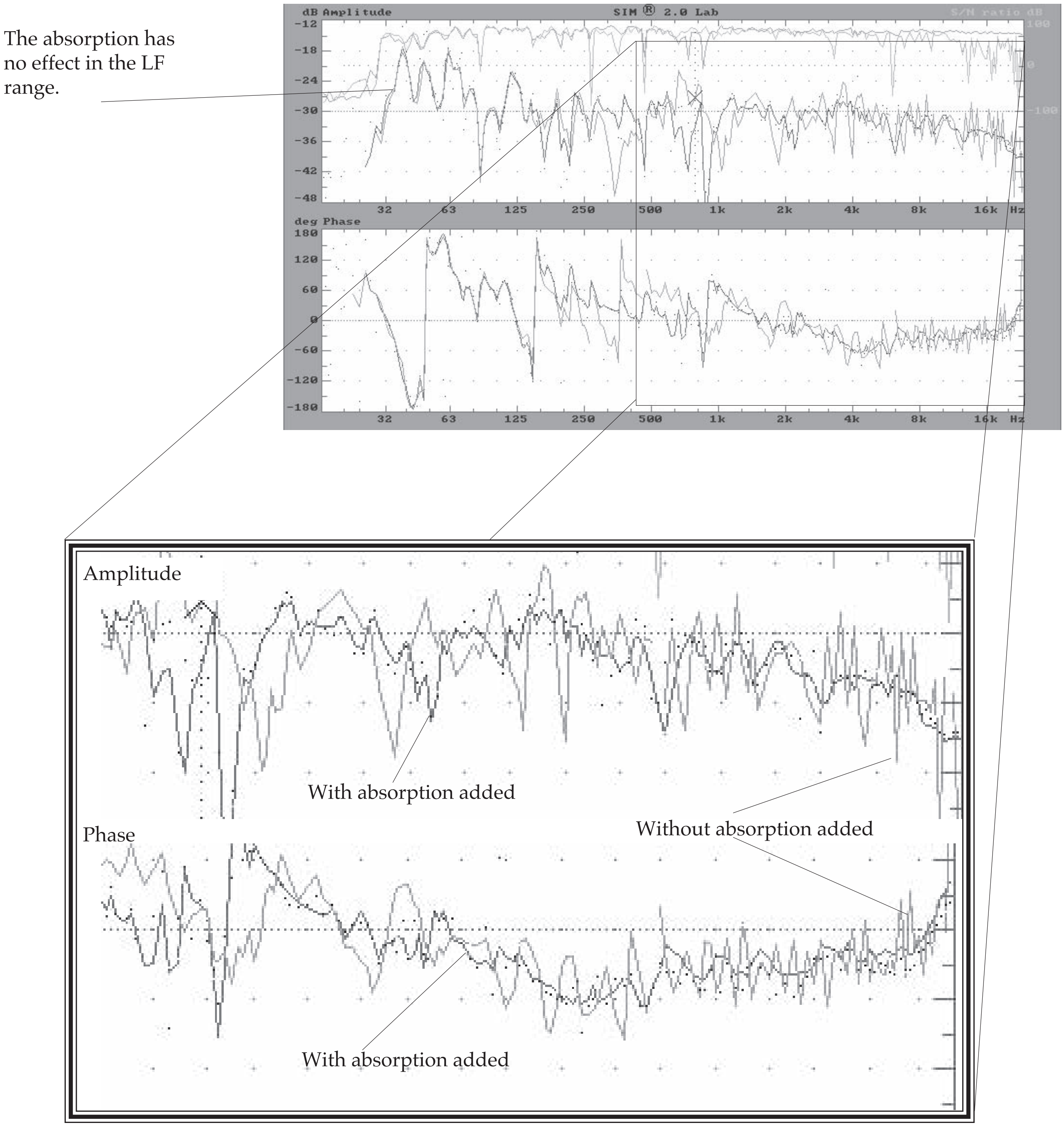


Fig 5.4b Effect of absorptive materials on frequency response.

5.5 Speaker Repositioning

Speaker positioning is critical to creating a system design that can adequately cover the audience area with a maximum direct-to-reflected sound ratio. Unfortunately, the ideal locations for speakers often turn out to be impractical due to sight lines, lack of hanging points, or obstructions. Compromise seems to be the status quo for speaker positions.

System subdivision provides a means for individually controlling the sound field in different areas of the room. Most sound reinforcement applications can benefit from separating the coverage into individual zones, each with individually adjustable speaker systems. This is the most effective way to maximize uniformity of coverage and minimize the interaction between speaker systems. However, this strategy will only succeed if the speakers are positioned properly.

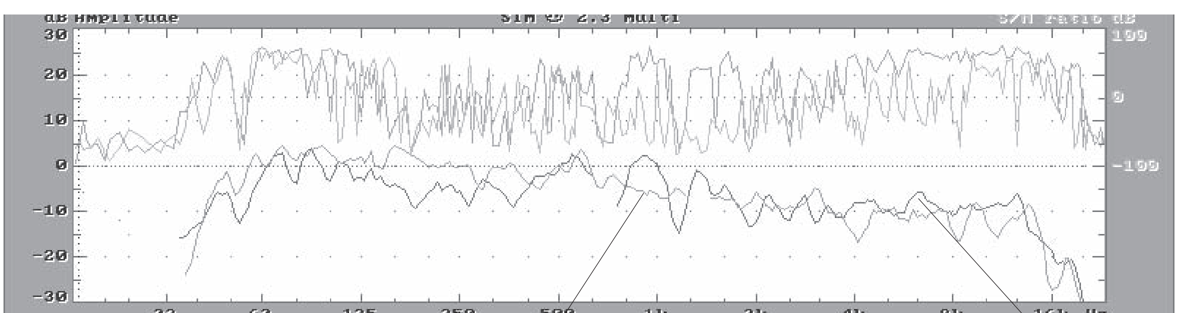
The speaker coverage can be verified by checking the edges of the coverage pattern and comparing them to the on-axis response at the primary position.

How to Verify Speaker Coverage

1. Place a measurement mic on-axis, measure the response, store it, and recall it onto the screen.

2. Move the mic to the intended edge of the coverage pattern and measure, making sure to maintain the same approximate distance from the speaker. This can be verified by checking the propagation delay time, which should be nearly the same.

3. The HF region of the second measurement should be 6 dB down from the original on-axis measurement. The HF region will roll off faster than the LF due to increased directional control at high frequencies and the lack of absorption of the lows. This is particularly evident with highly directional systems. Another key factor to consider is the S/N ratio. If this has fallen dramatically between the on-axis and off-axis positions, there may not be enough direct sound to maintain intelligibility.



Off-axis (45° off the horizontal center).
Trace position is boosted 6 dB.

On-axis at the primary
mic position.

Fig 5.5a Coverage verification.

These measurements were made in a reverberant room. In this example the off-axis curve has been raised 6 dB by a vertical scale offset. Therefore, when they appear equal on the screen the off-axis signal is –6 dB. Notice that the S/N ratio is reduced in the off-axis area. This has proven to be a very effective technique for checking coverage.

5.5 Speaker Repositioning

Once the coverage angle is measured we will come to one of the following conclusions.

The coverage angle is . . .

• **Too wide.** If there is too much spill out of the intended coverage area there will be excess interaction with the room or with other speakers. There will be a loss of S/N ratio due to increased noise (the interaction).

• **Too narrow.** If the speakers do not reach the required edges of the pattern, there will be gaps in coverage. There will be a loss in S/N ratio due to decreased signal (the lack of direct sound).

• **Optimized.** If the speaker's coverage equals the required pattern edge the coverage will be complete and the interaction minimized. The S/N ratio will be optimized.

Speaker positioning is more complex than simply reading a data sheet or plotting data with a protractor or prediction program. A single speaker may react predictably, but arrays of speakers are complex.

Field examples are shown in Figs 5.5a to 5.5d.

Techniques for Improving Coverage

- Repositioning or angular change of an individual speaker.
- Adjustment of the splay angle between cabinets in arrays.
- Amplitude tapering of arrays (see Section 3.6).
- Addition of fill systems to supplement the edges.

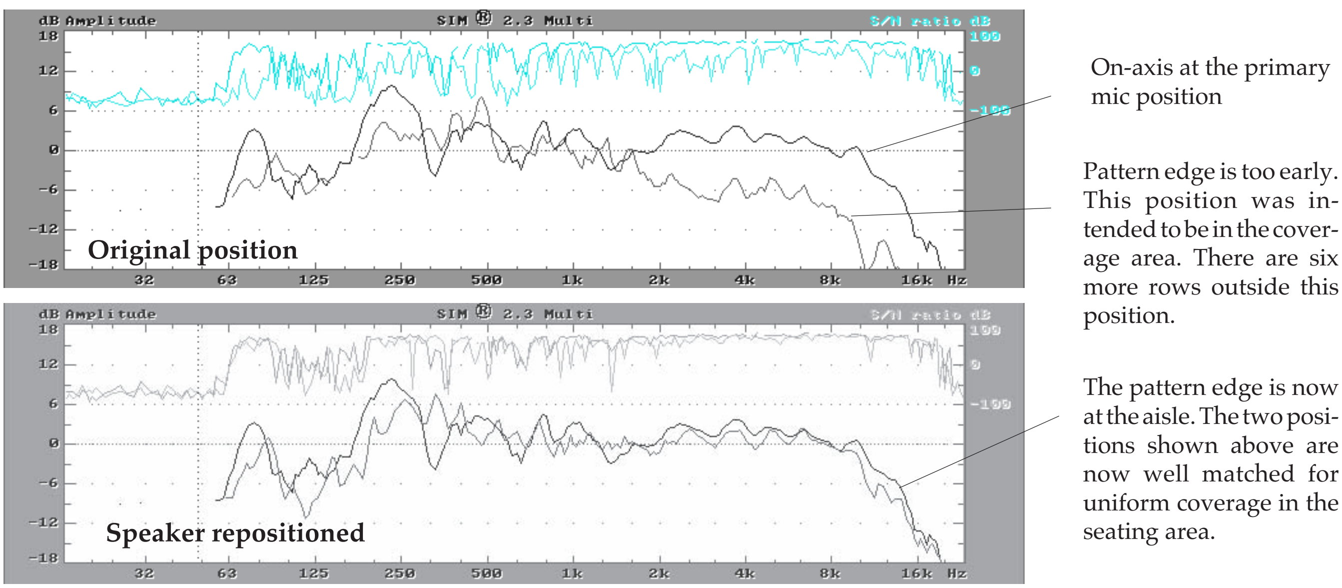


Fig 5.5b Speaker repositioning field example.

The above data illustrates the need for coverage verification. In the original position the speaker was aimed too far inward and a large number of seats fell outside of the coverage pattern. The speaker was the UM-1 which has a very sharp 45° pattern above 2 kHz. Using a secondary mic position we found that the coverage pattern ended six rows short of the aisle. Top screen: The speaker was repositioned so that the edge was at the last row. The result of the repositioning is shown in the lower screen.

5.5 Speaker Repositioning

The best estimate is made of the optimal position and angle before the rig goes up in the air, but unfortunately small errors can create big problems. This becomes more critical as directional control increases. We recommend that whenever possible, leave flexibility in the design of your systems for final angles and positions. The cost of flexibility is usually well worth the reduction of planning time, stress and above all, excuses.

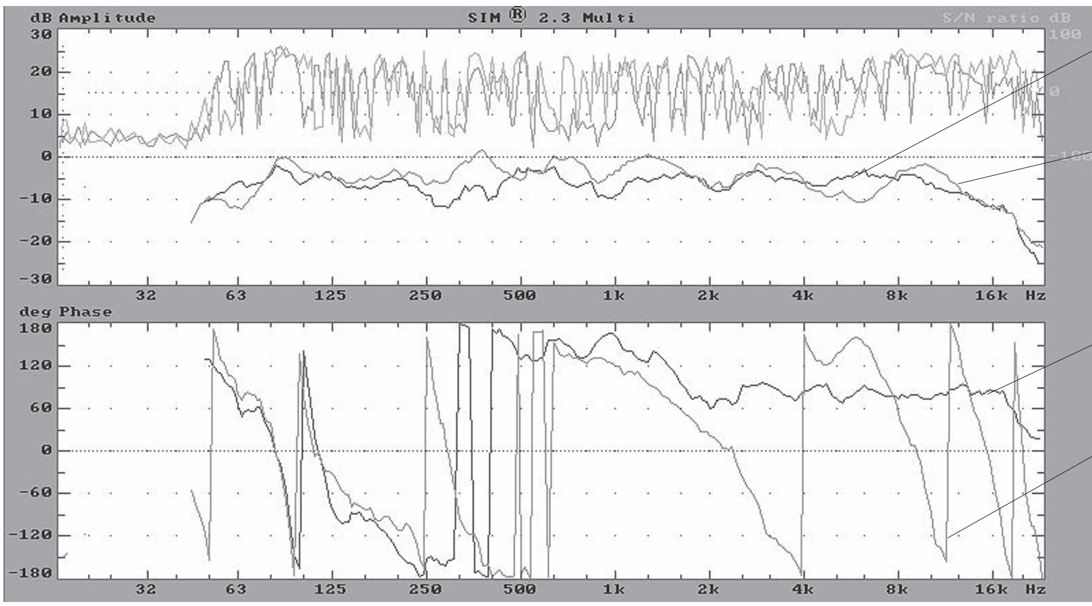


Fig 5.5c Speaker position example.

The balcony and mezzanine were covered by separate vertical rows of speakers. Horizontal coverage was fine on both levels, but when the vertical was checked it was found that the balcony system spilled too much into the mezzanine. The measurement of the mezzanine shows that the level from the two systems are equal. It was intended that they be well isolated. The longer throw required of the upper system created a downlobe. Notice also that the phase responses do not match (the balcony speakers are arriving late) creating the potential for cancellation. The solution was to raise the upper system's vertical angle.

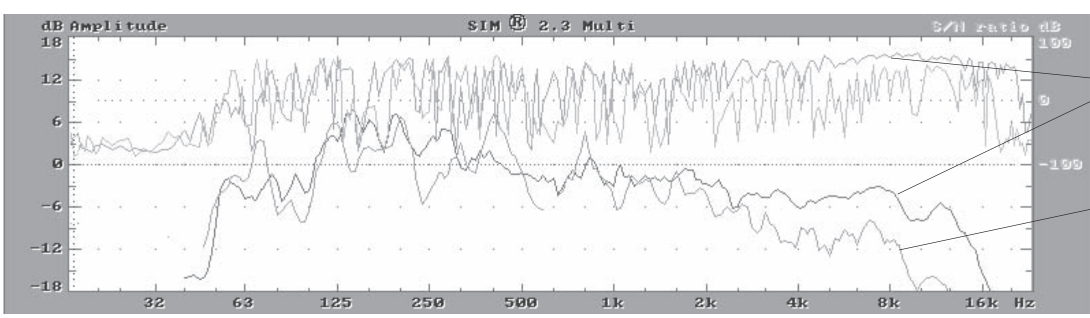


Fig 5.5d Speaker position example.

It was expected that the vertical coverage of this speaker would be too narrow to reach the top six rows of the balcony. Measurements were made to check this before the client invested auxiliary delay speakers. The data clearly shows that the HF range is outside the coverage pattern and that the S/N ratio is compromised. The solution was to add delays.

5.6 Gain Structure Adjustment

Gain structure adjustment can be used to compensate for the different sensitivities required when systems are assigned to cover different areas. By adjusting the relative levels between systems the interaction between them can be further minimized.

Gain structure adjustments will allow us to control two critical parameters:

• The relative levels of components in multi-way systems. Each system may be designed with differing proportions of upper range, mid-bass and subwoofer systems. Therefore, it is necessary to evaluate the relative levels of these systems on a case-by-case basis in order to achieve the proper addition of these systems through crossover.

• The relative level of subsytems. The relative level of each subsystem is adjusted for maximum overall uniformity and minimum interaction.

Measuring Sound Level

The most logical choice for measuring sound level— the sound level meter—turns out to be so poorly suited as to be almost useless. There are several reasons for this.

The sound level meter gives a single reading for the entire frequency range of the amplitude response. The spectrum of the different systems to be calibrated can differ dramatically. For example, a sidefill system may contain no subwoofers, relying upon LF energy from the main array. The sound pressure level meter (SPL) is not helpful for setting the level of multi-way components since we are primarily concerned with the level at the crossover area, not their entire ranges. Most importantly, however, is the SPL meter's inability to differentiate between direct and reverberant sounds. In the far-field of a reverberant hall the low frequency response will experience minimal attenuation while the high frequencies will drop as per the inverse square law. This causes even SPL readings over large distances, even though the sound quality and intelligibility have been destroyed in the far field. To set levels of subsystems in a way that reflects our sonic experience, we must see the full-range frequency response of the system over level. In addition, we must differentiate between the early-and late-arriving signals. Both of these aspects are incorporated in SIM System II.

Mix Console VU Meters

There is nothing more out-of-date in modern sound systems than volume unit meters (VU). We have state-of-the-art electronics designs that continue to use the best that the 1940s had to offer in signal monitoring. In the current age, VU meters lead to ridiculous ideas about gain structure and are a leading cause of excess noise in sound systems.

There are three obvious limitations to VU meters:

• Response time is limited by the ballistics of the meter.

• The usable range is limited to about 20 dB.

• The highest level shown is about 20 dB below the actual overload point.

We've come a long way in fifty years. A high-hat cymbal's transient response blows past a VU meter without moving it. We now have systems that have 110 dB of dynamic range.

In order to read a VU meter's response we must know precisely the transient nature of the applied signal. If it is an organ pipe at 100 Hz or a continuous sine tone the VU reading may be quite accurate. Not so with a high rise-time device like a snare drum. Then there is the fact that there is over 20 dB of headroom above the +4 reading that pins a typical VU meter. That organ tone can pin the meter continuously and yet never clip. Meanwhile, the snare is clipped if the meter occasionally moves up to –10 dB. This kind of inaccuracy can make mixers understandably nervous about using the last 20 dB of system headroom. Bring on the noise floor. Having VU meters that pin at +4 is like having a speedometer on your car that only goes up to 40 mph (64 km per hour).

Fortunately, many of the modern meters have an additional "peak" LED that accurately indicates when clipping actually occurs. This is the one to watch. Some manufacturers have moved over to peak reading LED bars that read up to the +24 region, providing a much more accurate picture.

5.6 Gain Structure Adjustment

Where Should Level be Adjusted?

In the average audio system there are a myriad of points in the signal chain where level can be adjusted. However, there are distinct advantages to tapering the subsystem levels near the end of the signal chain. The idea is simple.

For sub-systems that are driven with the same input signal:

- The console output drives should be the same.
- The level should be adjusted post-EQ.

Advantages to post-EQ level adjustments:

- All pre-EQ components will overload at the same point.
- All sub-systems will have maximum immunity from noise.

Disadvantages to pre-EQ level adjustments:

- Some sub-systems may be clipping at the console while others have huge amounts of headroom.
- The systems that are attenuated will be running near the noise floor of the console, digital delay line, and equalizer. This will result in higher noise levels at the amplifier inputs.
- The lower level systems will have less immunity from electro-motive interference (EMI) and radio frequency interference (RFI) as it traverses the house snake.

Have you ever been to a concert and walked from back to front, noticing that the frontfill speakers are hissing like crazy and yet the main system is quiet? How could it happen that the noise could be managed for a speaker throwing two hundred feet but not for the ones that only throw twenty feet? This happens when the gain structure is mismanaged so that the levels are reduced far upstream and the gain is made up at the amplifiers. In a case like this you will probably find a matrix output turned down 20 dB and the amplifiers and controllers up full. It is extremely common to find systems with all of the amplifiers up full. The most common reasons given for this are "to get the most power," or, "so we can be sure that nobody has messed with the settings."

First, the idea that setting your amplifier levels to maximum gives more power is totally erroneous (see Section 1.4, Amplifiers). The second reason is a lousy excuse too. It is like saying, "Lets set all the amps wrong so that we don't have to worry about someone coming in and setting the amps wrong."

Here is a simple example. You have two identical speakers to cover the main seating area and sidefill, respectively. The side area is half the depth of the main hall so we can expect to run the sidefill system 6 dB down from the mains. If we turn down the matrix out for the sidefill we pick up more noise through every succeeding device, only to amplify it at the same level as the mains. If we run everything equal up to the amp or CEU we will have less noise in the sidefill system with no loss in headroom.

If the system is set up so that the drive levels to the controllers or amplifiers are matched, then all systems will overload at the same time—you are either clipping or not—and the noise floor is below the overload point by the same amount for each.

The keys to gain structure management are:

- Get your gain as early as possible and keep it.
- Think of it like a freight train—not a roller coaster.
- Minimize gain requirements in the final stages where noise amplification will be the greatest.

5.7.1 Delay Setting: Introduction

Delay lines are now cheap and plentiful. They are an invaluable tool for designers, increasing speaker position flexibility and allowing fine tuning of speaker interaction to minimize combing, but *only if they are set accurately*. If set poorly, they will maximize combing. Remember that an error of just 1 ms decimates your response from 500 Hz and up. An error of 10 ms wrecks it from 50 Hz on up. The necessity of accurate tools to set delay times cannot be overstated.

The role of delay lines in our design is threefold:

1. To synchronize distributed speakers with the main system. This reduces comb filtering by increasing the direct-to-reverberant ratio in shadowed, off-axis or distant areas that cannot be covered by the main array.

2. To decrease the comb filtering in speaker arrays by minimizing time offsets in the overlap zones.

3. To steer the sonic image toward the performer and away from the speakers.

WARNING: It is a fairly common practice for people to intentionally set incorrect delay times in an attempt to prevent the image from appearing at the speaker. Often some 10 to 20 ms of excess delay is added to create the "Haas Effect." This practice dramatically reduces the S/N ration of the combined system and is not recommended. For more information, see my article "Haas Effect, Precedence Effect and Side Effects (*Mix Magazine*, June 1996).

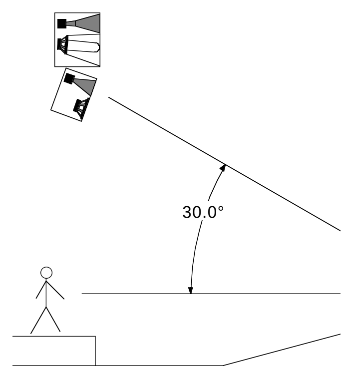


Fig 5.7a Vertical sonic image distortion.

In this example the image may appear as much as 30° above the performer. The greater the angle, the harder it will be to create realistic imaging. The image can be brought down if the performer's voice carries well or if lower speakers are added.

Sonic Imaging

The sonic image location is governed by the arrival time and relative level. Let's assume the intention is to create a sonic image at the acoustic origin (the stage, actors, musicians, etc.). Therefore, if the image appears elsewhere—displaced either horizontally, vertically (or both)— we can consider this to be *image distortion*. Our goal is to minimize image distortion without creating other damaging side effects (i.e., comb filtering). We are more sensitive to horizontal image distortion than vertical due to our binaural localization. This can be used to advantage in practical applications.

The sonic image can appear at any point on a line between two sources, depending primarily on first arrival, and on relative level. With three sources, the image can be positioned anywhere inside the triangle, giving us a method of vertical and horizontal image control.

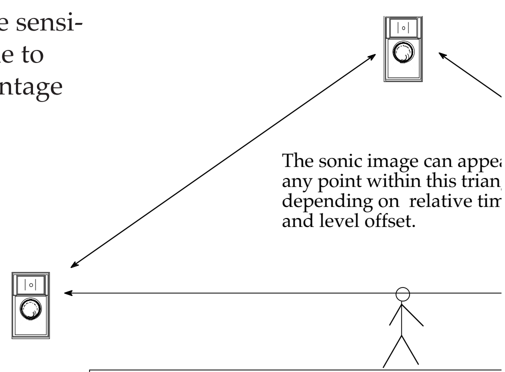


Fig 5.7b Sonic image triangle.

5.7.2 Choosing a Reference Speaker

In complex systems there are many options available. Which speaker is delayed to which? The choice of reference speaker (the source we will delay to) will affect the sonic imaging of the system as well as the interaction between systems. In an ideal scenario it is possible to view a distributed sound system as an exploded point source. The selection of reference speaker can simply be traced back to the center point. An example of such a system is Roger Gans's design for Walt Disney's Holiday on Ice.

Most systems do not have such clear-cut delay strategies. Typically they resemble elongated point sources in both the vertical and horizontal axes with the virtual point source far behind the stage. In such cases, it is necessary to carefully consider the order in which systems are synchronized, and which speaker will become the reference source.

The "Fictitious Source"

To create a sonic image on stage it will be necessary to place a sound source temporarily on the stage as a time reference. Such a reference speaker is referred to as a “fictitious source.” It should be placed at a point on stage that represents a typical upstage position for the actor. For bands, use the drum riser or the back line (whichever is louder), as the reference point.

Speaker systems in direct proximity to the stage—such as a front fill or on the lower proscenium—should be synchronized to the fictitious source.

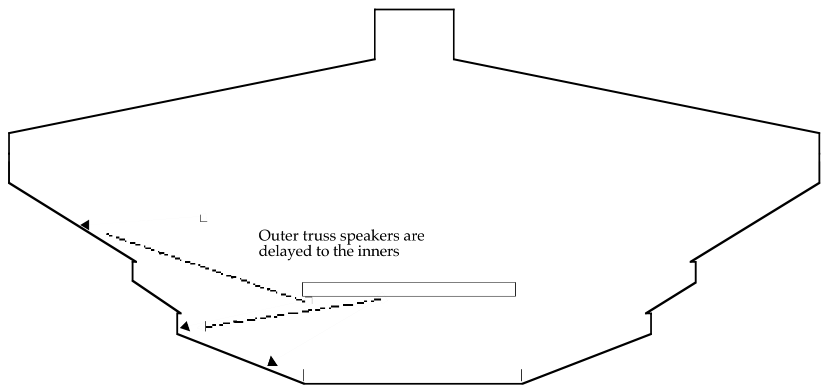


Fig 5.7c Disney on Ice delay alignment.

Location	Speaker	Delay Reference
1	Frontfill	Fictitious Source
2	Stalls (Orch)	Fictitious Source
3	Stalls Underbalcony	Stalls (Orch)
4	Circle (Mezzanine)	Stalls (Orch)
5	Circle Underbalcony	Circle (Mezzanine)
6	Balcony	Circle (Mezzanine)

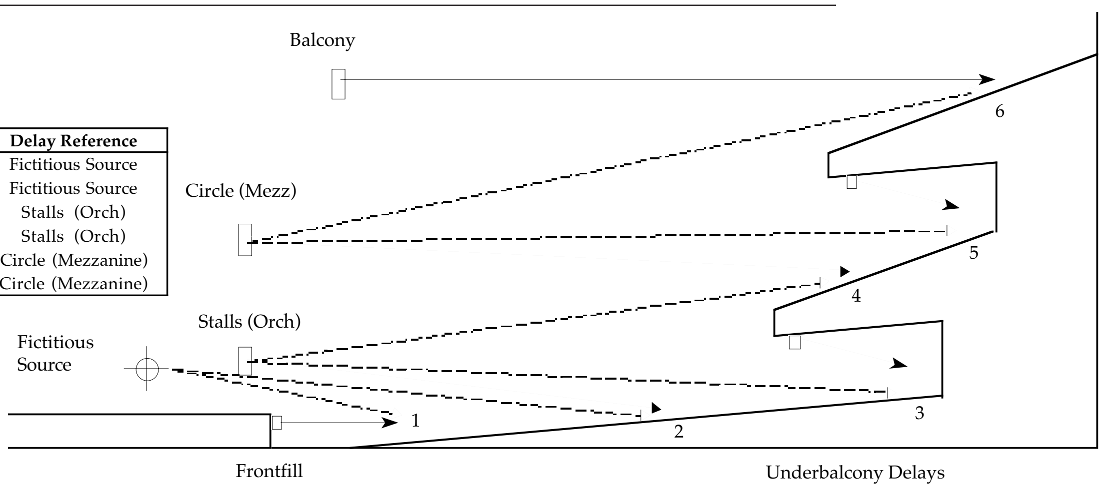


Fig 5.7d Musical theatre design example.

Musical theatre systems designed by Abe Jacob, Andrew Bruce, Tony Meola and others employ complex multi-tiered systems. As you move upward, each level is synchronized to the one below. This minimizes interaction and keeps the image down on the stage.

5.7.2 Choosing a Reference Speaker

As you move away from the stage it is important to evaluate the choice of reference speaker by answering the following questions:

- Where do you want the image to appear?
- What other speaker(s) have the most overlap into the delay speaker's coverage area?

If the overlapping speaker is positioned in the direction of the desired image, the decision is clear: delay your speaker to the overlapping speaker, as in the ice show design.

If the overlapping speaker is *not* positioned in the direction of the desired image, you will have to weigh the following: If you do not synchronize to the overlapping speaker, and instead sync to a less interactive source, you may be able to move the image as desired, but at the cost of decimating your frequency response by combing.

The actual procedures for setting the delays are simple when accurate tools, such as SIM, are applied. Most importantly, SIM actually measures whether or not the indicated delay has actually been achieved by the system delay. (Just because the front panel says 56 ms doesn't make it so!)

The procedure for setting the delays has five steps:

1. Place the mic in the primary position in the field of the delay speaker.
2. Measure the delay speaker's propagation time to the mic.
3. Select the reference speaker (as discussed above).
4. Measure the reference speaker's propagation time to the mic.
5. The difference between these two figures is then added to the delay line.

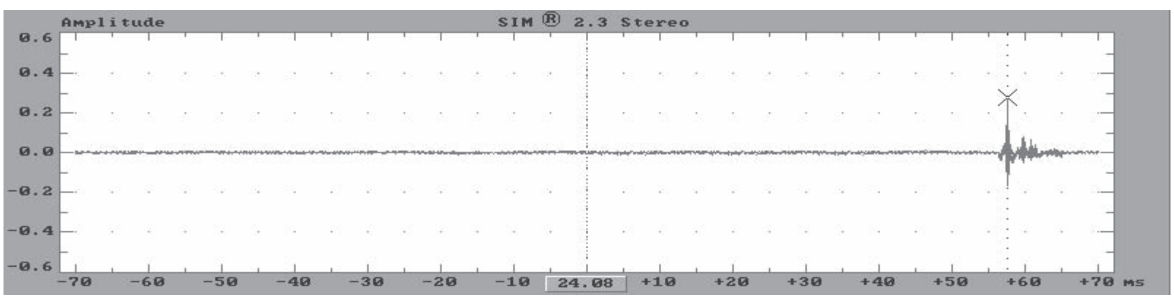


Fig 5.7e Delayfinder data indicates that 56 ms of delay must be added to this system. This is seen by the position of the impulse.

