Loudspeaker Array Case Study

The need for intelligibility

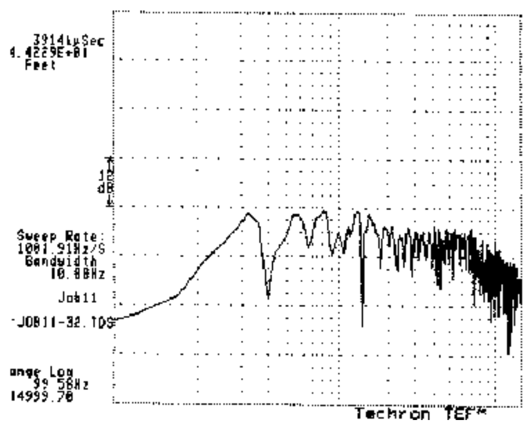
Churches, theatres and schools are the most demanding applications for speech intelligibility. The whole point of being in these facilities is communication through spoken word. There has been a definite trend towards using small, two-way concert sound speaker boxes for speech reinforcement loudspeaker clusters in facilities of this type, and the resulting speech reinforcement performance is often substandard.

Any single loudspeaker may measure to have exceptional response, and the sound quality of a single loudspeaker may be nothing short of hi-fidelity, but that will not be the case once arrayed. With these direct-radiator packaged speaker systems grouped in a cluster, the individual drivers can not be positioned to minimize time delays between devices and the resulting destructive interference, cancellation and response variations. These variations are important because the coverage uniformity in the 1,000Hz to 4,000Hz range is critical for good intelligibility and uniform frequency response from 250Hz to 1,000Hz is important for natural sounding speech.

Where the reverb time of the room extends beyond 1.5 seconds, the issue of coverage uniformity and related directivity control is even more critical as it affects both speech intelligibility and gain before feedback. These same variations may add a sense of spaciousness to music playback, but music playback is not the issue we are talking about here.

A Real-world Example

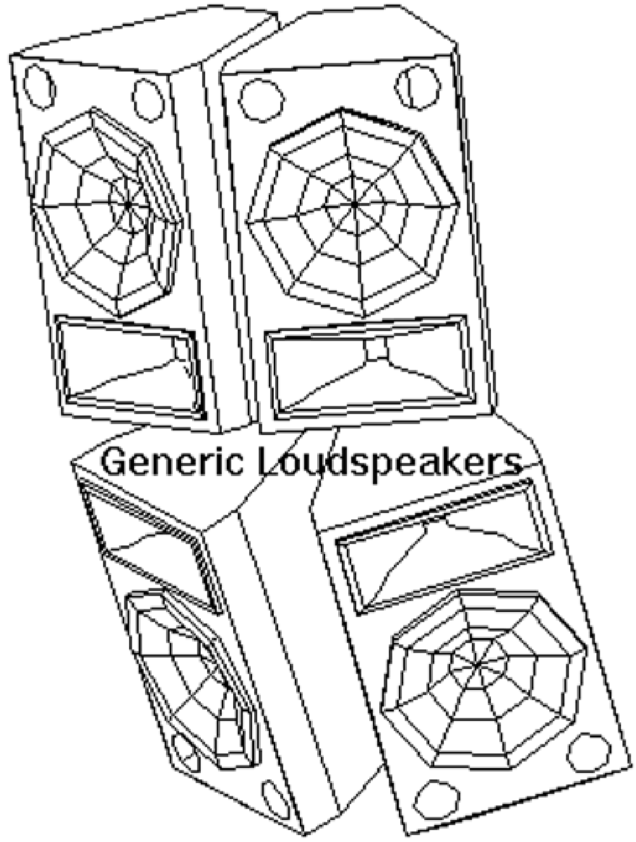
We had a chance to measure an example of just such an array; a two wide by two high cluster of 12" two-way boxes, very typical of the most popular of small concert sound boxes, in a typical mounting configuration. We hasten to add that we did not design this system, we were asked to make these measurements for the facility owner after a newly installed sound system did not meet the owner's expectations. In this example, the boxes were hung by a contractor using the manufacturer's processor, mounting hardware, and their recommended mounting configuration. The individual boxes sounded good and measured well when used one at a time. The response variations you will see here are not specifically the fault of the sound contractor that installed the system, or a problem with the loudspeaker quality, this is a fundamental problem created by the laws of physics.



