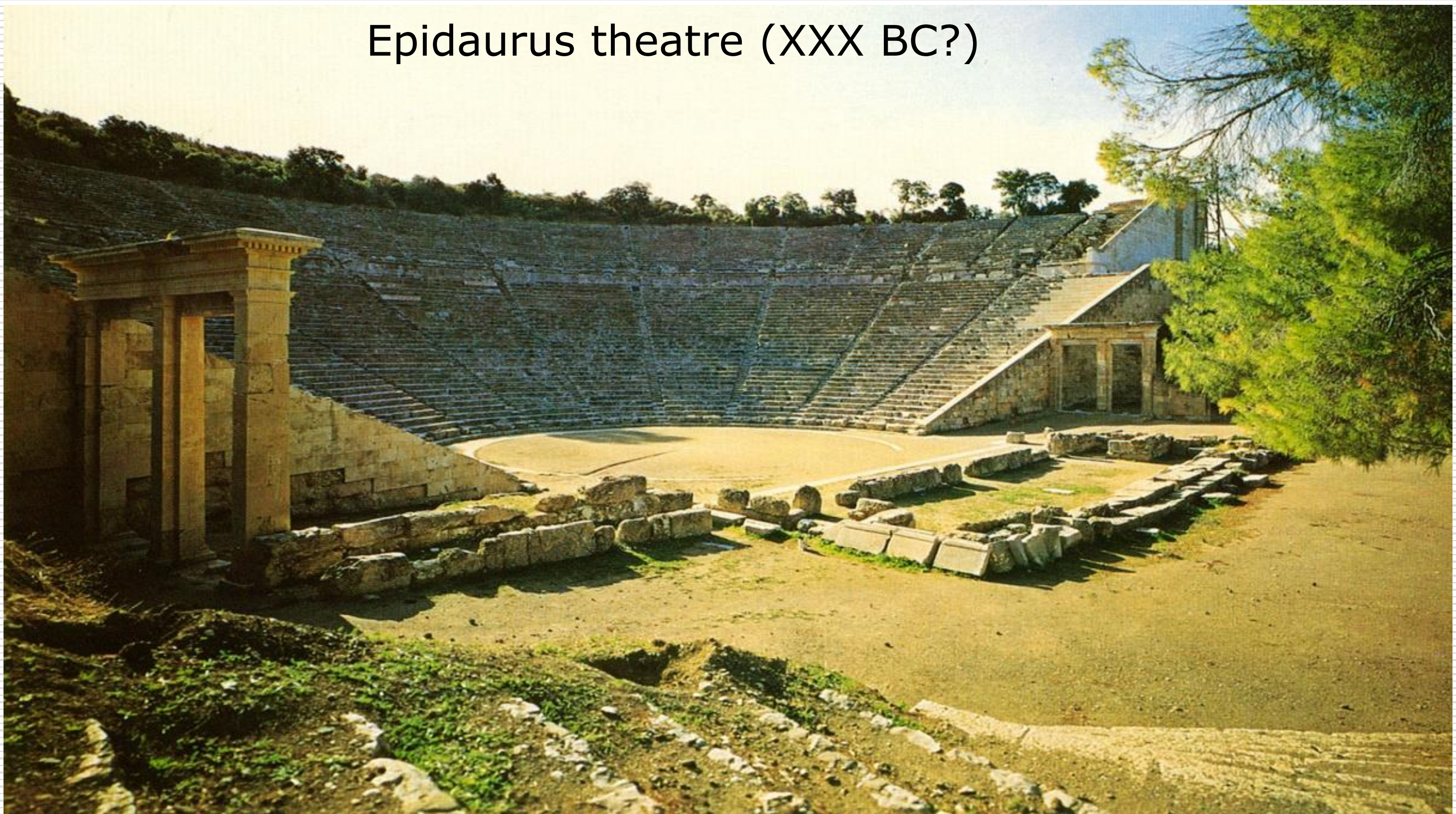
Difficult acoustic environments?
Maintaining voice intelligibility

Greece was a democracy – It's all about talking

Public discourse is the food and currency of democracy. Once Greece dispensed with kings, public consent was the name of the game. Public salesmanship – not coercion