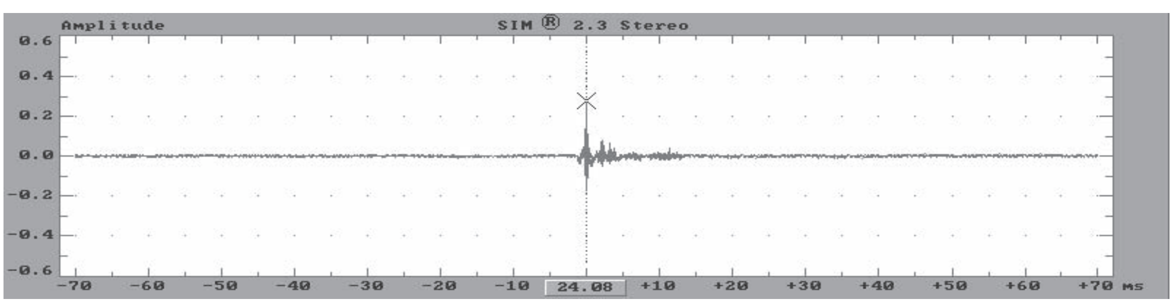


Fig 5.7f Delayfinder data indicates that the correct delay has now been added, thereby centering the impulse response.

5.7.3 Delay Tapering

The benefits of using delays in a distributed system are obvious. However, delays can also be a key factor in minimizing destructive interference between speakers in close proximity, such as horizontally and vertically arrayed point-source elements.

The amount of time delay applied here is small, usually 1 ms or less, but the benefits can be huge. Remember that 1 ms of time offset between two systems will decimate your response above 500 Hz.

The technique is called "delay tapering" and will provide substantial benefit if the following relationships exist between two speakers:

- They must have distinct coverage areas with controlled overlap.
- They cover different distances (or they must be run at different levels).

When multiple speakers are combined each listening position is different in terms of its distance from the speakers and the axial orientation to them. These two factors will have a large effect upon the frequency response since it will determine the depth of comb filter interaction and which frequencies will be boosted and cut.

The frequency response relationship between the speakers is a function of the listener's axial orientation to the two speakers. (The speaker that is more on-axis will have more HF energy.)

These effects can be controlled to some extent by adjusting the speaker position and relative level, both previously discussed . In addition, they can be controlled by precise adjustments of delay lines to synchronize the speakers at the point where comb filtering is the most extreme.

To best understand the concept of delay tapering consider the three types of center points between two identical speakers

- **Geometric:** The physical center between the speakers. This can only be changed by moving one or both speakers.
- **Power:** This is the point where the two speakers are equal in level. This would be at the geometric center unless one of the speakers is driven louder than the other. As the level is offset, the power center will move toward the speaker that is lower in level.
- **Temporal:** This is the point where the two speakers arrive at the same time. This would be at the geometric center unless one of the speakers is delayed relative to the other. As the time offset increases, the temporal center will move toward the later speaker.

In practice, the physical center is irrelevant in regard to interaction. The nature of the acoustical combination of the speakers is entirely a product of the power and temporal centers. The best acoustical addition and the maximum response uniformity will occur if the power and temporal centers are matched.

Observe the two identical speakers shown in Fig. 5.7g. The geometric center line is both the point of equal acoustical energy and time arrival. Therefore the systems will have maximum addition at this point. Points to the left and right of the center will exhibit comb filtering as you approach one or the other speaker. As you move further to the sides the comb filtering will be reduced due to axial attenuation in the more distant speaker.

In circumstances where there are asymmetrical coverage requirements, however, it may be advisable to attenuate the speakers with a shorter throw. This is shown in Fig. 5.7h. Notice that the point of equal acoustical energy and time arrival have become separated due to the decreased energy of the right speaker. The marked position represents the equal power point. This position will have substantial combing due to being offset in time while equal in level (see Section 2).

5.7.3 Delay Tapering

Fig 5.7i shows the effect of delaying the previously attenuated speaker "B." Delaying the speaker moves the time synchronous position in line with the equal energy. This will eliminate the combing.

You might ask, won't the center point of the system now comb? Yes, it will. But remember that the right speaker has been attenuated, so the depth of the combing is reduced. Once you have introduced multiple sources you will never be able to eliminate all comb filtering, however, you can reduce it substantially by targeting areas where the sources are equal in level, as above.

When is Delay Tapering Most Useful?

When the throw distance requirements of two coupled speakers are substantially different. For example, in a horizontal array where the distance to the center is long but the distance to the side is short.

Vertical Delay Tapering

Delay Tapering is also useful for aligning systems vertically, as in the case of downfill arrays. This is the most typical form of delay tapering because most vertical coverage requirements are grossly asymmetrical, i.e., the upper speakers must throw much farther than the lower ones.

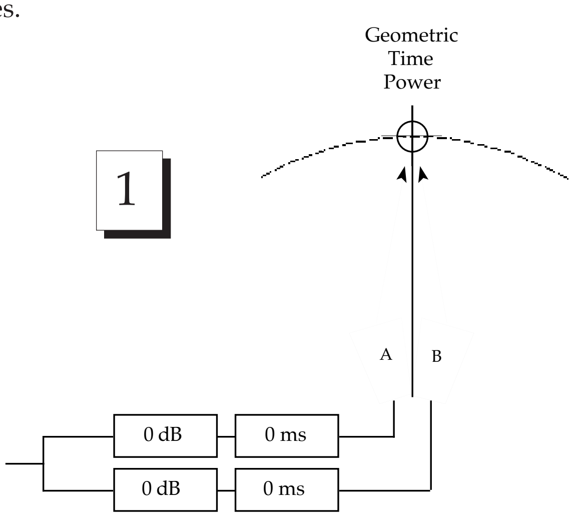


Fig 5.7g Delay tapering—All centers are together.

Identical speakers at same level and delay. The geometric, power and temporal centers are all the same. The interaction will gradually decrease as you move off center.

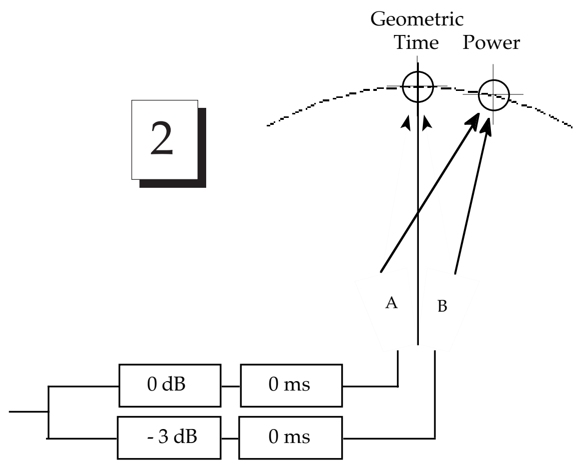


Fig 5.7h Delay tapering—Power center has moved.

The level of speaker "B" has been reduced for some reason (it has a shorter throw). The geometric and temporal centers remain in the center but the point of equal power moves toward the quieter speaker. There will be comb filtering at the power center due to the late arrival of speaker "A" into the area.

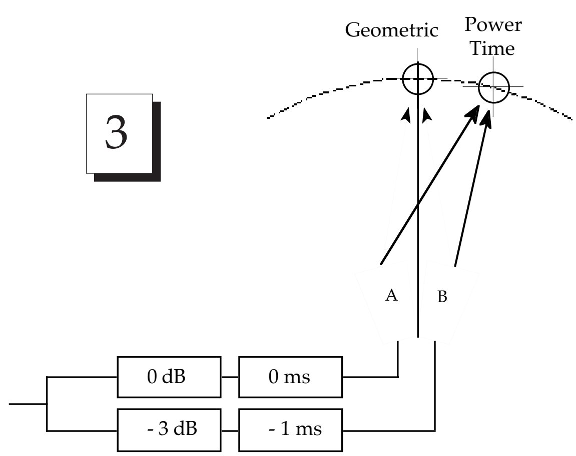


Fig 5.7i Delay tapering—Time center moved to power center.

Speaker "B" has been delayed so that the power and temporal centers have now come together again. The interaction will be minimized at the power center due to the synchronous arrival of the speakers.

5.8.1 Introduction

Complementary equalization is the last of the available options. While equalization can be very effective it is critical that it only be employed when it is the proper tool. Attempting to equalize for the effects of bad speaker positioning, misaligned delay lines, or interactions due to poor gain structure will inevitably be disappointing.

The emotional charge around equalization makes it stand out among the techniques for system alignment.

Rule number 1 is that everyone is an expert.

Rule number 2 is that everyone has an opinion on the choice of EQ settings. You'll get the advice of everyone from the sound crew, the lighting designer and the audience.

Rule number 3 is that everyone is against it. You'll hear statements like, "The speakers sounded great but I had equalized it," the message being that needing to EQ it somehow diminishes the end result. Equalization has many slogans and folklore.

This is the standard equalization motto:

"S/He who EQ least, EQ best."

"The best EQ is no EQ."

(But make sure there are plenty of them at the gig.)

Since equalizers are held in such low regard, you might think that you would find a lot of systems without them. Not likely. These slogans speak to the fact that equalization is a solution with a limited range. Too often equalization is embraced as the cure before the system is verified and without regard for the other techniques for alignment. This leads to disaster.

The reason we need EQ is that our frequency response is heavily modified between the original source and our ears. Let's follow the frequency response of an instrument as it makes the journey from the stage to our ears.

On Stage

- The microphone frequency response adds its signature.
- The proximity of the mic to the instrument affects the LF response of the mic (cardioid mics).
- Local reflections near the mic will cause combing.
- Stage monitors can potentially produce deep broadband combing if the monitors are too loud. (This is rare, of course.)

At the Mixer

- Summation of multiple sources at the mixer of a single instrument can cause combing, e.g., leakage of a guitar speaker into multiple mics.

In the House

- The speaker system frequency response.
- Multiple speaker interaction causes coupling and (potentially) combing.
- Speaker room interaction does the same.

It is critical to discern between these factors if we are to restore the original response through equalization. The key is that a clear line must be drawn between channel equalization and speaker system equalization. The resulting response of each mic is radically different when you add the above factors together. The equalization for one channel will not likely be valid for the next. For direct input channels factors, the stage factors are nonexistent.

5.8.1 Introduction

The key to creating an equalizable system is:

- Choose mics with a linear frequency response.
- Minimize leakage on stage by isolating mics and minimizing (or eliminating) stage monitors. If monitors are to used, they should be highly directional and kept as low in level as possible. (Good luck.)
- Minimize leakage by gating mics where possible.
- Minimize multiple speaker interaction as detailed in the previous sections.
- Minimize speaker/room interaction as detailed in the previous sections.

Complementary equalization, as the name implies, is a technique for setting the equalizer to create a complement *in both amplitude and phase* for the complex acoustical response of the room+speaker system. Since the effects are highly complex and can manifest themselves with variations in center frequency, bandwidth, and magnitude, it is necessary to use an equalizer that is capable of controlling all of these parameters independently. The Meyer Sound CP-10 Complementary Phase Parametric Equalizer was specifically designed for this task.

The equalizer response can be measured directly as shown in Fig 5.8a. This is of great importance since the front panel displays of equalizers can be misleading. This is particularly true for graphic equalizers, which do not factor in the interaction between bands. An example of this is shown in Section 1.6.

Recent innovations in equalizers have given rise to models that have computer generated representations of the frequency response, and, to varying extents, factor in the band interaction. While these are much more accurate than conventional equalizers, note that all of these units base their display on the *selected* equalization parameters, which, unfortunately may not be the *actual* response. This has been observed far too often by various SIM engineers. *The best technique is to measure the equalizer directly.*

5.8.2 Room/EQ/Result Measurements (SIM®)

Fig 5.8a shows the connections for the three transfer functions involved in complementary equalization.

Measuring the Room+Speaker Response

The source feeds the equalizer, the output of which is connected to the room+speaker system.

The EQ output signal is delayed and used as the reference against the microphone response. The analyzer displays the difference between these two signals, which is the frequency response of the room+speaker system.

Measuring the Equalizer Response

The EQ input signal is compared to its output. The equalizer has been adjusted to produce a dip corresponding to the peak in the loudspeaker system.

Notice that the analyzer can display an *inverse transfer function*, causing the equalizer's response dip to appear as a peak **(1/EQ)**. This greatly facilitates the system correction procedure: all the operator needs to do is to set the system equalizers so that the **1/EQ** display duplicates the major characteristics of the room+speaker display. A precise *complement* of the loudspeaker/room characteristic is thereby generated.

To see the effect of the equalization, measurements of the *result* are used.

Measuring the Result Response

The EQ input is compared to the microphone. The corrective effect of the equalization is shown in the frequency response display.

In practice, all three of the responses (room, EQ and result) are measured and displayed simultaneously as shown in Figs 5.8c to

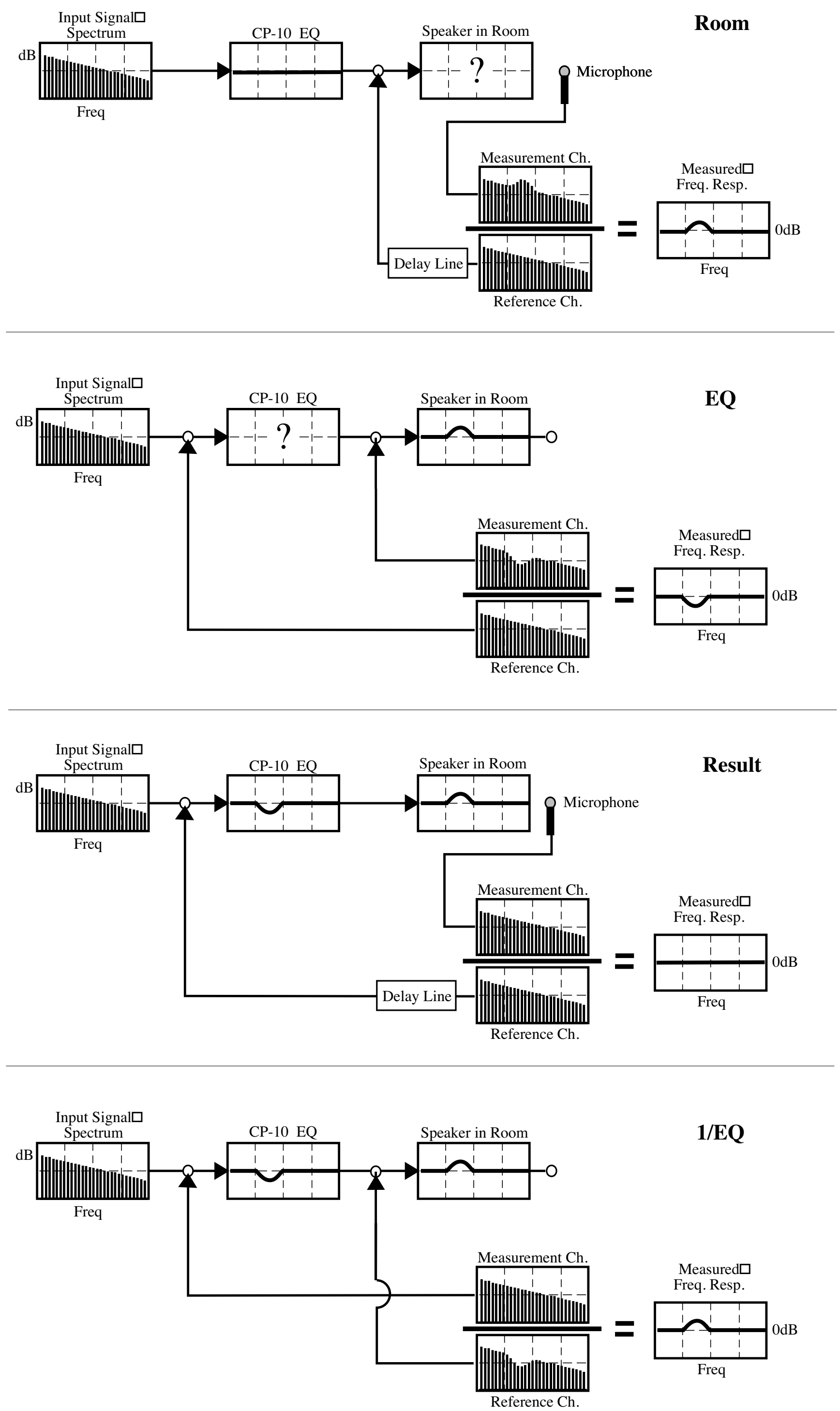


Fig 5.8a Room/EQ/Result Flow Block.

Drawing by Jamie Anderson

5.8.2 Room/EQ/Result Measurements (SIM®)

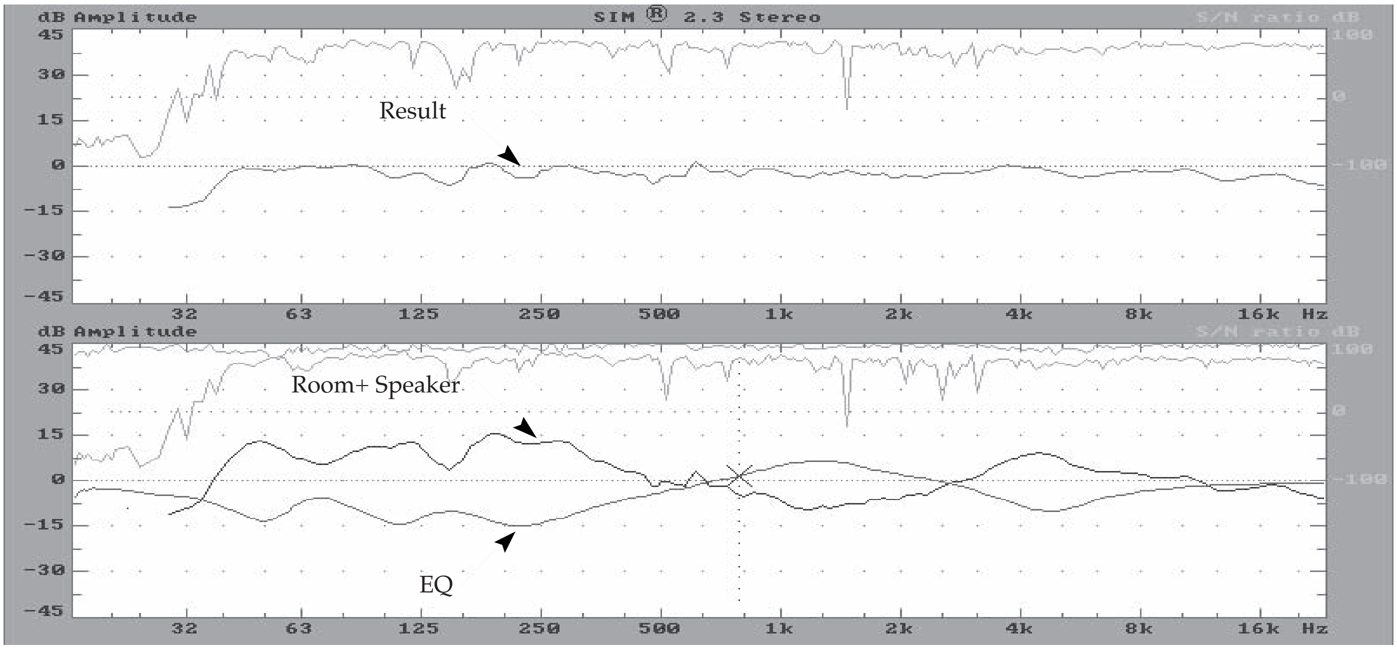


Fig 5.8b Group view of the three SIM transfer functions. The equalizer response is adjusted until it has created an inverse of the room+speaker response. This creates a flat response in the result.

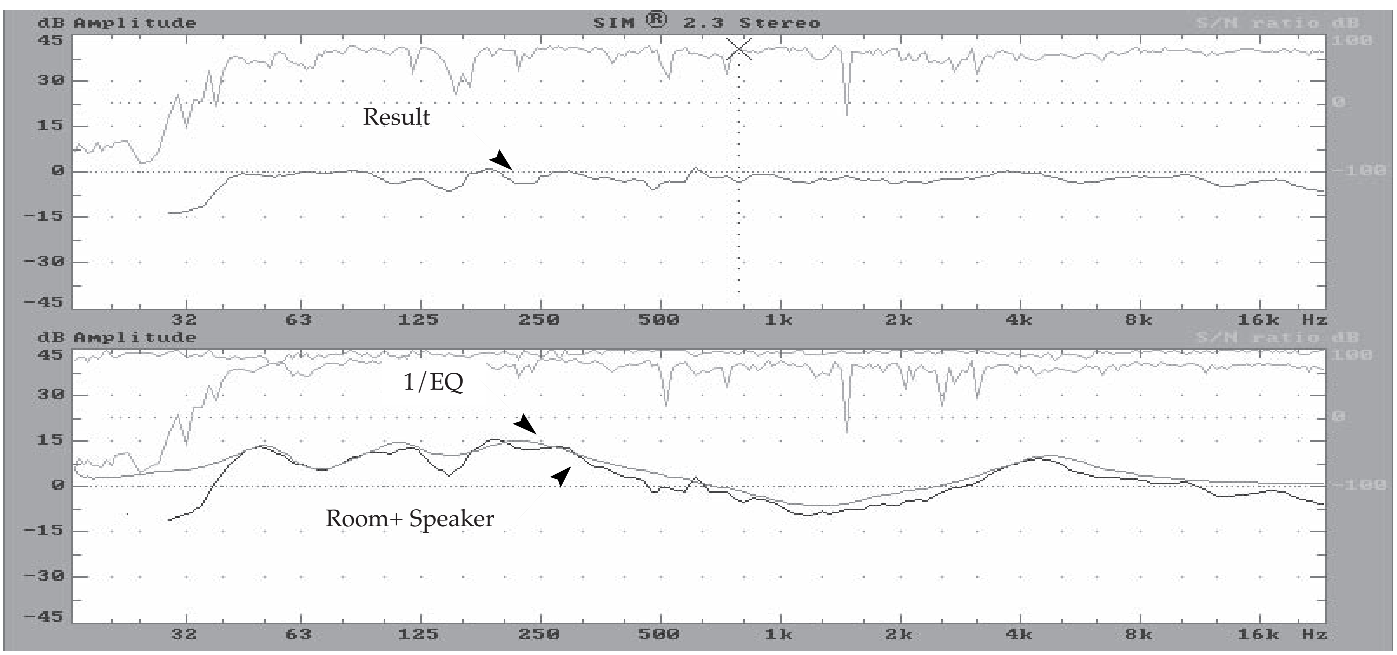


Fig 5.8c Group view of the three SIM transfer functions with the 1/EQ response. The process of creating a complementary response is aided by the visual inversion of the EQ trace. This allows you to simply match the equalizer response to the room+speaker response. This creates a flat response in the result.

5.8.3 Field Example

The data on this page shows the same sound system equalized by two different methods.

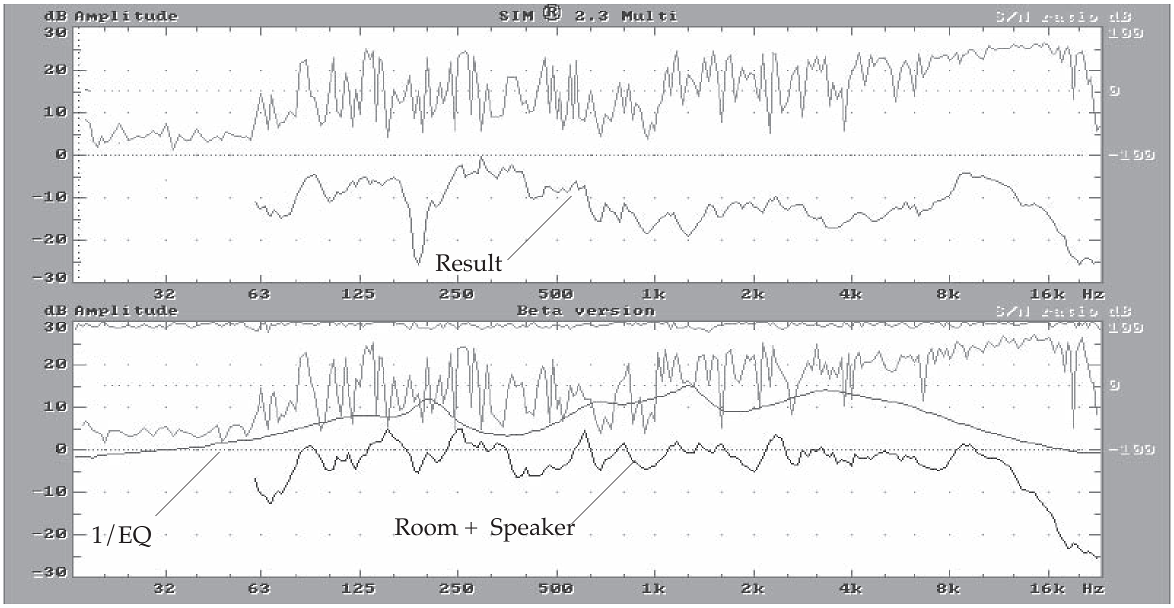


Fig 5.8d. The data here is of a poorly equalized sound system. The system was equalized primarily to suppress feedback of a podium mic. This was one of six subsystems, all of which had similarly set equalizers. The podium mic was not the only signal passing through this system. All other signals (such as the CD we listened to) sounded awful. The podium mic should have been equalized separately. When viewing the data notice that the inverse EQ does not at all correspond to the room+speaker response. The result response was worse than the original room+speaker response.

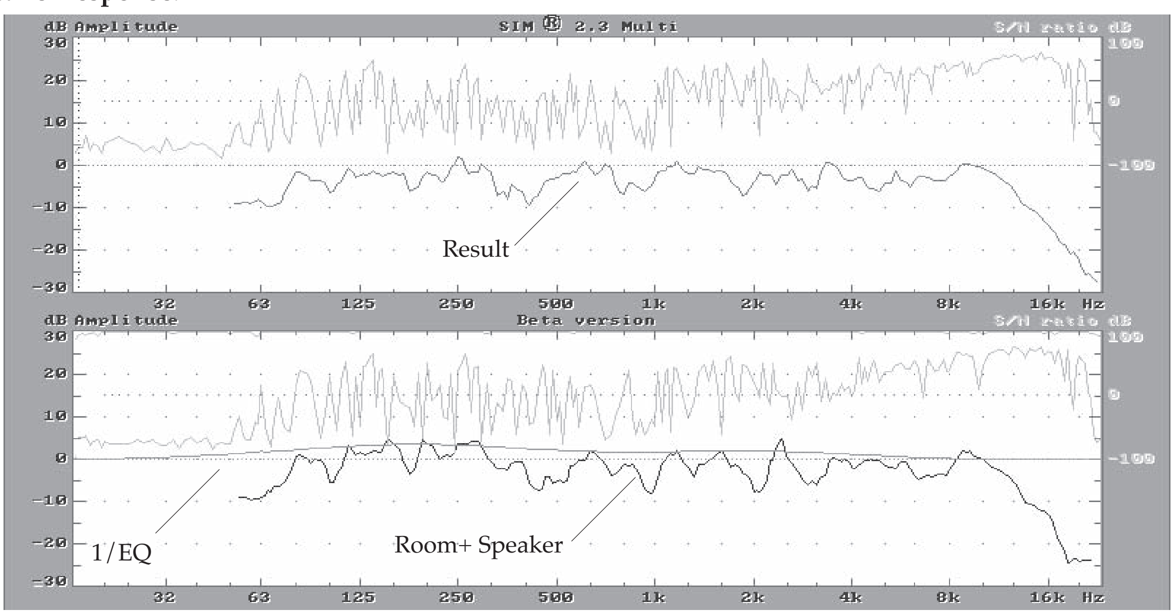


Fig 5.8e. This is the same system when equalized with SIM. The system response was greatly improved by equalizing the interaction of the speaker with the room and with each other, separate from the microphone.

5.8.4 Strategy

If you want to "EQ least," as in our previous slogan, here is how to do it: Clearly separate the equalization of the mics and speakers.

The strategy for equalization is to first deal with the speakers:

Step 1: Equalize the speaker system first. This system is to be aligned flat (or with some LF buildup as discussed on the next page). The speaker system is aligned without regard for the response of the mics. The system should sound normal for CDs or other direct input signals.

Now add the more complex acoustic input signals (mics).

Step 2: Equalize the individual mic channels (channel EQ or outboard).

Step 3: Add the stage monitors and adjust the channel EQs as required.

Step 4: Open up multiple mics and fine adjust the channel EQs.

Now both line and mic input channels will sound normal through the system.

The most common system alignment mistake is to align the speaker system response to a single performer's mic. This is most often either the kick drum or main vocal mic. The result is a system grossly out of frequency balance for all the other instruments, which must then endure radical channel equalization to get back to normal. System headroom is lost and channel EQ may not be sufficient to restore some channels.

The initial equalization of the speaker system can be performed objectively and accurately using SIM System II. The process is simply to overlay the 1/EQ trace onto the unequalized response in the room until a complement is created. It's that simple. The complexity increases when the measurement mic is moved and ambiguous readings are seen.

The key to speaker system EQ is don't overdo it. If you have been careful to use the previous techniques—such as precise speaker positioning, delay and amplitude tapering, and system subdivision—there should not be too much left for the equalizer to do.

Your goal is to set the table for the mixer so that minimal EQ is required on each of the 100 or so channels in a modern mix situation. The best tactic is to knock out the largest problems with a few carefully placed filters and let them mix!