Response of single speaker measured on axis

The Laws of Physics

The problem is related to the physical distance between individual loudspeaker components. The special signal processor/crossover that comes with the loudspeaker can only correct for time offset of the signal between the low and high frequency drivers along a single speaker system axis. The signal delay in the processor cannot adjust for every possible off-axis listening angle, especially in the vertical plane, where the physical offset between high and low frequency devices changes the signal delay for various listening positions by fractions of a millisecond. The processor certainly can't fix the additional multiple signal delay variables thrown in by having additional boxes added to the mix.



In the four box cluster shown here, there was approximately 30" between the box centres, both horizontally and vertically, and this was the manufacturer's minimum interference configuration (That's a half wavelength at 225Hz, one wavelength at 450Hz, two wavelengths at 900Hz). At the crossover point, where the horn and the 12" bass speaker in each box are operating at equal levels, there are a total of eight devices operating at the same level with multiple wavelength spacing between all eight sound sources. This is the offset distance vertically, horizontally and diagonally between all devices. Where the coverage of the boxes overlap, there will be significant lobing of the coverage and comb filtering caused by these physical-offset-induced signal delays.

The problem is compounded by the inability of the small format high frequency horns that are used in these boxes provide high directivity in the vertical plane near the crossover frequency (the laws of physics again). The actual vertical coverage of the horn can be well over 100 degrees (and can approach 180 degrees) at the crossover point, ensuring vertical overlap of coverage between boxes, and destructive interference and cancellation. Above the crossover point, the comb filtering will fall in the critical speech intelligibility range, and the resulting response notches are often in excess of 1/3 octave wide.

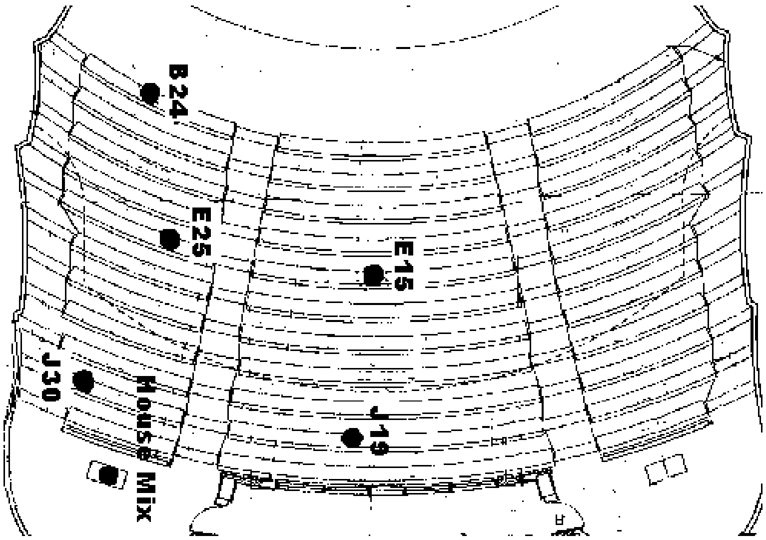
Below the crossover point, the 12" bass drivers gradually broaden their coverage angle until the boxes each become omnidirectional. In the frequency range between 250Hz and 1000Hz, very large variations in level will be found, with the highest level exactly on the centre axis of the cluster, and spurious lobes all around that centre hot spot. This produces plainly audible sound quality variations with seating position, and may produce feedback prone positions under the cluster.

