Difficult acoustic environments?
Maintaining voice intelligibility

Measurement Conventions

Speech transmission index, Alcons and all the others

Measurement Conventions

% Alcons, STI or C50?

%ALcons = Articulation Loss of Consonants expressed as a percentage (< 10 % intelligibility is very good)

STI = Speech Transmission Index Speech Intelligibility expressed by a single number value

C50 = Ratio of total energy occurring in the first 50ms to the total sound energy of the impulse response (>3dB is good)

Measurement Conventions

Which is better?

% Alcons was developed primarily as predictive technique

STI was developed as a measurement method

C50 is another system for determining clarity

As time has progressed there have been attempts to automate the process

Measurement Conventions

Speech Intelligibility may be expressed by a single number value. Two scales are most commonly used: STI and CIS (Common Intelligibility Scale)

STI predicts the likelihood of syllables, words and sentences being comprehended

Common Intelligibility Scale (CIS), based on a mathematical relation with STI - This measure is called the Speech **Intelligibility** Index, or SII.

Measurement Conventions

STI	0 - 0.3	0.3 - 0.45	0.45 - 0.6	0.60 - 0.75	0.75 - 1.0
	unacceptable	poor	fair	good	excellent
ALcons	100 - 33%	33 - 15%	15 - 7%	7 - 3%	3 - 0%

%ALcons < 10 % intelligibility is very good
%ALcons < 15 % intelligibility is acceptable
%ALcons > 15 % intelligibility will be a problem

C50 - Clarity index is the ratio of early to late sound energy in a room impulse response, expressed in decibels. The variants of **clarity index C50** and **C80** are commonly used in room acoustics, where 50 ms and 80 ms are taken as the respective boundaries between early and late energy.

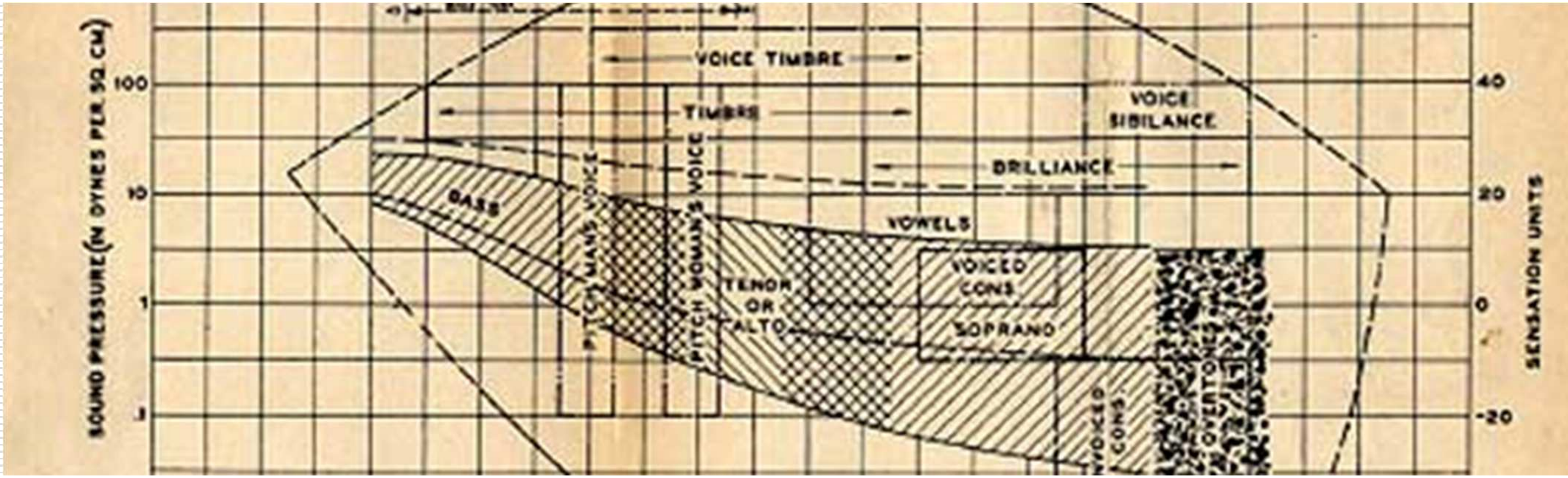
Measurement Conventions

Testing now and then

Measurement Conventions

Testing now and then

Sensation Units?



There are now a few systems to automate the process

Also plenty of detractors that claim some top venues fail the ISO3382 standard tests

ISO3382 analyses are based on a crude model of hearing but it suits automating the measurement system in most environments

People will argue the human ear is still the best measurement device for this purpose

“An opera without words is just a silent movie”

Standardising the measurements

RASTI : Rapid Speech Transmission Index : is an objective way of measuring speech intelligibility. It is measured by placing a loudspeaker, which transmits sound from the location of a person speaking, and a microphone where the listeners are situated

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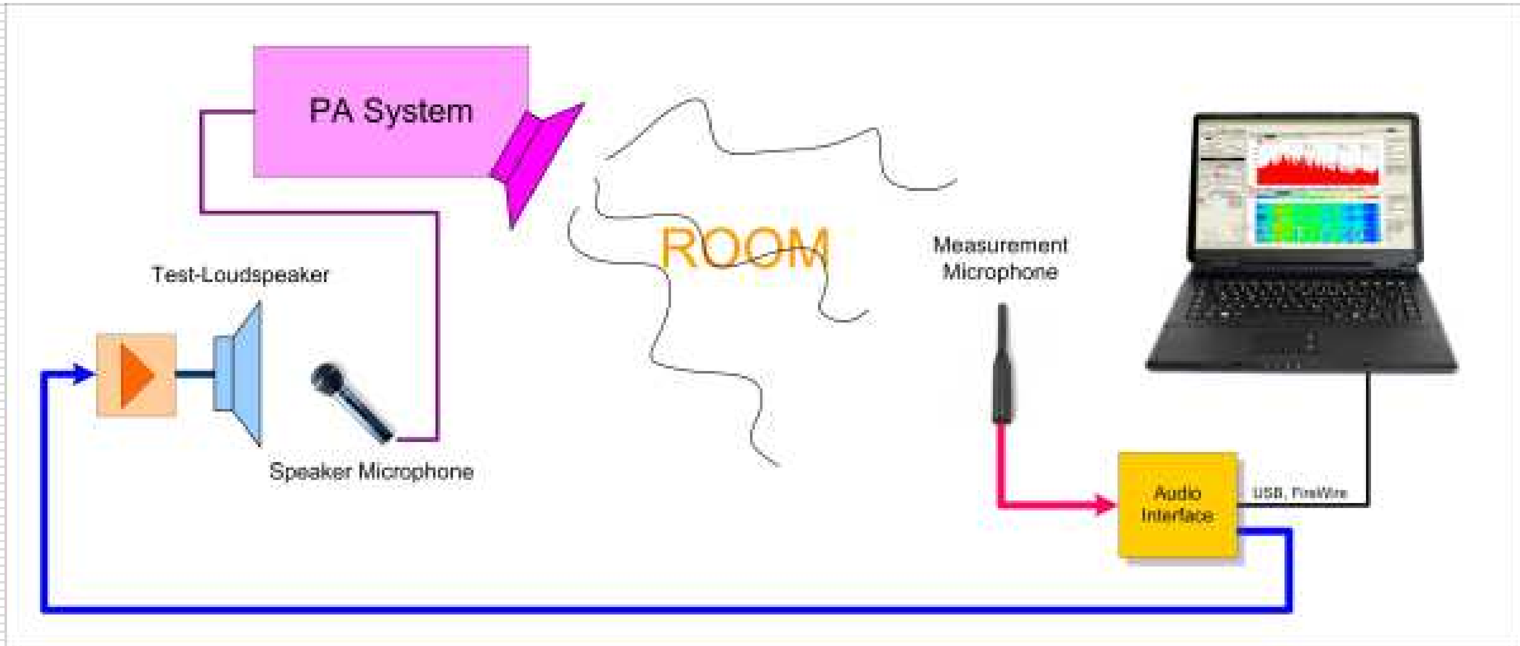
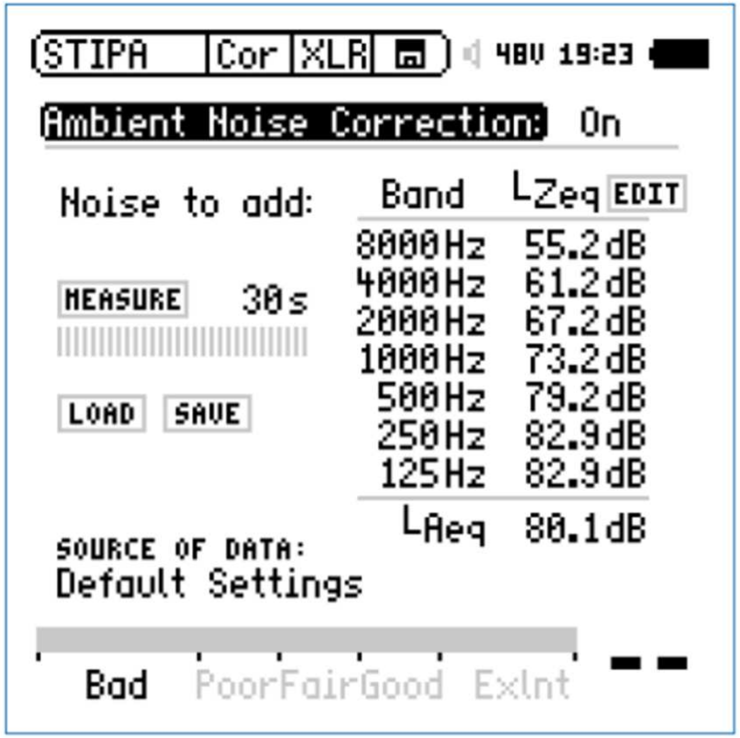


Influence of Reverberation on speech

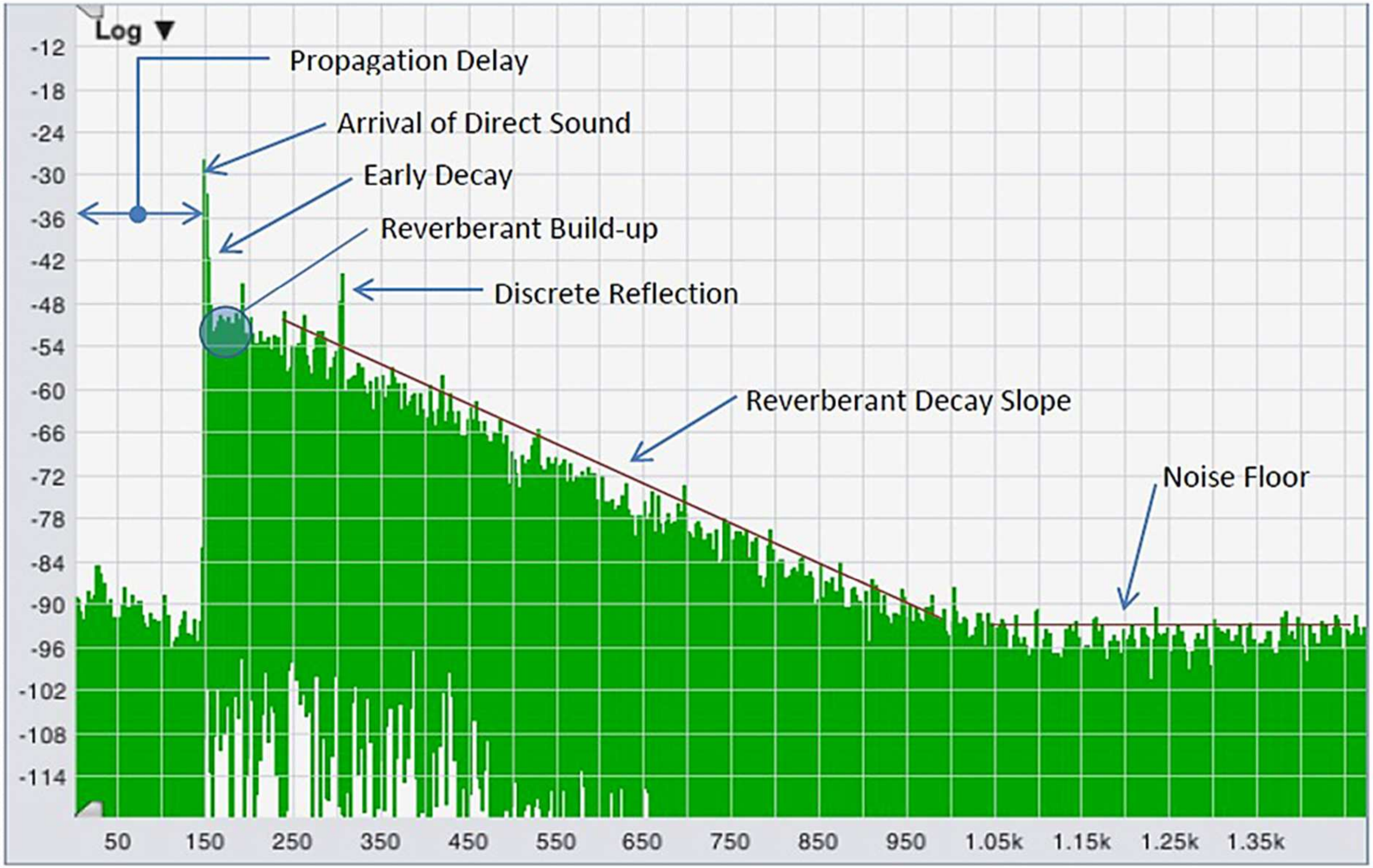
Modulation Transfer Function (MTF)



Introducing the venue

For our purposes we will focus on the room and it's reaction to an installed speaker system

Influence of Reverberation on speech



Was this the birthplace of audio as a Geometric problem?