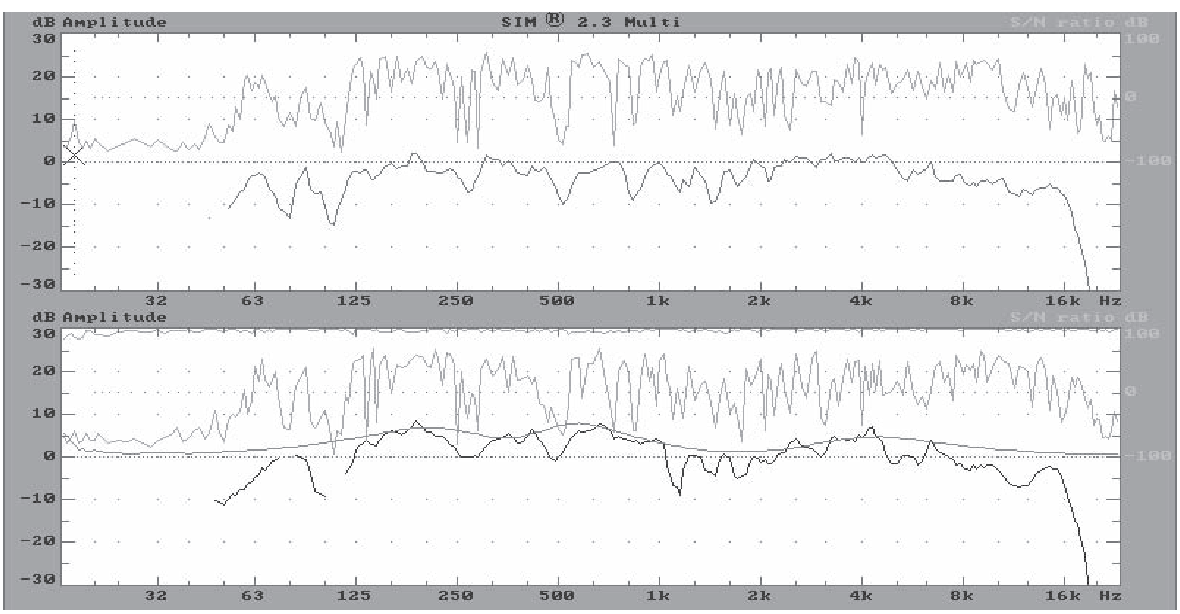


Fig 5.8f Complementary equalization example.

5.8.5 Should the System be Set Flat?

What does "flat" mean? Does it mean that the EQ is set flat, as in, "I turned the system on and it sounded great flat." Or is it "I equalized the system response and it sounded great flat." From our point of view the former is "unequalized" and the latter is "flat."

Now let's try it again: What does flat mean for a speaker system in an arena? It depends on how it's measured.

As measured by a real-time analyzer (RTA), "flat" would mean the system has no low end and brittle highs. This is due to the RTA's inability to differentiate between direct and reverberant sound, (see section 1.9). This is why most mix engineers cringe at the prospect of their system being flat.

As measured by an analyzer that attempts to simulate the speaker's anechoic response, the low end could be anything, since these systems tend to ignore the LF response. This is because these analyzers are usually run with short time records in order to see only the direct sound, removing early reflections and all of the LF coupling from the measurement.

As measured by SIM, "flat" would mean that the system sounds natural, with the room and speaker interactive effects reduced to a minimum. The SIM frequency response measurement shows the response as we hear it: an integration of the direct and early reverberant signal.

The whole flat discussion is virtually a non-issue. The frequency response can be viewed as three parts: the response below 100 Hz, the midrange, and above 8 kHz. Each section should be flat with the expected rolloff at the extremes. Many engineers like a rise in the LF below 100 Hz. Give them what they want by raising the level in that area, but keep the region as a whole flat. The VHF can be trimmed to taste with a simple shelving filter.

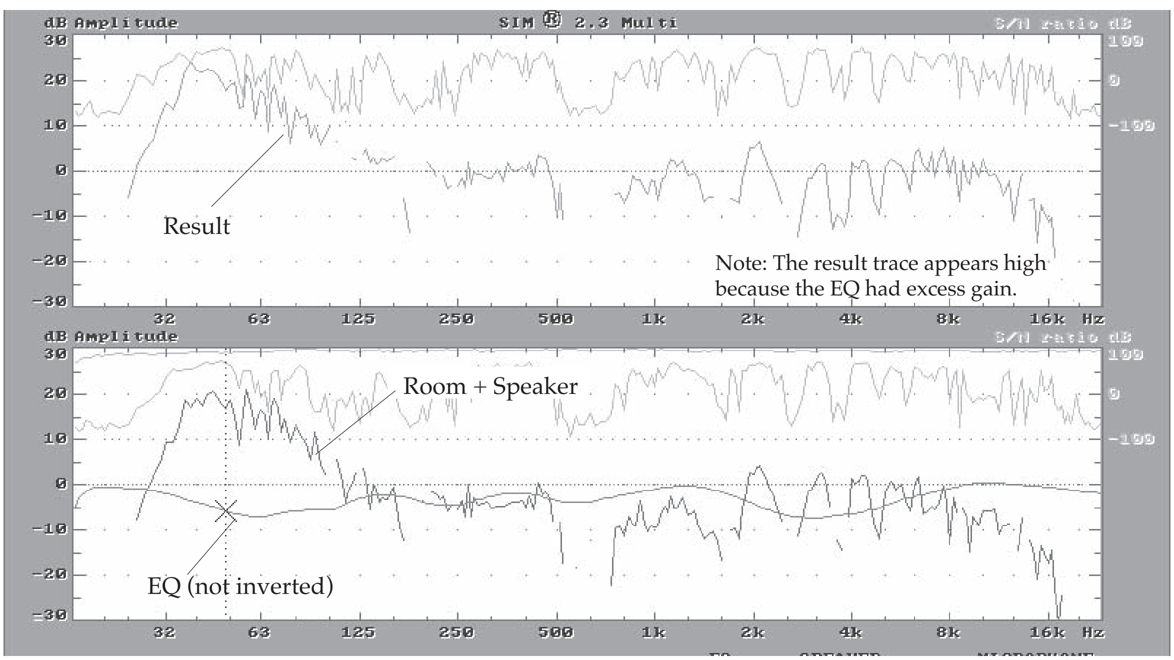


Fig 5.8g Is this system flat?

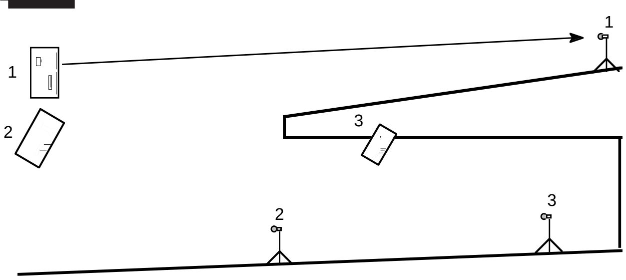
Here is a classic case of the "What is flat?" discussion. From the client's point of view the system is down 4 dB at 50 Hz. He is very concerned that I not take any more out of the low end or it will sound thin. His perception of the system's response is based on the panel settings of the graphic equalizer, which was set with a 4 dB cut at 50 Hz, independent of what his ears were telling him. From the viewpoint of the measured response, the system's 22 dB peak at 50 Hz (relative to 200 Hz) has been reduced to a mere 18 dB peak. The huge LF peak was due to setting the drive to the subwoofers 20 dB above the other speakers in this competitor-manufactured system. The 18 dB boost was left in and, I can assure you, the system *did not* sound thin.

5.9.1 Alignment Procedures: Introduction

Introduction

Because of the complexity of analyzing even a simple sound reinforcement system it is necessary to break the process into steps. These steps will serve primarily to differentiate the effects of the acoustical factors, such as interaction and reflections, and our techniques for optimization. For example, it would be reckless to begin with multiple speaker system interaction before looking first at the individual subsystems.

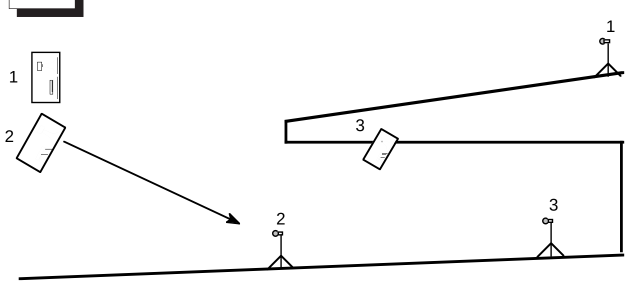
1



Single system analysis of the upper system.

The upper system is measured in its intended coverage area. Level and equalization adjustments are made as required.

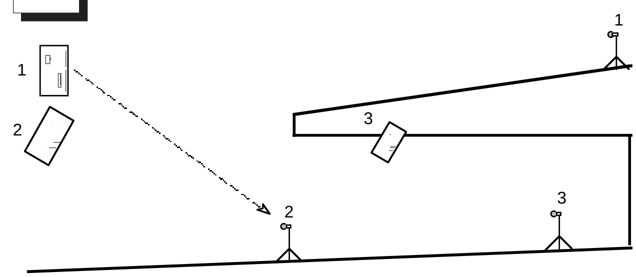
2



Single system analysis of the lower system.

The lower system is measured in its intended coverage area. Level is adjusted so that the lower system provides the same acoustic level downstairs as the upper system does in the balcony. Complementary equalization is applied as required. Although not shown, the same is done for the underbalcony delay system.

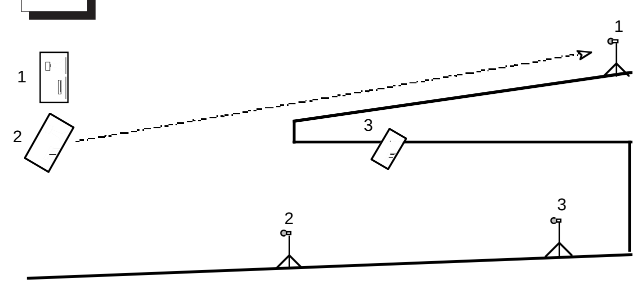
3



Lobe study analysis of the downlobe from the upper system.

The downlobe from the upper system is analyzed downstairs and compared to the single system response. Lobe study analysis will illustrate the amount of isolation and time offset between the signals from the upper and lower systems. If a delay line is available, the lower system should be delayed to align it to the upper system downlobe.

4

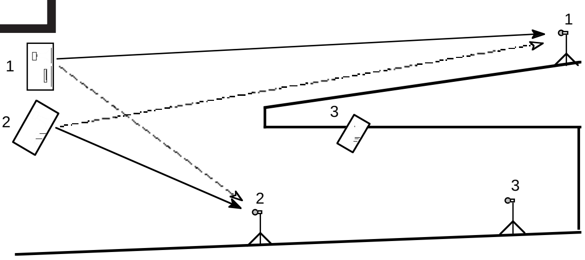


Lobe study analysis of the uplobe from the lower system.

The uplobe from the lower system is analyzed in the balcony and compared to the single system response. Lobe study analysis will illustrate the amount of isolation and the time offset between the signals from the upper and lower systems. The isolation should be much greater since the lower system has a shorter throw and therefore should be driven at a lower level.

5.9.1 Alignment Procedures: Introduction

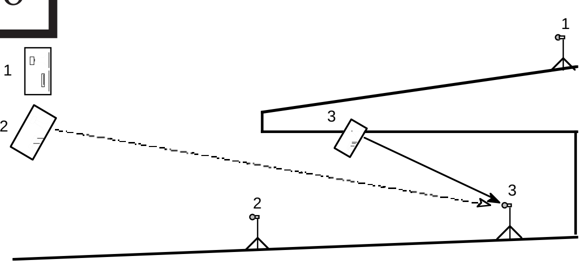
5



Combined system analysis of the upper and lower systems.

The combined response is measured at both locations and each compared to its respective single system responses. EQ and level adjustments are applied as required to restore the combined response to match the single system response.

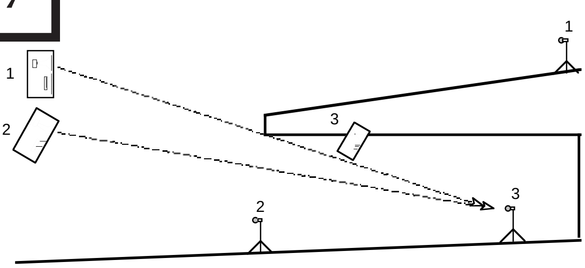
6



External delay alignment of the underbalcony.

The relative time arrivals are measured and the underbalcony delay adjusted as required.

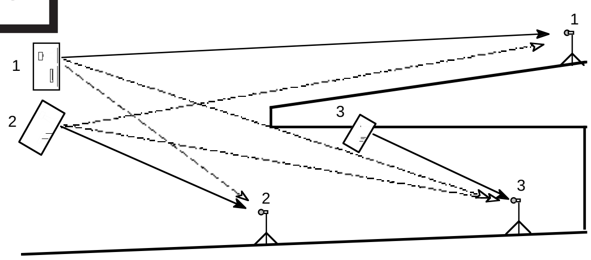
7



Lobe study analysis of the main cluster into the underbalcony area.

A lobe study is made below the balcony to determine the nature of the response of the main cluster under the balcony. This will determine the delay time and how much S/N ratio improvement can be achieved by the underbalcony speakers.

8



Combined system analysis of all systems.

The combined response can be viewed at all locations and checked for uniformity of response and S/N ratio. Level and EQ adjustments can be made until an optimum balance is achieved.

9

Analysis of the combined system with the audience present.

The final step is to prepare for the performance. If the present microphone positions are unacceptable, new ones must be found. Data is then accumulated for each position to create a complete record of the empty hall response. During the performance the live data can be compared to the empty hall data and adjusted as necessary.

The above procedures allowed us to break apart the three principle mechanisms:

- Speaker/room interaction (single system procedures).
- Speaker/speaker interaction (lobe study and combined system procedures).
- Dynamic conditions (show procedure).

5.9.1 Alignment Procedures: Introduction

The following data is a field example of the basic single system/lobe study/combined cycle. This was a case of underbalcony delays combined with a main system.

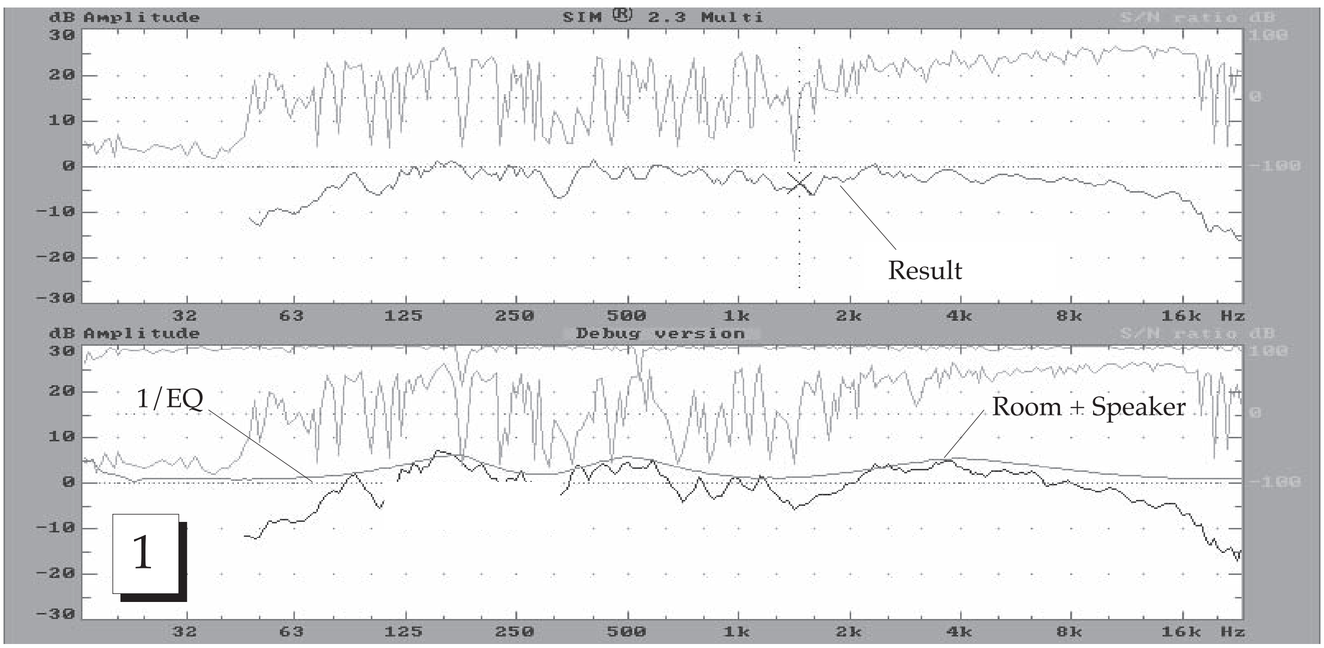


Fig 5.9a Single system analysis of the underbalcony system.

The underbalcony system is measured in its intended coverage area. Level and equalization adjustments are made as required. Notice that the S/N ratio values are generally high, indicating good intelligibility.

5.9.1 Alignment Procedures: Introduction

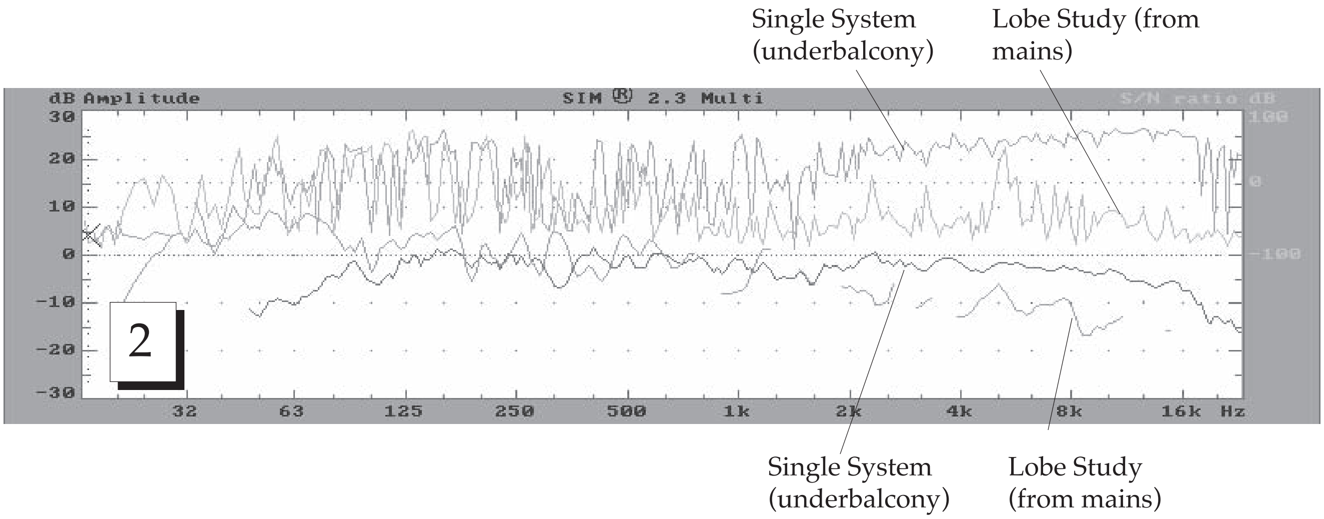


Fig 5.9b Lobe study analysis of the signal from the main system.

The energy from the main system is analyzed under the balcony and compared to the single system response. The lobe study analysis indicates that the main system has very poor S/N ratio under the balcony and very uneven frequency response compared to the underbalcony speakers in the same area. The underbalcony delay system will be capable of improving both of these parameters when combined with the mains.

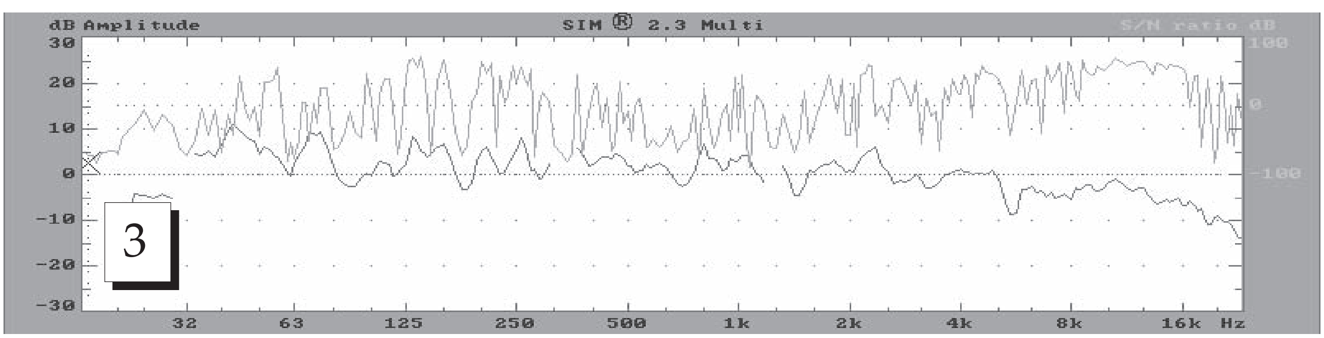


Fig 5.9c Combined system analysis of the main and delay systems.

The combined response is measured under the balcony and compared to the original single system response. EQ and level adjustments are applied as required to achieve improved S/N ratio and linearity.

5.9.1 Alignment Procedures: Introduction

The alignment procedure is best described by using a representative sample. The following sample describes the alignment of the orchestra level vocal system of a typical West End or Broadway musical theatre sound design.

The system is monaural with separate inner and outer mains, frontfills and underbalcony delays.

The "frontfill" system is a split-parallel array of small UPM-1s inset into the stage lip. These will cover the first three to four rows and help to keep the sonic image centered.

The "stalls inner" system is a point-destination infill system designed to cover the center of the orchestra area.

The "stalls outer" system is a wide split-parallel array that will cover the majority of the orchestral seating area.

The "underbalcony" system is a wide split-parallel/split-point source array that will cover the last four to six rows under the balcony.

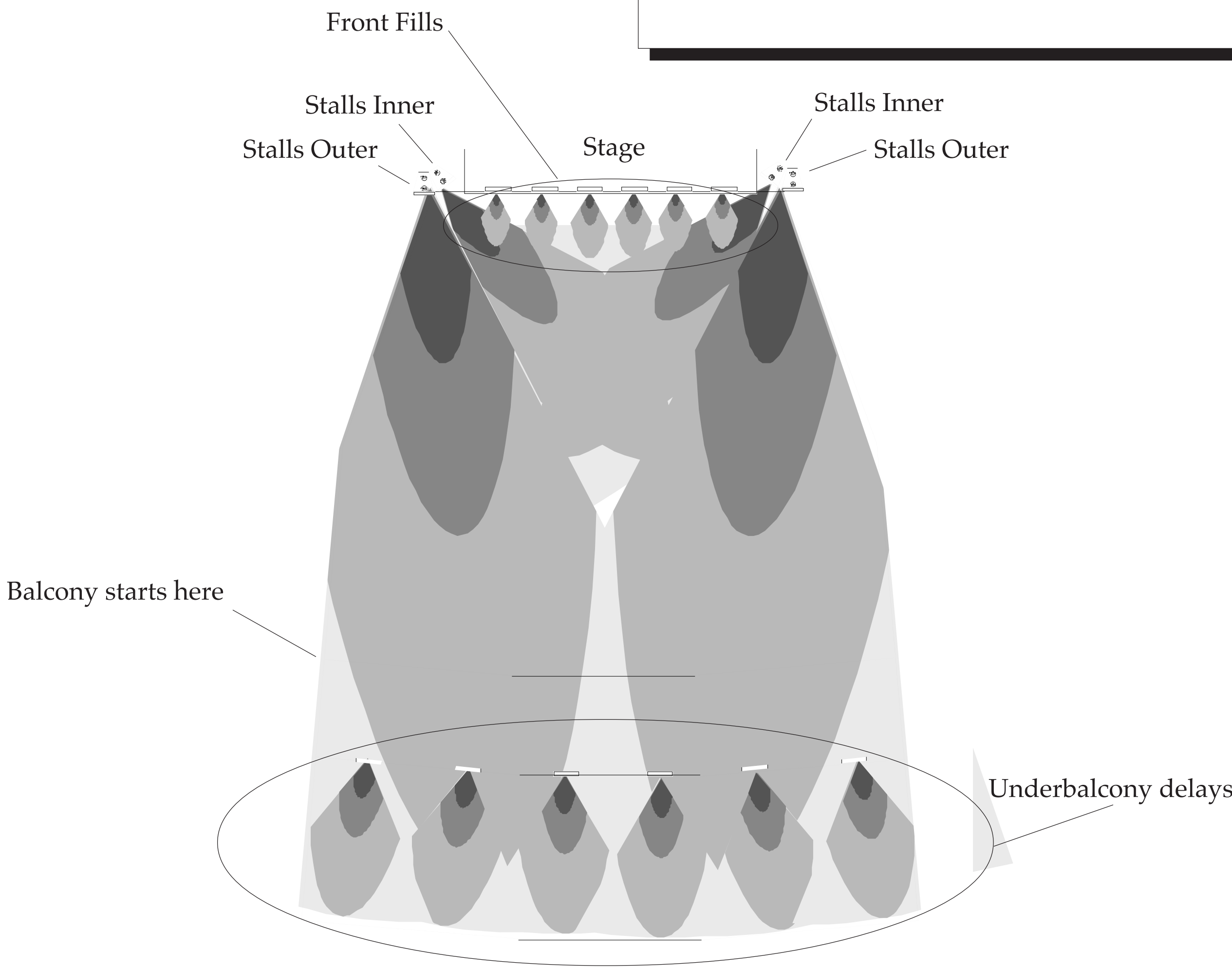


Fig 5.9d

5.9.1 Alignment Procedures: Introduction

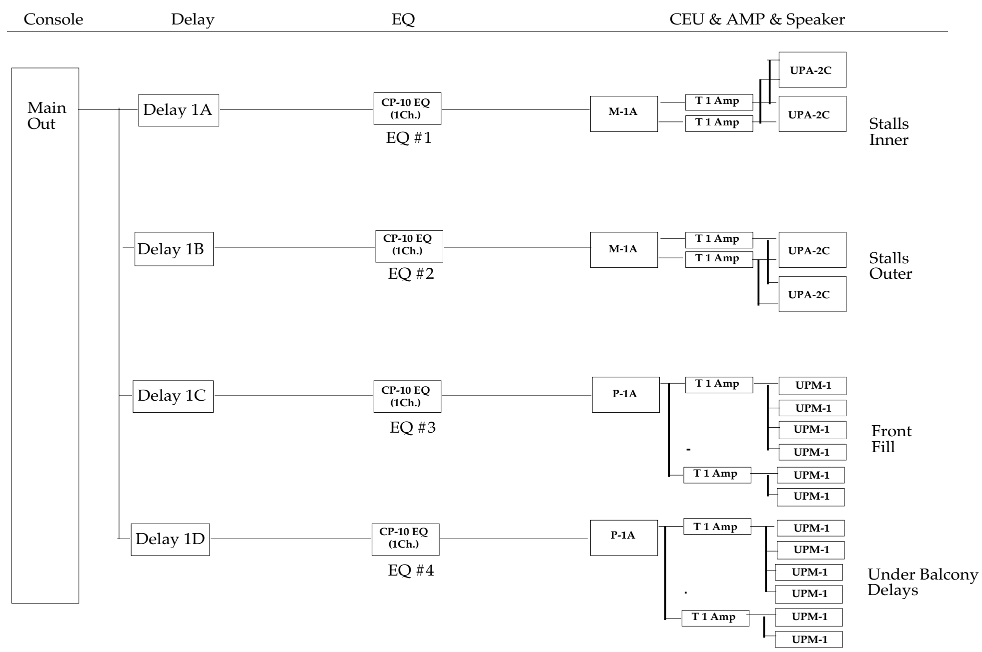


Fig 5.9e

The original flow block diagram from the mixer to the speakers is shown above in fig 5.9e. Below (see fig 5.9f) is the flow block after SIM System II has been interfaced into the system. Each of the EQ inputs and outputs are routed through the SIM 2403 interface network and returned.

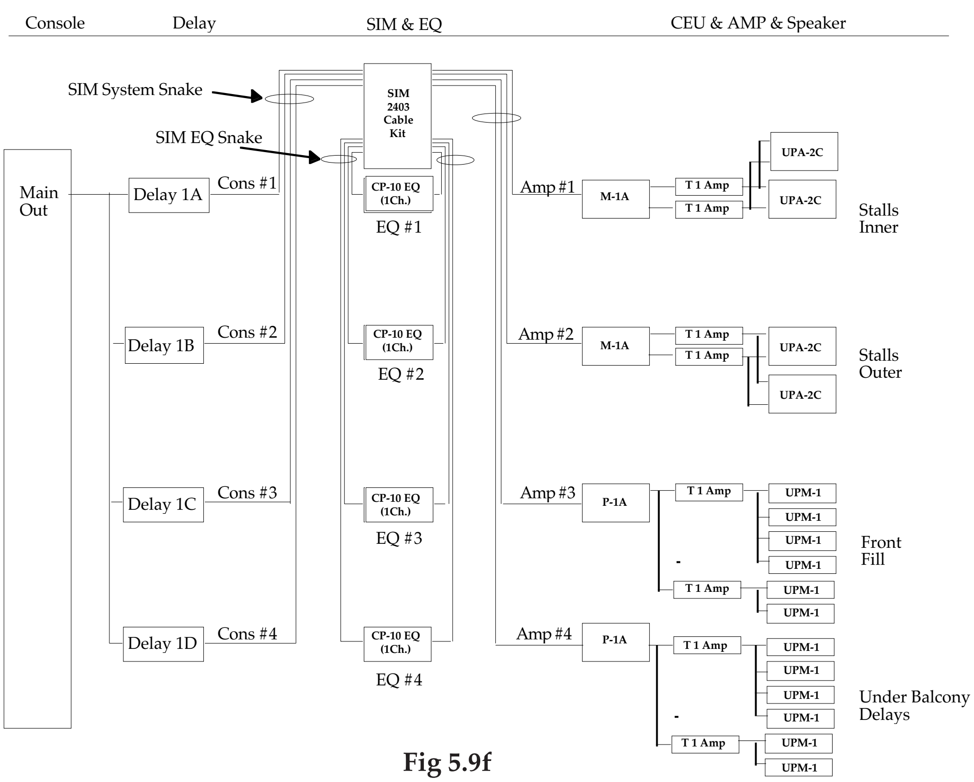
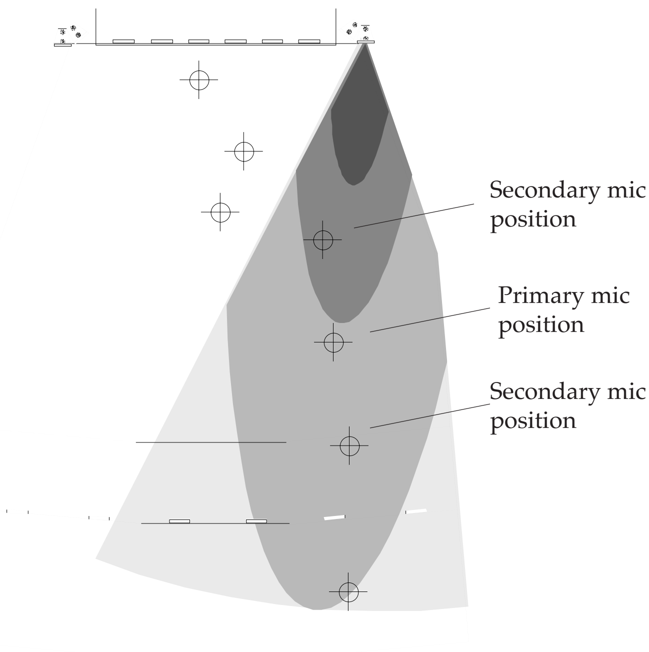
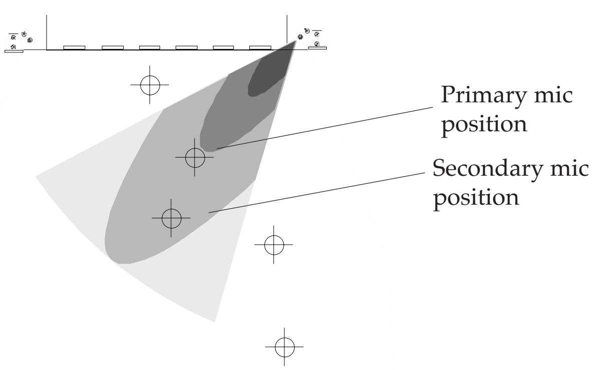


Fig 5.9f

5.9.2 Single Systems

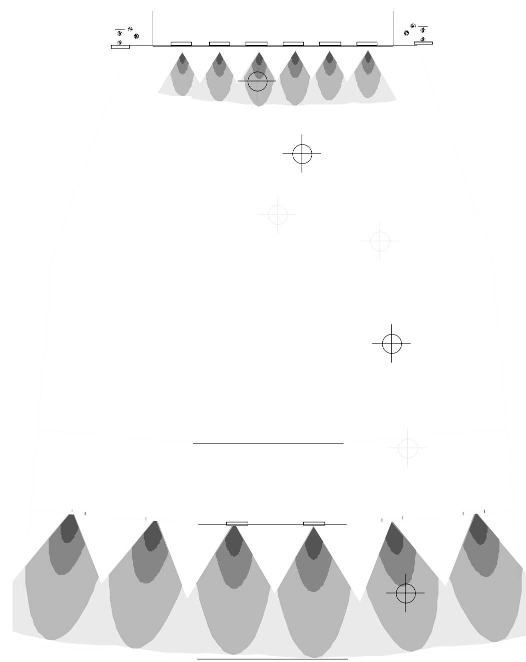


Step 1: The stalls outer system is measured alone as a single system. The primary mic is positioned at one-half the depth of coverage, on-axis. Two secondary mics are placed in alternate positions. Because the system is a split- parallel array only one of the speakers is used for the initial alignment. Coverage verification, level setting and EQ can be done at this time.