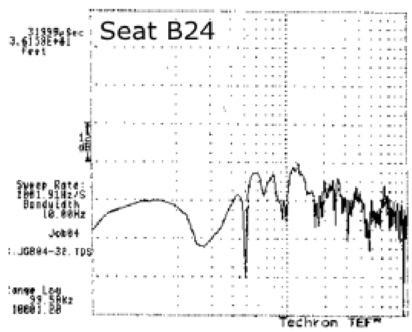
Measured coverage

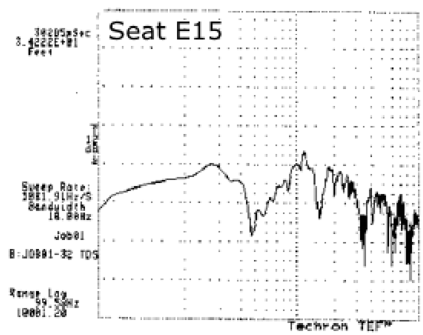
Let's take a look at the measured coverage of the main floor and balcony of a real room equipped with this four box cluster. These frequency response graphs were acquired with a TEF analyzer, the time window set for 10 milliseconds to capture the direct sound from the speaker with 100Hz resolution. The lowest frequency shown in all the graphs is 100Hz, the highest either 10kHz or 15kHz. The major vertical divisions are 12dB. Note that these response graphs represent an actual measured installation, not a computer model...this is real life folks!

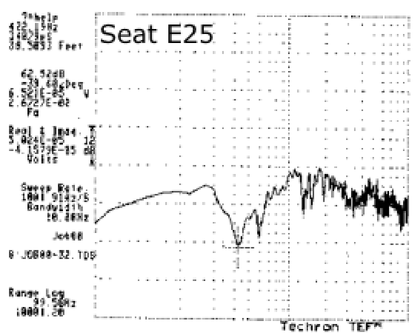
Graphs begin on next page.