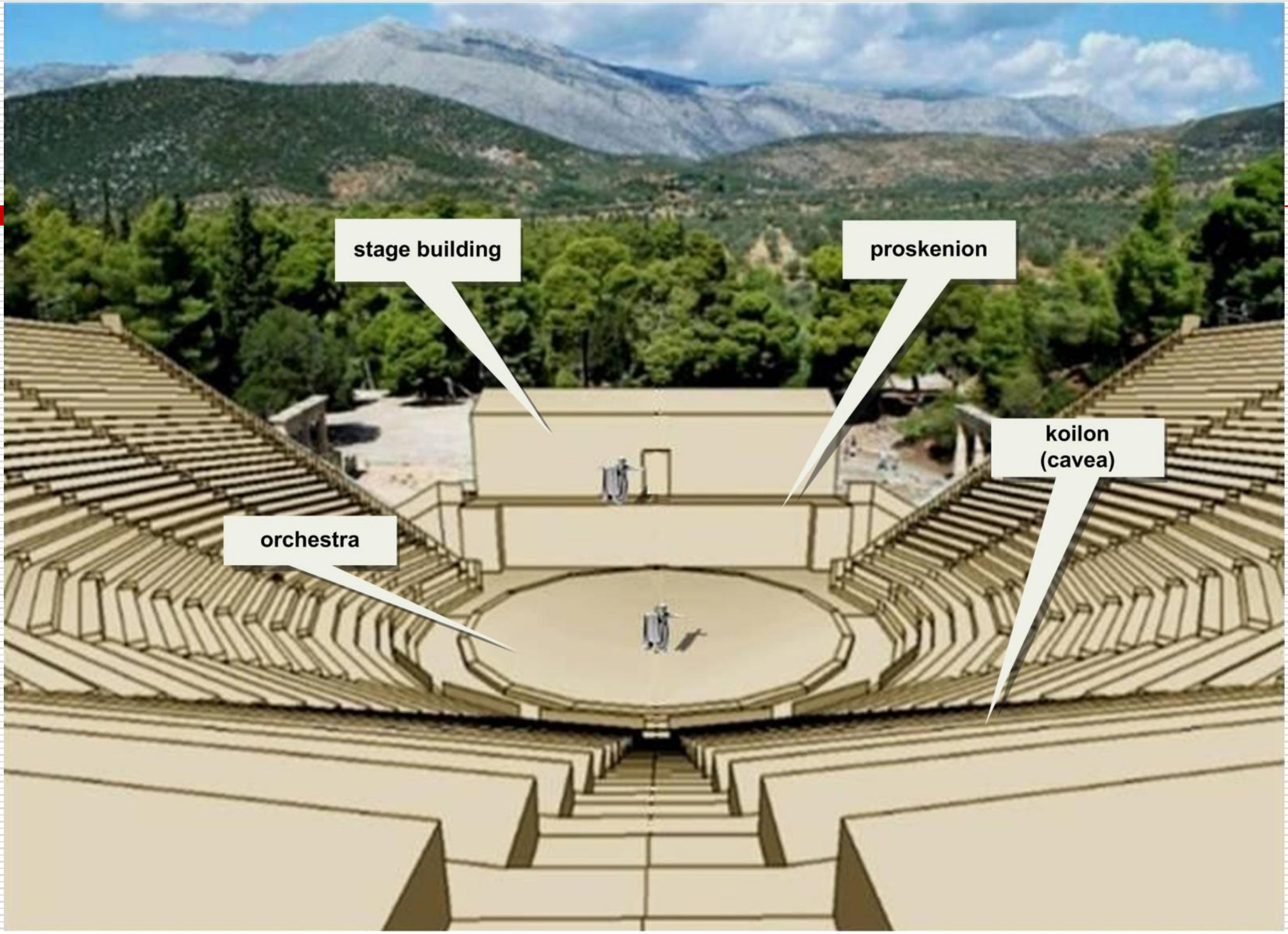
Epidaurus theatre (XXX BC?)