Step 2: The stalls inner system is measured similarly to the outer, above. The level of the inner system is set so that the inner and outer systems are matched at their respective primary positions. This will create equal average levels across the orchestra.

Step 3: The frontfill system is measured alone. Coverage is verified and the level and EQ are set. The primary mic is placed on axis to one of the speakers. While it can be helpful to measure with just a single speaker on (as above) it is usually not practical due to the speakers being recessed in the stage.



Step 4: The underbalcony fills area measured alone where the level and EQ are set. The same practical considerations prevail as in the frontfills, above.

Store all of the above frequency-response data. The single system alignment phase is complete.

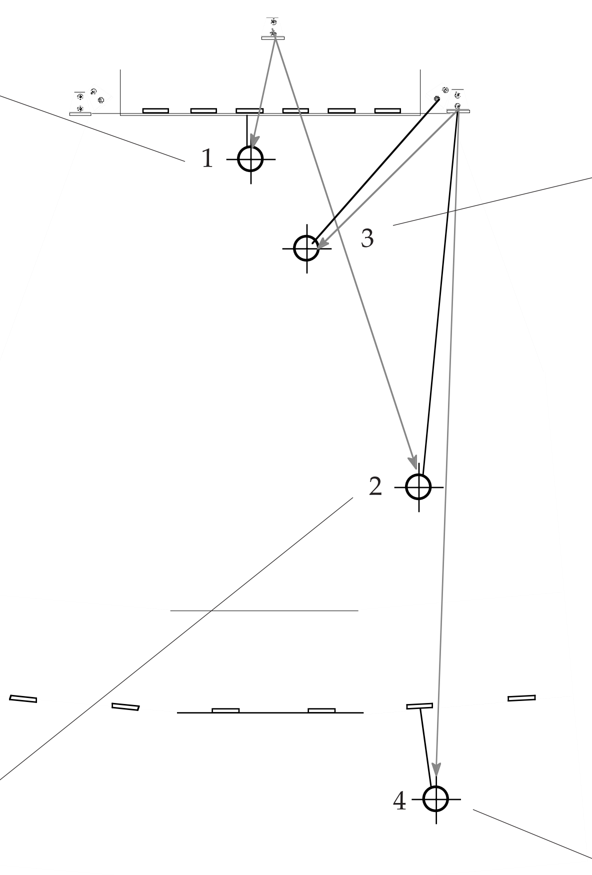
5.9.3 Setting Delays

The next step involves setting the system delays. For each of the four systems, a reference must be chosen so that the subsystems will combine optimally and the sonic image appears at the stage.

Step 1: Set frontfill delay time. Use a fictitious source as a reference to keep the image on the stage.

Step 3: Set stalls inner delay time. It is not possible to synchronize both the inner and outer systems to the fictitious source in their respective primary positions. The difference in delay times would be 10 to 20 ms. (the inner gives a much shorter time). This difference is much too great for speakers in direct proximity and would cause large-scale comb filtering. Since only one of the systems can synchronize to the fictitious source, the outer should be chosen. This is because the imaging will be best if the outer is synchronized and the inner is later than the sound from the stage.

Use the delay tapering technique to synchronize the stalls inner to lobe from the stalls outer. This will provide maximum coverage uniformity.



Step 2: Set stalls outer delay time. Use a fictitious source as a reference so that the image moves onto the stage.

Step 4: Set the underbalcony delay time. Use the stalls outer system as reference for the under balcony speakers. The actors will not project to the rear of the hall, therefore the fictitious source is not applicable.

5.9.4 Lobe Study and Combined Systems

The single system procedures provided a view of each speaker system's interaction with the room, or in the case of an array, with its own speakers as well. This is a best-case scenario for these systems with no interference from other systems. To complete the alignment we must combine this subsystem and the others with a simple clear goal in mind: To make the combined system response in each area as close as possible to its original single system response.

In other words: *Keep the damage to a minimum.*

Recall the discussion of speaker interaction in Section 2. This is the essence of combining systems. The key factors are the relative time and level offset between systems.

Combining systems will be a positive experience if:

- The subsystems are close in time where they are close in level.
- The subsystems are well isolated when they are not close in time.

Combining systems will be a negative experience if:

- The subsystems are not close in time when they are close in level.

How can we ensure the positive? The key is a study of the relative amplitude and phase responses over frequency in the interactive area.

Remember that "close in time" is measured in degrees of phase, not milliseconds. A time offset of 2 ms is very close for 50 Hz (36°) but miles away for 5 kHz (3600°). Remember also, that "close in level" is not measured on a full-range SPL meter. The relative levels between the systems changes radically over frequency due to the axial characteristics of the speakers.

In the best-case-real-world scenario, the elements of a combined system will be:

- Close in time and level in the low frequencies, which produces efficient power addition over a wide area.
- Well isolated in the high frequencies, since it is virtually impossible to keep the time offsets minimized over a wide area.

Lobe Study Analysis

We can make informed decisions in this regard if we can isolate each system's contribution into the interactive area. We already have the single system response which gives us the response of the speaker in its intended coverage area. The response of the neighboring system into this area is termed in SIM jargon a "lobe study," since we are examining what should be the off-axis area of the neighboring speaker. The lobe study data can be compared to the single system data and analyzed over frequency. The relative time (phase) and level offset (amplitude) and their relative S/N ratio are then revealed.

Lobe study analysis can provide interesting (and sometimes surprising) information.

• The degree of isolation between systems is often much less than expected. It can actually be a negative number, especially at low frequencies. In other words, the interference from neighboring speakers can may be stronger than the signal from the system intended to cover in that area.

• If there is insufficient isolation between the systems it may be necessary to reevaluate speaker position, delay or relative level.

• Comparing the amplitude and phase responses of the single system and lobe study data provides a preview of the frequencies where additions and cancellations will occur and their extent.

The greater the time offset the more difference there will be in the phase response. The frequency range of high interaction extends lower and lower as offset increases. Areas where the phase responses converge will have maximum addition, while those areas that are 180° apart will exhibit maximum cancellation.

The extent (size of the peaks and dips) of the interaction will be heavy in areas with less than 3 dB of isolation. The interaction will be light in areas with greater than 6 dB of isolation and negligible, in terms of alignment decisions, if greater than 10 dB.

• The S/N ratio reading provides further key information. The lobe study data might reveal, for example, that under the balcony there is comparable level from the mains (usually too much in the low end) but poor S/N ratio, compared to that of the single system data from the nearby underbalcony speakers. This is not surprising, since improved S/N ratio is the principal reason for installing underbalcony speakers.

5.9.4 Lobe Study and Combined Systems

A comparison of the S/N ratio of the single system and lobe study data will preview the extent to which your underbalcony speakers will improve (or degrade) the response. Most underbalcony speaker designs are "split-parallel" or "split point-source" array types, (see Section 2.2). Such arrays are effective when the overlaps are minimized. However, if the depth of coverage is too deep, they become highly interactive. In such cases the S/N ratio of the lobe study (usually from a point-source array) may be much higher than that of the highly interactive underbalcony speakers. In other words: The speaker-to-speaker interaction of the underbalcony speakers may be more destructive than the speaker to room interaction of the distant main system. In such cases, adding the underbalcony speakers can only make things worse and, therefore, must be repositioned deeper into the room or turned off. This actually does occur and it can make for a *very interesting* political situation.

Low-frequency leakage from a neighboring system can dominate the response in the local area. It is important to know the source of this low-frequency energy since attempts to remove it by turning the knobs of the local system's equalizer would be futile and potentially destructive. The actual solutions for this leakage are actions taken with the neighboring system's position, level, equalization or architecture. It is often better to know what you have and live with it, rather than making the situation worse with ineffective changes to your system.

Lobe study analysis sets the stage for the next step: combining the systems and allowing you to know what is about to happen.

There is an old saying: "If it's not broke, don't fix it."You can add to that one: "If you can't fix it, don't break it."

Lobe Study analysis tells you what you *can* and *cannot* fix.

5.9.5 Combining Systems

The data we have previously obtained gives us a very good impression of what will happen when the systems are added together. The combined system response is a summation of the single system and lobe study responses. The response has three basic variations:

1. Areas where the local system is well isolated from the neighboring system will have minimal changes from the *single system* response. This can be verified by comparing the combined systems data to the stored single system data for each system. This is the best-case scenario and is very achievable in the midrange and above where speaker directional control is at is best. Since this was the original goal, your work is done.

2. Areas where the local system is dominated by the neighboring system will have minimal changes from the *lobe study* response. This can be verified by comparing the combined systems data to the stored lobe study data for each system. This is typical in the low-mids and below for fill systems (side, down, front, underbalcony, etc.) that supplement a large main system. The solutions for this type of combination are relatively few. Presumably the LF response for the main system is set for the desired response in its local area. If the LF in the mains is reduced to accommodate the fill systems it will upset the spectral balance out front and will not help the fills anyway. Why? Because the problem here is the difference between the LF response in the main area and the fill. Reducing it in the mains will not change this. The real solution is to find a way to make the LF response of the mains more directional.

The most often touted solution is to reduce the LF response in the fill system with a low-cut filter. If the LF response fill system is already more than 6 dB below the mains, you can reduce it another 20 dB and see almost no effect on the combined response. The only real effect this practice has is to reduce the system's combined maximum power capability and to create the unnatural experience of being able to localize reverberant displaced low-end separated from the mids and highs. To experience this yourself visit your local arena with its 1950s-era (but still installed) flying junk yard of distributed horns.

Another idea would be to raise the level of the fill system so that its mid-and high-frequency areas rise up to the leakage from the mains. This will indeed linearize the response, but the side effects include excess level in the fill area and the possibility of leakage back in to the mains area. Which do you like better, the LF range too loud or everything too loud? Keep things in perspective. If improvements in the fill areas damage the main seating area, it is not worth the risk. Damage assessment to the mains can be checked by comparing the main's single system response to its combined system response.

If the fill system is dominated by the mains and the mains cannot be changed, it is usually best to leave it alone and move on to matters that can be fixed.

3. Areas where the systems are close in level will take on a new frequency response not found in either the single system or the lobe study alone. The response is the complex product of the amplitude and phase summation of the two systems. It will couple and comb depending on the time and level offsets between the system (see Section 2 for a complete discussion of this). This is where the complex decisions are made.

The solution tools are still the same: architectural, position, delay, level, and EQ.

Let's assume that the architectural and position options are already exhausted.

Delay Tapering

The phase relationship between the systems can be fine-tuned by the delay tapering technique (described in Section 5.7.3), which uses small amounts of delay to minimize speaker interaction. Delay tapering is particularly effective in situations such as downfills and sidefills where the audience is slightly closer to the fills but within the leakage area of the mains.

Level Tapering

In the single system procedure we adjusted the speaker system levels to achieve maximum uniformity of sound level at the primary mic positions. This level set the drive level necessary to achieve maximum uniformity *if the systems have no interaction.* This starting level is the highest level. When we add the subsystems together we should expect only to turn some of the systems down, due to interaction. Remember that if it doesn't get louder when we add systems together, you had better check polarity. The addition of speakers together may allow us to reduce the drive level of a fill system due to the overlapping energy from the mains. Such a level reduction of the fill system

5.9.5 Combining Systems

can be an effective means of reducing interaction while maintaining a consistent combined level. Bear in mind that a 3 dB reduction in the fill system level will not reduce the combined response by the full 3 dB because it only reduces one of the systems coupled into the listening area. Further reductions of the fill system will have even less effect, and the combined response will begin to resemble the lobe study more than the desired single system. If the systems were set for equal level contours to begin with, level tapering is effective to about –4 dB after which the S/N ratio and HF response of the combined response will likely degrade.

Equalization

Equalization can be most effective in cases where a combined response change manifests itself in both systems simultaneously. This typically occurs when similar systems are throwing comparable distances, such as proscenium upper and lower systems throwing to orchestra rear and mezzanine seating areas. This is easy to spot. Check both combined responses against their respective single system responses. If they both have the same change, select a filter on both systems that will bring it down. The CP-10 Lo-Cut function has proven very effective for this.

When systems are close in level it becomes very important to check for changes in the combined response of both systems. A fix for one area has to be checked to ensure that it will not damage the other.

If a peak shows up in one of the combined responses but not the other, caution must be exercised. Equalization will have only minimal effect since the equalizer only affects one of the sources of the interaction. Equalization of the combined response is similar to the level tapering described above, except that it refers to tapering the level in a specific frequency range. It also is effective to around –4 dB from its original single system setting.

Additional Combinations

The two systems can now be considered as one. Each additional speaker system is added together until a composite response of all the speaker systems is formed. Each combination goes through the cycle of lobe study and combined system measurements and adjustments. As each system is added, its effect is charted against the previous set of measurements, allowing us to single out the effect of each additional speaker system.

The order in which systems are combined requires careful consideration.

• Start with the most dominant speaker system. This is the system that has the longest throw and therefore will run at the highest sensitivity.

• If there are multiple systems with similar sensitivities, choose the one that most closely resembles the point where you would prefer to place the sonic image. This typically means the stage.

• Once the "main" speaker system is chosen, the speaker that will have the most interaction with the mains should be combined with it. This usually refers to the speakers that are closest in proximity.

•These are now joined together as a system and should always be driven together for all subsequent additions.

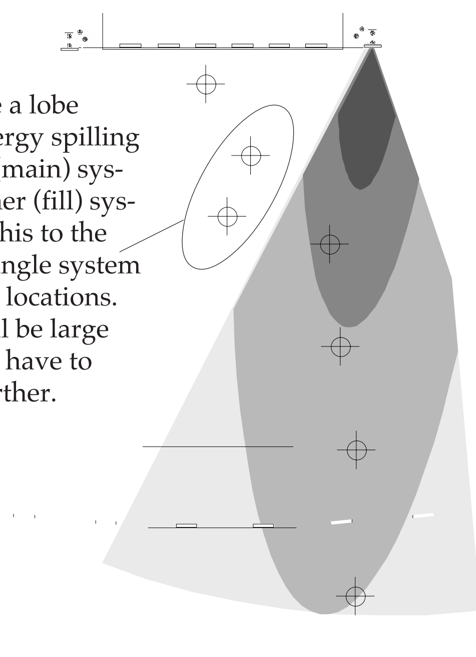
• If there are other sets of speakers that are closely paired it may be best to combine them as a pair before adding them to the main set, for example, a system that contains inner and outer systems on the floor and in the balcony. Each in/out pairs should each be combined before the upper and lower systems are combined.

Important Note: The combined system procedure refers to systems that are driven with the same input signal. Such systems exhibit position-fixed interaction patterns that can be accurately measured, and to some extent, controlled. Systems that contain different drive signals will exhibit a constantly changing interaction due to the uncorrelated spectra being sent by the systems. Effects speakers are an extreme case of this and should not be considered part of a combined system with, for example, a vocal system. Stereo is a special case since it typically involves material that is semi-correlated due to items that are panned to the center. The primary concern in combining stereo systems is verification of polarity—the systems must add, not cancel. Attempts to equalize for the combined effects will aid the center signals to the detriment of the discrete signals.

5.9.5 Combining Systems

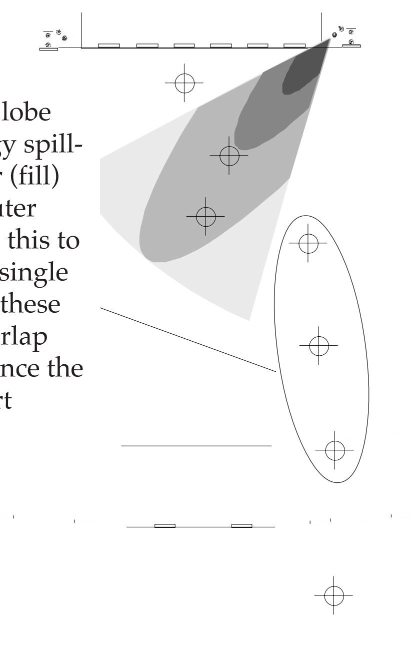
1

Step 1: Measure a lobe study of the energy spilling from the outer (main) system into the inner (fill) system. Compare this to the inner's stored single system data from these locations. The overlap will be large since the outers have to throw much further.



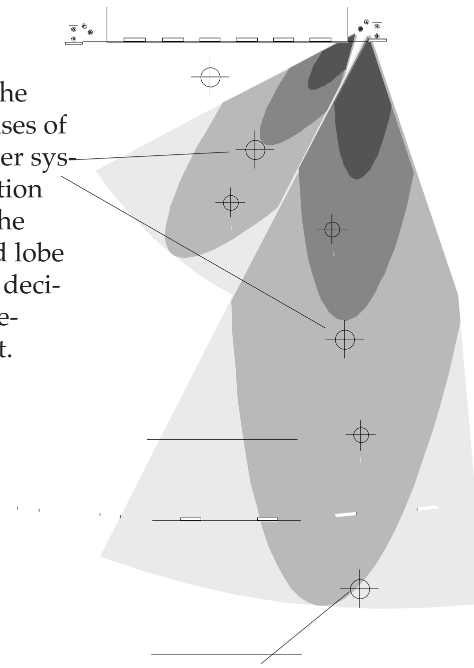
2

Step 2: Measure a lobe study of the energy spilling from the inner (fill) system into the outer (mains). Compare this to the outer's stored single system data from these locations. The overlap should be small since the inners have a short throw.



3

Step 3: Measure the combined responses of the inner and outer systems at each location and compare to the single system and lobe study data to aid decision making as described in the text.

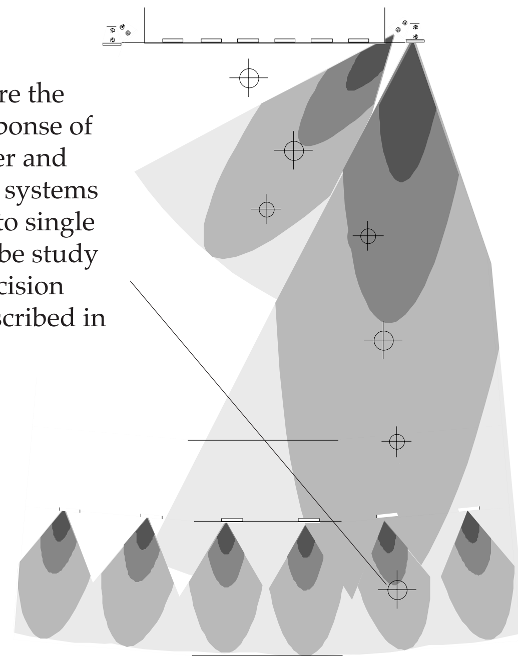


4

Step 4: Measure a lobe study of the energy spilling from these systems under the balcony. Compare this to the underbalcony system's stored single system data from these locations. Monitor closely the signal-to-noise ratio under the balcony and optimize.

5

Step 5: Measure the combined response of the inner, outer and underbalcony systems and compare to single system and lobe study data to aid decision making as described in the text.



5.10.1 Example System Alignment: Introduction

The following chapter illustrates a case study of an alignment performed in 1994. This was chosen as a representative system of medium complexity, and an alignment where clear examples of the process abound. The venue was a church that seats approximately 3,000 people. The system consists of ten UPA-1As divided into four subsystems.

The four subsystems:

Main Center:	4 UPA-1As
Main Side:	2 UPA-1As (1 per side)
Delay Center:	2 UPA-1As
Delay Side:	2 UPA-1As (1 per side)

These are shown in the plan view below (Fig 5.10a).

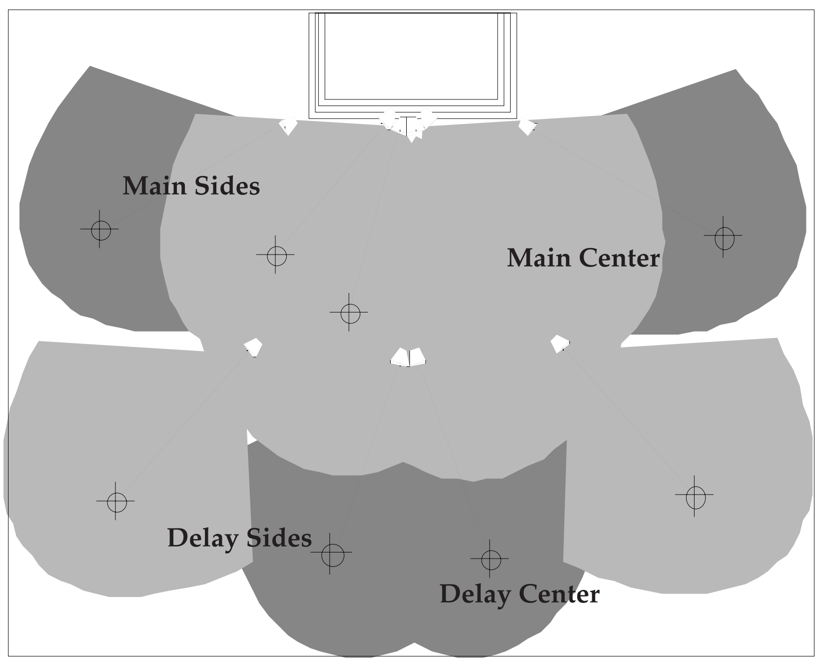
The system had been installed several years earlier. In 1994 it was felt that improvements would need to be made, but it was unclear whether the system needed to be (1) redesigned, (2) realigned or (3) replaced.

Option 1, redesigning the system, was explored in two phases:

1. Repatching of the system equalizers (the subsystem EQs were in series with that of the main center system).

2. The addition of a delay line to move the sonic image of the main system back to the podium area.

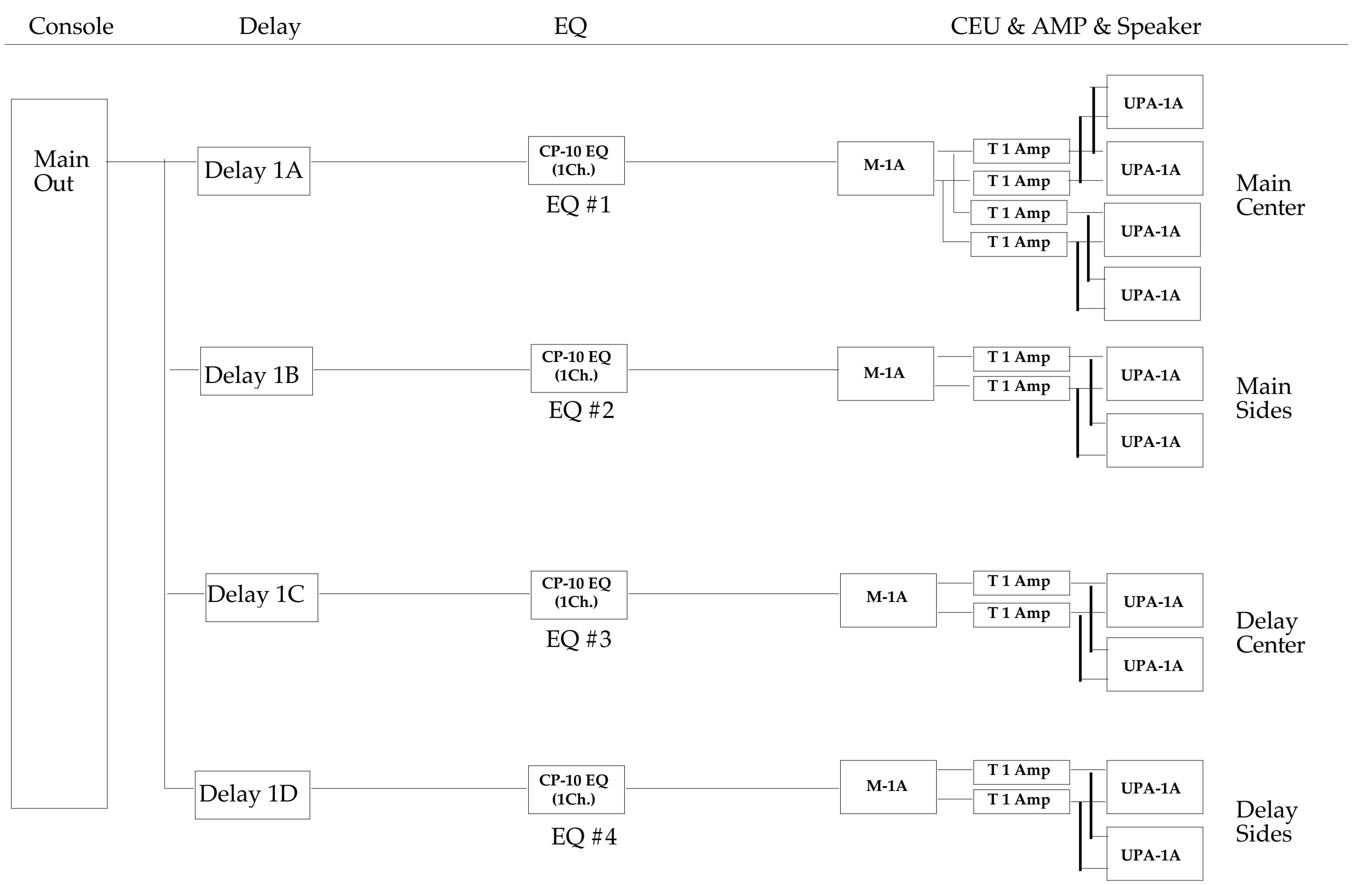
Option 2, realigning the system, was implemented by measuring and aligning the system with SIM System II, which put option 3 to rest.



5.10a Plan view of the hall showing the intended speaker coverage zones.

5.10.1 Example System Alignment: Introduction

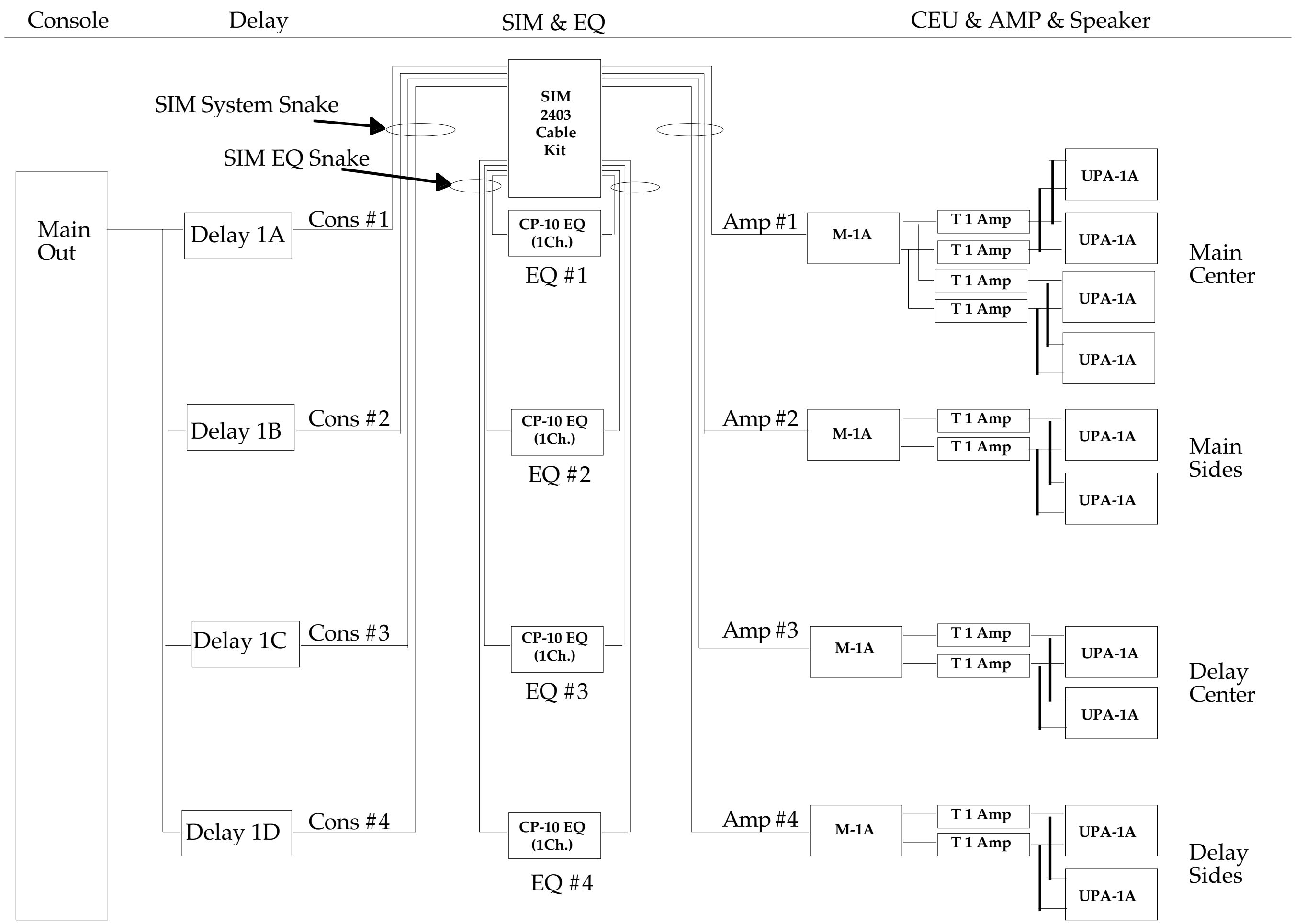
The system signal flow is shown in Fig 5.10b. Each of the subsystems has independent EQ and delay lines.