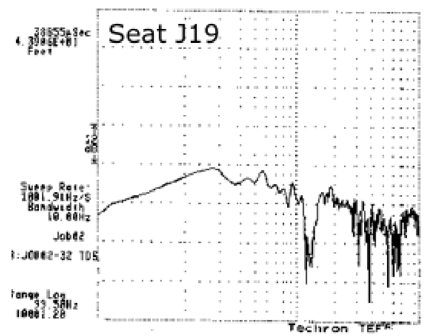
Main floor coverage

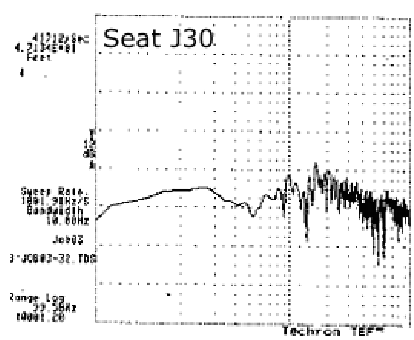


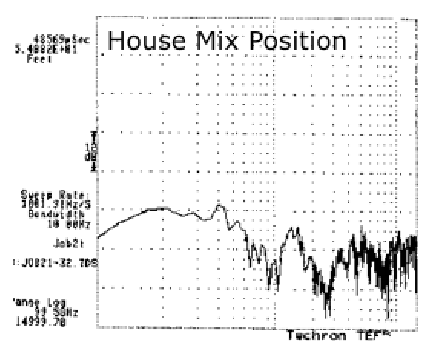




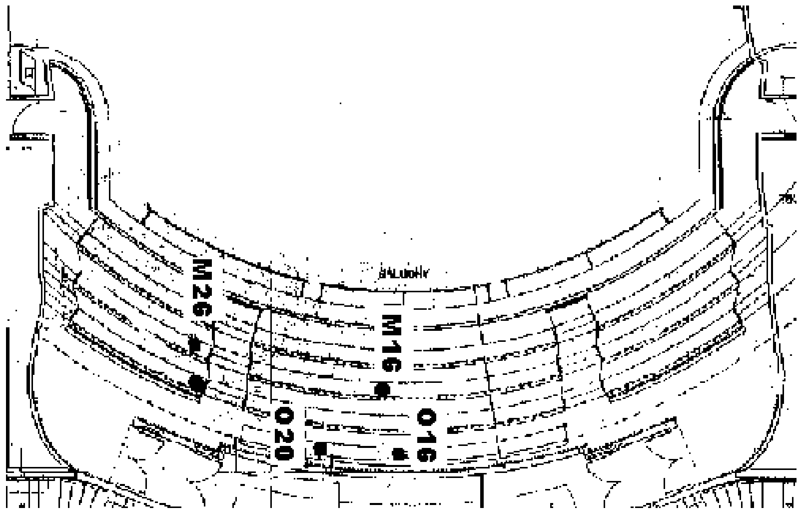


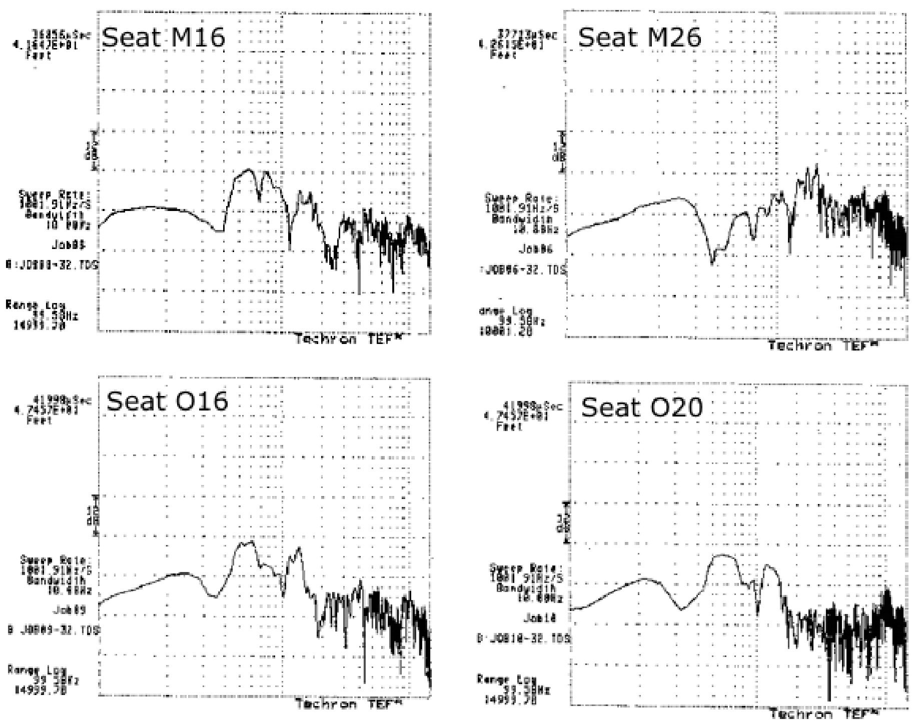






Balcony coverage





So why don't you just equalize it?

Note the extreme variations in frequency response through the seating area, every seat looks and most importantly, sounds different. This room had a very short reverb time, so these measurements accurately represent the audible variations in sound. Some of these response curves have over 24dB of maximum to minimum variation.

Consider the system equalizer that is intended to improve the quality of the sound. All four loudspeakers are connected to the same signal chain, so any change in equalization will affect all four speakers. Which of these frequency response curves would you correct for? A couple of the seats are not too bad, a couple of them (including the house mix position!) are very poor. If you tried to correct the bumps and dips in one location, you would actually be making them worse in other areas. This is really a non-equalizable problem caused by physical distance offset of all the drivers (this is what is known as a non-minimum phase problem caused by a significant signal delay). Even though the loudspeakers are designed to be arrayed horizontally, there is no way to get around the physical offset problem vertically with small packaged two way direct-radiator systems. There are no delays or equalizers that will fix this problem (until someone develops a four dimensional signal processor).