In the early days, knowledge of acoustics and building design did the job. The word Auditorium is a Latin word meaning **'a place of hearing'.**

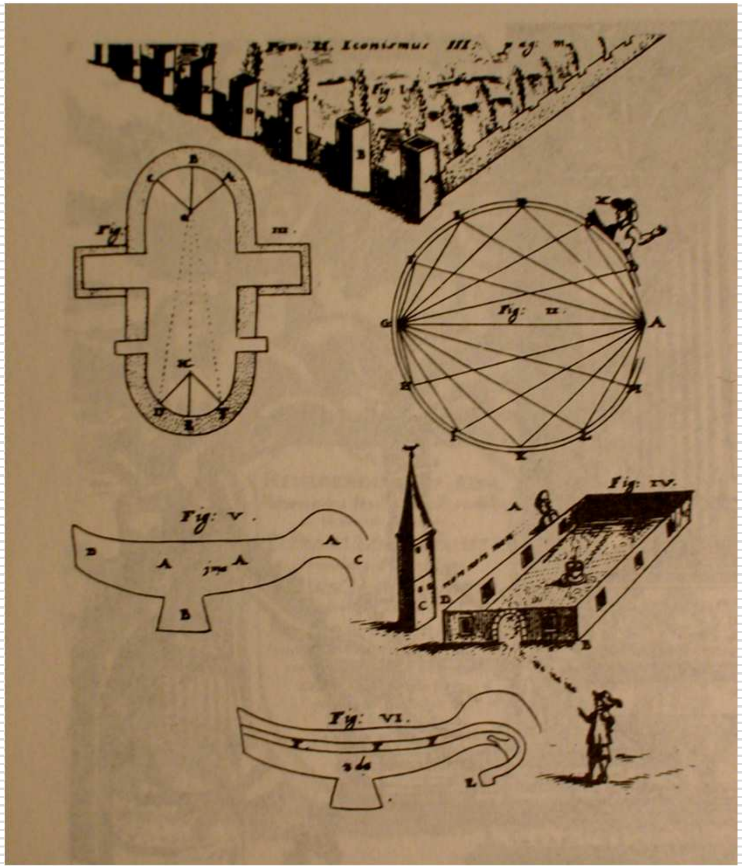
Trivia question:

When was the first recorded published work that exclusively focused on acoustics for the purpose of understanding the propagation of sound waves?

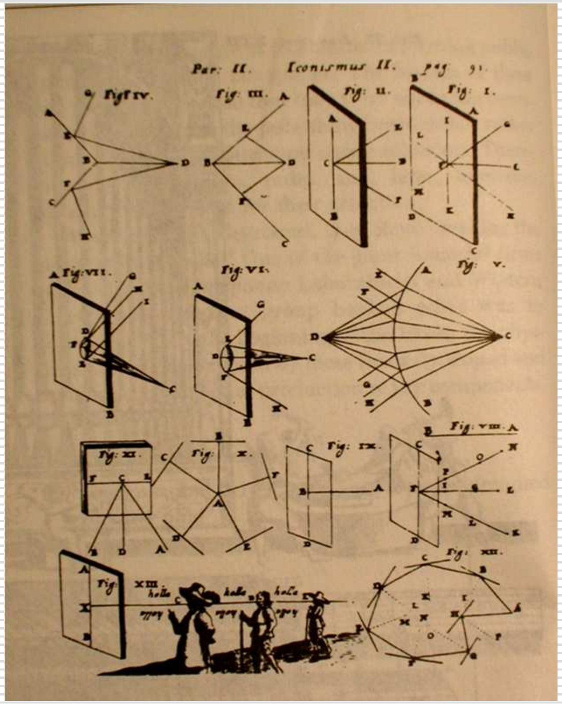
Answer:

- 1657. The book *Magiae Universalis* by Gaspare P Schotto. (1608 – 1666) Written in Latin and published in Germany.
- Gaspare was one of the few early researchers on acoustics whose work is still around today.
- He was the first recorded person since the early Greeks to consider sound behaviour and distribution as a largely geometric problem.

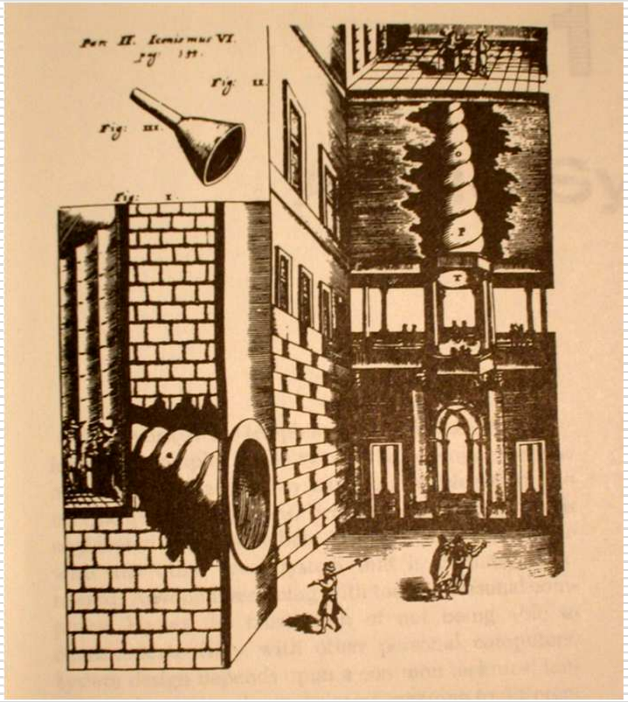
Illustration of reflections and sound wave behaviour within a semicircular reflective environment such as under a dome or within a parabolic stone wall.



Probably not the first to treat Acoustics as a problem in geometry

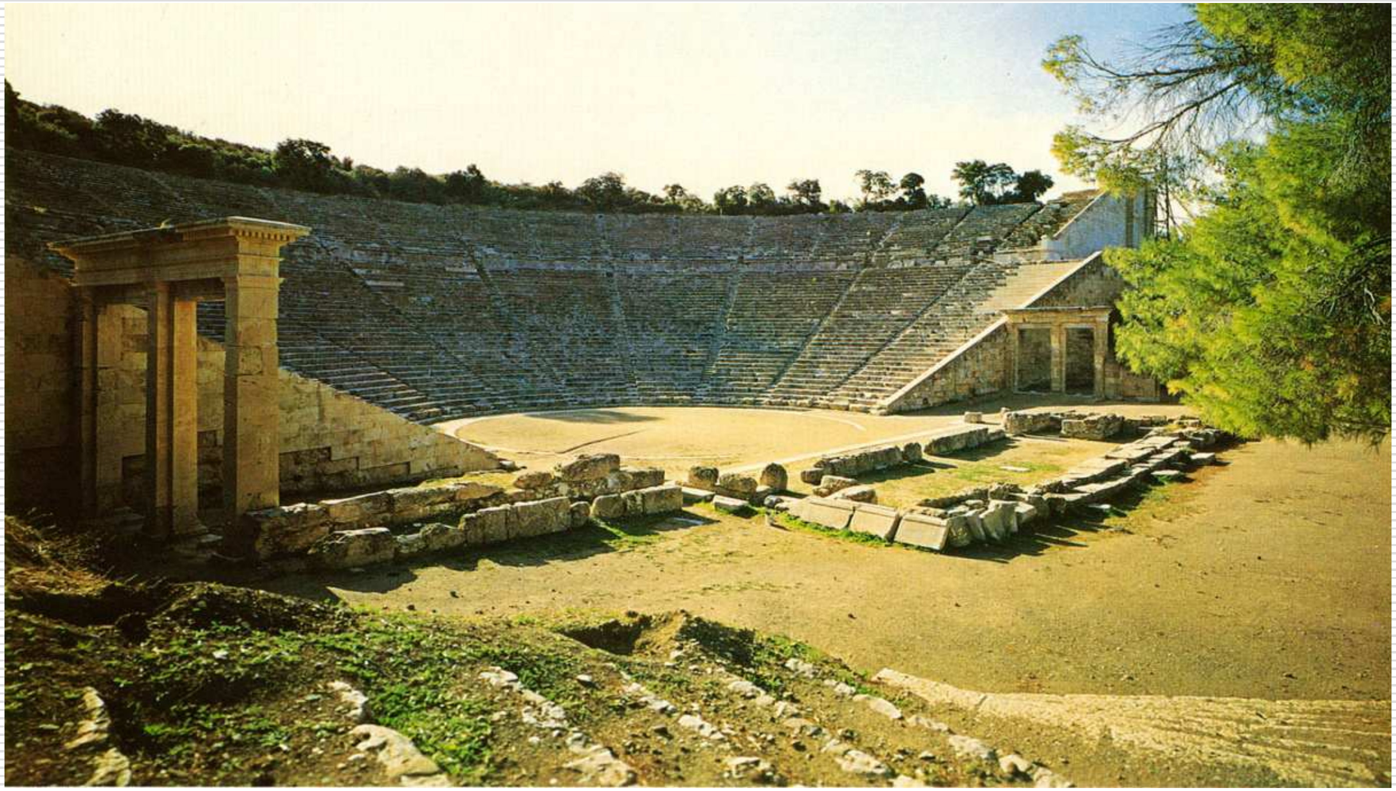


A graphic representation of the rules regarding reflective sound expressed as a geometric problem.



A post medieval mechanical sound transmission and amplification system featuring a large horn to direct and amplify speech and music.

The acoustic principles of the Greek theatre verses the Gothic Cathedral (600AD ~ 1970AD)



The Gothic message … The atmospheric

- Cologne Cathedral was commenced in about AD 1250 and completed 500 years later

The last 100 years

- Modern sound reinforcement began in 1915 thanks to political necessity when a telephone receiver and a phonographic horn was used to announce a USA presidential inauguration to a large crowd.

"If the first attempt in 1915 was primitive, by 1920 the convention had become a 'big PA'

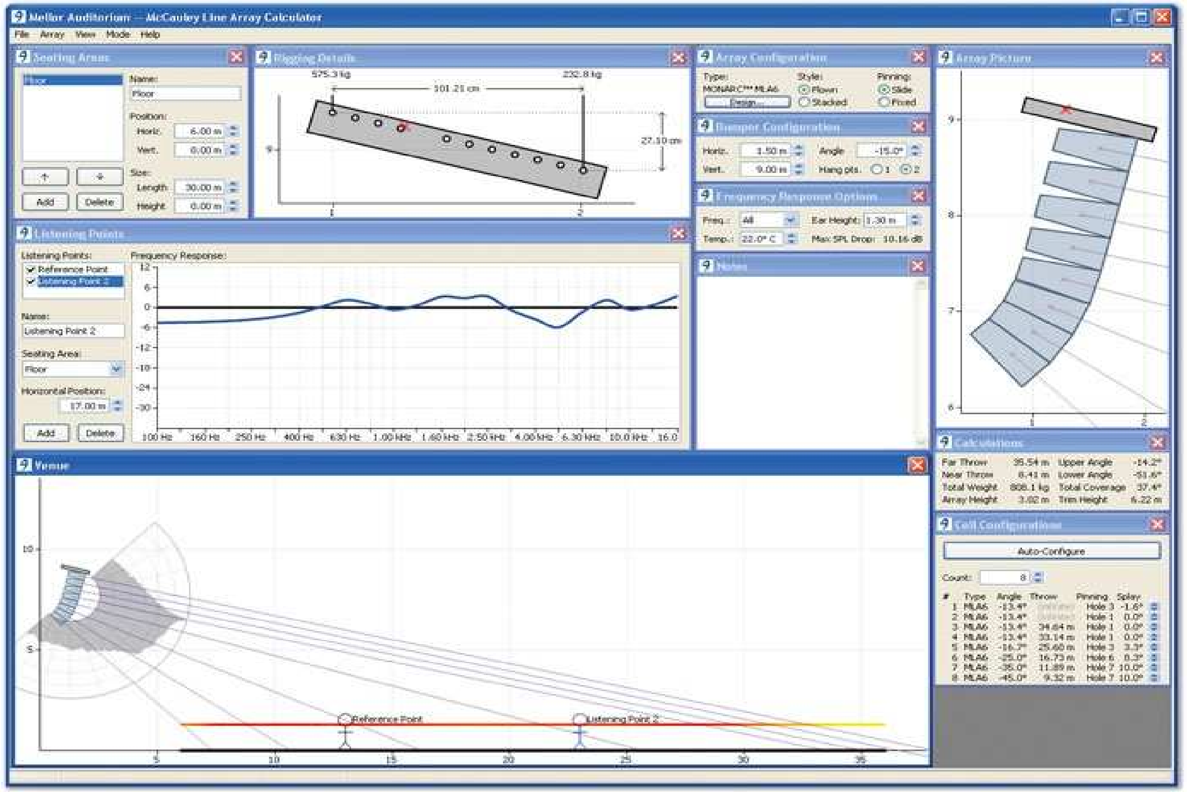
The present Era

The present Era

The present Era

The present Era

The line array concept has almost become a faith when it comes to the “go too” installation concept – and for some good reasons

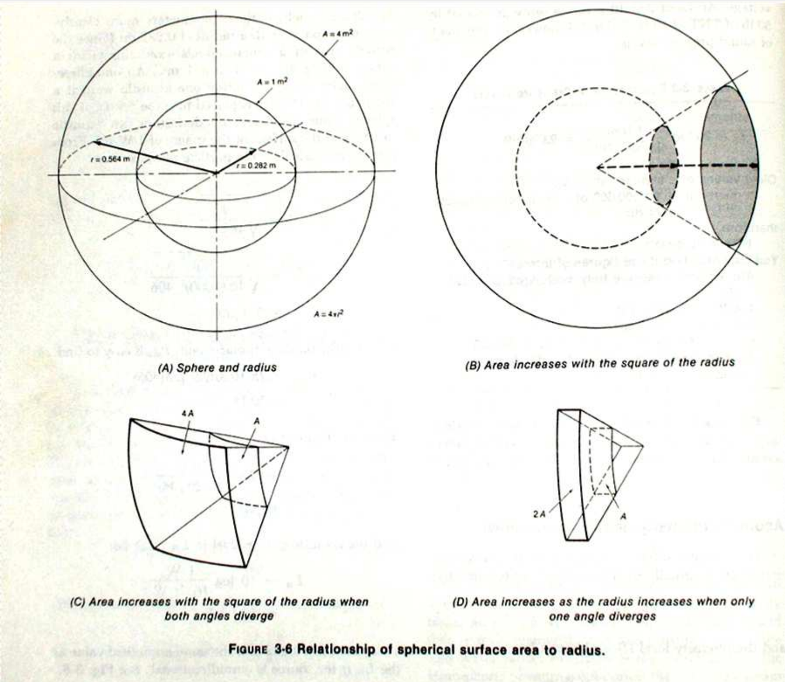


Is The Hype Justified?

Yes and no
(Horses for courses)

The inverse square law

- For decibels or dB to be useful for predicting sound coverage, you need to understand the **inverse square law**
- It is a ratio that a change of one unit more or less in quantity, will double the result.