5.10b Flow block diagram of the system from mix console to the speakers.

5.10.1 Example System Alignment: Introduction

SIM System II was interfaced as shown in Fig 5.10c. Each of the four channels is routed through the SIM 2403 interface network and returned.



5.10c Flow block diagram of the system after SIM System II is patched in.

5.10.2 Setup

There are three setup panels that can be edited. Before the alignment process begins the operator will configure these panels to conform to the actual system patch (or patch the system to conform to the software panel). The setup can then be saved and recalled later.

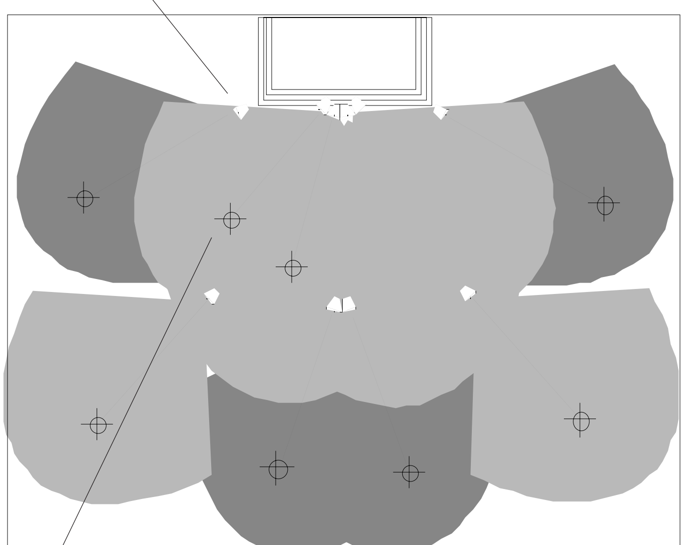
Setup Speakers

The setup speakers panel is used to show the location and channel of each equalizer/speaker subsystem. Each equalizer channel routed through the SIM system will be represented here. The operator is free to edit the names as desired.

Switcher address: 0.......
Label Speakers:

Eq ch	NAME	Type	Location
1			
2	Front	4 x UPA-1A	Center cluster
3	Front Side	1 x UPA-1A / side	at stage ends
4	Mid Delay	2 X UPA-1A	center of house
5	Side Delay	1 x UPA-1a /side	in house at sides
6			
7			
8			

Use PgUp/PgDn to select address. Press <Ctrl+Enter> when done.



Setup Microphones

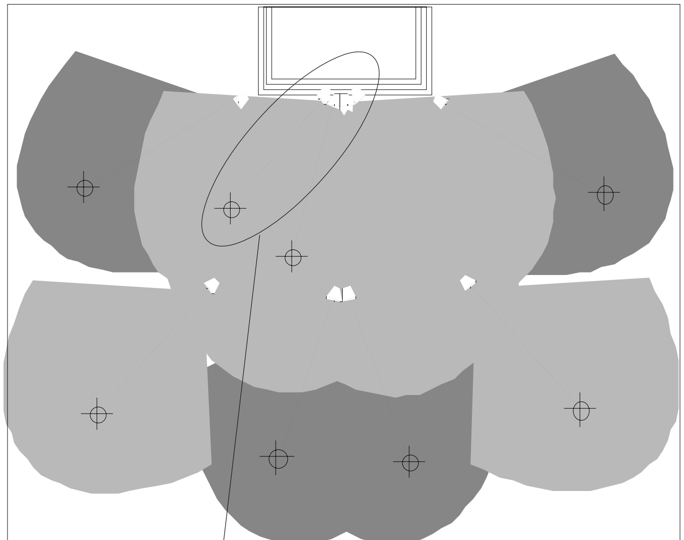
The setup microphones panel is used to record the roles and positions of the measurement microphones. In this case, eight mics were used, two for each of the four speaker system types. The operator is free to edit the names as desired.

Label Microphones:
Switcher address: 0.......

Mic ch	NAME	Type	Location
1	Front A	B&K 4007	Mix position
2	Front B		off center of cluster
3	FR Side L		on axis
4	FR Side R		on axis
5	Delay A		On axis
6	Delay B		secondary pos
7	Del Side A		on axis
8	Del Side B		secondary pos

Use PgUp/PgDn to select address. Press <Ctrl+Enter> when done.

5.10.2 Setup



Setup Branches

The setup branches panel is constructed by building "Branches"—combinations of speakers and measurement mics. The channels used in the mic and speaker setup panels can be combined as required and the branches named accordingly. The setup branches panel also maintains the delay time for each branch and other parameters, such as level offset.

B R A N C H   S E T U P   P A N E L

	BRANCH	EQ adr	EQ chn	SPEAKER Name	MICROPHONE adr	MICROPHONE chn	MICROPHONE Name	propagation delay ms	trace offset dB	compare branch
1	Main_A	0	2	Front	0	1	Front A	46.86	0.00	7
2	Main_B	0	2	Front	0	2	Front B	39.24	-2.00	6
3		0	0		0	0		0.00	0.00	3
4		0	0		0	0		0.00	0.00	4
5	MN_SideA	0	3	Front Side	0	3	FR Side L	38.96	-1.00	5
6	MN_SideB	0	3	Front Side	0	4	FR Side R	40.42	-2.00	5
7	Delay_A	0	4	Mid Delay	0	5	Delay A	47.18	3.00	7
8	Delay_B	0	4	Mid Delay	0	6	Delay B	47.42	3.00	8
9	DL_SideA	0	5	Side Delay	0	7	Del Side A	38.76	5.00	9
10	DL_SideB	0	5	Side Delay	0	8	Del Side B	41.30	5.00	10
11		0	7		0	0		0.00	0.00	11
12		0	8		0	0		0.00	0.00	12
13		0	8		0	0		0.00	0.00	13
14		0	8		0	0		0.00	0.00	14
15		0	7		0	0		0.00	0.00	15
16		0	8		0	0		0.00	0.00	16

Tab: next field. Shift+Tab: prev field. Press <Ctrl+Enter> when done...

Setup branches panel is where measurement branches are created from equalizers, speakers and measurement mics.

5.10.2 Setup

Logging of the original EQ settings.

The single system EQ settings using SIM.

Checking for isolation between systems.

Combining the systems.

Group / Branch	1	2	3	4	5	6	7
1 Main_A	SingleSys orig	SingleSys 1st pass			Comb Sys with sides		
2 Main_B	SingleSys orig	SingleSys 1st pass					
3							
4							
5 MN_SideA	SingleSys orig	SingleSys first pass	Lobe mains	Comb Sys 7 ms offse	Comb Sys sync to ma		
6 MN_SideB	SingleSys orig	SingleSys 1st pass					
7 Delay_A	SingleSys orig	SingleSys 1st pass		Lobe mains+side	Comb Sys +mains+sid		
8 Delay_B		SingleSys					
9 DL_SideA	SingleSys orig	SingleSys 1st pass	Lobe mains	Lobe 234 del cn	Comb Sys 2345		
10 DL_SideB	SingleSys orig						
11							
12							
13							
14							
15							
16							

Press a key to continue...

Fig 5.10d Data panel.

The data panel is a spreadsheet that shows the location of each of the data files. Each branch has eight memory blocks, each of which hold nine traces: amplitude, phase and signal-to-noise ratio of the room, EQ and result. The labeling of the data panel makes it easy to follow the alignment procedure. The measurement types are shown as well as excerpts from the note pad. In this example you can see the following:

Memory group 1 holds the original EQ curves found in the system from the previous alignment.

Memory group 2 holds the initial equalization using SIM while measuring the individual responses of the speakers *before* combination.

Memory group 3 holds "lobe study" information. This is the leakage of the main cluster into the coverage area of the main sides (horizontal leakage) and the center delays (vertical leakage).

Memory group 4 has an initial combination and additional lobe studies.

Memory group 5 shows the final combined response of the speakers.

5.10.3 Equalizing the Main Cluster

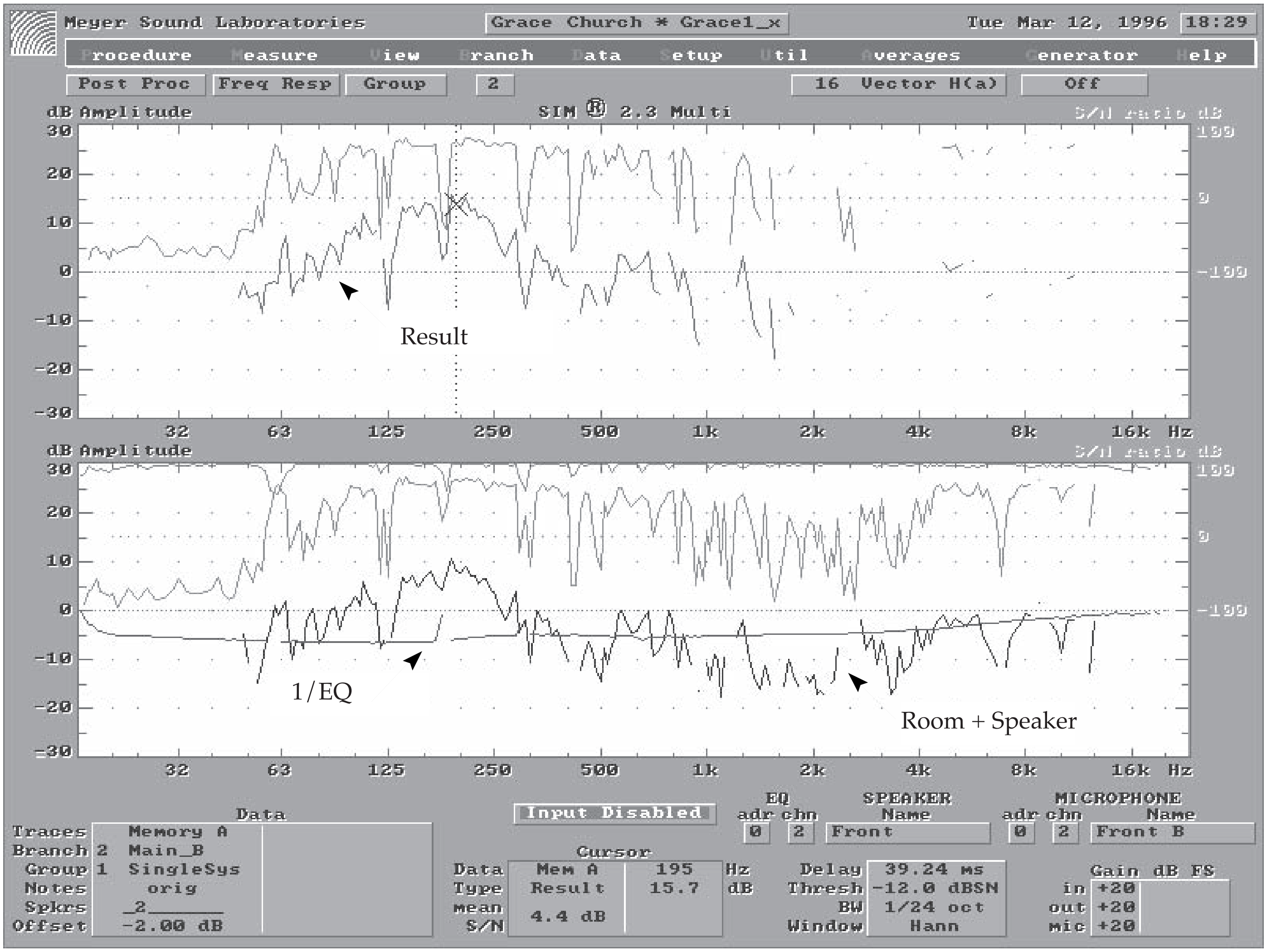
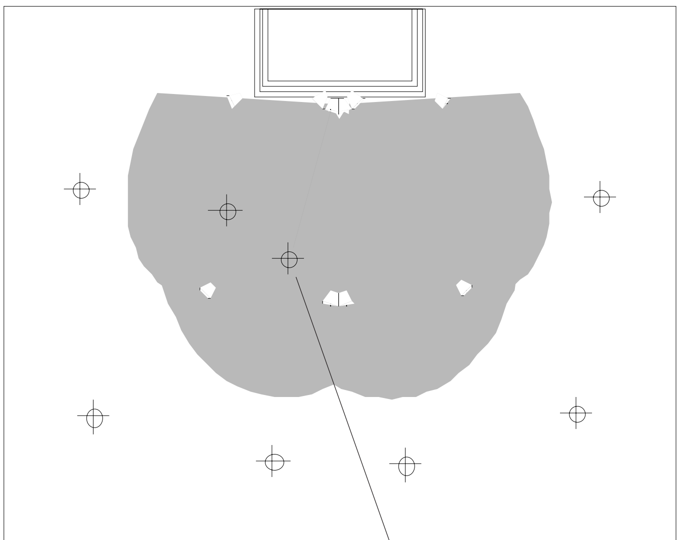


Fig 5.10e Original equalization of the main cluster.

The original response of the equalizer is shown above, overlaid against the unequalized response of the Room+Speaker. The displayed EQ trace is actually 1/EQ (inverted) as typically displayed on SIM System II. The closer this 1/EQ trace resembles the room + speaker trace the more linear the response will be. The result trace is shown in the upper screen, where the effectiveness of the equalization can be seen. Notice, however, that the actual shape created by the EQ is virtually flat and bears no resemblance to the shape of the response of the speaker in the room. The 10 dB peak from 80 Hz to 200 Hz is not even touched and is in fact actually worsened. (Remember that the eq is inverted.) The cause of this peak was the coupling of four front-loaded low-frequency drivers in a point-source array, and to some extent, the room interaction.

Note: The blank area in the HF Frequency response occurred because the data was stored a little too quickly. The HF samples had not fully accumulated. It is not an indication of any problem with the system and should be disregarded in this evaluation.

5.10.3 Equalizing the Main Cluster



Plan view of the hall indicating the mic position and speaker status during the measurements shown opposite.

	Branch	Group 1	Group 2	Group 3	Group 4	Group 5
1	**Main A**	Single System	Single System			Combined Sys
		Original EQ	1st Pass			With Mn Sides
2	**Main B**	Single System	Single System			
		Original EQ	1st Pass			
3						
4						
5	**Main Side A**	Single System	Single System	Lobe Study	Combined Sys	Combined Sys
		Original EQ	1st Pass	From Mn Center	7 ms offset	synched to Mains
6	**Main Side B**	Single System	Single System			
		Original EQ	1st Pass			
7	**Delay Cent A**	Single System	Single System		Lobe Study	Combined Sys
		Original EQ	1st Pass		From all Mains	With all Mains
8	**Delay Cent B**		Single System			
			1st Pass			
9	**Delay Side A**	Single System	Single System	Lobe Study	Lobe Study	Combined Sys
		Original EQ	1st Pass	From Mn Center		All Speakers
10	**Delay Side B**	Single System				
		Original EQ				

Data panel indicating the source of the traces shown on the opposite page.

5.10.3 Equalizing the Main Cluster

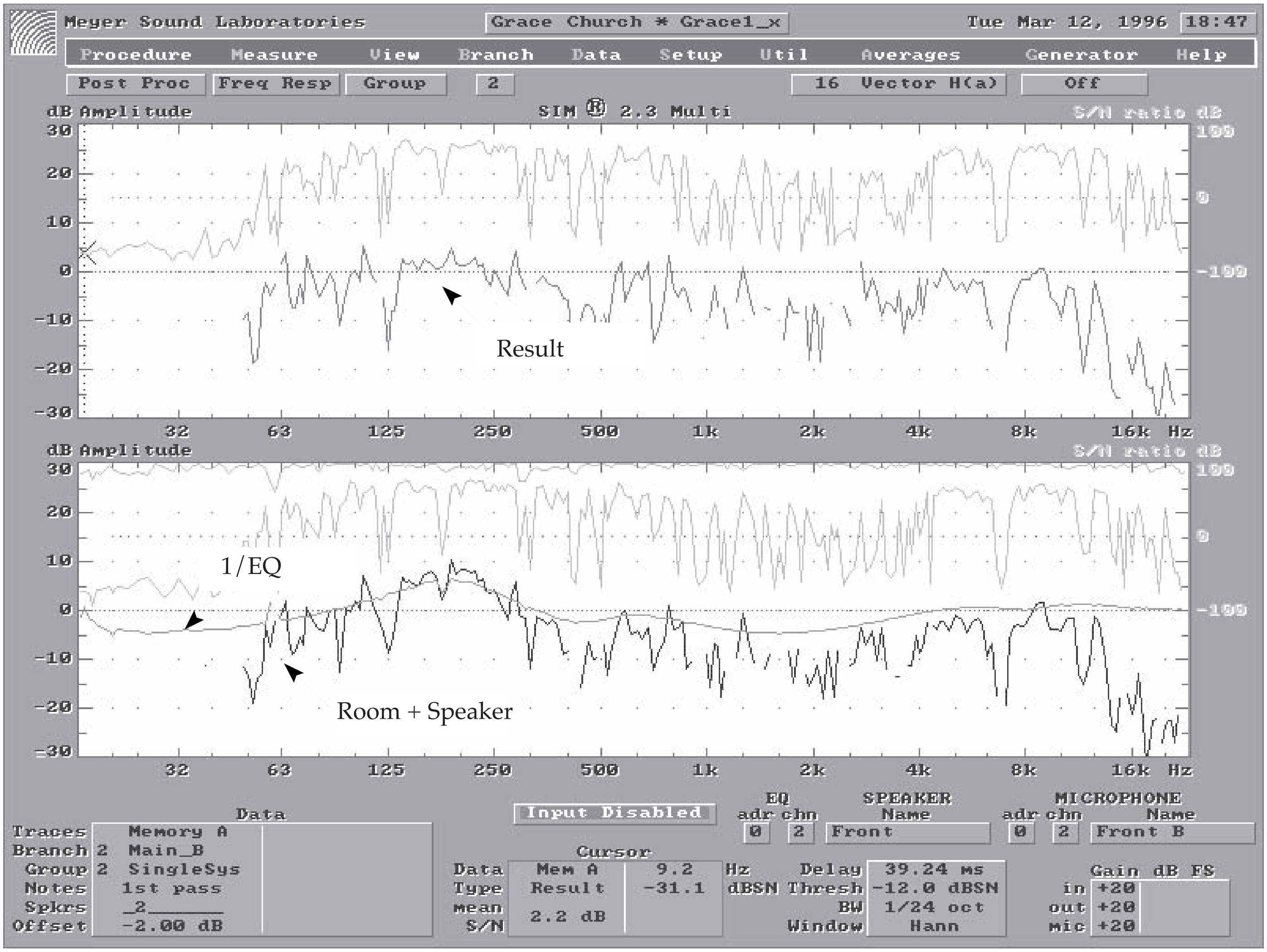
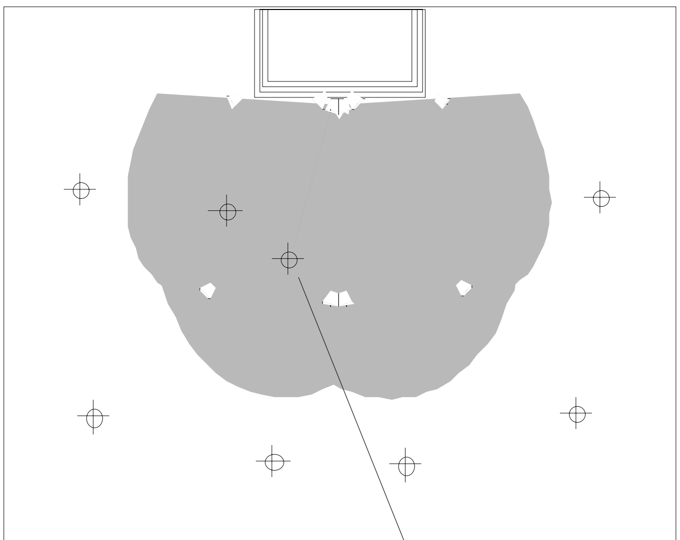


Fig 5.10f Equalization with SIM System II and the CP-10.

The system was then analyzed with SIM. The traces above show the effect of precisely contoured equalization. It may surprise you to notice the broad gentle slopes of the EQ curve. Precise and effective equalization *does not* mean narrow band equalization. It was created with four carefully chosen filters with bandwidths ranging from 1/5 to 1/2 octave. The shape of the EQ curve follows the general trends of the room+speaker trace. This results in a very linear result trace as shown in the upper screen.

5.10.3 Equalizing the Main Cluster



Plan view of the hall indicating the mic position and speaker status during the measurements shown opposite.

	Branch	Group 1	Group 2	Group 3	Group 4	Group 5
1	**Main A**	Single System	Single System			Combined Sys
		Original EQ	1st Pass			With Mn Sides
2	**Main B**	Single System	Single System			
		Original EQ	1st Pass			
3						
4						
5	**Main Side A**	Single System	Single System	Lobe Study	Combined Sys	Combined Sys
		Original EQ	1st Pass	From Mn Center	7 ms offset	synched to Mains
6	**Main Side B**	Single System	Single System			
		Original EQ	1st Pass			
7	**Delay Cent A**	Single System	Single System		Lobe Study	Combined Sys
		Original EQ	1st Pass		From all Mains	With all Mains
8	**Delay Cent B**		Single System			
			1st Pass			
9	**Delay Side A**	Single System	Single System	Lobe Study	Lobe Study	Combined Sys
		Original EQ	1st Pass	From Mn Center		All Speakers
10	**Delay Side B**	Single System				
		Original EQ				

Data panel indicating the source of the traces shown on the opposite page.

5.10.4 Polarity Reversal Discovered

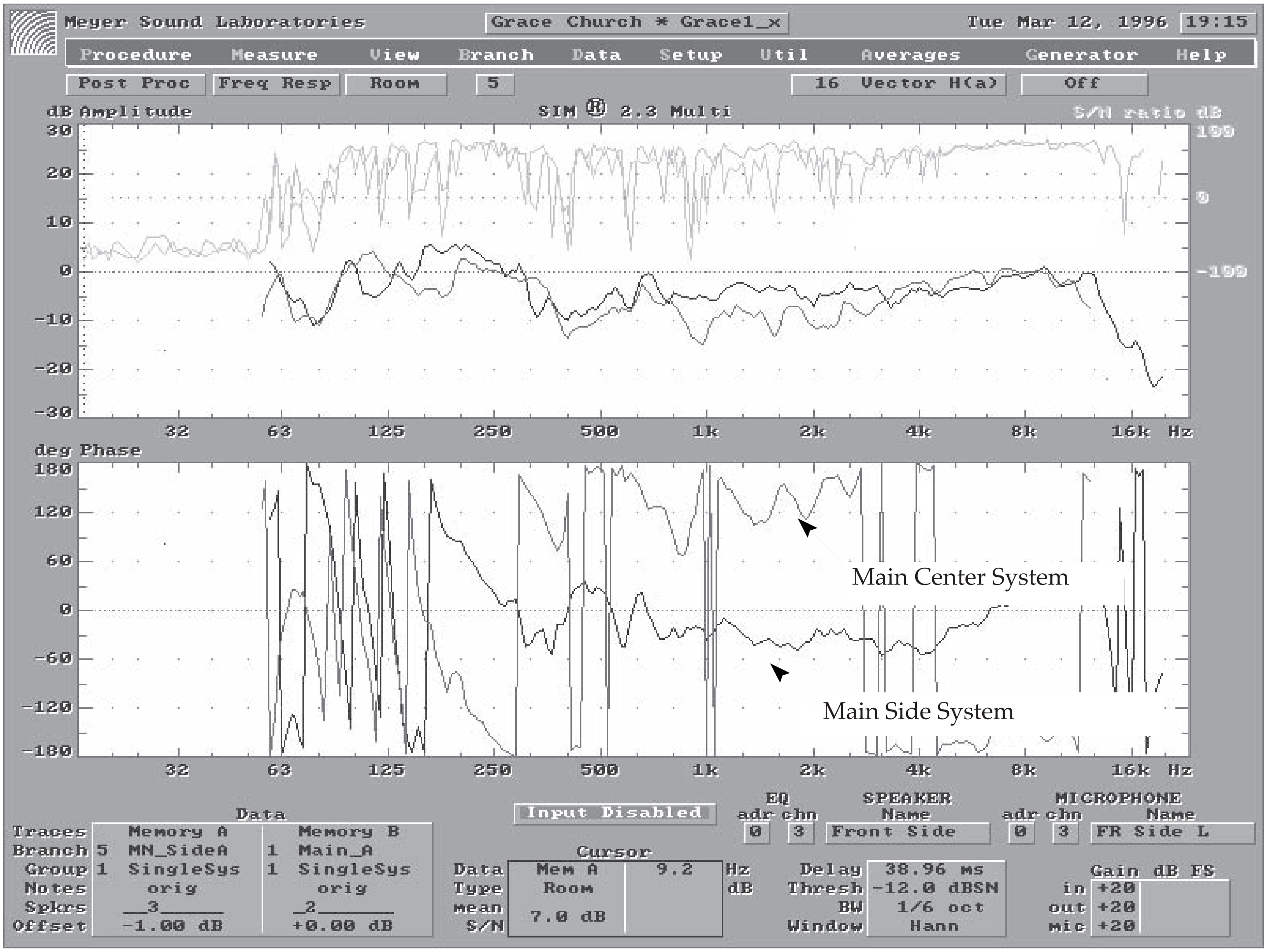
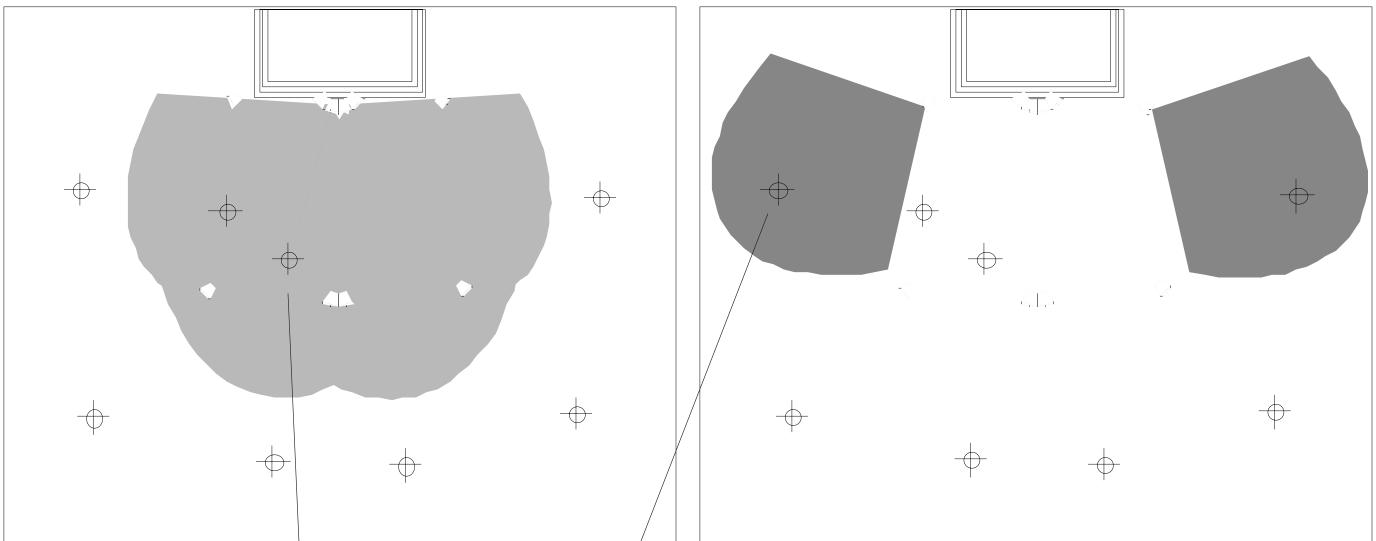


Fig 5.10g Polarity reversal discovered.

After finishing the initial equalization of the main cluster we moved on to the main side speakers. The phase trace revealed that the main and side clusters were polarity reversed from each other. A quick look at the delay systems revealed that the main cluster was reversed from all of the others. Upon further examination it was revealed that the main cluster amplifiers were a different brand than the rest of the system, a mix of pin 2 and pin 3 hot amplifiers. The polarity reversal is obvious when viewing the phase trace above. Notice that the two traces maintain a constant phase relationship of 180°.

5.10.4 Polarity Reversal Discovered



Plan view of the hall indicating the mic position and speaker status during the measurements shown opposite.