Greece was a democracy – It's all about talking

A temple has a different purpose and ambiance

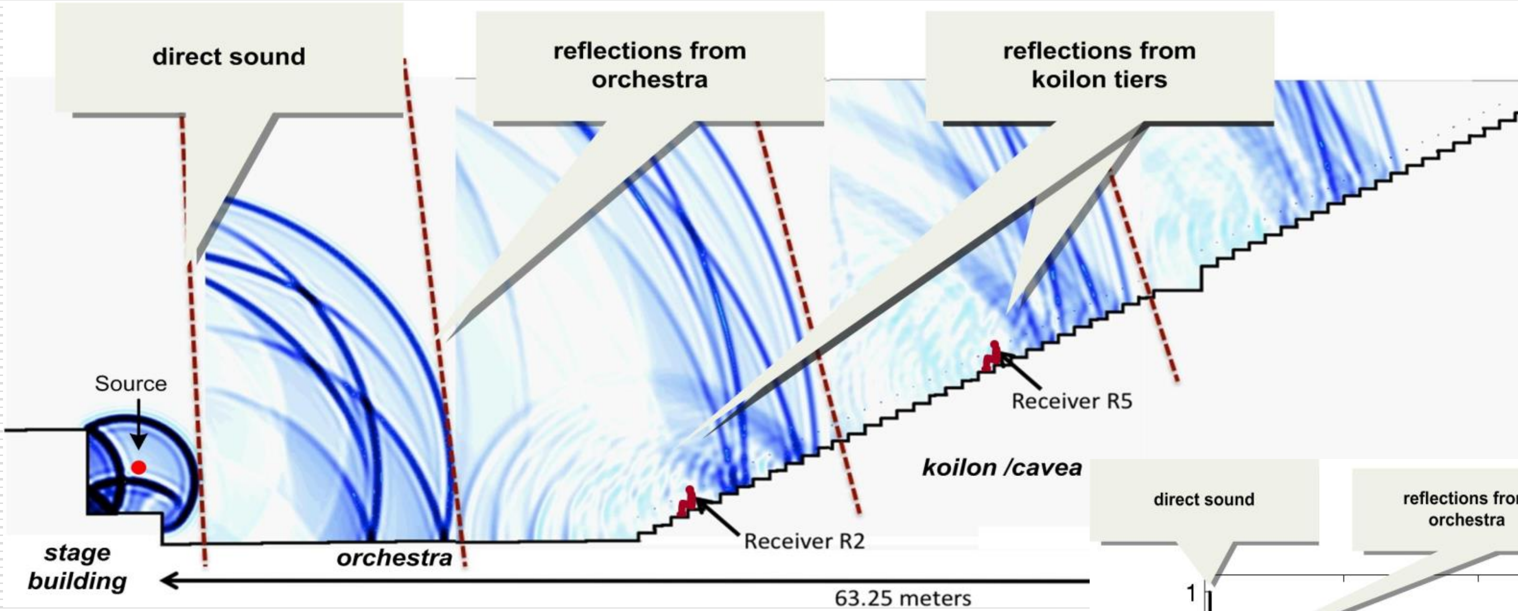
2,500 years ago +

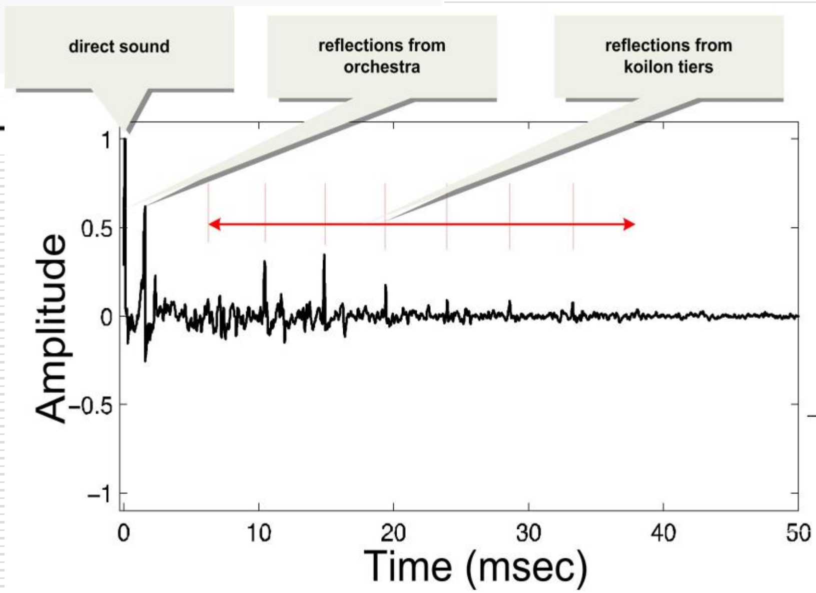
The word Auditorium is a Latin word meaning 'a place of hearing'.