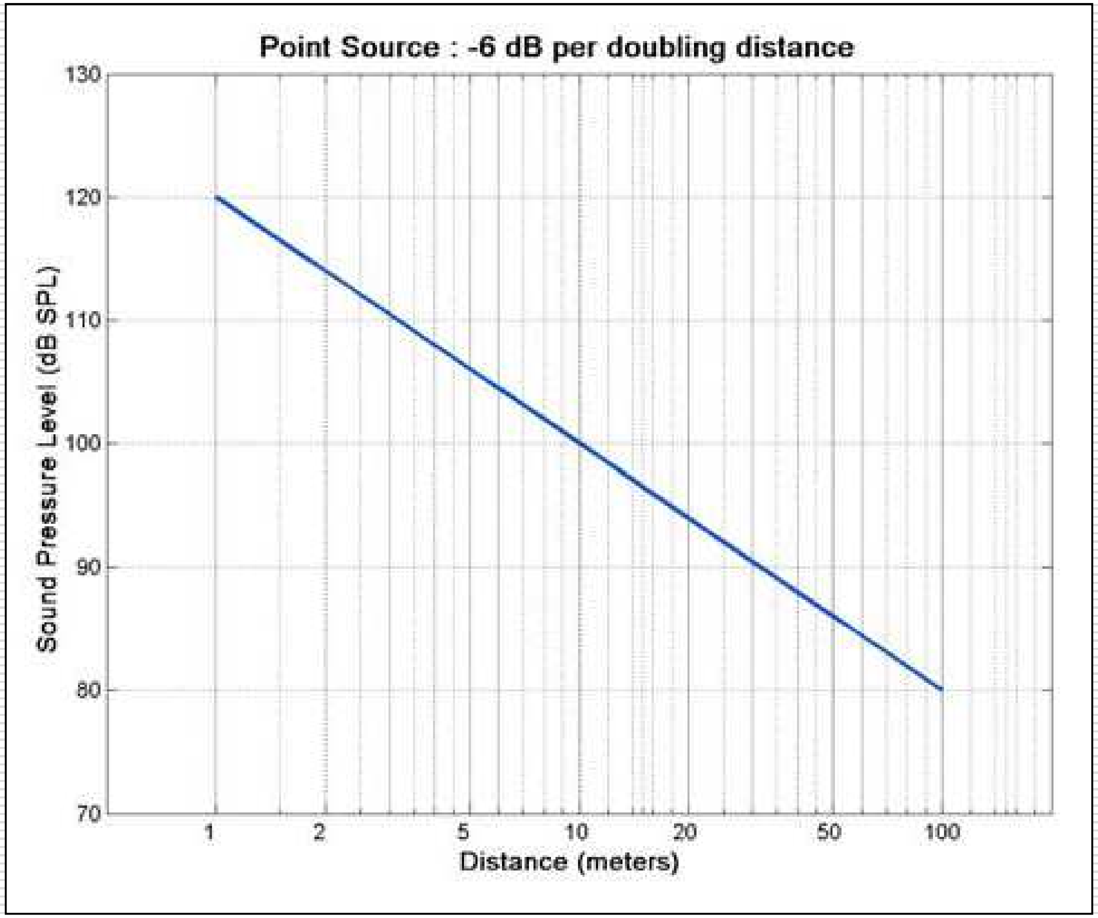
A sound source in an open space streams out uniformly in all directions.

Imagine a light bulb and measure the light intensity at 1 metre.

At 2 metres the light intensity will be one quarter the light level at 1 metre.

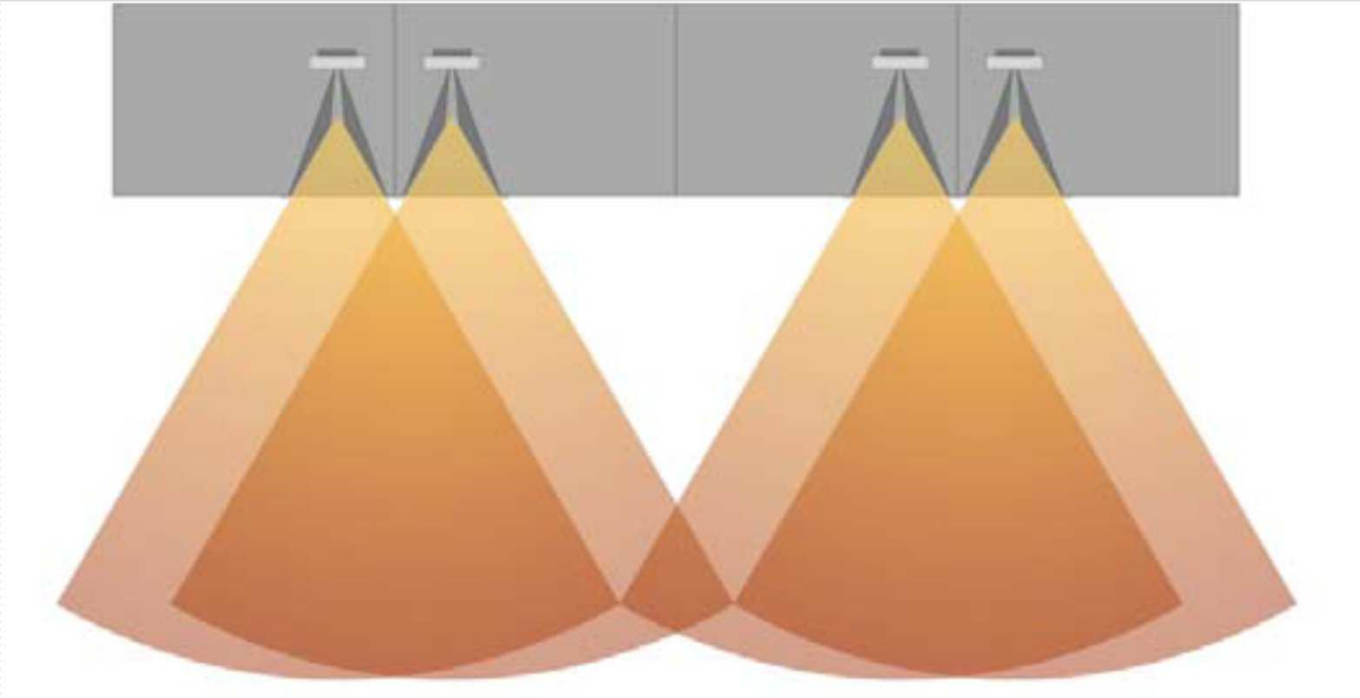
Why a multi speaker array?

- The extremely high output of modern sound reinforcement loudspeakers is required because most of them radiate as point sources and therefore obey the “inverse square law.”
- That is, their output drops –6 dB for each doubling of distance.



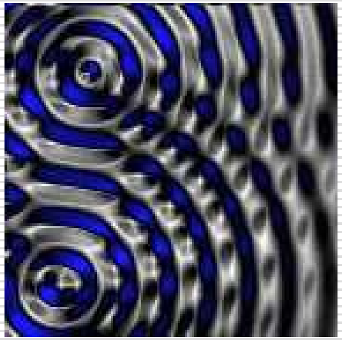
The ear and the big PA (Just two boxes together big)

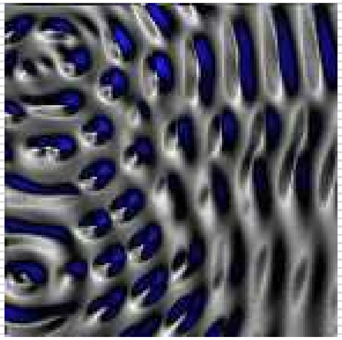
- Our first concert situation is an array of two full range boxes each side of the stage at an outdoor festival. It sounds like a simple situation.



Rule # 3

- **The successful audio engineer will work with the acoustic environment, not against it. To do this successfully, he will understand the behaviour of sound waves and how we hear.**





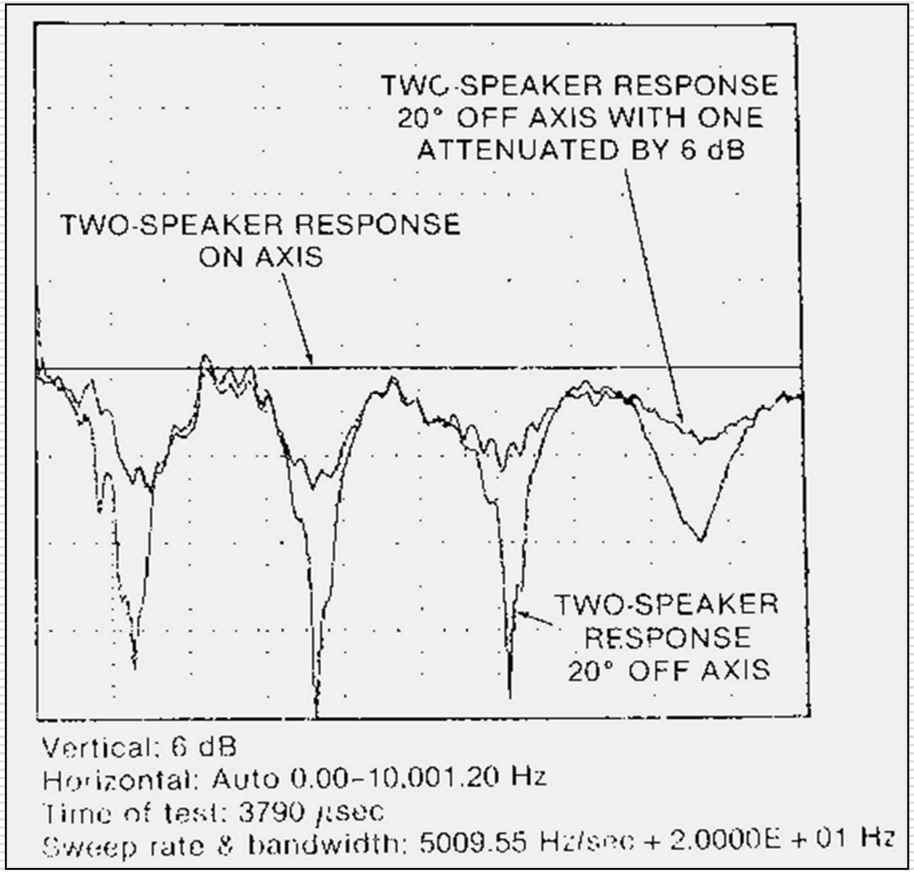
Practical Example

- throwing a stone into a pond. Even waves spread out in all directions, their energy dissipated uniformly
- throw a handful of gravel instead of a small stone, there will be a disturbed wave flow right from the first splash.
- Imagine one speaker source as the small stone and an array of speakers as the hand full of gravel.

Comb Filtering

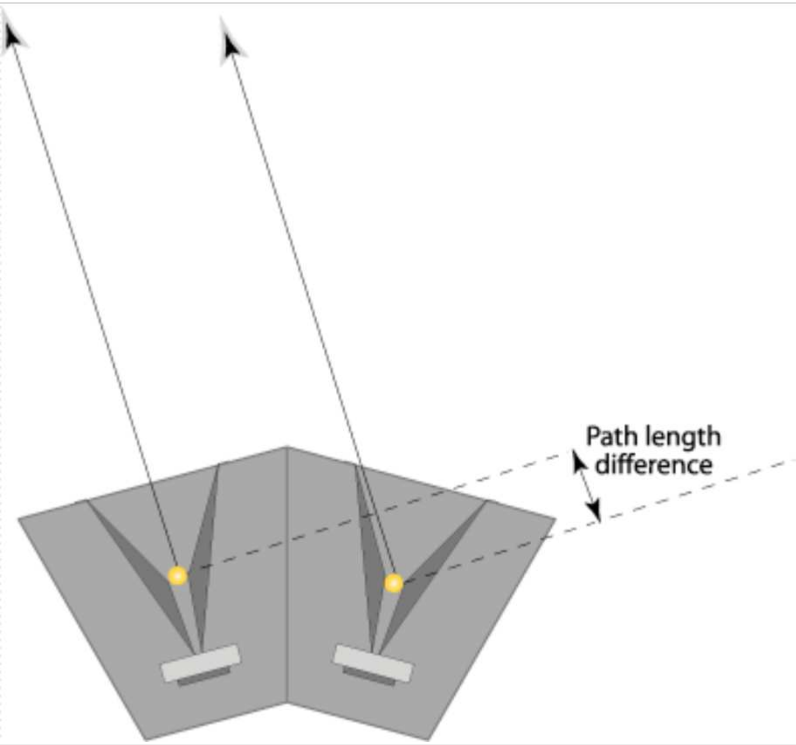
- There is a practical clue for a sound guy about arraying a big PA, can you figure it out?

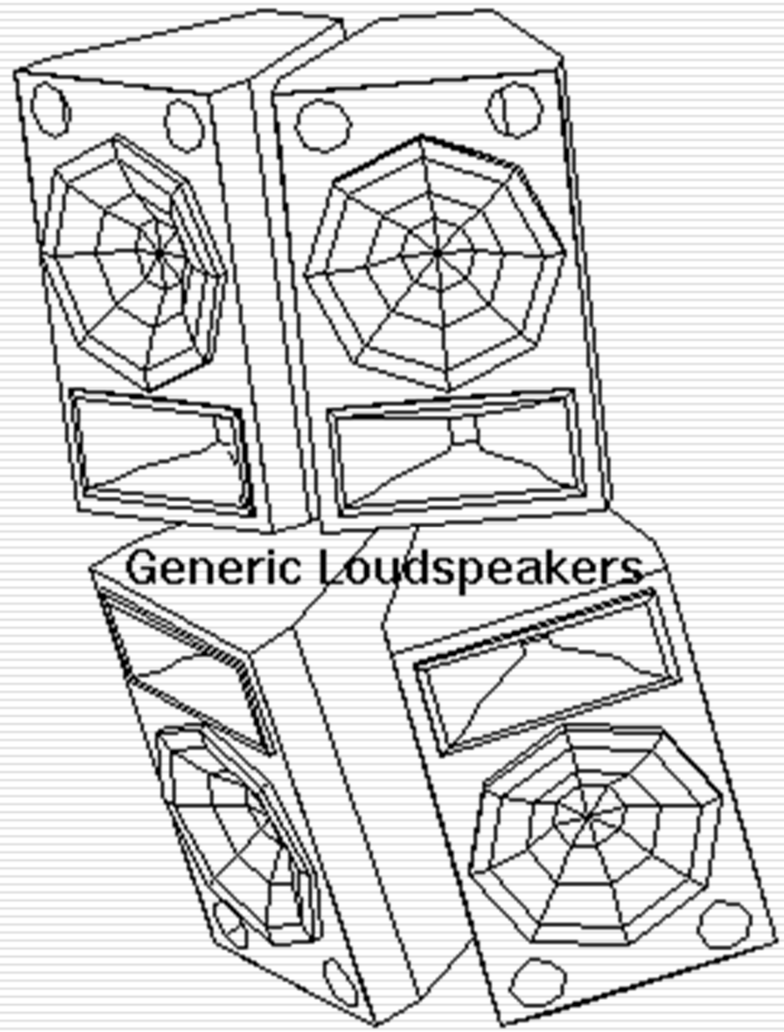
- We now have an understanding of the dB scale, frequency and wave length, loudness contours, the inverse square law and comb filtering,   let's apply this to a real PA array installation.