	Branch	Group 1	Group 2	Group 3	Group 4	Group 5
1	Main A	Single System	Single System			Combined Sys
		Original EQ	1st Pass			With Mn Sides
2	Main B	Single System	Single System			
		Original EQ	1st Pass			
3						
4						
5	Main Side A	Single System	Single System	Lobe Study	Combined Sys	Combined Sys
		Original EQ	1st Pass	From Mn Center	7 ms offset	synched to Mains
6	Main Side B	Single System	Single System			
		Original EQ	1st Pass			
7	Delay Cent A	Single System	Single System		Lobe Study	Combined Sys
		Original EQ	1st Pass		From all Mains	With all Mains
8	Delay Cent B		Single System			
			1st Pass			
9	Delay Side A	Single System	Single System	Lobe Study	Lobe Study	Combined Sys
		Original EQ	1st Pass	From Mn Center		All Speakers
10	Delay Side B	Single System				
		Original EQ				

Data panel indicating the source of the traces shown on the opposite page.

5.10.5 Equalizing the Main Side System

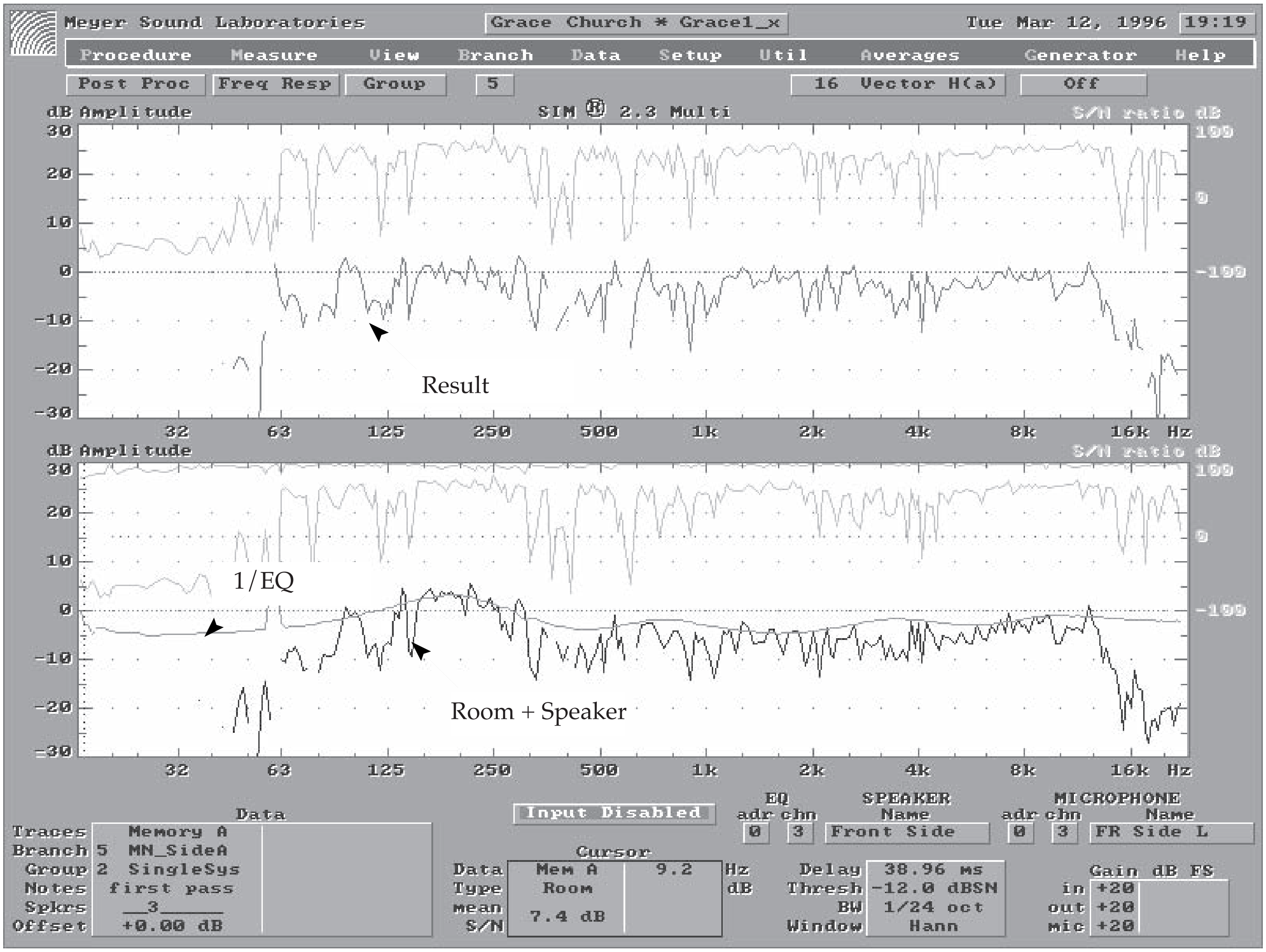
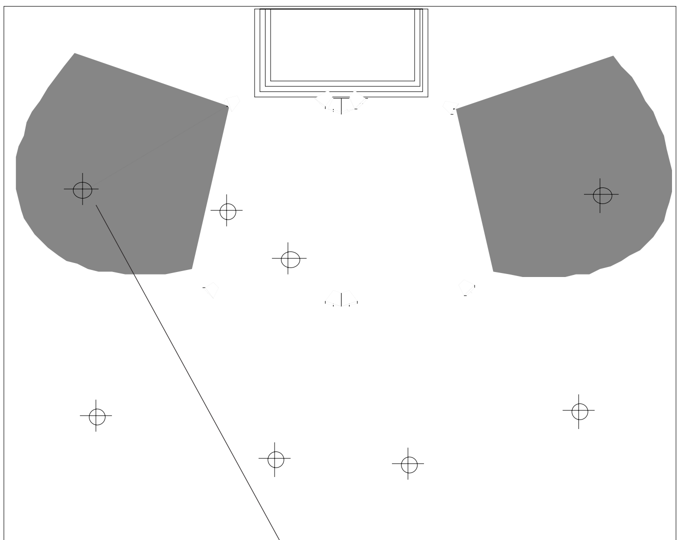


Fig 5.10h Equalizing the main side system.

The main side speakers are a single UPA-1A per side. The interaction with the room dominates the minimal multiple speaker interaction. This results in a much flatter room-plus-speaker response to start with compared to the four speaker main cluster. Therefore less equalization is required.

5.10.5 Equalizing the Main Side System



Plan view of the hall indicating the mic position and speaker status during the measurements shown opposite.

	Branch	Group 1	Group 2	Group 3	Group 4	Group 5
1	**Main A**	Single System	Single System			Combined Sys
		Original EQ	1st Pass			With Mn Sides
2	**Main B**	Single System	Single System			
		Original EQ	1st Pass			
3						
4						
5	**Main Side A**	Single System	Single System	Lobe Study	Combined Sys	Combined Sys
		Original EQ	1st Pass	From Mn Center	7 ms offset	synched to Mains
6	**Main Side B**	Single System	Single System			
		Original EQ	1st Pass			
7	**Delay Cent A**	Single System	Single System		Lobe Study	Combined Sys
		Original EQ	1st Pass		From all Mains	With all Mains
8	**Delay Cent B**		Single System			
			1st Pass			
9	**Delay Side A**	Single System	Single System	Lobe Study	Lobe Study	Combined Sys
		Original EQ	1st Pass	From Mn Center		All Speakers
10	**Delay Side B**	Single System				
		Original EQ				

Data panel indicating the source of the traces shown on the opposite page.

5.10.6 Combining the Main Systems

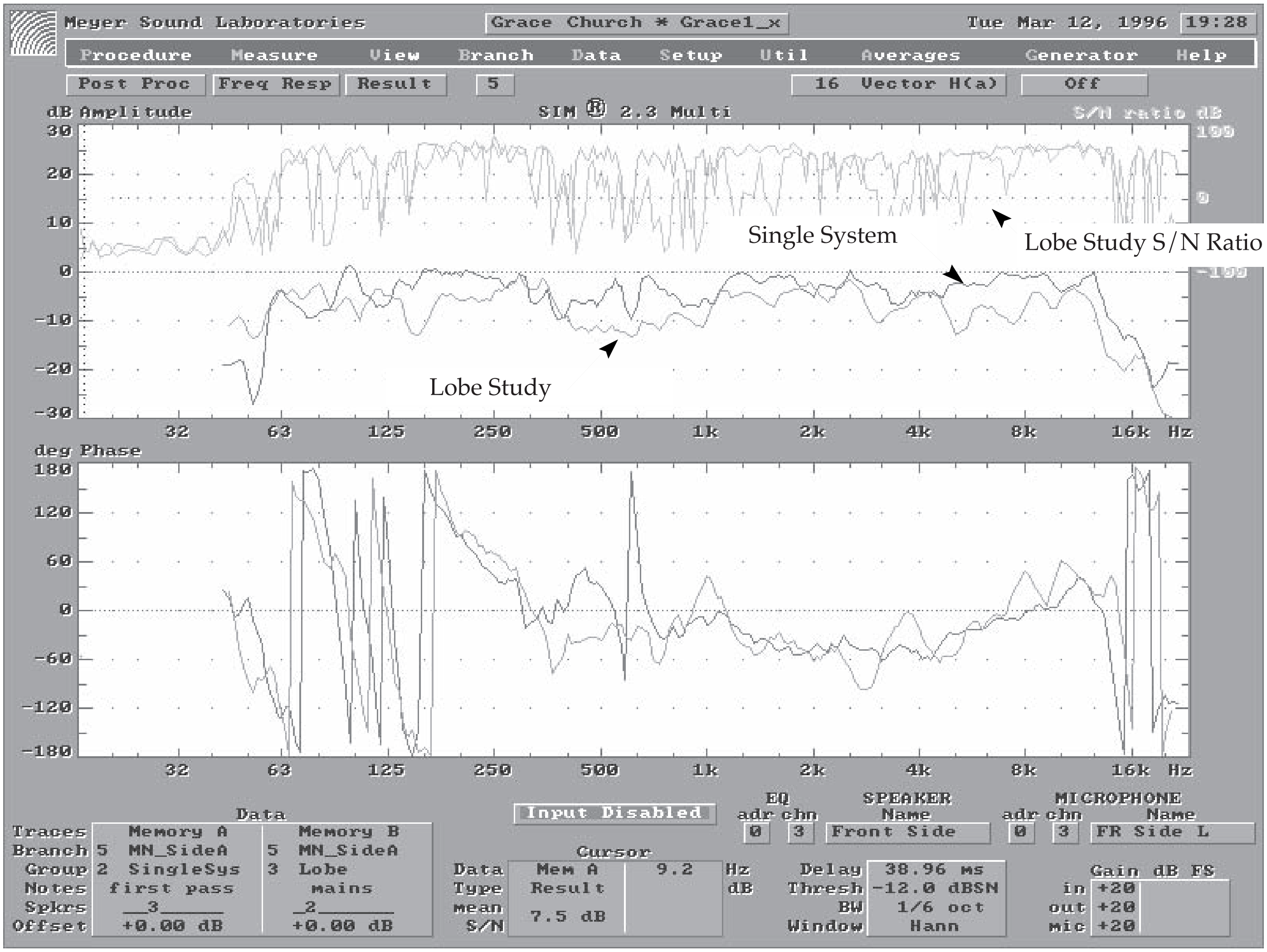
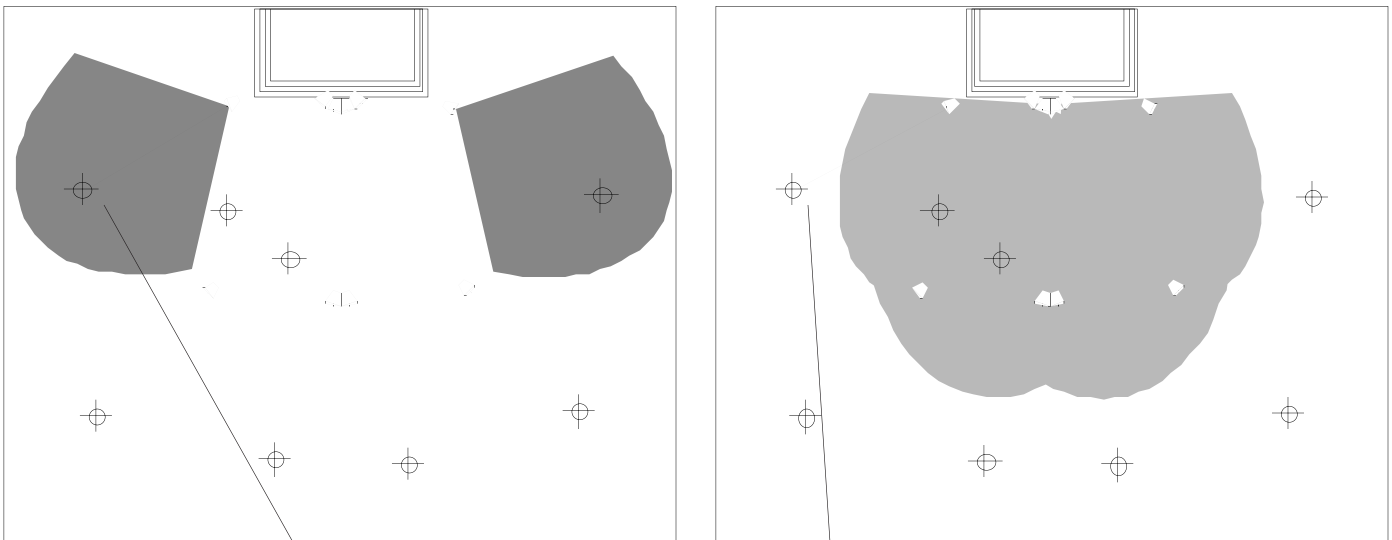


Fig 5.10i Checking for isolation between the main and side systems.

The main cluster must cover the largest section of the hall. The remaining subsystems will need to accommodate any overflow coverage from the main cluster. The traces above illustrate a comparison of the signal arriving in the side area from the side and center cluster, respectively. The traces were individually measured so that we can see how much overflow is coming from the center relative to the side speaker. The above trace shows that there is only minimal isolation. In particular there is a lot of overlap in the range between 2 kHz to 5 kHz where the UPA-1's pattern is quite wide.

5.10.6 Combining the Main Systems



Plan view of the hall indicating the mic position and speaker status during the measurements shown opposite.

	Branch	Group 1	Group 2	Group 3	Group 4	Group 5
1	**Main A**	Single System	Single System			Combined Sys
		Original EQ	1st Pass			With Mn Sides
2	**Main B**	Single System	Single System			
		Original EQ	1st Pass			
3						
4						
5	**Main Side A**	Single System	Single System	Lobe Study	Combined Sys	Combined Sys
		Original EQ	1st Pass	From Mn Center	7 ms offset	synched to Mains
6	**Main Side B**	Single System	Single System			
		Original EQ	1st Pass			
7	**Delay Cent A**	Single System	Single System		Lobe Study	Combined Sys
		Original EQ	1st Pass		From all Mains	With all Mains
8	**Delay Cent B**		Single System			
			1st Pass			
9	**Delay Side A**	Single System	Single System	Lobe Study	Lobe Study	Combined Sys
		Original EQ	1st Pass	From Mn Center		All Speakers
10	**Delay Side B**	Single System				
		Original EQ				

Data panel indicating the source of the traces shown on the opposite page.

5.10.6 Combining the Main Systems

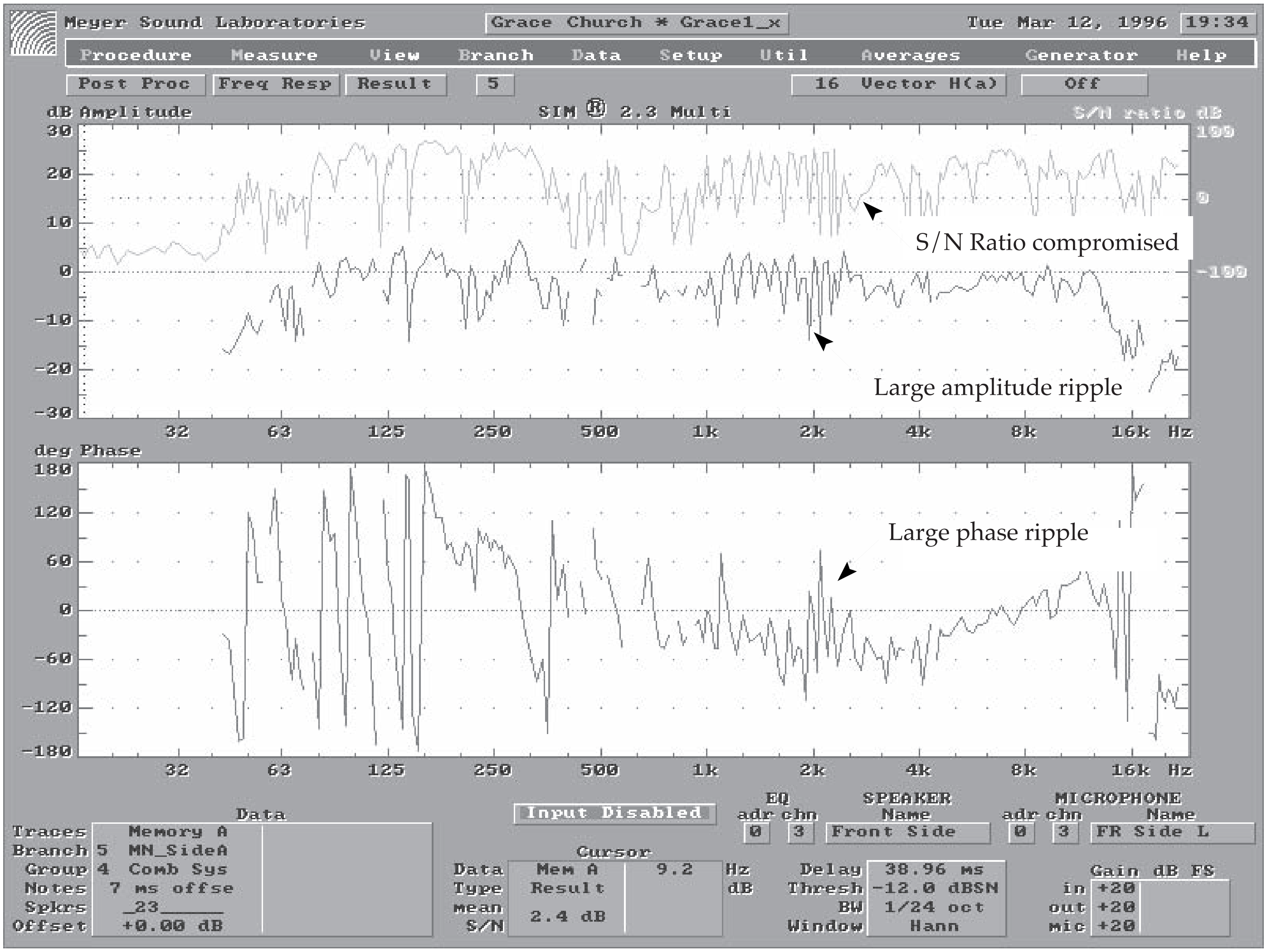
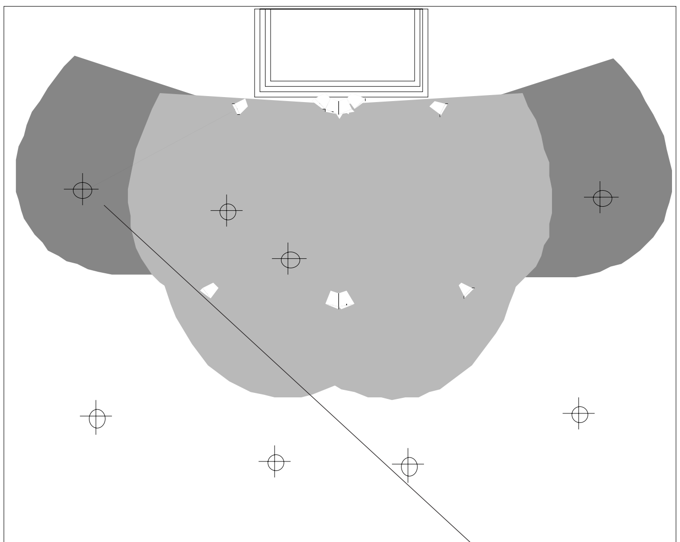


Fig 5.10j Combining the main cluster and side systems.

When the systems were combined there was a large change in the frequency response due to the strong interaction between the systems. The systems did not add well since the mains speakers arrived 7 ms late into the side area. The result is shown above. Note the loss in midrange S/N ratio and the large frequency and phase response ripple.

5.10.6 Combining the Main Systems



Plan view of the hall indicating the mic position and speaker status during the measurements shown opposite.

	Branch	Group 1	Group 2	Group 3	Group 4	Group 5
1	**Main A**	Single System	Single System			Combined Sys
		Original EQ	1st Pass			With Mn Sides
2	**Main B**	Single System	Single System			
		Original EQ	1st Pass			
3						
4						
5	**Main Side A**	Single System	Single System	Lobe Study	Combined Sys	Combined Sys
		Original EQ	1st Pass	From Mn Center	7 ms offset	synched to Mains
6	**Main Side B**	Single System	Single System			
		Original EQ	1st Pass			
7	**Delay Cent A**	Single System	Single System		Lobe Study	Combined Sys
		Original EQ	1st Pass		From all Mains	With all Mains
8	**Delay Cent B**		Single System			
			1st Pass			
9	**Delay Side A**	Single System	Single System	Lobe Study	Lobe Study	Combined Sys
		Original EQ	1st Pass	From Mn Center		All Speakers
10	**Delay Side B**	Single System				
		Original EQ				

Data panel indicating the source of the traces shown on the opposite page.

5.10.6 Combining the Main Systems

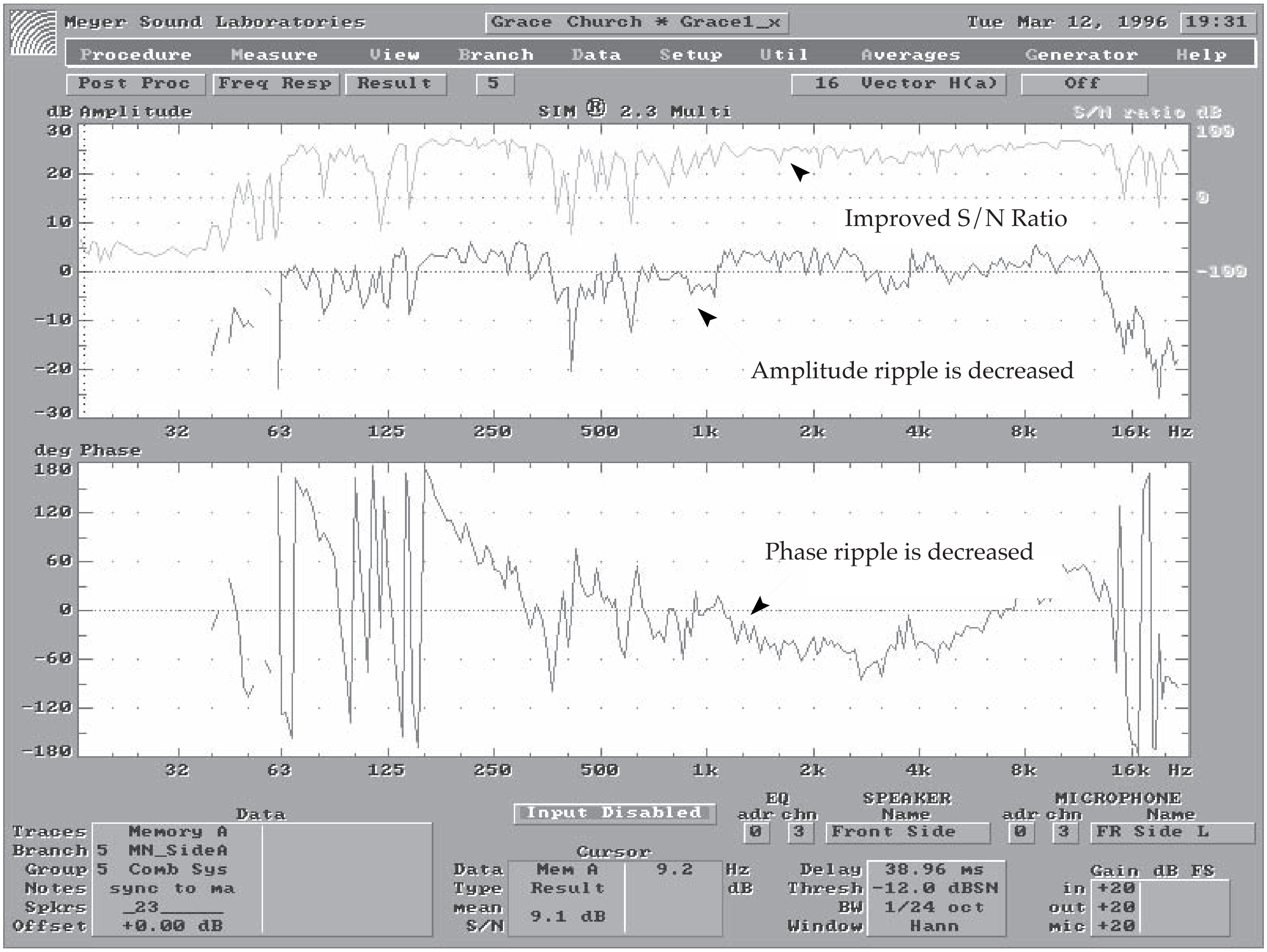
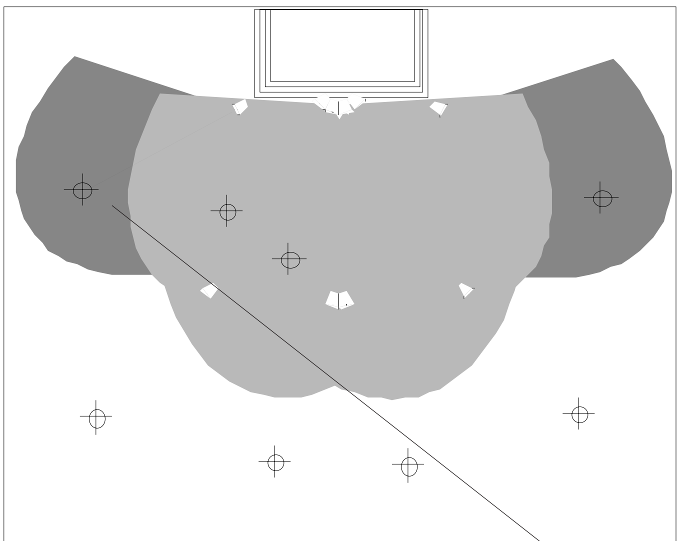


Fig 5.10k Delay tapering of the side system.

The original system patch was configured so that the main and side speakers had the same delay time. The system was repatched so that the side could be delayed separately. The delay offset was determined by using the procedure for external delay. The sides were delayed 7 ms, resulting in a much smoother combination of the two systems. Compare the traces shown above with the previous set taken before the delay was added. The above data shows improved S/N ratio and reduced amplitude and phase ripple.

5.10.6 Combining the Main Systems



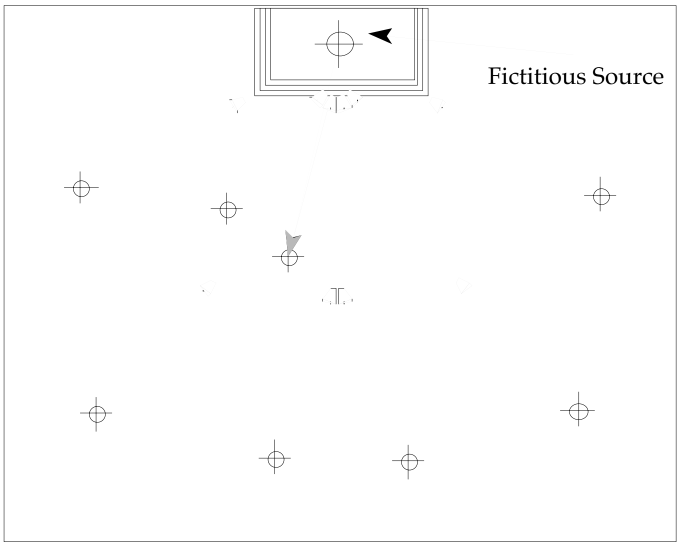
Plan view of the hall indicating the mic position and speaker status during the measurements shown opposite.

	Branch	Group 1	Group 2	Group 3	Group 4	Group 5
1	**Main A**	Single System	Single System			Combined Sys
		Original EQ	1st Pass			With Mn Sides
2	**Main B**	Single System	Single System			
		Original EQ	1st Pass			
3						
4						
5	**Main Side A**	Single System	Single System	Lobe Study	Combined Sys	Combined Sys
		Original EQ	1st Pass	From Mn Center	7 ms offset	synched to Mains
6	**Main Side B**	Single System	Single System			
		Original EQ	1st Pass			
7	**Delay Cent A**	Single System	Single System		Lobe Study	Combined Sys
		Original EQ	1st Pass		From all Mains	With all Mains
8	**Delay Cent B**		Single System			
			1st Pass			
9	**Delay Side A**	Single System	Single System	Lobe Study	Lobe Study	Combined Sys
		Original EQ	1st Pass	From Mn Center		All Speakers
10	**Delay Side B**	Single System				
		Original EQ				

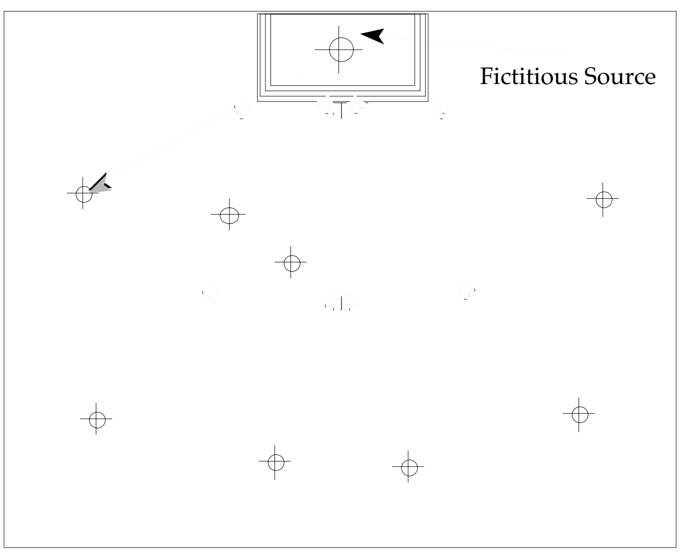
Data panel indicating the source of the traces shown on the opposite page.