Some of the modern terminology dates back to BC

Epidaurus theatre



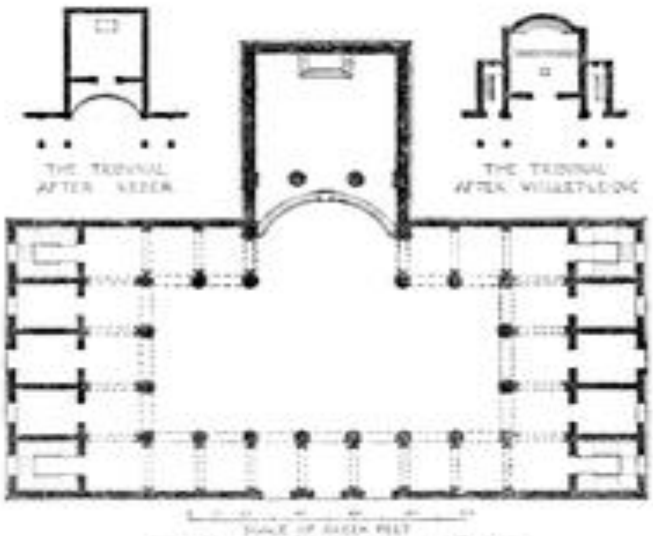


Greek - Roman Theatre Glossary

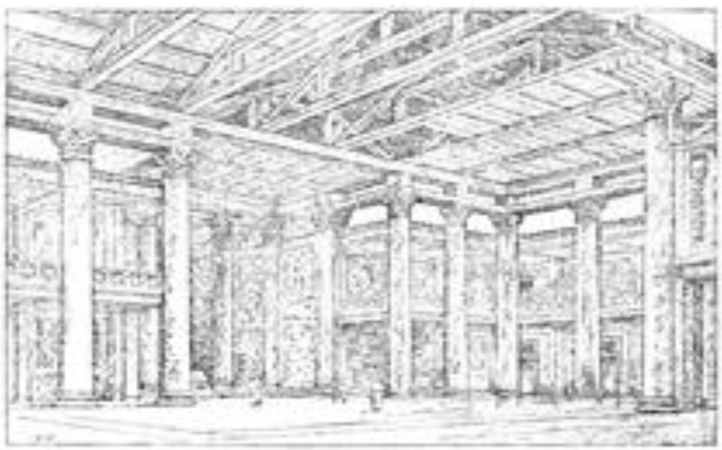
(At least 50 specialised words – suggests a knowledge of the subject

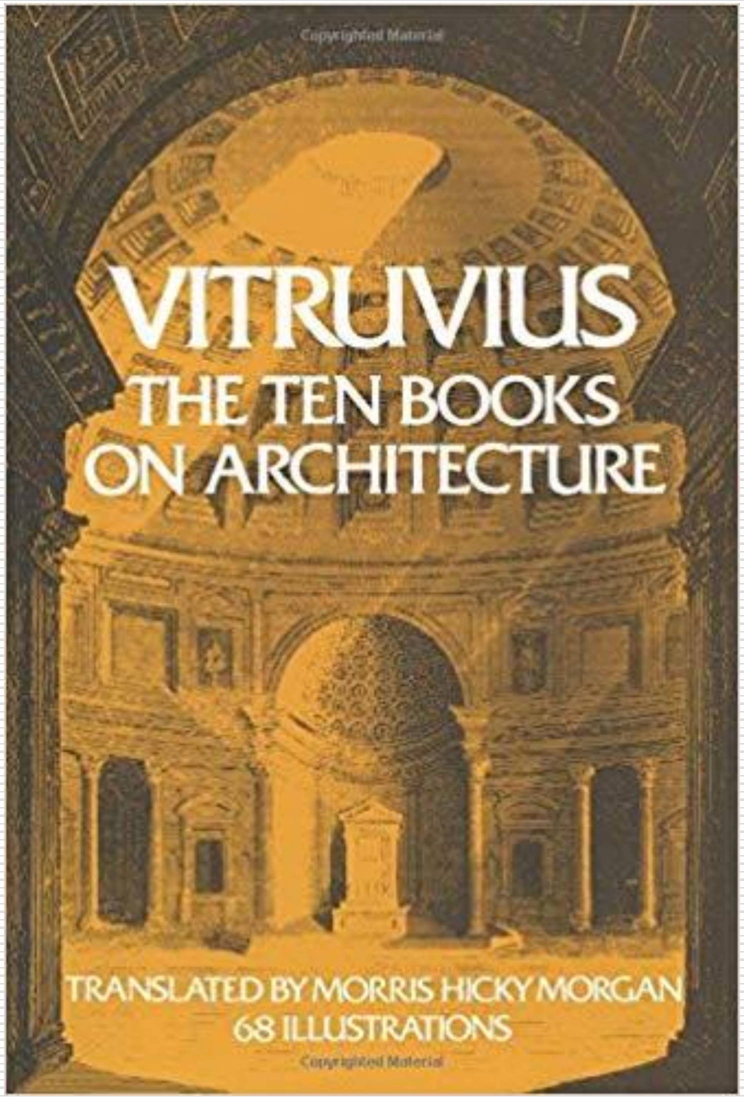
At that time, the building was the sound system