So let's add more boxes to our two box array? Maybe that will help

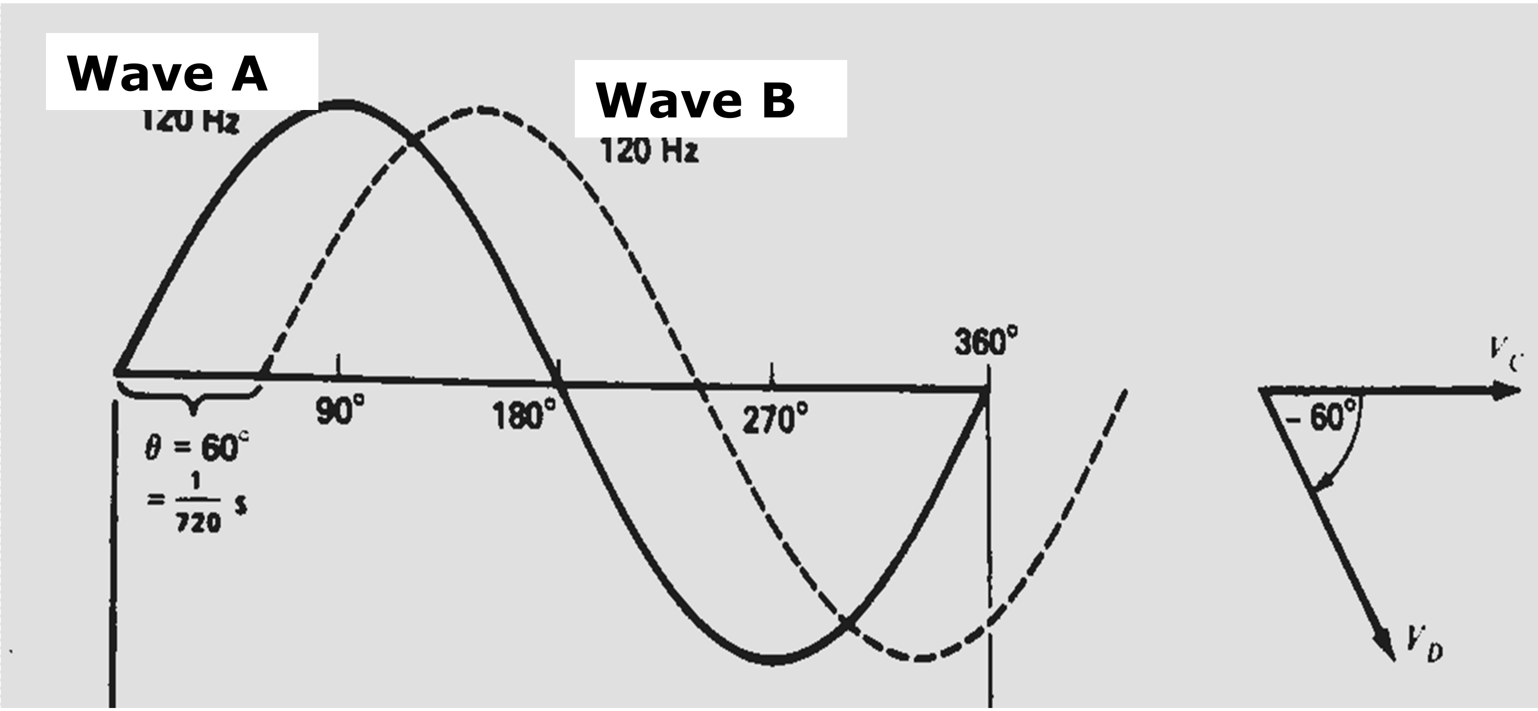
- We have gone from throwing four pebbles into the pond, (two per hand full), to eight.





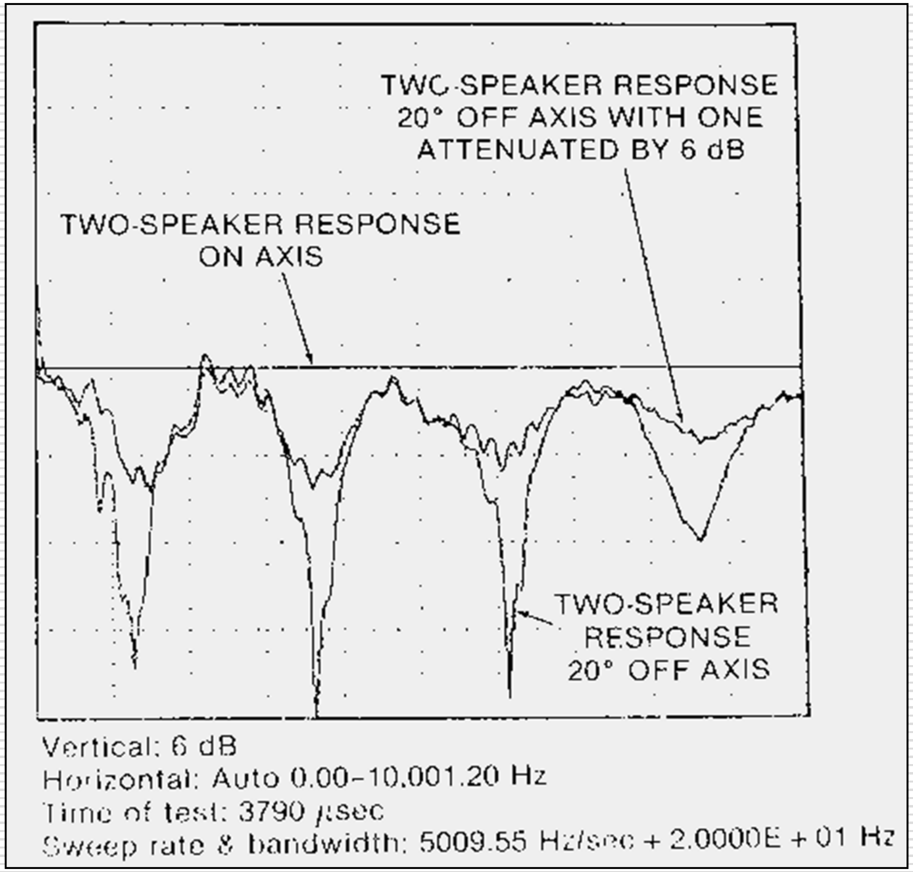
A graphic representation

- **Wave A** is now cutting across **Wave B** out of time sync. The slight time variation (60 degrees) will mean that there is a pressure lag between the waves and they will be working against each other. It's like a tug of war with everybody pulling at a slightly different time.



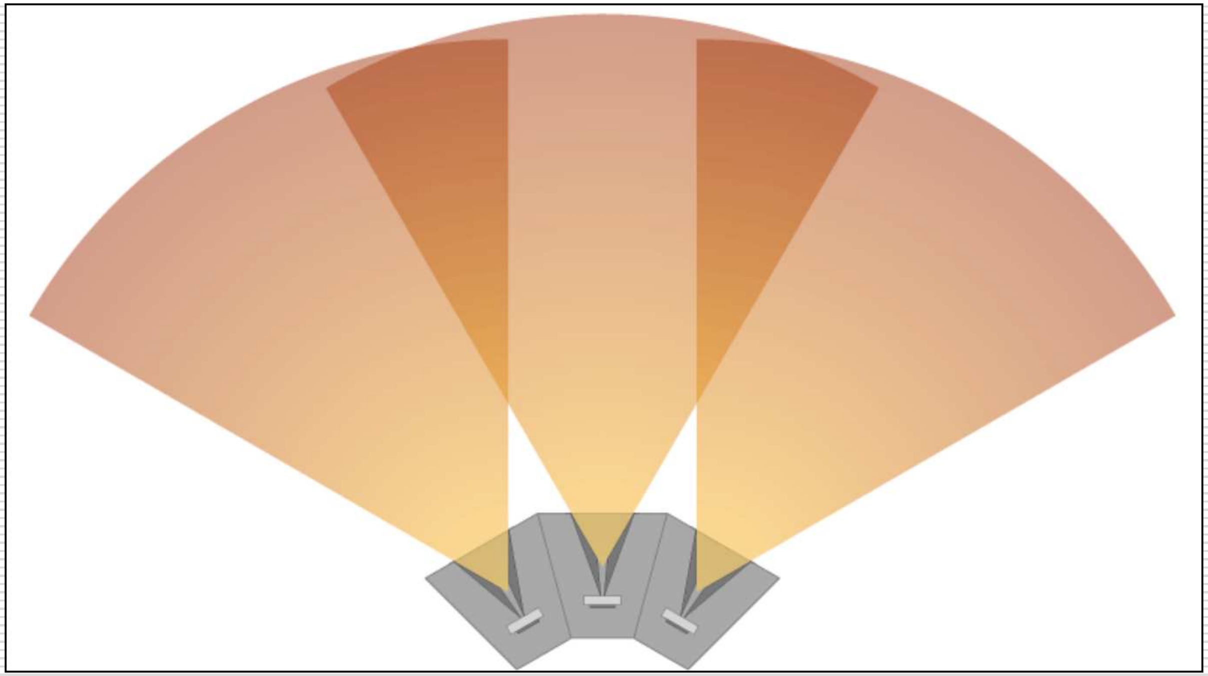
What about..?

- The fancy System Controller
- Using equalisation



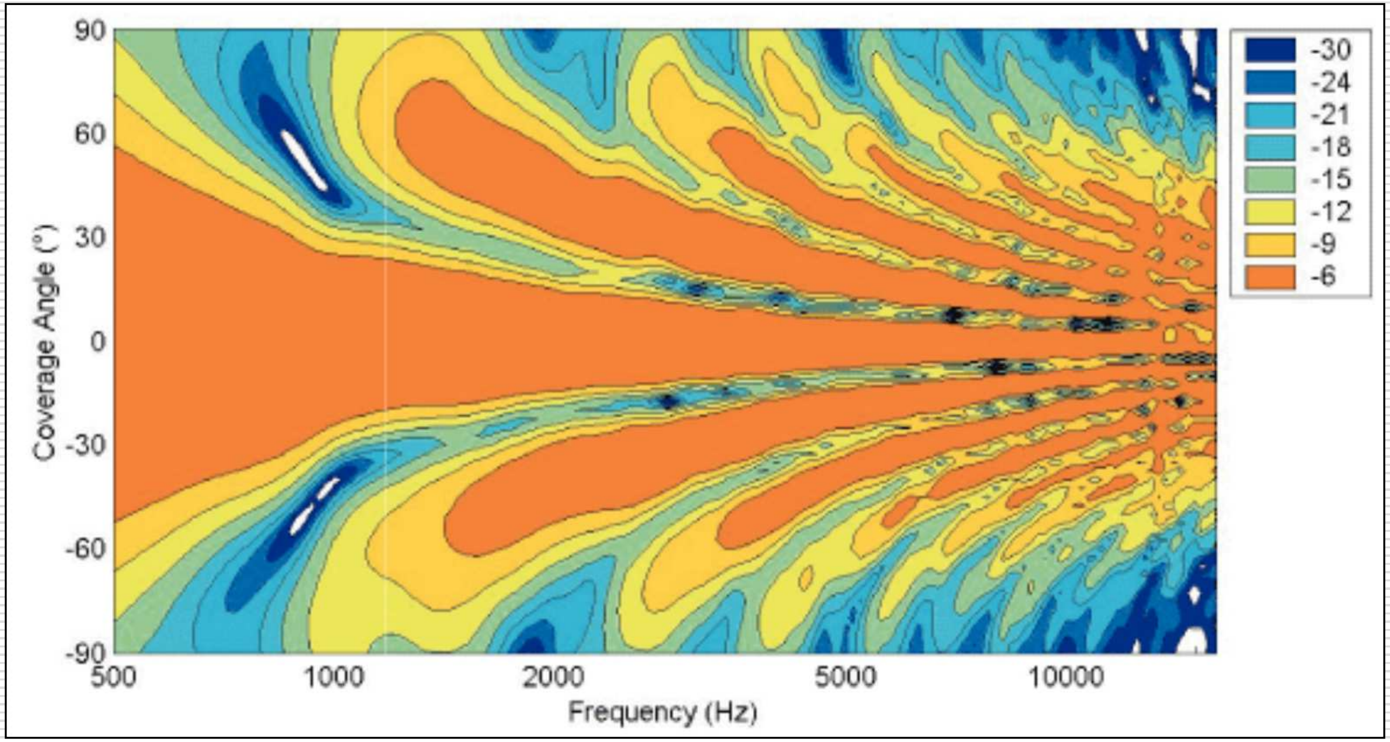
What if we try a different type of horizontal array?

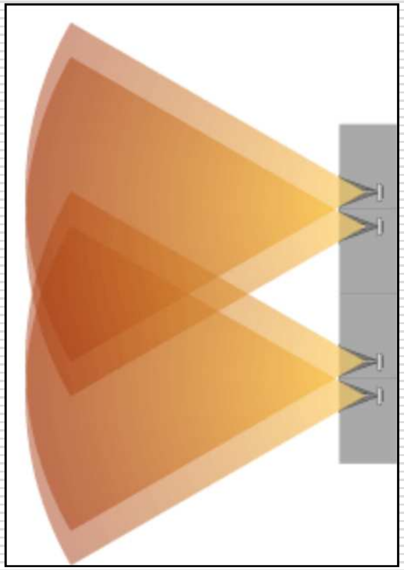
- In this particular configuration, the comb filters and peaks are most severe below the crossover point, where there are three 12" loudspeakers and horns widely arrayed.



Even 2 Boxes Have Problems

- And the view from above a two box array. A predictable mass of ‘fingers’ ready to make the system lumpy and drive an army of complaining punters to the mixing desk.





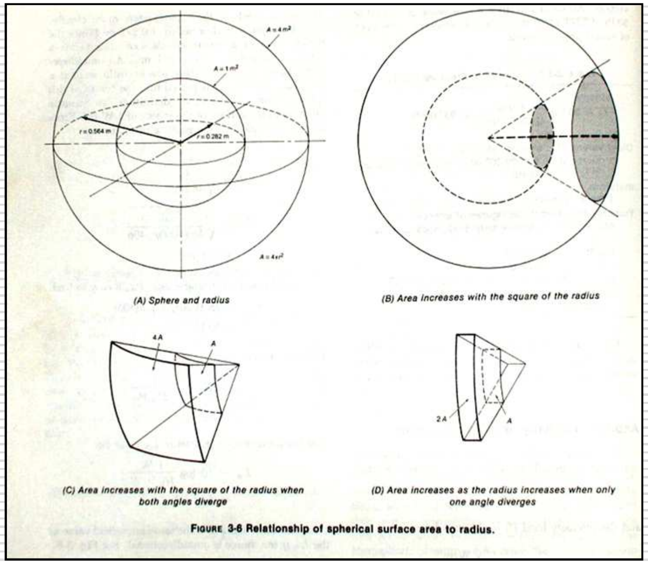
Line Arrays

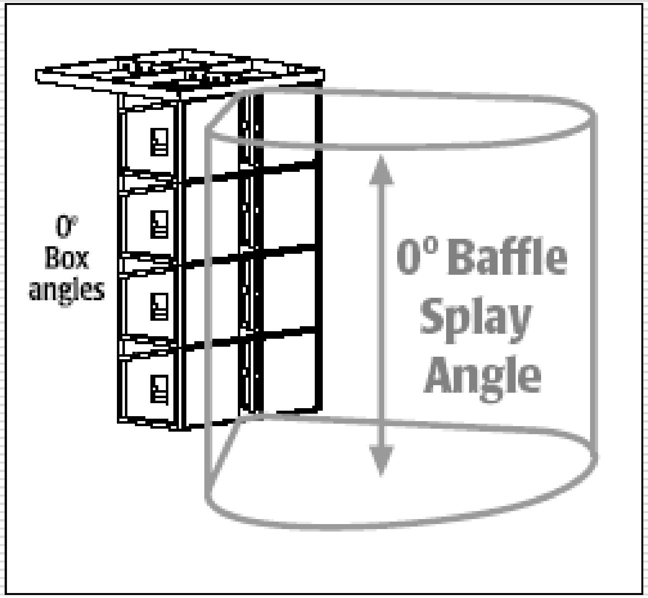
- The problems and limitations of the conventional speaker array were very clearly understood by the mid 80s.
- A solution lay buried in theoretical texts from the 40's and 50's. It was called the 'Line Source' effect

A concert this century. Meyer Sound MD3 subs, Milo (line array) and MSL4 hung for outfill. Nothing is on the ground except the sub bass

So what is a line array and how does it work?