5.10.7 Setting Delays

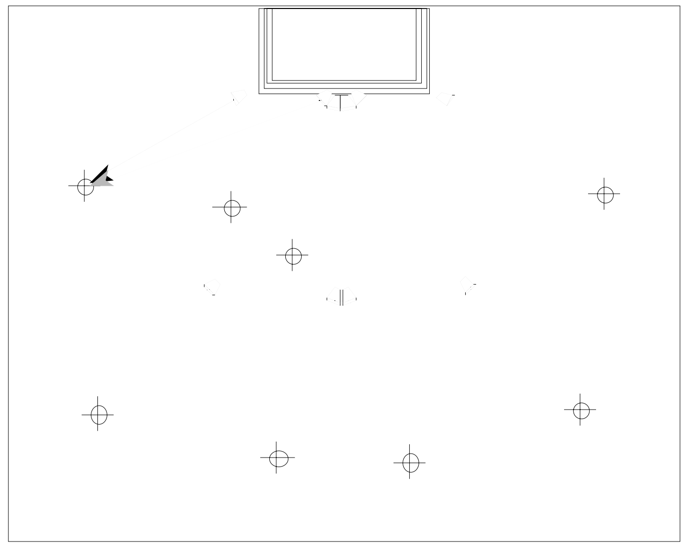
Delay Setting proceeded in five stages beginning with the systems nearest the stage and moving outward. The process is described below.



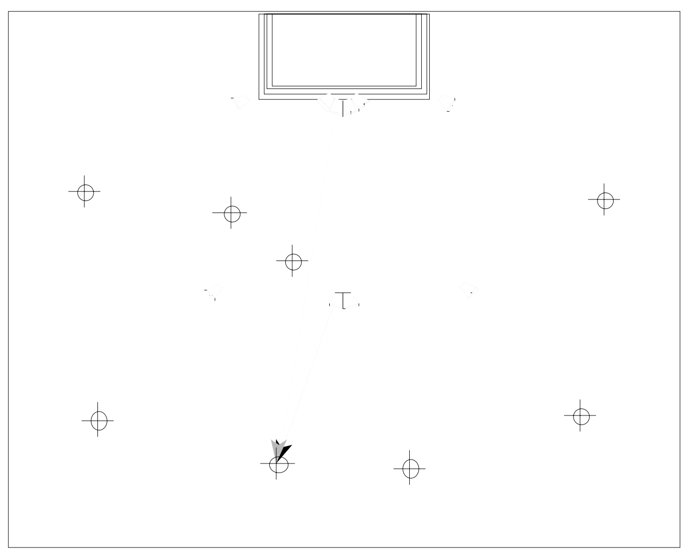
Step 1: Delay the main center system to synchronize to the fictitious source. This helps to create a sonic image in the area of the minister.



Step 2: Delay the main side system to synchronize to the fictitious source. This was chosen because the fictitious source would help create a sonic image in the direction of the stage. This proved unworkable, however, due to the interaction of the main sides with the main center system, as shown in the previous section.

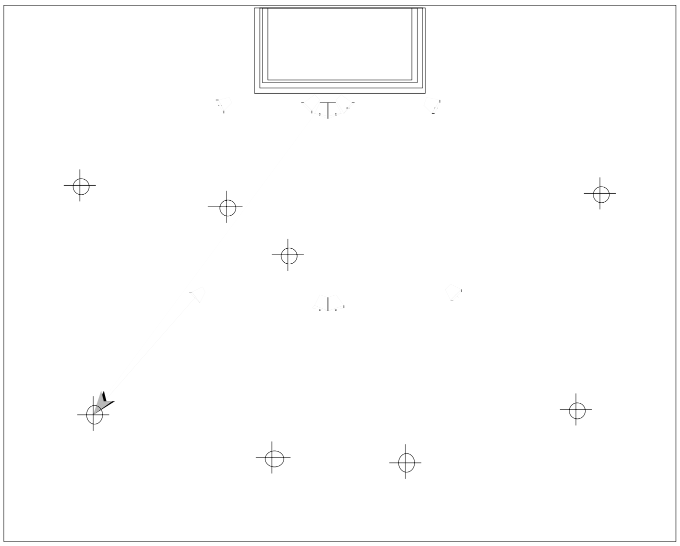


Step 3: Delay the main side system to synchronize to the signal from the main center system. This proved far superior for intelligibility than the previous setting.



Step 4: Delay the delay center system to the main center system. This was chosen because the main center was the second strongest signal in the delay center area and created a sonic image in the direction of the stage.

5.10.7 Setting Delays



Step 5: Delay the delay sides to the main center system. This was chosen because the main center was the second strongest signal in the delay side area and created a sonic image in the direction of the stage. The interaction with the delay center was negligible.

5.10.8 Equalizing the Delay Side System

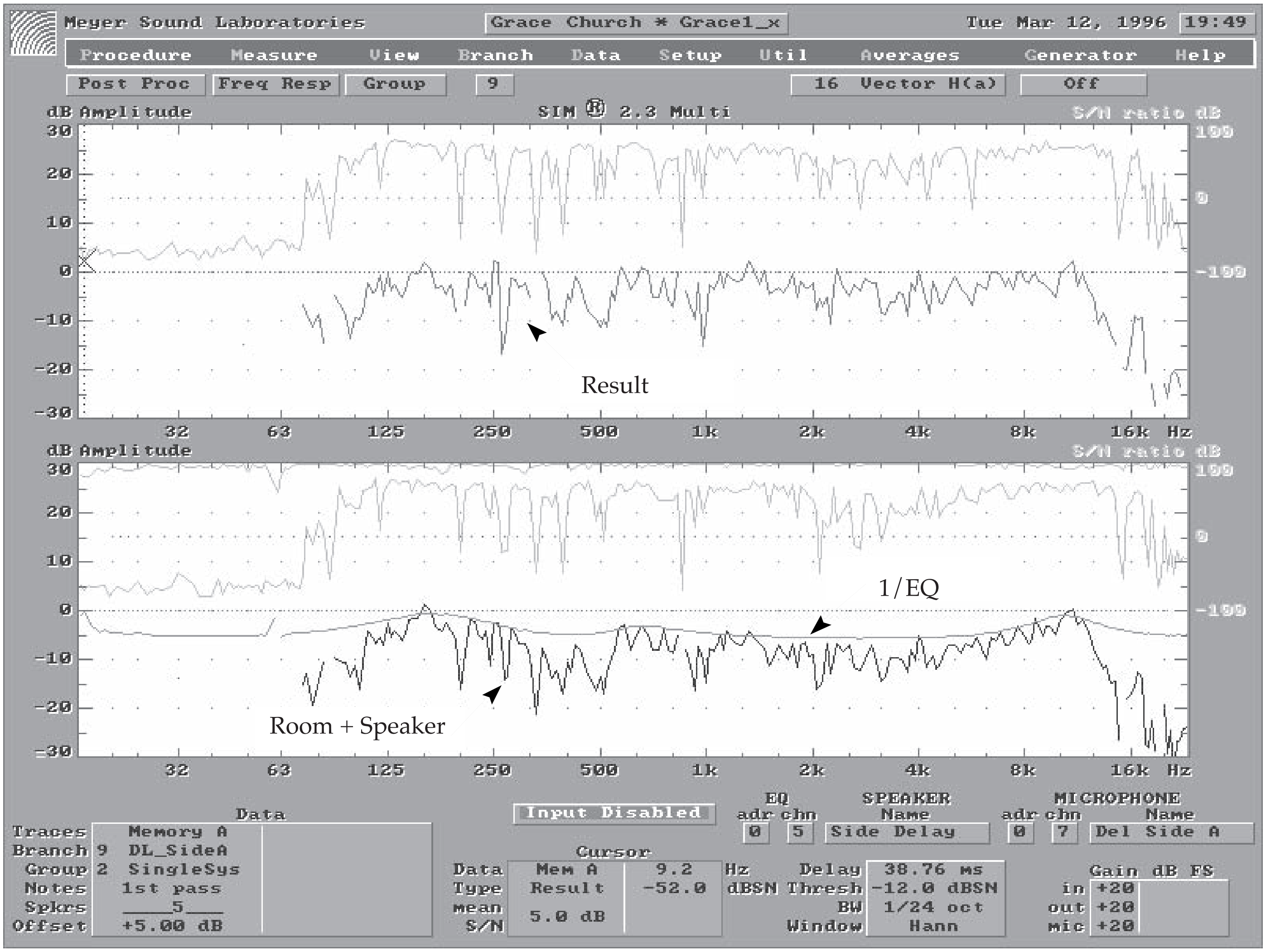
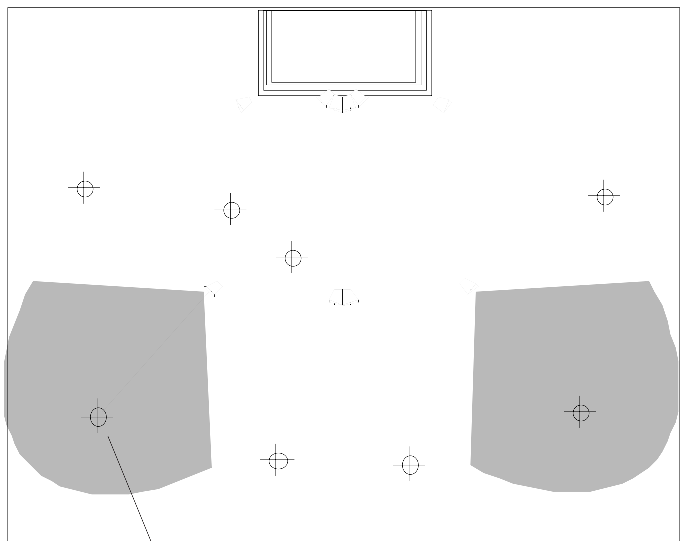


Fig 5.10l SIM equalization of the side delay speakers.

The side delays consisted of a single speaker per side, therefore there was only a minimal LF buildup in the response. This system required the least equalization. This is not surprising since there is no speaker/speaker interaction but instead only speaker/room interaction.

This response is a good example of the UPA-1A's characteristic 10 kHz peak. This is an efficiency peak in the 1401A driver response. The peak can be optionally removed by moving the jumper wire on the Y-1PD network in the cabinet.

5.10.8 Equalizing the Delay Side System



Plan view of the hall indicating the mic position and speaker status during the measurements shown opposite.

	Branch	Group 1	Group 2	Group 3	Group 4	Group 5
1	**Main A**	Single System	Single System			Combined Sys
		Original EQ	1st Pass			With Mn Sides
2	**Main B**	Single System	Single System			
		Original EQ	1st Pass			
3						
4						
5	**Main Side A**	Single System	Single System	Lobe Study	Combined Sys	Combined Sys
		Original EQ	1st Pass	From Mn Center	7 ms offset	sync,d to Mains
6	**Main Side B**	Single System	Single System			
		Original EQ	1st Pass			
7	**Delay Cent A**	Single System	Single System		Lobe Study	Combined Sys
		Original EQ	1st Pass		From all Mains	With all Mains
8	**Delay Cent B**		Single System			
			1st Pass			
9	**Delay Side A**	Single System	Single System	Lobe Study	Lobe Study	Combined Sys
		Original EQ	1st Pass	From Mn Center		All Speakers
10	**Delay Side B**	Single System				
		Original EQ				

Data panel indicating the source of the traces shown on the opposite page.

5.10.9 Combining the Delay Systems

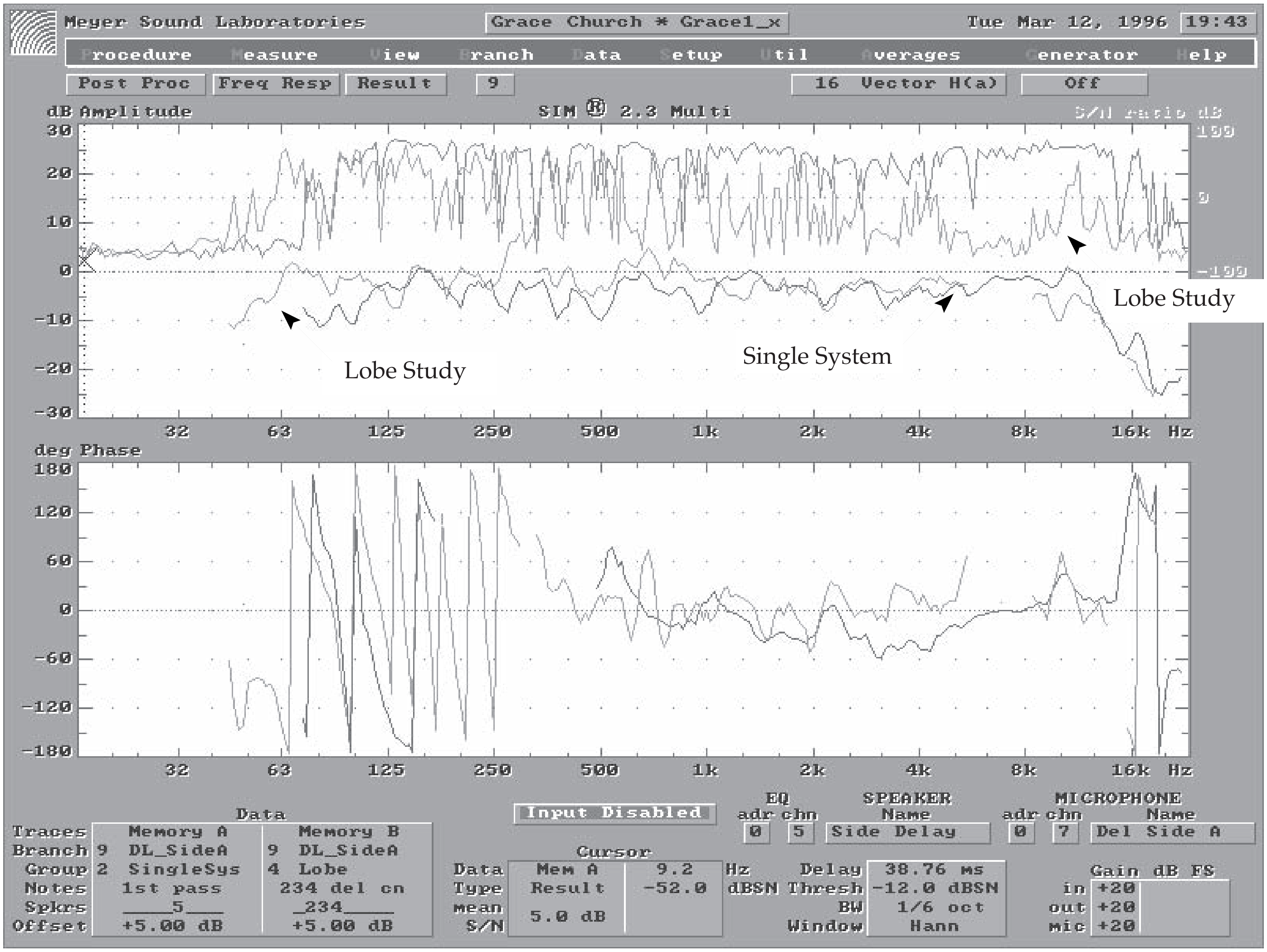
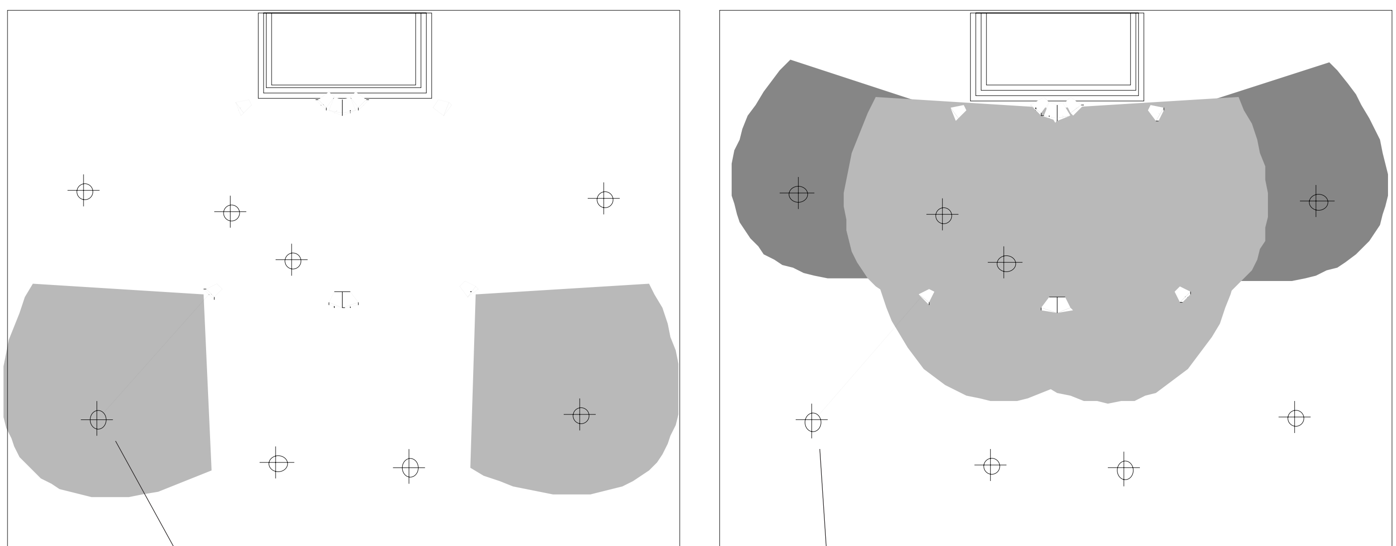


Fig 5.10m Evaluating the need for the delay speakers.

One of the questions that comes up in the evaluation of a design is: "Do we really need these delay speakers?" This question is answered in the data above, which compares the signal arriving from the main speakers, and delay speakers arriving into the side delay coverage area. If the response from the main speakers is good, this would indicate that the delays were not required. If the response is poor then we know that the delays are required.

A comparison of the amplitude response alone shows that the frequency response and level are well matched. This would seem to indicate that the delays are not needed. The S/N ratio traces, however, are grossly different, indicating that the response from the main system is totally unintelligible in the side delay area due to a low direct-to-reverberant ratio. The delays will be required to raise the intelligibility (not the level).

5.10.9 Combining the Delay Systems



Plan view of the hall indicating the mic position and speaker status during the measurements shown on the opposite page.

	Branch	Group 1	Group 2	Group 3	Group 4	Group 5
1	**Main A**	Single System	Single System			Combined Sys
		Original EQ	1st Pass			With Mn Sides
2	**Main B**	Single System	Single System			
		Original EQ	1st Pass			
3						
4						
5	**Main Side A**	Single System	Single System	Lobe Study	Combined Sys	Combined Sys
		Original EQ	1st Pass	From Mn Center	7 ms offset	sync,d to Mains
6	**Main Side B**	Single System	Single System			
		Original EQ	1st Pass			
7	**Delay Cent A**	Single System	Single System		Lobe Study	Combined Sys
		Original EQ	1st Pass		From all Mains	With all Mains
8	**Delay Cent B**		Single System			
			1st Pass			
9	**Delay Side A**	Single System	Single System	Lobe Study	Lobe Study	Combined Sys
		Original EQ	1st Pass	From Mn Center		All Speakers
10	**Delay Side B**	Single System				
		Original EQ				

Data panel indicating the source of the traces shown on the opposite page.

5.10.9 Combining the Delay Systems

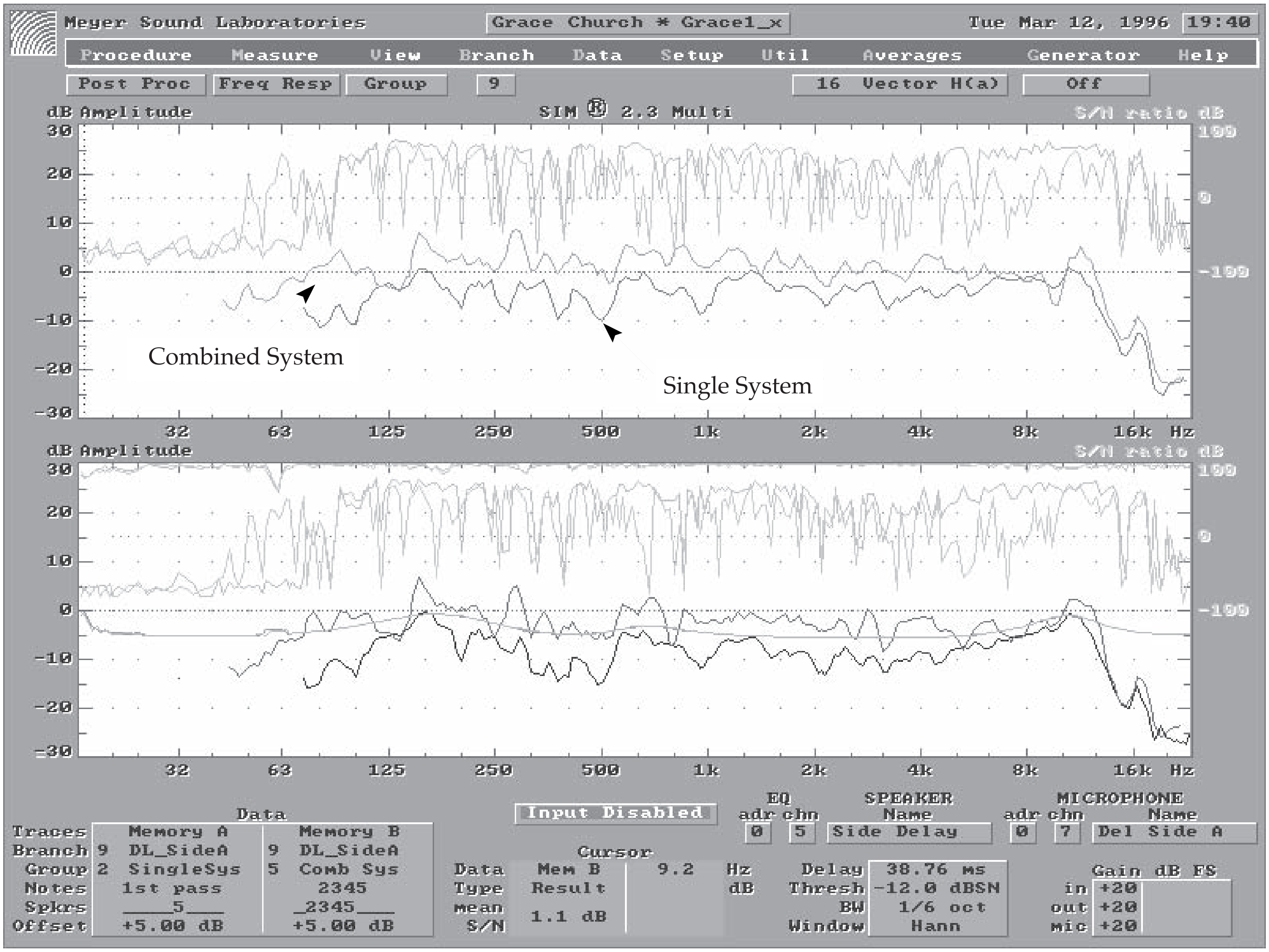
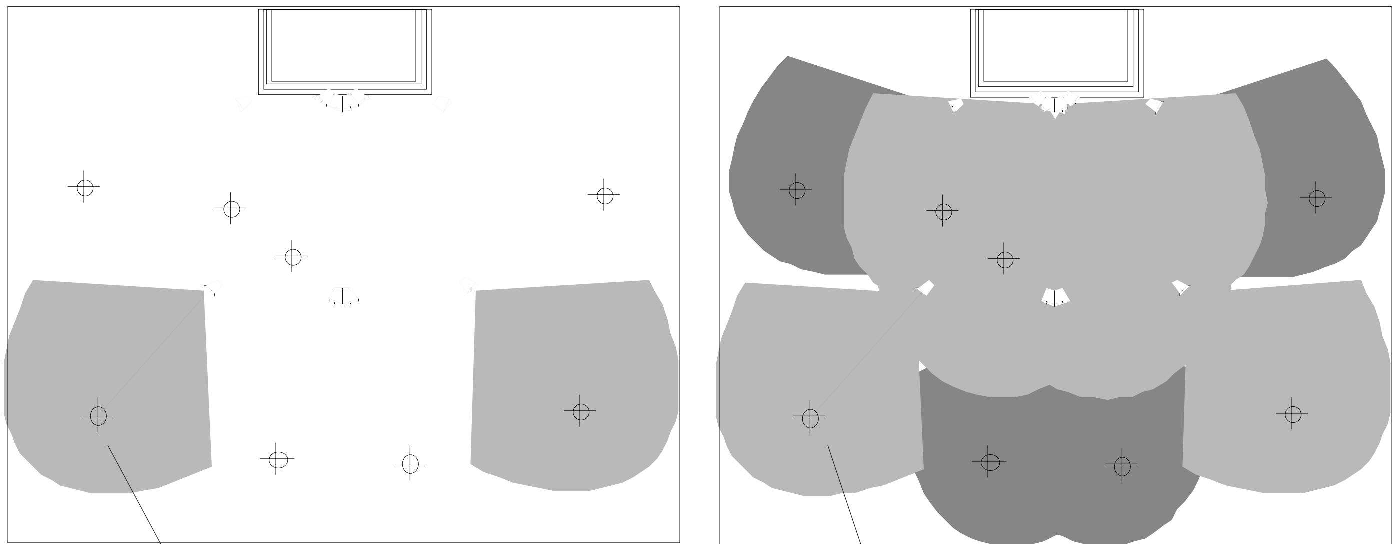


Fig 5.10n Combining the delays with the mains.

The combined response of the delays and mains is shown above compared to the original response of the delays alone. Notice that the overall level has risen but the overall linearity has been maintained. The level of the delay speakers was set so that the S/N ratio in the delay areas was comparable to that in the main system area. This helps to provide consistent intelligibility and minimizes the tendency to image toward the delay speakers. If the level of the delay speakers is set too high, the improvement in intelligibility will be offset by the distraction of poor imaging. If the level is set too low, the intelligibility will suffer. Some might advocate the practice of intentionally adding excess delay to the delay speakers. This results in moving the sonic image away from the delay speaker. Unfortunately though, it also decreases the S/N ratio due to combing. This then causes the delays to get turned up louder (since their intended purpose is increased S/N ratio) and you get poor imaging and low S/N ratio.

5.10.9 Combining the Delay Systems

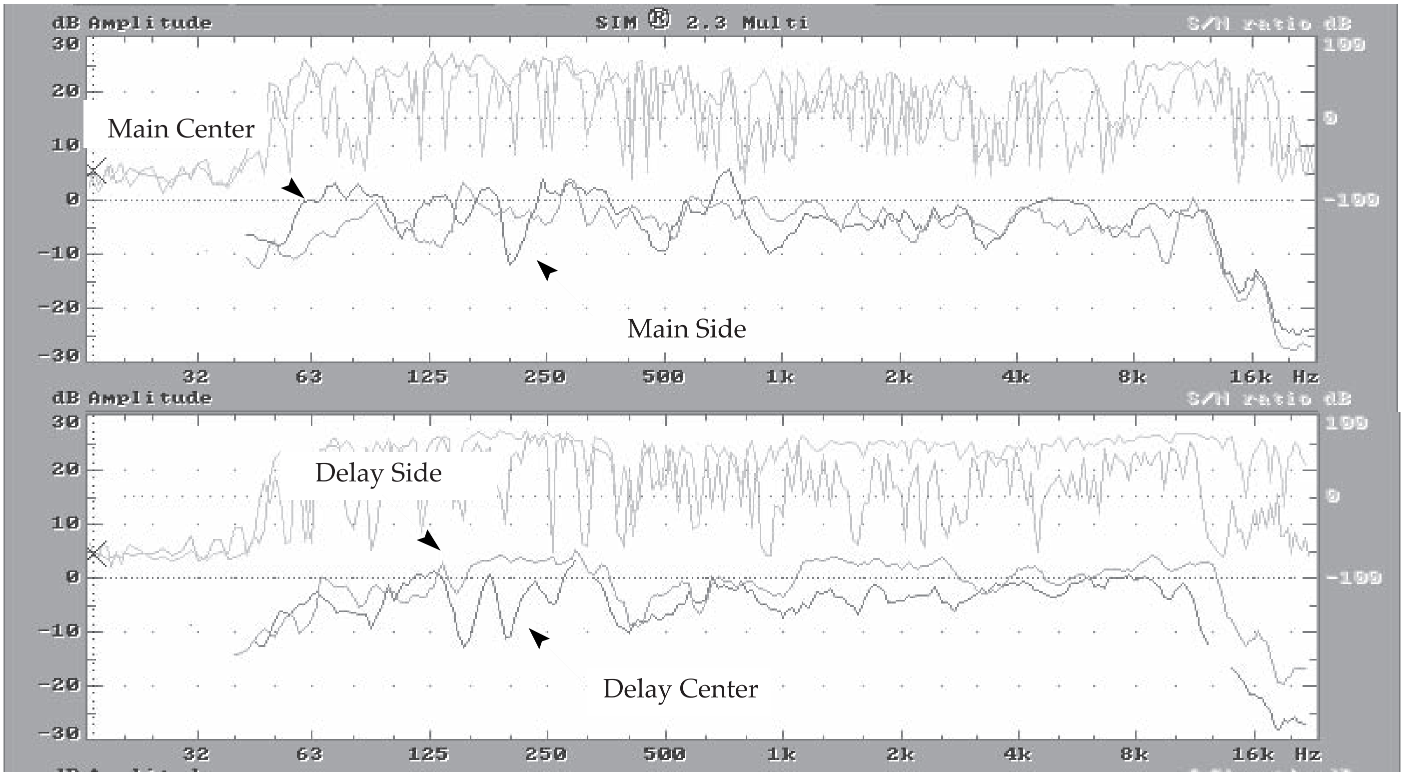


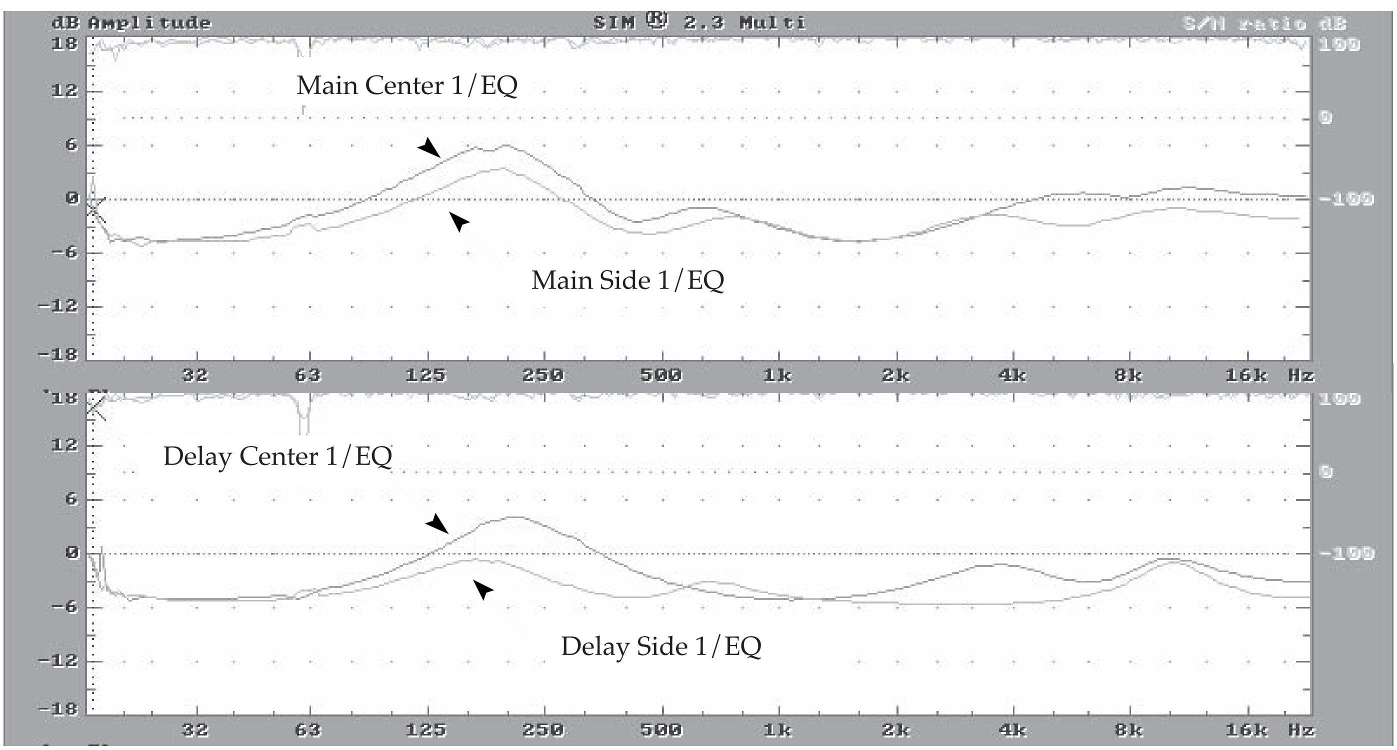
Plan view of the hall indicating the mic position and speaker status during the measurements shown opposite.