Acoustics – A maths problem since 2,500+ BC

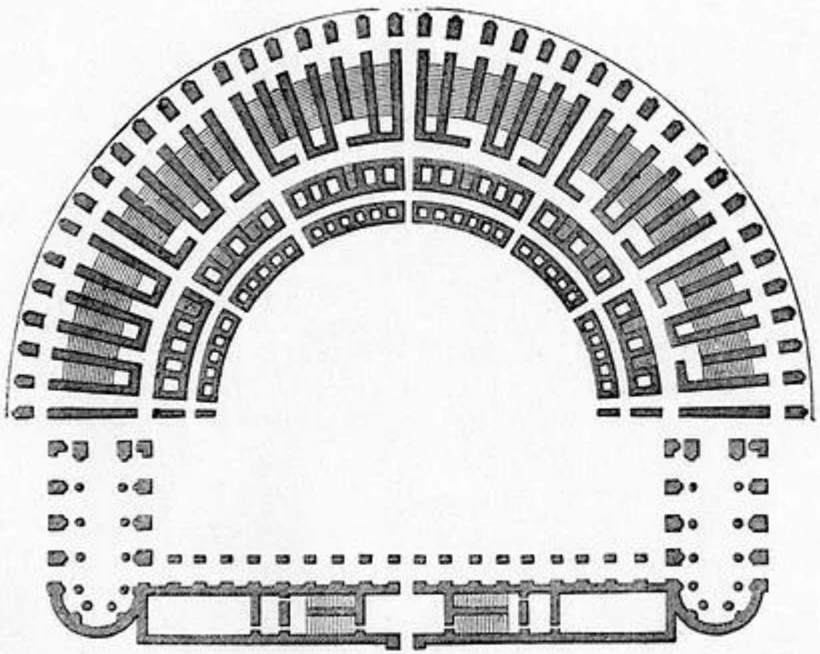


VITRUVIUS' BASILICA AT FANO