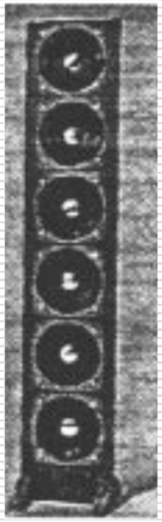
- To understand this answer, we need to go back to the *inverse square law and our spherical propagation of sound model*.
- With a tall line of transducers, the source of the wave is emanating from a line source instead of a point source.





So what is a line array and how does it work?

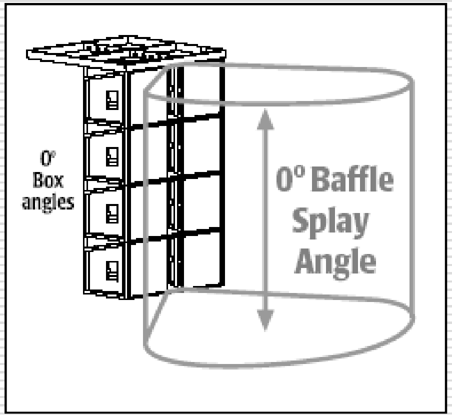
- The line source propagates the wave so it emanates like a cylindrical shape instead of a spherical. An extensive and very influential work on this subject was first published in the early 50's by Harry F. Olson in the book *Acoustical Engineering*.



An early line source box and Harry F. Olson. He held over 100 patents.

In a line source, the inverse square law no longer applies, or does it?

- A line source of wave propagation has the characteristic of having its SPL fall off at a rate of 3 dB per doubling of distance, (as oppose to 6dB)
- This is a well known characteristic of infinite line sources
- Because the surface area of the expanding cylinder is inversely proportional to distance, NOT distance squared



A cylindrical wave front is the goal of the line array. The loss is proportional with the distance, not distance squared.

What happens to a line array in the real world?

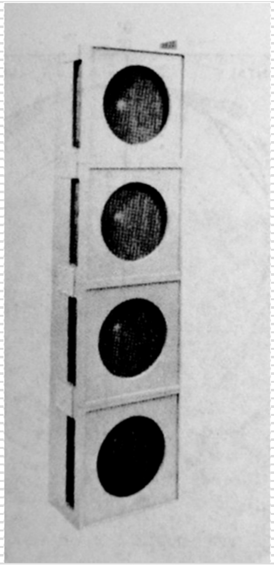
- As soon as we start stacking speakers in a close proximity to each other, we are again back in the land of interference effect, phase shift and the associated geometries.

In this set up, there's no shortage of bass boxes. We have a long array of sub bass in the vertical and horizontal next to the floor.

Forget the need for the 'cylinder' when there are loads of boxes below 100 Hz. Coverage will still not be even because it will still be subject to interference effect between the two sides.

More tech

- the terms **line array** and **line source** are not synonymous.
- A reasonable approximation of a "cylindrical wave front" requires a line height of about 4x the wavelength. So for 50 Hz, we need a 26.6m tall line



Remember, the original research and formulas are intended for line source boxes that were a long line of small speakers, all the same producing a relatively narrow band of frequencies

Another history lesson

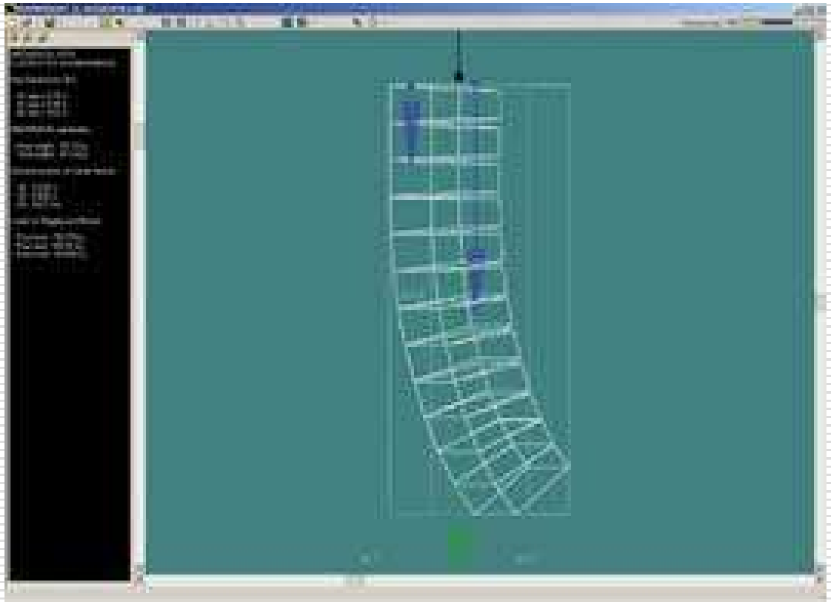
- An unknown (outside of France), French company called L-Acoustics burst on to the scene with the first really practical system marketed as a line array back in the early 90's.

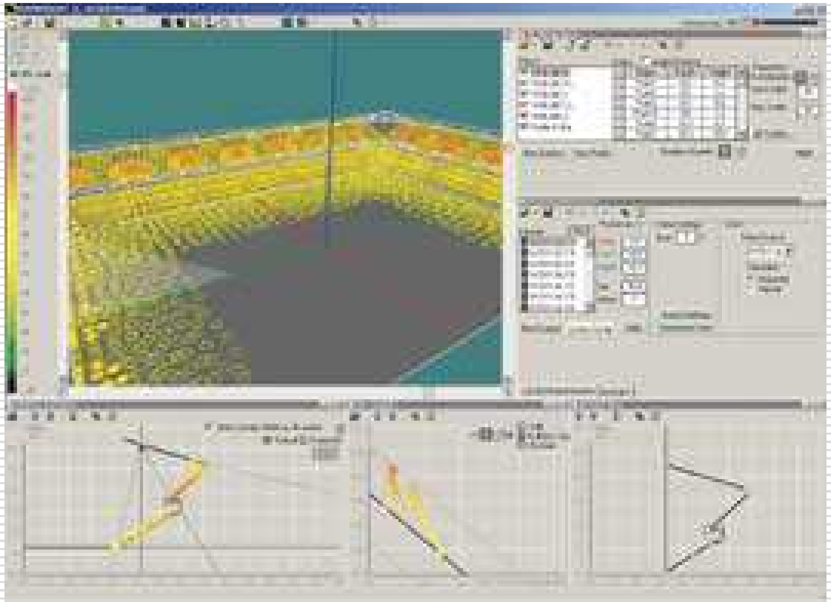
A V-DOSC line array

Christian Heil

Now with Software

- Part of the technology included the L-Acoustics array and modeling software

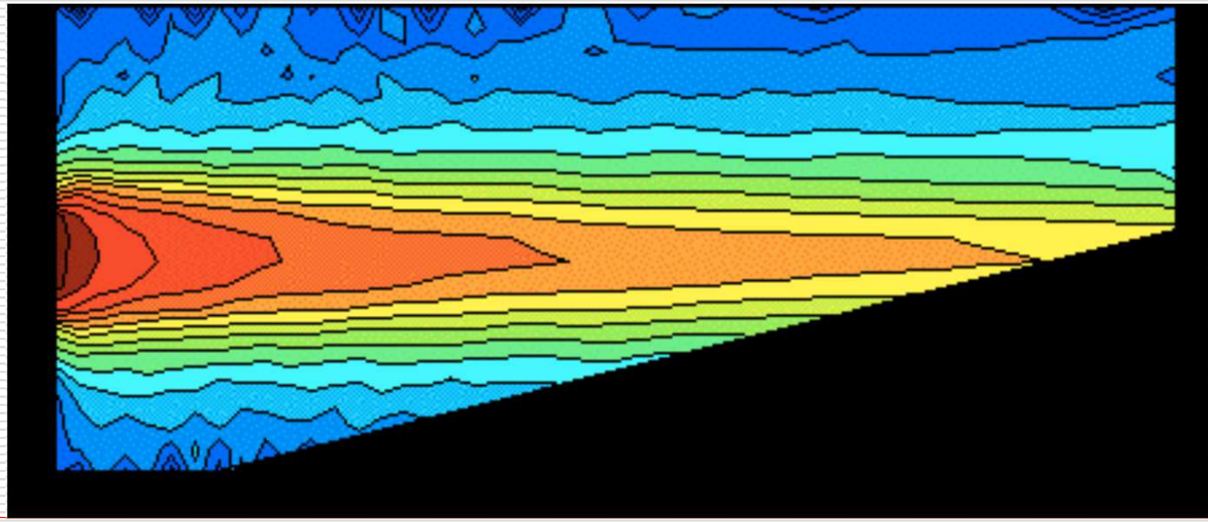




So long as you followed the plan, you came out ok.

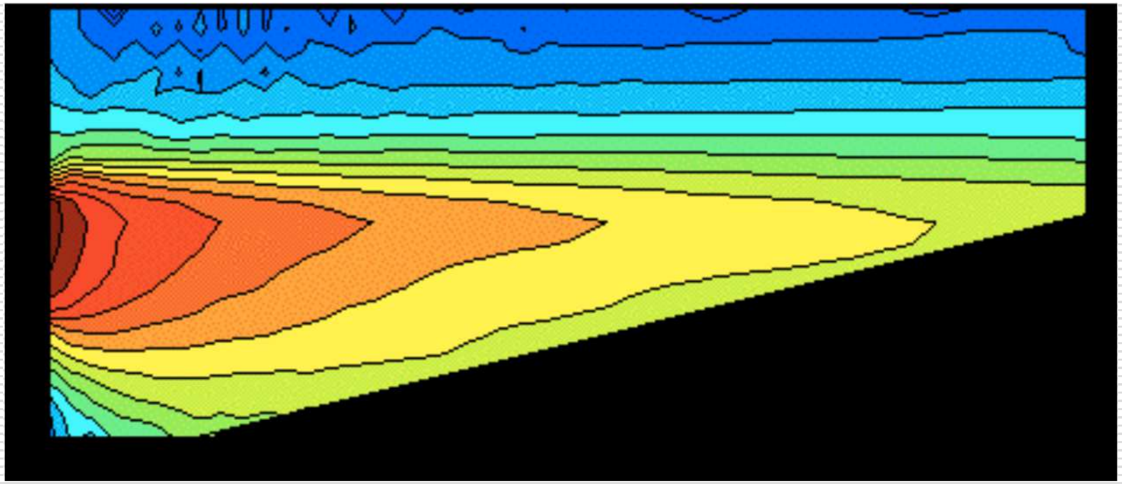
Straight Array

- An unshaded straight vertical line array is a ‘tunnel’ of focused high SPL. It is still prone to interferences but certainly delivers where it’s pointed.
- A straight array. It’s still not perfect, but it is a big improvement on the big ground stack. The graduations indicate dB SPL, not frequency response.



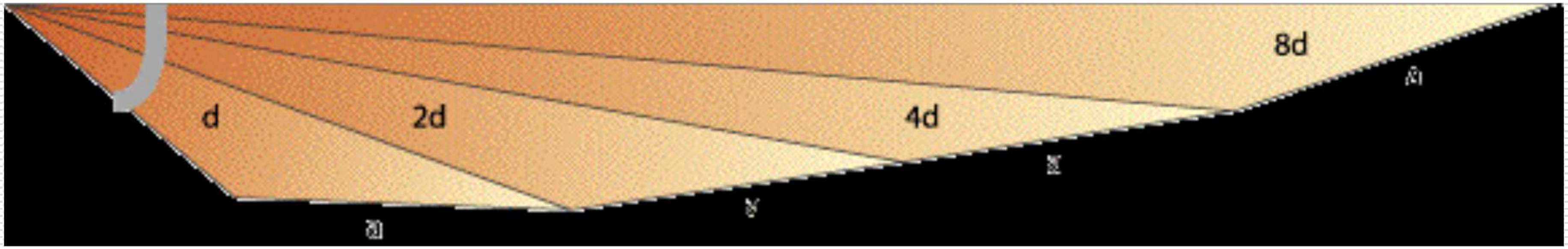
J Shaped Arrays

- The J-shaped array can produce reasonably even front to rear SPL (volume) using "angular shading" or "amplitude shading."