	Branch	Group 1	Group 2	Group 3	Group 4	Group 5
1	Main A	Single System	Single System			Combined Sys
		Original EQ	1st Pass			With Mn Sides
2	Main B	Single System	Single System			
		Original EQ	1st Pass			
3						
4						
5	Main Side A	Single System	Single System	Lobe Study	Combined Sys	Combined Sys
		Original EQ	1st Pass	From Mn Center	7 ms offset	sync,d to Mains
6	Main Side B	Single System	Single System			
		Original EQ	1st Pass			
7	Delay Cent A	Single System	Single System		Lobe Study	Combined Sys
		Original EQ	1st Pass		From all Mains	With all Mains
8	Delay Cent B		Single System			
			1st Pass			
9	Delay Side A	Single System	Single System	Lobe Study	Lobe Study	Combined Sys
		Original EQ	1st Pass	From Mn Center		All Speakers
10	Delay Side B	Single System				
		Original EQ				

Data panel indicating the source of the traces shown on the opposite page.

5.10.10 All Systems Combined





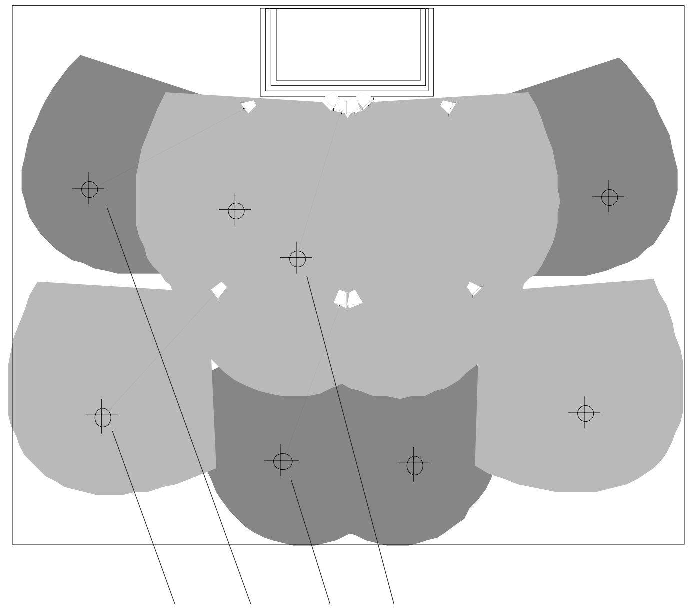
Note : The equalizer traces are inverted (1/EQ).

5.10.10 All Systems Combined

The Full System Response

Once the systems are fully combined we can take a look at each position and compare. The hope is that the systems will be closely matched in level and frequency response. In this case the frequency responses were well matched but it was noticed that the level of the main side system was higher than the others. This was remedied by a level adjustment at the M-1A controller.

Notice, also, the differences in the response of the four equalizers. Each subsystem required unique equalization in order to have a matched result response. The main and delay center systems required more EQ in the low midrange since they were multiple speaker point-source arrays. The side systems were single speaker split-point source elements with much less speaker interaction.



Plan view of the hall indicating the mic position and speaker status during the measurements shown opposite.

	Branch	Group 1	Group 2	Group 3	Group 4	Group 5
1	**Main A**	Single System	Single System			Combined Sys
		Original EQ	1st Pass			With Mn Sides
2	**Main B**	Single System	Single System			
		Original EQ	1st Pass			
3						
4						
5	**Main Side A**	Single System	Single System	Lobe Study	Combined Sys	Combined Sys
		Original EQ	1st Pass	From Mn Center	7 ms offset	sync,d to Mains
6	**Main Side B**	Single System	Single System			
		Original EQ	1st Pass			
7	**Delay Cent A**	Single System	Single System		Lobe Study	Combined Sys
		Original EQ	1st Pass		From all Mains	With all Mains
8	**Delay Cent B**		Single System			
			1st Pass			
9	**Delay Side A**	Single System	Single System	Lobe Study	Lobe Study	Combined Sys
		Original EQ	1st Pass	From Mn Center		All Speakers
10	**Delay Side B**	Single System				
		Original EQ				

Data panel indicating the source of the traces shown on the opposite page.

Speakers	Field Installable?	Covered under Warranty?
UM-1A to 1C Upgrade	Yes	No
UM-1B to 1C Upgrade	Yes	Yes
UPA-1A to 1C Upgrade	Yes	No
UPA-1B to 1C Upgrade	Yes	Yes
MSL-3 to MSL-3A Upgrade	Yes	No
S500 to S500A Upgrade	Yes	No

Electronics	Field Installable?	Covered under Warranty?
CP-10 Gain Adjust Modification	Yes	No
Linear Pot Modification	Yes	No
(For M-1, M-3, B-2, B-2A, B-2AEX)		
EX Card for B-2 and B-2A	Yes	No
P-1 to P-1A Upgrade	Yes	No
HD-1 Low Noise Modification	No	No
SIM 2403 Mute Circuit Modification	No	Yes
SIM 2403 Address Switch Modification	No	No
SIM 2403 ESD Modification	No	Yes
M500 to M500A Upgrade	No	No
MS1000 to MS1000A Upgrade	No	No

Software/Hardware	Field Installable?
SIM v2.0 to v2.3s Upgrade	Yes
SIM v2.3s to v2.3m+s Upgrade	Yes
SIM v2.0 to v2.3m Upgrade	Yes

Speakers

UPM-1: Original and current version. Use P-1A or MPS-3 CEU.

MPS-355: Contractor version of UPM-1. Similar acoustic performance. Lower cost enclosure without trapezoidal shape. Use P-1A or MPS-3 CEU.

MPS-305: Reduced size power and bandwidth version. Uses only a single five-inch LF driver. Must be used with the MPS-3 controller only. Lo-Cut switch should be engaged for maximum dynamic range.

Compatibility Issues: The UPM-1 or MPS-355 speakers are frequency response compatible.

The MPS-305 is an 8Ω load with reduced LF response (the others are 16Ω). These should be powered separately from the UPM-1 or MPS-355 speakers.

Controllers

P-1: Original controller for UPM-1. These have been superseded by the more sophisticated P-1A.

P-1A: Improved frequency response and limiting. P-1A gives extended LF response with a peak sliding filter that rolls off low end when low-frequency power exceeds the threshold.

MPS-3: Stereo version of P-1A. Improved frequency response linearity over the P-1A. Cost effective for multi-channel applications. This unit has a slightly higher noise floor than the P-1A.

CEU Upgrades: P-1 to P1-A Upgrade Kit.
Field installable PCB.

Compatibility Issues:The P-1A and MPS-3 are not frequency response compatible. If they are used together the system should be equalized separately.

The P-1A controller can be used for the UPM-1 or MPS-355 speakers.

The MPS-3 Controller can be used for any of the above speakers.

Speakers

UM-1: Original UM-1.

UM-1A: Diaphragm modification to 1401A and network change to Y1-PB for improved HF response. Networks were shipped in "10 kHz boost" position.

UM-1B: HF driver changed to 1401B and network to Y-1PC. Extended HF response to 20 kHz. Not compatible with UM-1s, UM-1As or UM-1Cs. Networks were shipped in the HF peak position leaving a resonant peak at or above 16 kHz.

UM-1C: A throat extender was added and the network changed for improved compatibility with UM-1A. HF response remains extended from UM-1A. Networks are shipped in the flat position. Can be changed to 16 kHz boost position. The boost position is useful when the UM-1C is used for long-throw PA applications.

Speaker Upgrades: UM-1B to 1C conversion kit is available free of charge. UM-1A to 1C conversion kit can be purchased.

Compatibility Issues: UM-1 response will not match any of the other models.

UM-1A with HF network "Flat "(not as shipped) is very close to UM-1C with HF network "Flat" (as shipped).

UM-1B response will not match any of the other models. The "UM-1A" HF network setting has not proven to be sufficiently compatible with the UM-1A. Not recommended.

UM-1C with HF network "Flat" (as shipped) is very close to UM-1A with HF network "Flat" (not as shipped).

Controllers

UltraMonitor™: The original controller for the UM-1. Had a switch labeled "–20 dB" which reduced the limiting threshold by 20 dB. This was designed so that stage monitors could be "rung out" without causing ear damage.

M-1: The introduction of the UPA-1 required a slight redesign. The –20 dB switch was changed to "Safe" which reduced the limiting threshold by 6 dB. This served the function of increasing system reliability in long-term high-power applications and reducing the audibility of amplifier clipping. Frequency response was not changed.

M-1A: Identical to M-1 except that the level potentiometer was changed to a linear taper. This was due to the high variability in component tolerance of the log pots. The linear pot is calibrated in dB attenuation. It is easier to match levels between units and their restricted range tends to keep the CEU operated in its optimal range.

CEU Upgrades:

For M-1s only: U22 hum reduction ground modification. This reduces the hum on pin 2 of the HF output channel. These are done to any units returned for service.

M-1 to M-1A conversion kit can be purchased (linear pot upgrade).

Compatibility Issues:

UltraMonitor and M-1: Factory-only upgrade to reset limit threshold for "Safe" switch.

M-1 and M-1A: The only compatibility issue is the front panel level control. Linear pot upgrade is recommended.

Speakers

UPA-1: Original UPA-1.

UPA-1A: Diaphragm modification to 1401A and network change to Y1-PB for improved HF response. Networks shipped in "10 kHz boost" position.

UPA-1B: HF driver changed to 1401B and network to Y-1PC. Extended HF response to 20 kHz. Not compatible with UPA-1s, UPA-1As or UPA-1Cs. Networks were shipped in the HF peak position leaving a resonant peak at or above 16 kHz.

UPA-1C: A throat extender was added and network changed for improved compatibility with the UPA-1A. HF response remains extended from UPA-1A. Networks are shipped in the flat position. Can be changed to 16 kHz boost position. The boost position is useful when the UM-1C is used for long-throw PA applications.

Speaker Upgrades: UPA-1B to 1C conversion kit is available free of charge. UPA-1A to 1C conversion kit can be purchased.

Compatibility Issues: UPA-1 response will not match any of the other models.

UPA-1A with HF network "Flat"(not as shipped) is very close to UPA-1C with HF network "Flat" (as shipped).

UPA-1B response will not match any of the other models. The "UPA-1A" HF network setting has not proven to be sufficiently compatible with the UPA-1A. Not recommended.

UPA-1C with HF network "Flat" (as shipped) is very close to UPA-1A with HF network "Flat" (not as shipped).

Controllers

UltraMonitor™: The original controller developed for the UM-1. Had a switch labeled "–20 dB" which reduced the limiting threshold by 20 dB.

M-1: The introduction of the UPA-1 required a slight redesign. The –20 dB switch was changed to "Safe" which reduced the limiting threshold by 6 dB. This served the function of increasing system reliability in long-term high-power applications and reducing the audibility of amplifier clipping. Frequency response was not changed.

M-1A: Identical to the M-1 except that the level potentiometer was changed to a linear taper. This was due to the high variability in component tolerance of the log pots. The linear pot is calibrated in dB attenuation. It is easier to match levels between units and their restricted range tends to keep the CEU operated in its optimal range.

CEU Upgrades:

For M-1s only: U22 hum reduction ground modification. This reduces the hum on pin 2 of the HF output channel. These are done to any units returned for service.

M-1 to M-1A conversion kit can be purchased (linear pot upgrade).

Compatibility Issues:

UltraMonitor and M-1: Factory only upgrade to reset limit threshold for "Safe" switch.

M-1 and M-1A: The only compatibility issue is the front panel level control. Linear pot upgrade is recommended.

Speakers

650: Original single eighteen inch driver version. Permanent install only. Developed for the movie *Apocalypse Now.*

650-R2: Original and current high-power road version subwoofer. Frequency response range is 30 to 100 Hz.

USW-1: Developed for use with arrays where minimal size is required. Can be flown. Frequency response range is 40 to 100 Hz.

MSW-2: Single MS-18 driver. Companion to MSL-2A (has the same dimensions). Frequency response range is 35 to 100 Hz.

Controllers

650-EM: The original controller for the 650. The first version used predictive limiters and therefore required individual calibration of the controller for each model (and voltage gain) of power amplifier. Later version included sense lines. This was the first product to utilize "SpeakerSense™."

B-1: Successor to 650-EM. Stereo version with similar frequency response. The unbalanced phone jacks outputs proved unpopular, and the single level control for both channels was unable to maintain stereo tracking. This product was quickly discontinued.

B-2: The true successor to 650-EM. Limiting threshold could be controlled by a continuously variable potentiometer with a range of 12 dB.

B-2A Has two summing input channels so that stereo systems with monaural low-frequency information will only need a single B-2A CEU. Additional pole was added to the high pass filter at the bottom end to increase excursion control.

EX excursion card. The trend toward higher power output in modern power amplifiers was most evident in user's desire to drive the subwoofer cabinets with amplifiers with power ratings above those recommended in our literature, creating the need for more sophisticated circuitry to protect the drivers. This began with the introduction of a field-installable retrofit PCB, the "EX" circuit which incorporates an excursion limiter that reacts quickly enough to protect the driver when power amplifiers up to 720 watts into 8Ω are used.

Controllers

B-2AEX: The B-2AEX is identical to a B2-A with the "EX" card.

B-2EX: The B-2EX differs from the B-2AEX in that the excursion protection circuit can be switched out. The limiting threshold can be set to "Safe" which reduces the RMS limiting threshold by 6 dB and engages the excursion limiter. When not in "Safe" the RMS limiter is set to full power and the excursion limiter is defeated. The "Bass Extender," a phase delay network that was in the previous B-2 series products, was deleted for the B-2 EX.

Upgrades : B-2. Hum reduction ground modification, sense line oscillation reduction modification. This is done on any units returned for service.

EX card: can be added to any B-2 or B-2A.

Linear pot upgrade: The B-2EX has a linear taper level control. B-2, B-2A and B-2AEX can be retrofit with the linear pot upgrade.

Compatibility Issues: Frequency response compatibility is not an issue since it has only changed slightly through the entire evolution with steeper highpass filters employed on the B-2A and steeper yet on the B-2AEX This affords improved excursion protection without significantly modifying the audible response.

Important note: To maintain frequency response note the following:

- *The B2-EX frequency response matches the response of all other controllers in the EQ setting, NOT the flat setting.* This change came as the result of recent high-resolution SIM measurements that reveal that the speaker response is flatter in the EQ position.
- Limiting has been changed by addition of the EX circuit.
- 650-EM, B-1, B-2, B-2A *without EX card* are all limit circuit compatible with B-2EX (when not in "Safe").
- B-2 and B-2A *with EX card* are limit circuit compatible with the B-2AEX and B-2EX (in "Safe").
- B-2AEX and B-2EX are limit circuit compatible with B-2AEX "Power" control at 12:00 (–6 dB) and B-2EX in "Safe."

Note: 650-EM and B-1 will *not* take EX cards.

Speakers

500: Low-cost two-way passive system.

500A: Improved HF driver.

500RW: Stage monitor version.

501: Single eighteen-inch subwoofer. Passively crossed.

502: Different enclosure design than the 501. Deeper and shorter but has same volume and tuning.

518: Smaller enclosure than the 502.

Upgrade: 500 to 500A conversion kit available. Not available for 500-RW.

Compatibility Issues: 500 and 500A products are not compatible.

CEU plus Amplifier

M500 amplifier: Original amplifier plus CEU for 500 series speakers.

M500A amplifier: Improved amplifier and CEU for 500A speakers.

Upgrade: M500-M500A conversion kit available.

Compatibility Issues: 500 and 500A products are not compatible.

Speakers

MSL-3: Original MSL-3 system with MS-P4 piezoelectric tweeter array.

MSL-3A: Improvements in HF driver response allowed for the deletion of the MS-P4 array. This yields improved VHF directional control over the MSL-3.

Upgrades:

MSL-3 to MSL-3A conversion kit. Field installable driver retrofit to convert MSL-3s to MSL-3As.

Compatibility Issues: The MSL-3 and MSL-3A will sound quite different above 8 kHz, especially in regard to directional control. If used together, they should be run on separate CEUs. An M-3T will be required for the MSL-3s. The *TC*, *HF* and *EQ* circuit (see below) should be *in* on all MSL-3s and *out* on all MSL-3As.

Controllers

M-3: Original controller for the MSL-3.

TC-3: Time correction unit for piezoelectric tweeter array in the MSL-3. To be used in series with the M-3.

TC-3A: An improved version of the TC-3 with bypass relays and a dynamic range switch to optimize the signal-to-noise ratio.

M-3T: The TC-3 circuitry was incorporated into the M-3. The TC circuit may be bypassed. HF EQ switch included for improved linearity of HF response for MSL-3 in the 8 kHz range. Includes amplifier voltage gain detection circuit. Mute relays are engaged if the amplifier gains is out of the acceptable window (10 to 30 dB). This feature protects the speaker in cases of excess amplifier voltage gain or when the sense lines are inadvertently disengaged. The gain detection circuit is not linked to the limit circuit. It responds only to the voltage gain in the power amplifier. Excess voltage gain causes the limiting circuits to lose effectiveness, compromising driver protection. The limiting thresholds are identical for all models of the M-3 CEU. The gain detectional circuit can be defeated by a field adjustable internal jumper switch.

M-3A: For use only with MSL-3A. The deletion of the MS-P4 made the TC circuit obsolete and it was subsequently removed.

Upgrades: Linear pot upgrade kit is available for M-3s.

Compatibility Issues: The M-3A is functionally identical to the M-3 with these exceptions:

1. The M-3A has amplifier voltage gain detection circuitry (as in the M-3T).

2. The M-3A has the linear taper level control (M-3 has log).

The M-3T can create identical frequency response to M-3 and M-3A by switching *TC, HF* and *EQ* switches *out*.

SIM System II Hardware

2403 Interface Network: The original and current multi-channel switcher for SIM. There are two versions of channel cards. Earlier version did not have a DC offset trimpot for nulling DC on the microphone SPL meter. Early version had tack-on resistors.

Upgrades: Channel PCB mute modification. The original mute circuit can cause distortion measurement errors when engaged with high-level signals. This is described in detail in the field bulletin SIM 2403 Mute Circuit Modification. The modification is performed at Meyer Sound free of charge.

SIM Address Modification: A front panel rotary switch is installed to allow easy access to modifying the SIM address.

ESD Immunity Modification: The 2403 is susceptible to electrostatic discharge (ESD) problems resulting in inadvertent relay triggering. This problem is solved in two steps:

1. Power supply PCB modification (performed at Meyer Sound free of charge).

2. A modification of the existing interface cabling.

Compatibility Issues: None.

CP-10 Parametric Equalizer

CP-10 (pre-1990): Units manufactured before May 1990 had 6 dB of voltage gain when driven push-pull and received by a balanced input. A “cascading jumper “ (an unbalancing cable) was included with the unit, which prevented additional 6 dB buildups to occur when connecting units in series.

CP-10 (post-1990): Later units have a switch that adjusts the gain of the CP-10 for each type of drive. To obtain unity gain through the CP-10:

•If driving and receiving the unit balanced, the switch should be set to "balanced"

•If driving *or* receiving the unit balanced, the switch should be set to "unbalanced"

Note: The CP-10 outputs are *always driven balanced* and the switch affects only the system gain.

CP-10S (1994–present): Security version of CP-10 with screwdriver operated trimpots for all settings.

Upgrade: CP-10 Gain Adjust Modification Kit is available for all pre-1990 CP-10s.

Compatibility Issues: Frequency response of all CP-10 iterations is identical.For gain compatibility between pre- and post-1990 CP-10s the newer unit should be set to "unbalanced."

MST-1

Speakers

MST-1: Original and current. The MST-1 is a high-power tweeter bank (30 piezoelectric tweeters). Highly directional for long-throw HF applications. Protection circuit is internal to the enclosure.

Controllers

T-1: Original controller for MST-1.

T-1A: Improved version has a higher crossover point for improved system linearity.

Upgrade: None available.

Compatibility Issues:T-1 and T-1A are not compatible.

HD-1

HD-1: Original and current version.

Upgrade: Low noise modification. This is an electrical modification that reduces the noise floor of the HD-1 by several dB for critical listening situations.

MS-1000 Power Amplifier

MS-1000 Amplifier: Original power amplifier.

MS-1000A Amplifier: Improvements made regarding in-rush current, mechanical stability and circuit reliability.

MS-10 Amplifier: Modified MS-1000A configured to conform to Type 3 Amplifier specifications for use with MSL-10 systems. 16 dB voltage gain. Amplifier is bridged.

Upgrade:MS-1000 to MS-1000A conversion kit available.

Compatibility Issues:MS-1000 and MS-1000A products are compatible.

Section 7

Appendix

7.1.1 Combining Externally-Powered with Self-Powered Speakers

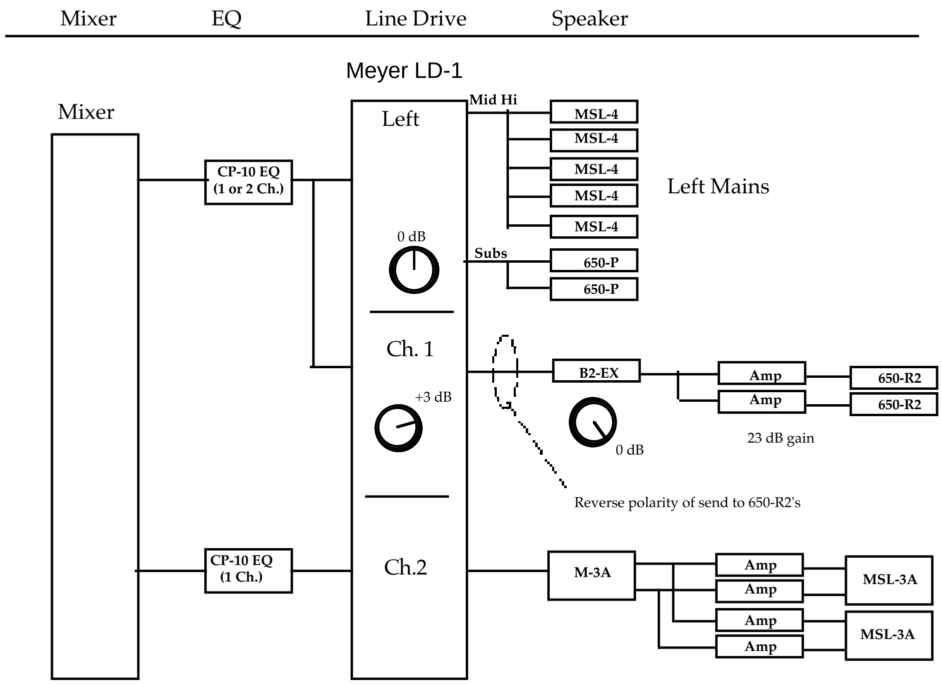


Fig 7.1a Combining externally-powered with self-powered speakers.

The 650-P is not a direct equivalent to the 650-R2. The 650-P was designed to maximize compatibility with the MSL-4. The 650-R2 was designed for compatibility with the MSL-3 and UPA. The result is a change of polarity between the two systems. The voltage gain of the 650-P is equivalent to a 26 dB amplifier with the CEU at 0 dB attenuation.

European Standard

The European amplifier voltage standard is 23 dB (14x).

In order to make the 650s compatible:

1) Reverse polarity of the 650-R2s.

2) Verify that the amplifier voltage gain is 23 dB.

3) Set the B-2 series CEU level to 0 dB.

4) Set the LD-1A drive of the 650-R2s to +3 dB.

The phase response correction of the MSL-4 is much more sophisticated than that of the MSL-3 or UPAs. The MSL-3 should not be placed side-by-side with MSL-4s. They can be used as a vertical downfill where the phase relationship is already physically displaced. They should always be equalized separately.

American Standard

The American amplifier voltage standard is 26 dB (20x).

In order to make the 650s compatible:

1) Reverse polarity of the 650-R2s.

2) Verify that the amplifier voltage gain is 26 dB.

3) Set the B-2 Series CEU level to full (0 dB).

7.2 Meyer Sound Design Verification Checklist

Parameter	Recommended Settings	If recommendations are not followed	Reference
Amplifier Maximum Output Power	Type 1 = 350 @ 8Ω Type 2 = 700 @ 8Ω	Overpowering of speakers reduces reliability. Underpowering reduces max SPL.	Section 1.4.1
Max Power for HF & LF Channels	Matched	If power capabilities are not matched the max SPL of the system will be compromised, (if HF too low) or the reliability decreased (if LF too high)	Section 1.4.6
Amplifier Voltage Gain	23 dB (Europe) 26 dB (all others)	V gain must be standardized for all of your amps. Should be within the range of 20-30 dB.	Section 1.4.2
V Gain for HF & LF Channels	Matched	If V gain is not matched the crossover will not add properly. Result will be phase cancellation at crossover.	Section 1.4.5
Amplifier "Hot" pin	Pin 2 is the standard 2 or 3 is OK - just make sure you know!	If pin configuration is not standardized, polarity reversals will result.	Section 1.4.7
CEU Checklist			
Parameter	**Recommended Settings**	**If recommendations are not followed**	**Reference**
Level Control Setting	0 dB to -12 dB.	If attenuation is too high the stages that drive the CEU may overload	Section 1.2.5
Sense LED	Lights green with moderate signal input	If not lighting green there is a "sense fault". (1) Sense line not hooked up, (2) Amp gain out of range (10-30 dB), (3) HF & LF Sense lines reversed.	Section 1.3.6
Limit LED	Lights occasionally under high power signal conditions	Limiters should not light continuously. This will cause driver overheating.	Section 1.3.6
Safe Switch	"In" if system is to be operated by amateurs.	Decreased life expectancy of drivers	Section 1.3.6
	"In" if the amplifier maximum power ratings are above the recommendation.	Decreased life expectancy of drivers	Section 1.3.6
	"In" if the system is to be run into continuous overload.	Decreased life expectancy of drivers	Section 1.3.6
	"Out" if the system is run responsibly and maximum dynamic range is desired.	You are giving up usable dynamic range.	Section 1.3.6
Lo cut Switch	"Out" if no subwoofers are used.	Thin sound	Section 1.2.6
	"Out" if main cabinets are flown and subwoofers on the ground.	Distinct MF and LF sound images. Not as natural sounding.	Section 1.2.6
	"In" if main cabinets are directly coupled to subs.	Boost in low midrange. Easily EQ'd.	Section 1.2.6
Speaker Checklist			
Parameter	**Recommended Settings**	**If recommendations are not followed**	**Reference**
LF Polarity Battery Check	LF driver will move forward when positive DC voltage is applied to the "+" pin.	LF polarity reversal	Section 4.9.2
Crossover addition	There should be measurable addition through crossover if the HF and LF drivers both have proper polarity	HF polarity reversal causes crossover cancellation	Sections 4.9.3 & 4.10
Multiple speaker addition	When two speakers are placed adjacent they should add 6 dB at the center, indicating that polarity of all drivers are matched.	Broadband cancellation indicates that the two speakers are fully reversed. If only the LF adds then one of the HF drivers is reversed. (And vice-versa).	Section 4.9.3
Subwoofer Addition	Subwoofer addition is dependent upon the relative placement of the Subs to the other enclosures. Try both polarities and see which one adds best at crossover. This should be done on a case by case basis.	Crossover cancellation.	Section 4.9.6

Table 7.3 Meyer Sound checklist.

This chart serves as a quick reference for design and verification of your speaker system.

About the Author

Bob McCarthy has been involved in professional sound reinforcement since 1978. An audio engineering graduate of Indiana University, he has toured around the world with major recording artists. He served as Director of SIM Engineering for Meyer Sound and was instrumental in the design of SIM and the CP-10 Complimentary Phase Equalizer. He has regularly taught the SIM Engineering training course since 1986. He currently works as an independent consultant specializing in audio system design and alignment. He lives with his wife and two children in St. Louis, MO.