In this particular configuration, the problems are most severe below the crossover point, where there are four 12" loudspeakers widely spaced. The 12" speakers cannot be positioned to minimize the interference between them because the boxes are pre-configured. By the time you get the high frequency horns pointed where they have to be pointed, the woofers are typically in the worst possible relationship to each other. The laws of physics triumph every time. In fact in this particular instance, the only available correction turned out to be turning off two of the 12" speakers.

With the amount of DSP horsepower that is currently available you could get fancy with frequency shading and using multiple amplifier channels to drive each device separately and adjust level and frequency response individually for each device, but that will take some sophisticated measurement equipment to set up.

What are the alternatives?

A high performance loudspeaker cluster can still be constructed in a cost-effective manner from high frequency horns and low frequency boxes (affectionately known as a Junkyard Cluster). It is also possible to use fully horn loaded full-range loudspeaker systems that provide broadband directivity control. It is possible to combine these two types of devices as well.

The important element of the speaker cluster design process is to determine the bandwidth (frequency response) and the maximum sound level requirements throughout the listening area and then select the minimum number of devices that can provide the needed level, bandwidth and uniformity of coverage. This is where the science has to be applied, as the selection and alignment of speaker devices in the cluster is critical to the completed cluster's performance. It is vitally important that the system designer and installer fully understand the coverage characteristics of loudspeakers used in a cluster, and the effects of grouping devices together. It is also important to understand the effects of physically mis-aligned loudspeakers, and to be able to measure and adjust the time-offset of devices. The small packaged boxes described above may relieve the buyer of the need to think about the selection of devices, but that doesn't alter the final results as we've shown here. Cluster design can't be left to a manufacturer's marketing department!

How did we get here?

The original market for small packaged speaker systems grew from the sound system rental market, where it was important that the systems be easily transportable, reliable, and could make music sound nice for the client. The systems were typically used in more reverberant spaces where the reverberant field in the room would "fill in" the response variations. This works fine for music playback, but does not work for speech reinforcement.

An ideal speech reinforcement system should be invisible to the listeners, the voice should sound completely natural and not sound amplified, it should just be louder to effectively move the talker and listener closer together. An ideal system would also preserve or improve the direct-to-reverberant ratio of the reinforced sound, but that's another issue. The marketing departments have convinced many consultants, clients and contractors that these small systems are a panacea, a universal problem solver for sound systems. There are many examples of installations using this type of speaker system for speech reinforcement, and yet when you listen to a voice through the final result, the deficiencies are often very apparent. Somewhere in all of this marketing push, people have quit listening to what the finished systems sound like.

Don't get fooled

Many of these small rock and roll speaker rigs sound very impressive when demonstrated with a CD of pre-recorded music. Most listeners are much more forgiving of response variations when listening to music. If you're listening to a system intended for speech reinforcement, insist that it be demonstrated with a live microphone in the room. Have someone you know speak into the microphone while you walk around. Does their voice sound the same in all the seats? Does it sound like the natural voice of the person talking or does it sound different? A proper speech reinforcement system should sound perfectly natural, not bright or sibilant and not boomy or bassy. Make sure that it doesn't sound "better" than the person speaking, as that gets tired pretty quickly and it won't work for all voices. Is there reasonable gain before feedback? If the speaker system is hung above the microphone, are there bass lobes aimed at the microphone position that will induce feedback? Above all else, is the system intelligible in all the seats? It is important to ask the right questions during a demonstration of a speech reinforcement system, this is not the same as buying a stereo system. When you're buying a speech reinforcement system, listen for how well it delivers the speech intelligibility your audience needs and don't get fooled by marketing and music.