Creating even coverage – shading SPL

- Rather than turn down boxes, what about gradually spreading the energy over a larger area?
- The concept of angular volume shading is a line of equal pressure where the pressure is distributed over a wider angle as the listener gets closer

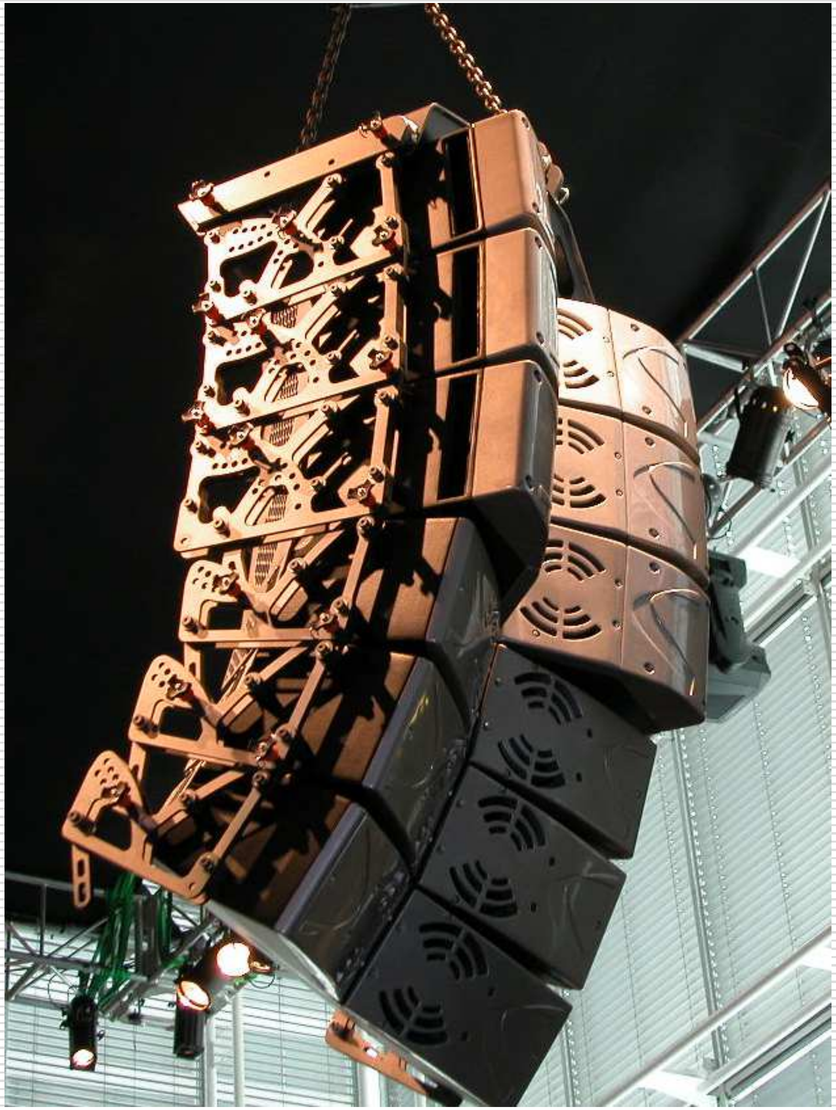


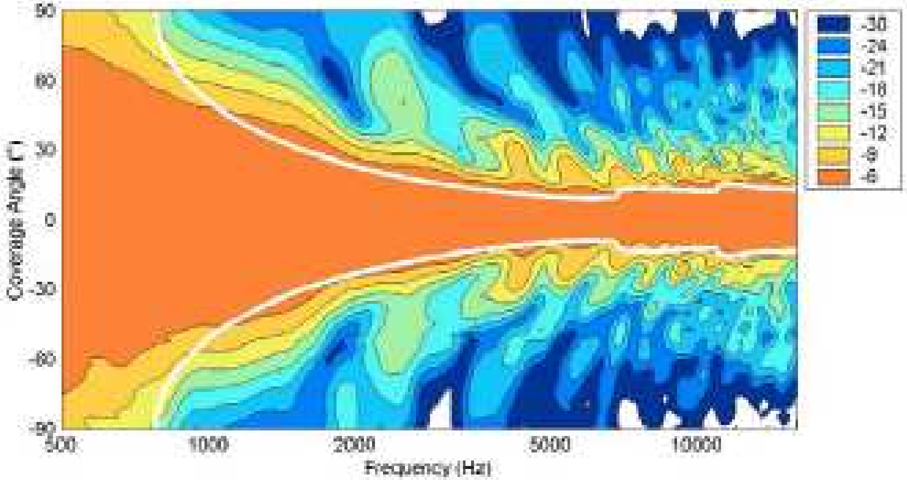
Then in 2003, something really new happened; the tangential array. (A tangent is two lined that touch but not intersect)

The **Nexo Geo** is the first tangent array.

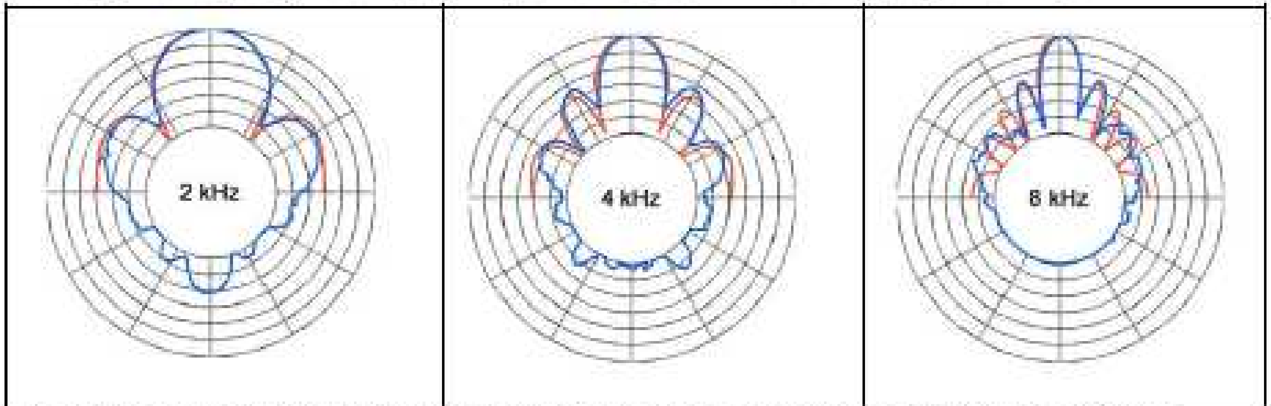
With patented technology if eliminates the ½ wavelength interference problems of all other arrays.

There is not one single development that makes it work, rather there are three patents and a few good ideas thrown in.





Actual coverage of a 30° coupling plane GEO reflective wavesource as compared with a theoretical "ideal isophase ribbon" of the same length and angular arc. While the GEO wavesource does not perform major miracles (note that beamwidth is still proportional to frequency), we can see that it reproduces the ideal isophase ribbon with a high degree of accuracy. The pattern produced is as close to the mathematical ideal as the actual device is to the mathematical model. Note that 30° coverage is achieved at 2000 Hz, even though this horn is sized for use alongside an 8-inch diameter cone woofer. Also note the smoothness of the coverage vs. frequency, indicating that the power response will have an equally smooth taper.



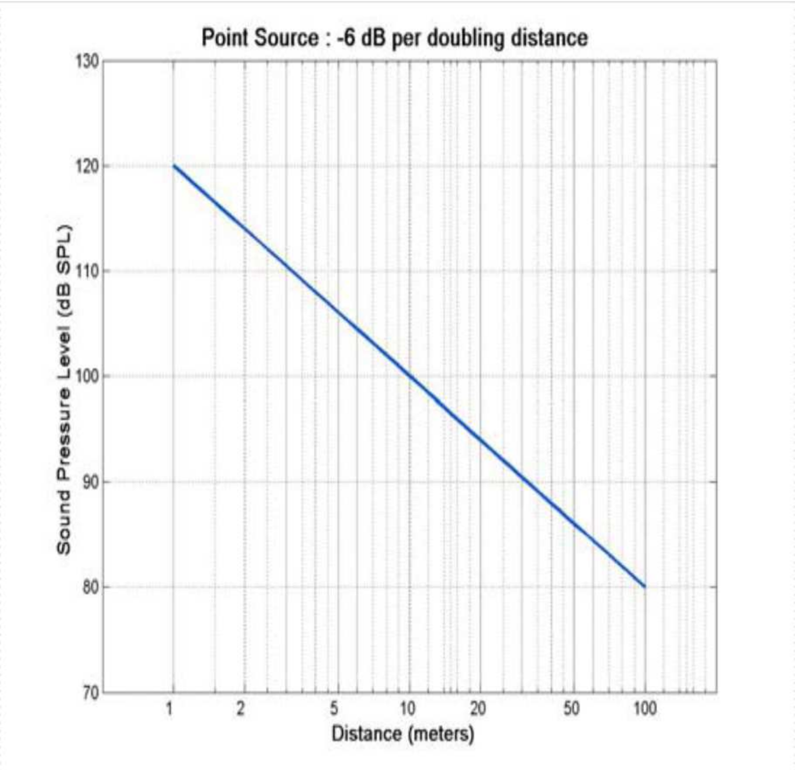
A more conventional polar plot: pure sine wave (not smoothed) measurements of the 5° GEO S805 wavesource (blue traces) vs. the theoretical 250mm 5° isophase ribbon (represented by the red traces). The geometrical model is an extremely accurate predictor of real world performance.

So to the question of room acoustics and intelligibility

**In a venue space that is not a standard rectangle,
not raked seating and surrounded in reflective surfaces,
is the line array a one size fits all answer?**

What is a delayed system?

- Back at the inverse law chart we started with a sound wave SPL of 120 dB.
- Even though the Alpha box is capable of 145dB peaks, at 40-50 meters we will have a big drop in level.



The Answer

- is to add more boxes just before the physical point where the volume drops to too low a level. This is the **delayed system**

Pictured is a front of house supported by a delayed system behind the mixing position.

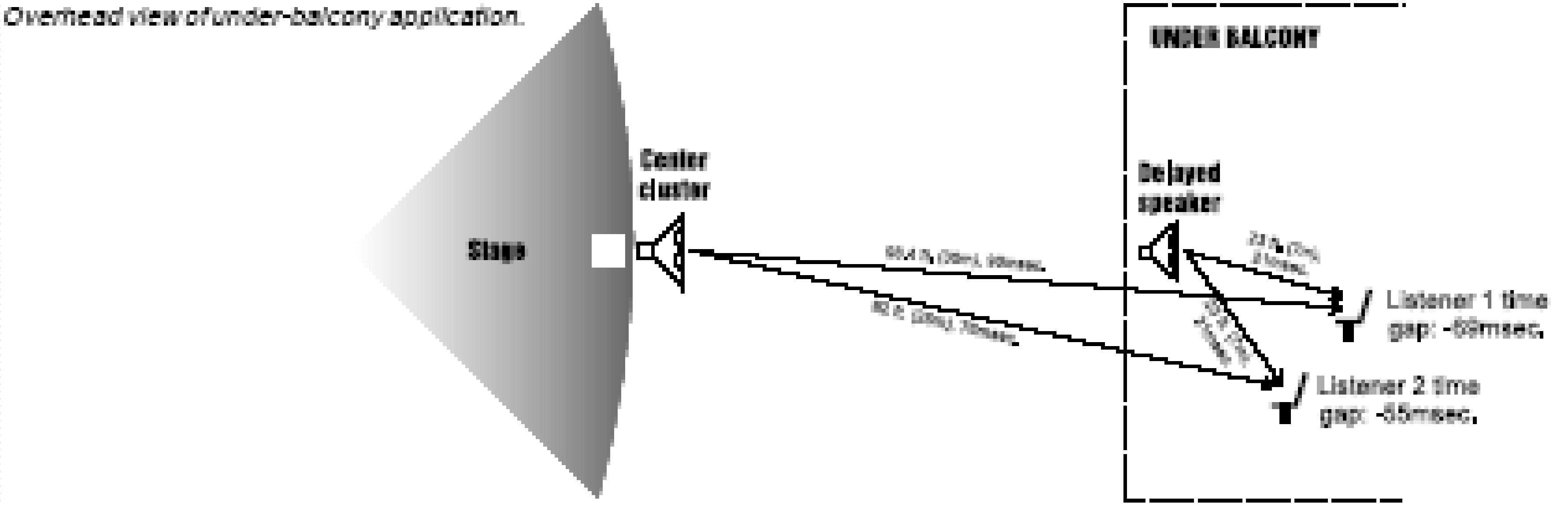
It doesn't need to be big to need a delay.

A front of house supported by one or two delay lines fixes the coverage and SPL problem.

New age sensitive cow persons. *These performers are easy to reproduce through most PA systems. Some folk and jazz acts are much more difficult*

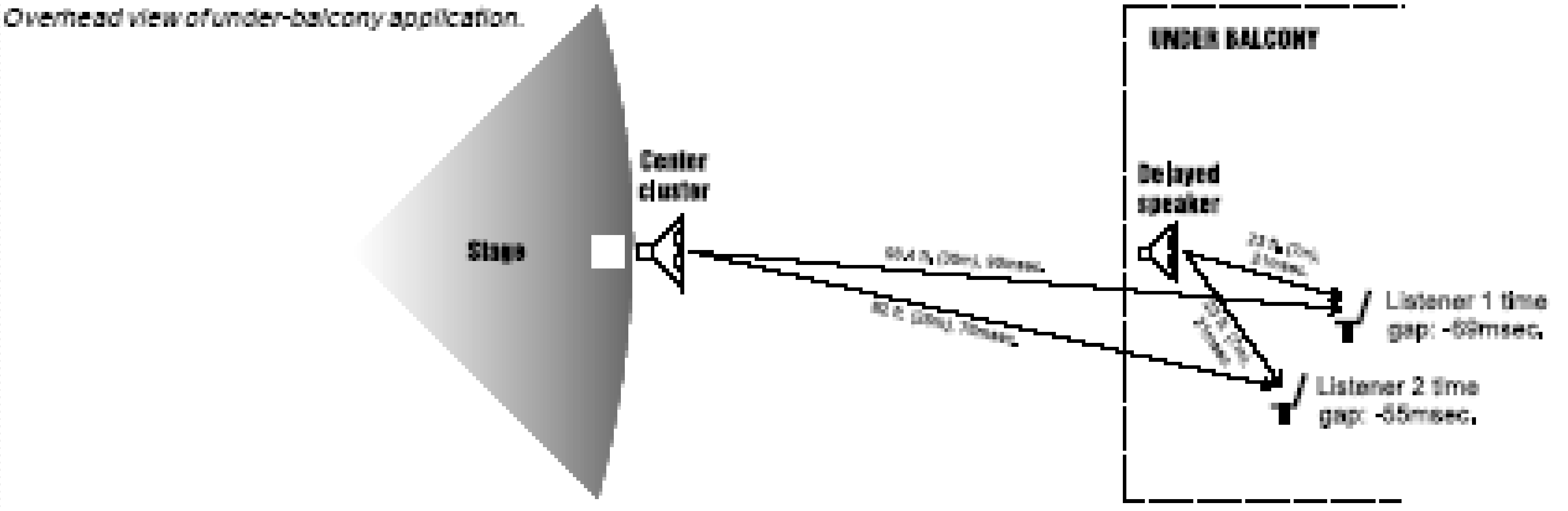
How is it done without the whole area being drowned in echoes and reflections?

- Loudspeaker Synchronization with a digital delay



Using a digital delay or programmable speaker controller to fix array problems

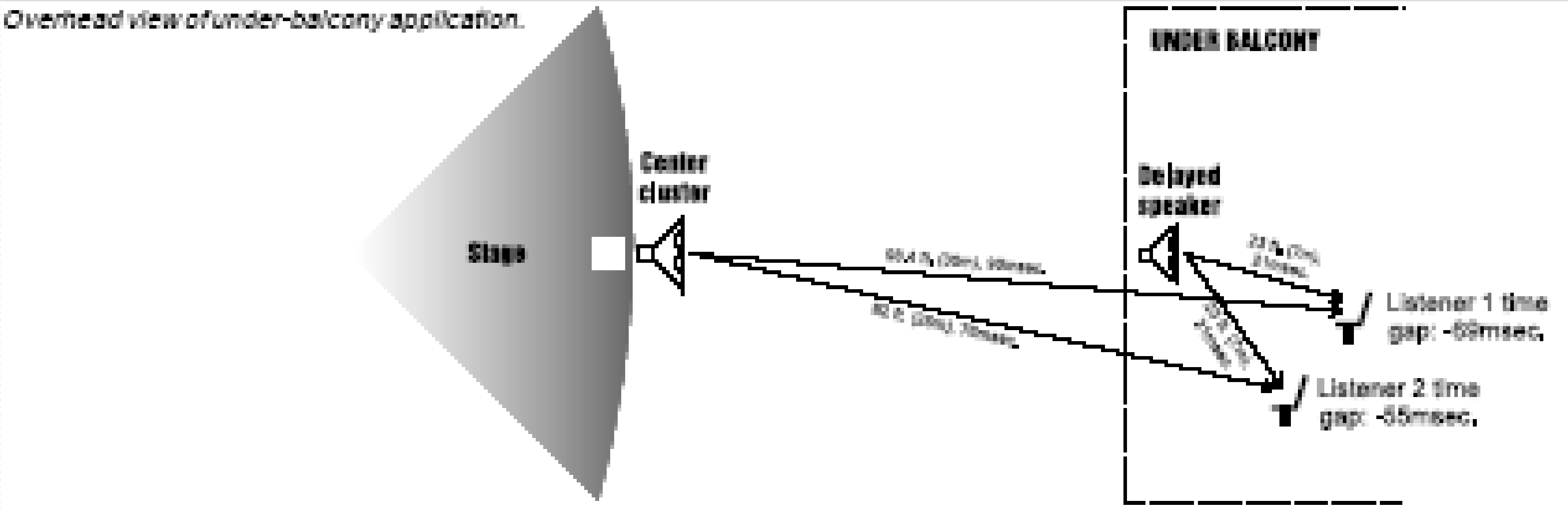
- You will notice in the diagram, the delay speakers are pointed in the same direction as the FOH speaker. It is important not to have delayed speakers pointed across the wave front of the preceding sound source. If you think you need to angle a box to get coverage, put up more boxes; don’t set your satellites at an angle across the main wave.



How to set the delay times

- In the early days, a sound guy would be running around the building with a tape measure but now there is an easier way thanks to a guy called Haas.
- **The Precedence Effect: Aligning the Acoustic Image**
- The phenomenon that makes two distinct sounds heard less than 35 ms apart seem like only one sound is call the Haas Effect.

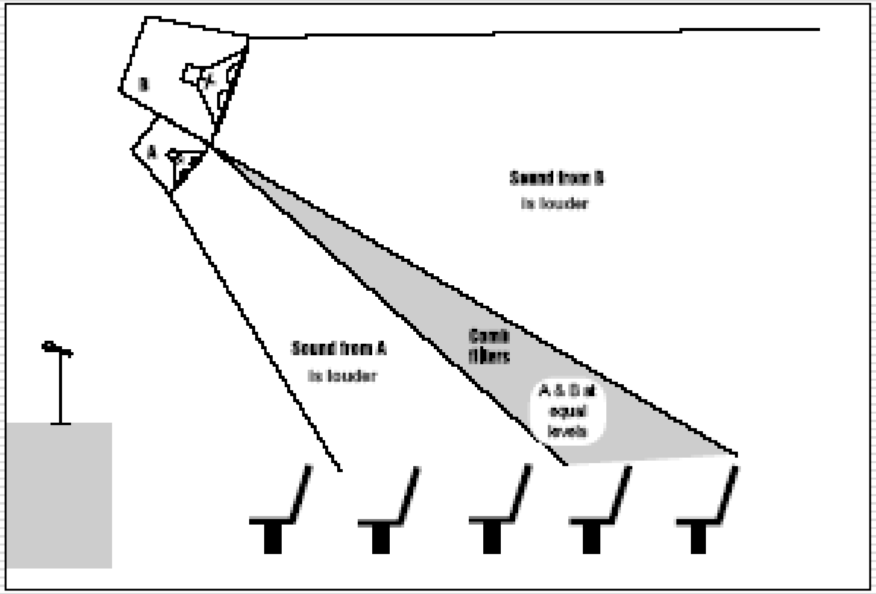
Back at the venue

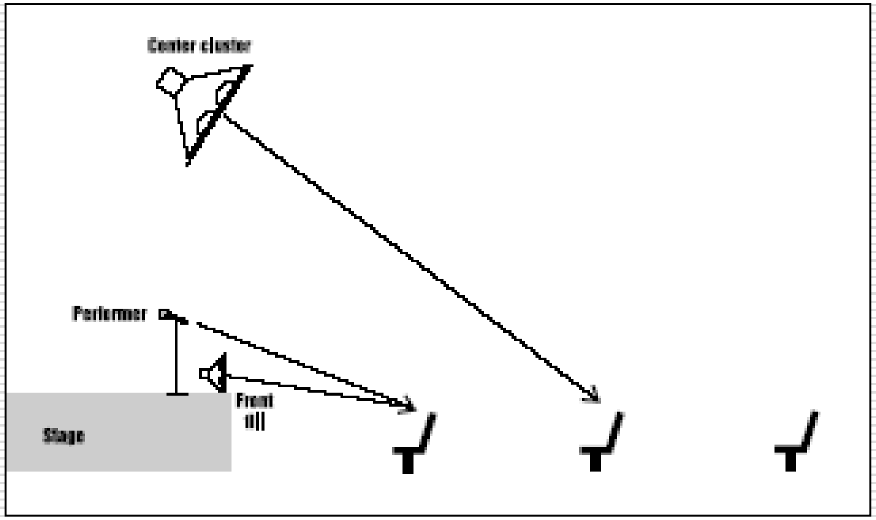


We must delay the sound from the under-balcony speaker to synchronize the signals. Do we set the digital delay to 76 or 84 milliseconds? Obviously, the geometry will not allow us to exactly synchronize every location under the balcony; we have to compromise.

Technique to time out-fill, or under-hang components of a primary array

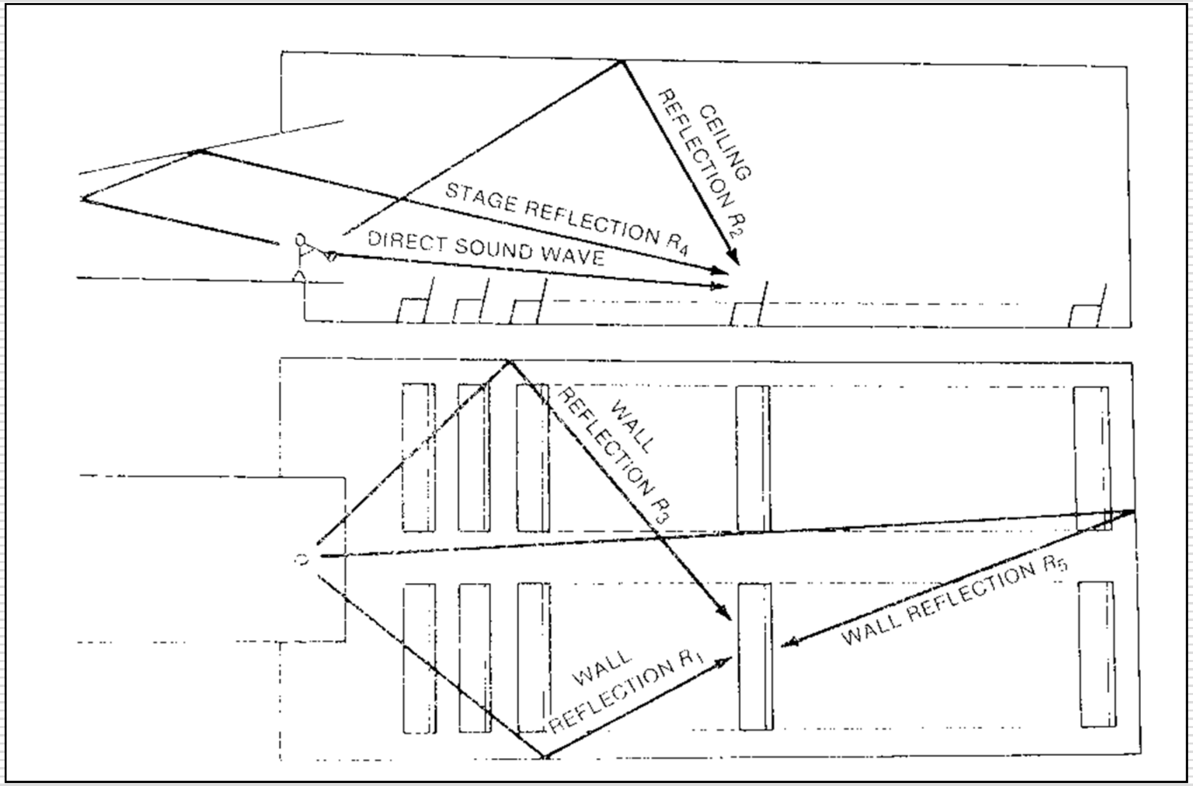
- Remember we can also lower the comb filtering effect by turning down the ‘fill’ speakers 4-6 dB.
- In the situation where front fill is required, the front fill is closer than the flown cluster and must be delayed. The sound operator will estimate the distance to the flown boxes and add 15-20 ms. the image will then appear to come from the flown cluster and stage.





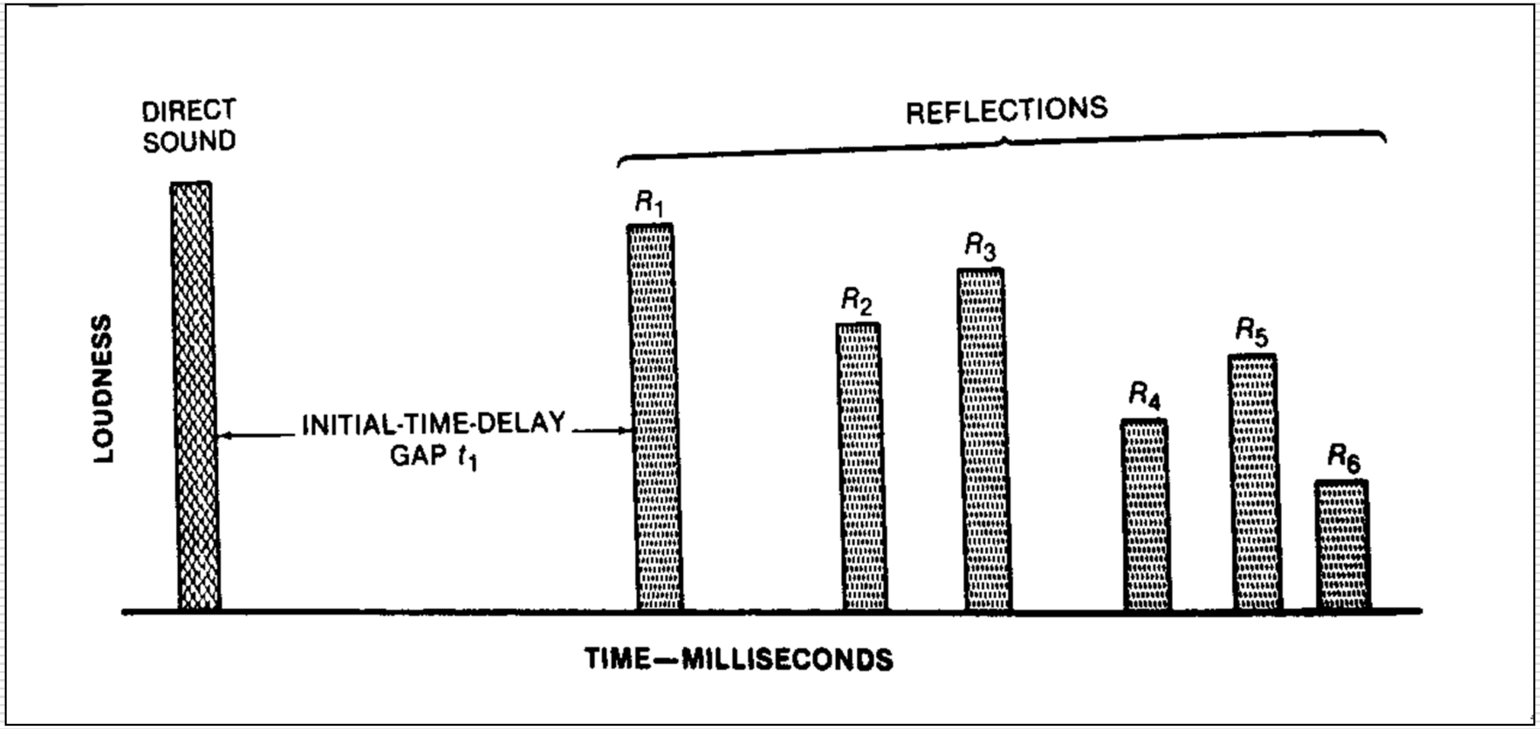
What is reverb? (Latin word reverbarare – to beat back)

- Echo (first reflection) and reverberation can get confused when in fact they are related. Lets see what happens when we generate some sound in a small hall.



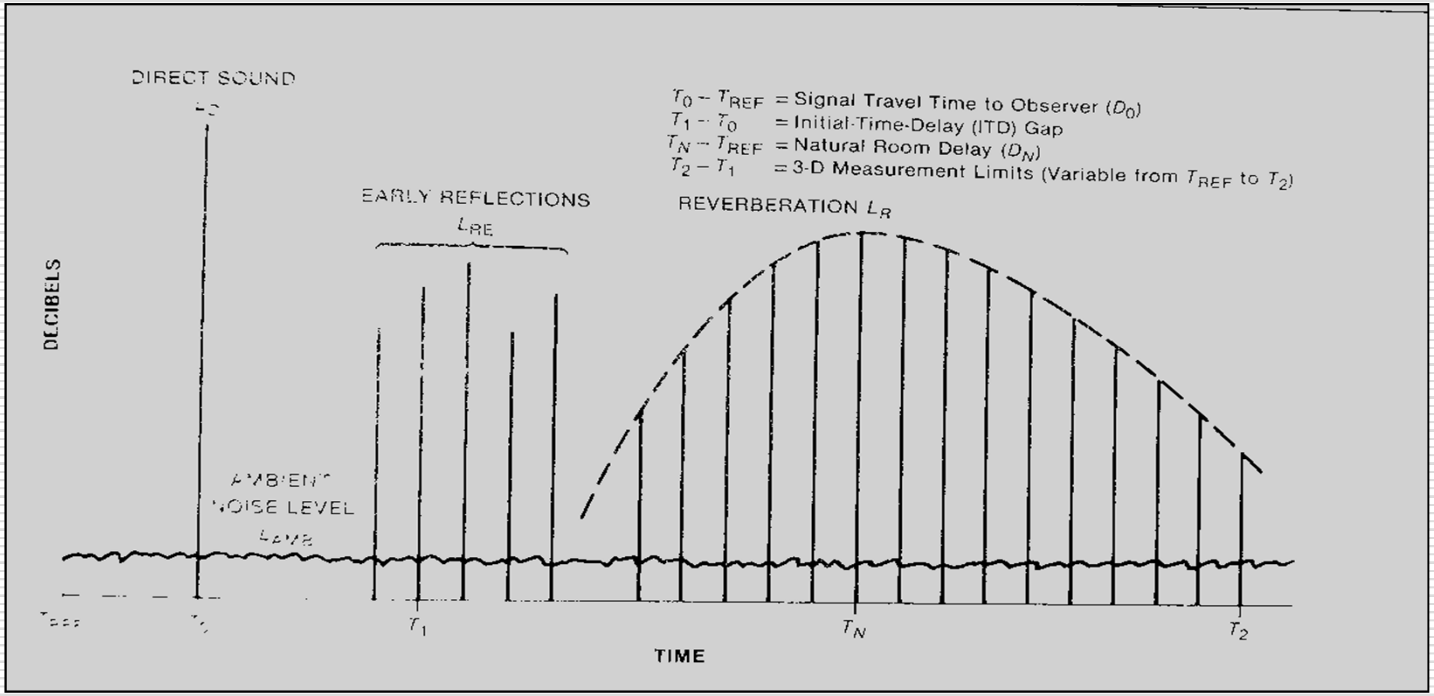
Early Reflections

- After the direct sound, there is a small delay followed by early reflections.



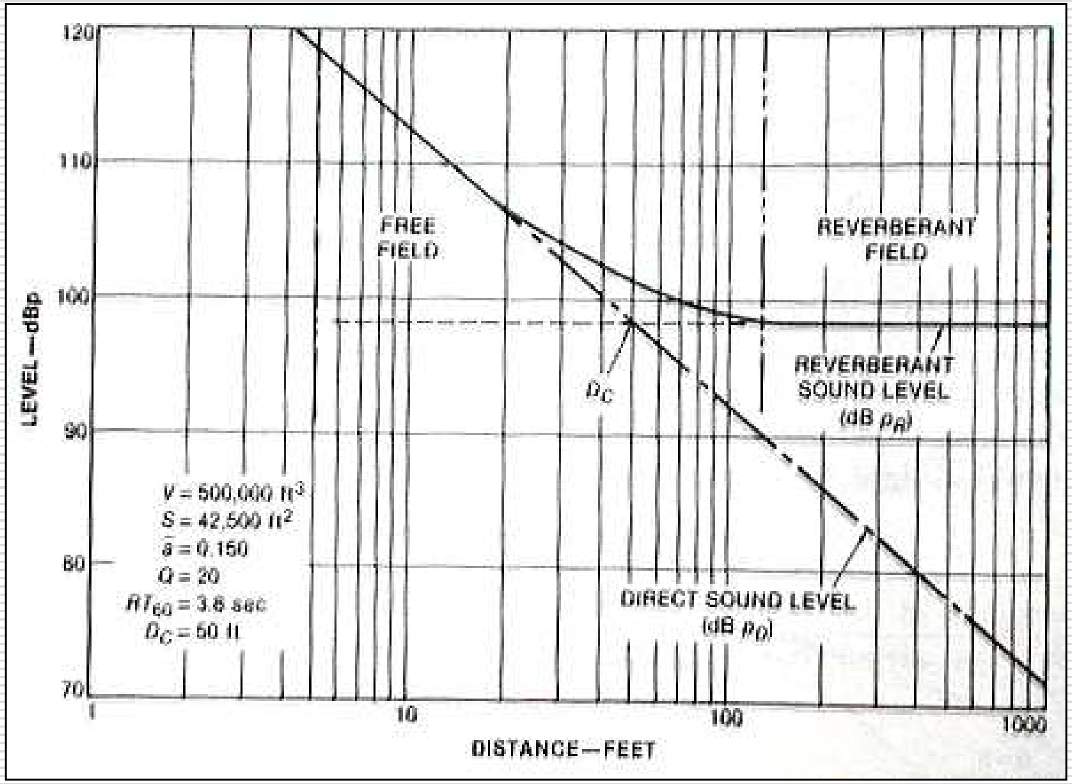
The Reverberant Field

- The reflections collapse into a random field of micro echoes in a 'big swooch' as the energy dissipates through out the room.
- **RT60** is the time that it takes the reverberant sound to decay 60 dB.



What is critical distance?

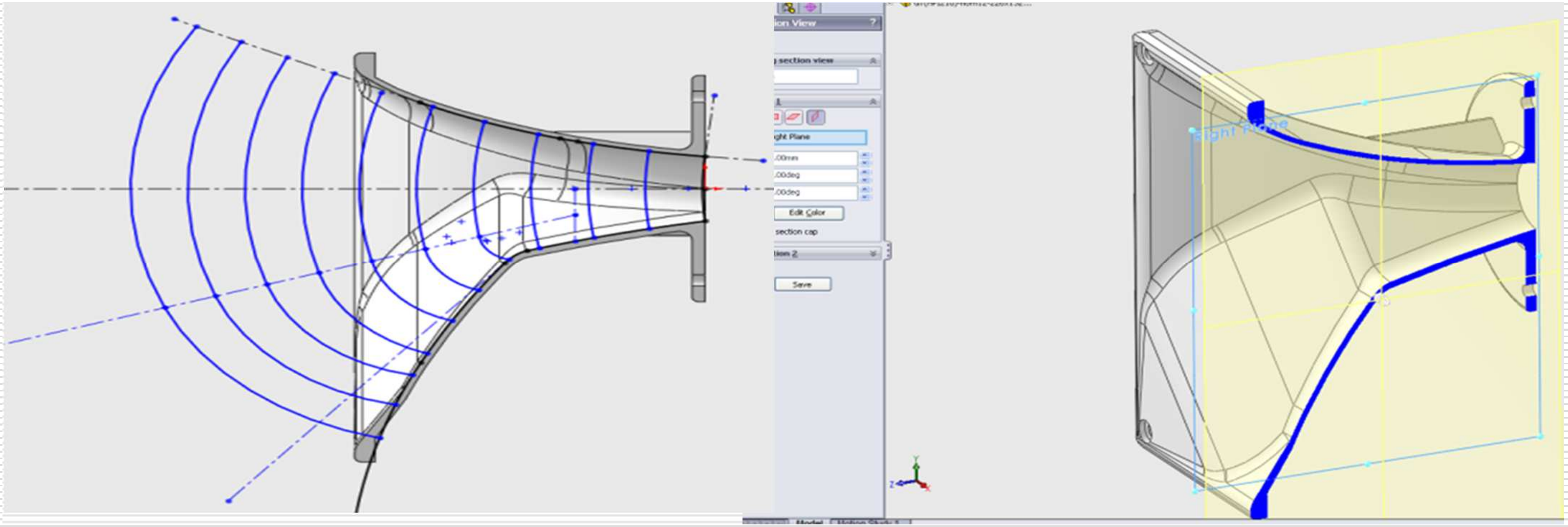
- Critical distance is that point in the room where the reverberant sound and early reflections are at the same SPL as the direct sound from the FOH speakers.



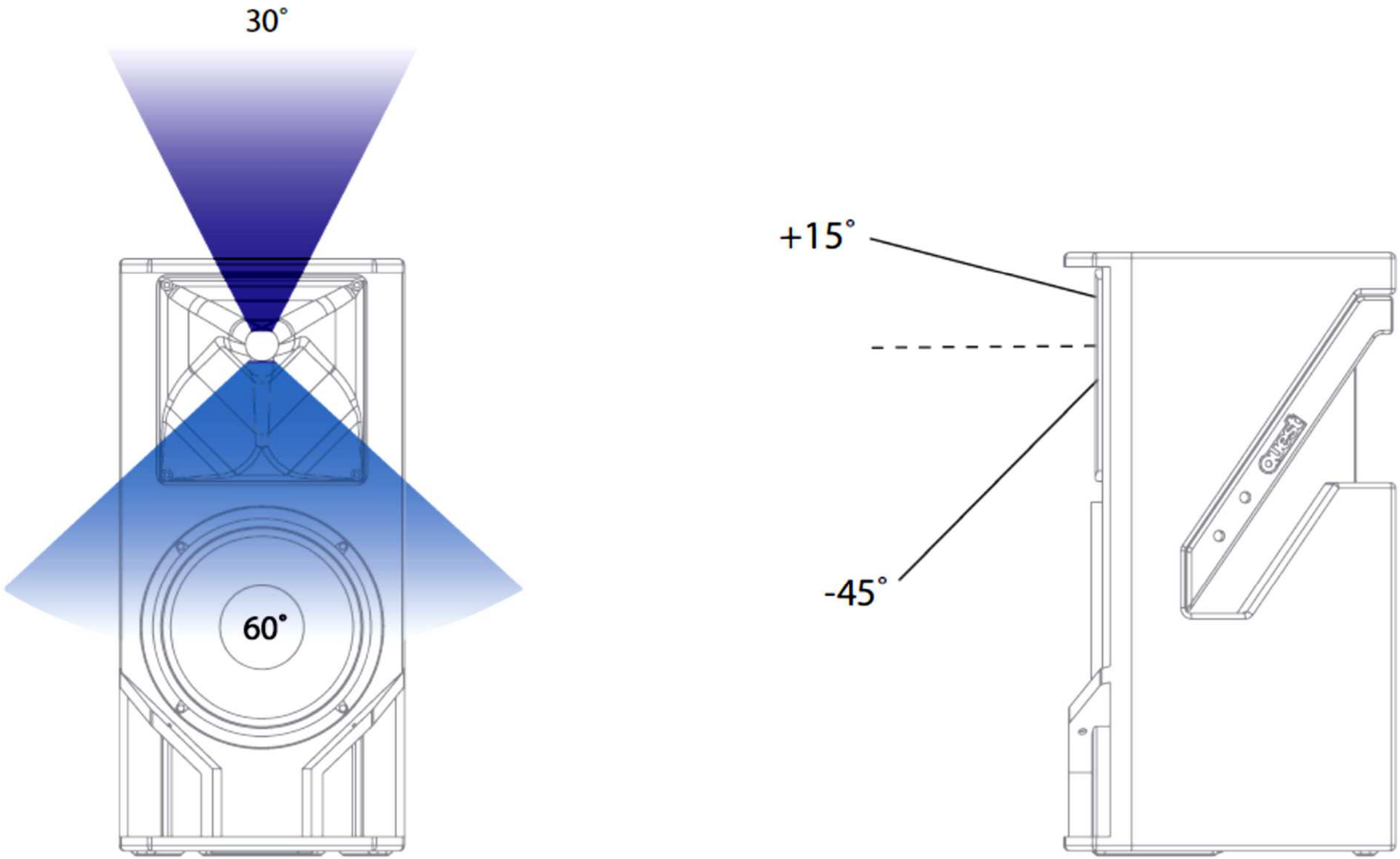
Intelligibility as an energy problem

Point source asymmetrical distribution in a horizontal array

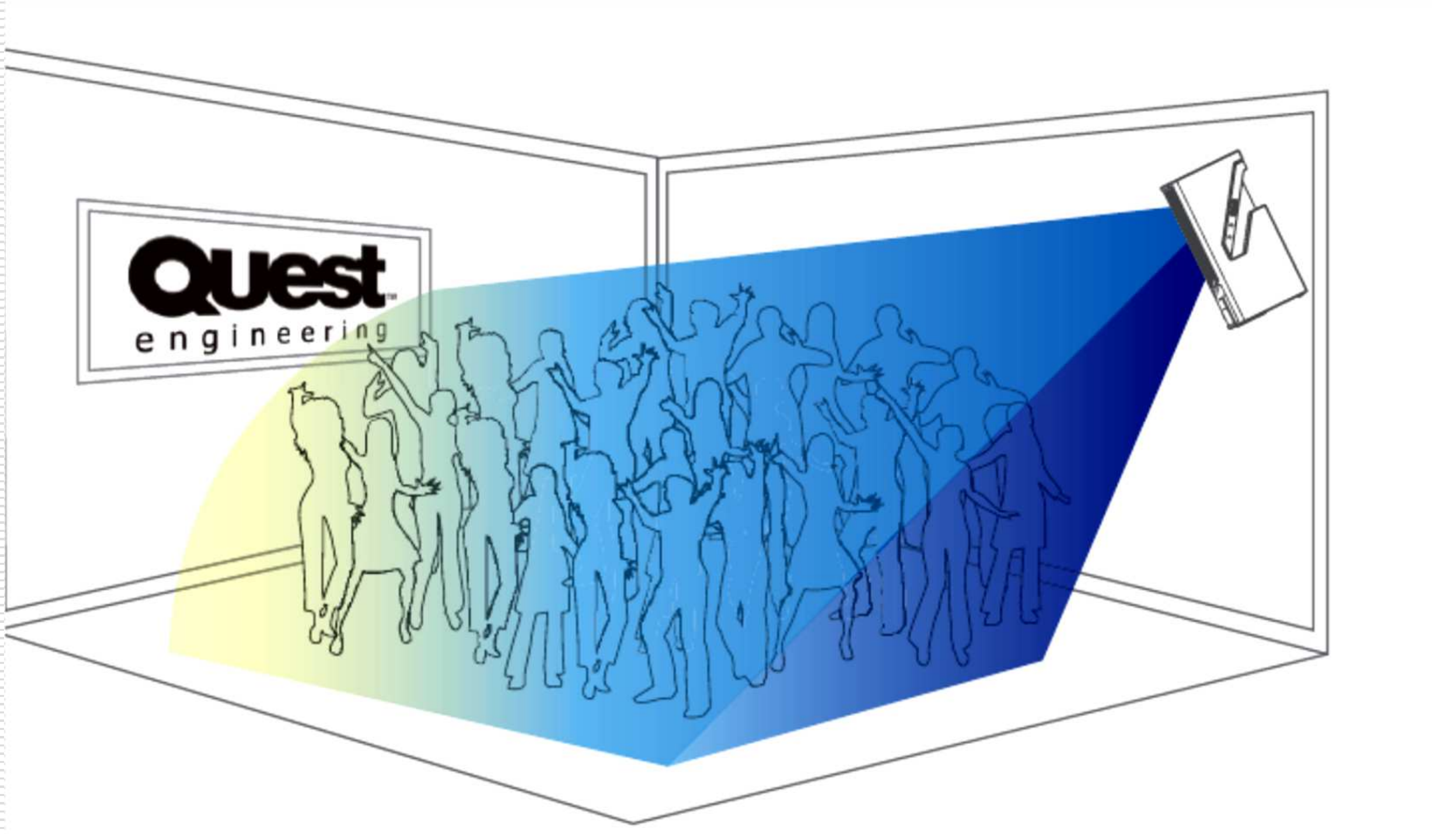
Asymmetrical distribution in the vocal range



Asymmetrical projection that is reasonably phase coherent



Uniform coverage is possible under most circumstances



Solutions need to consider the real world

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Light weight is important for when designers haven’t made structural provision for a PA

So despite the potential for comb filtering, lobbing and other undesirable effects, does it work?

Time for a test

Ultimately is needs to work in the real world