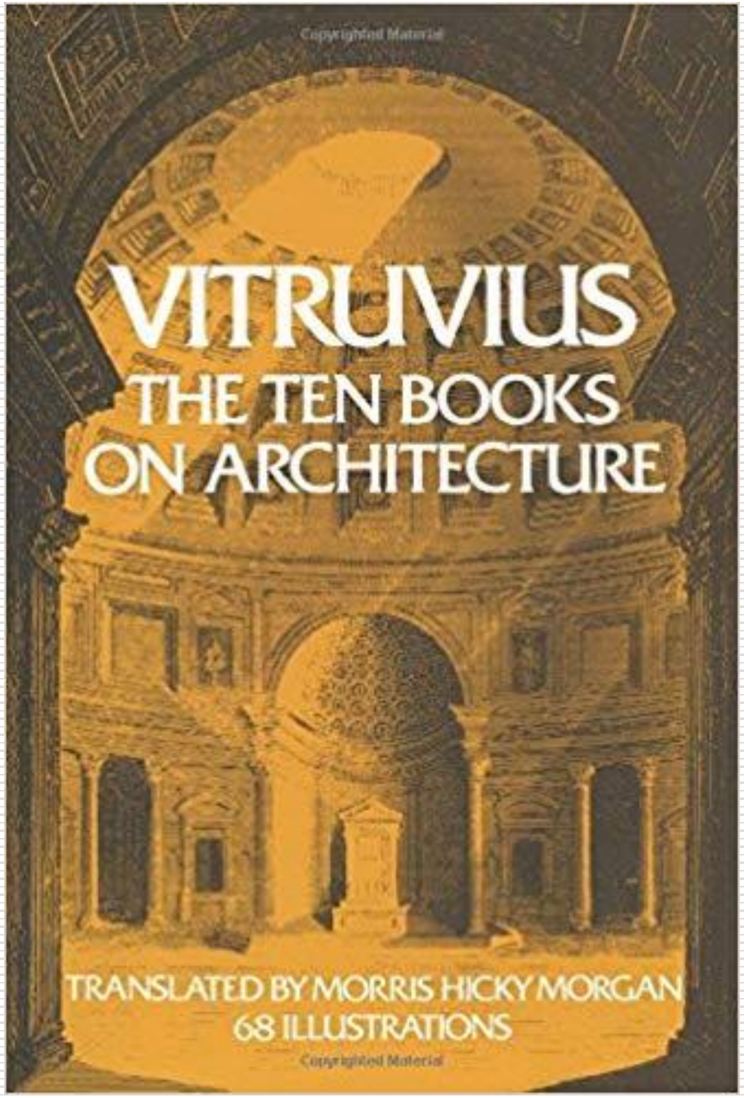




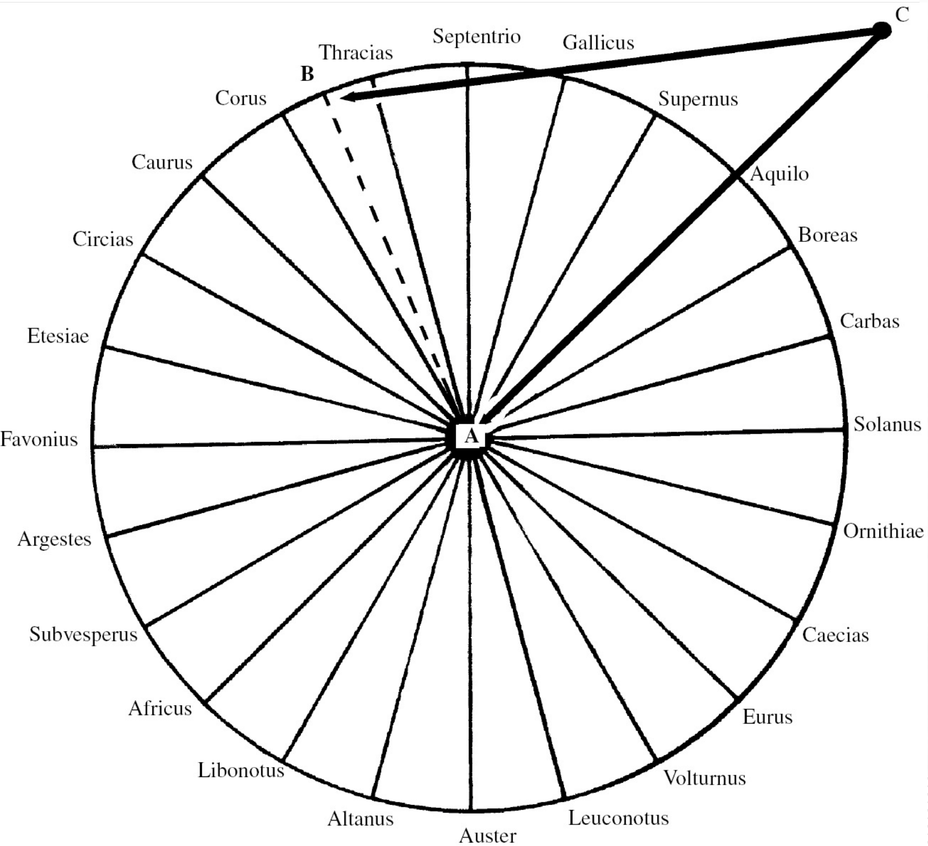
Acoustics – A maths problem since 2,500+ BC

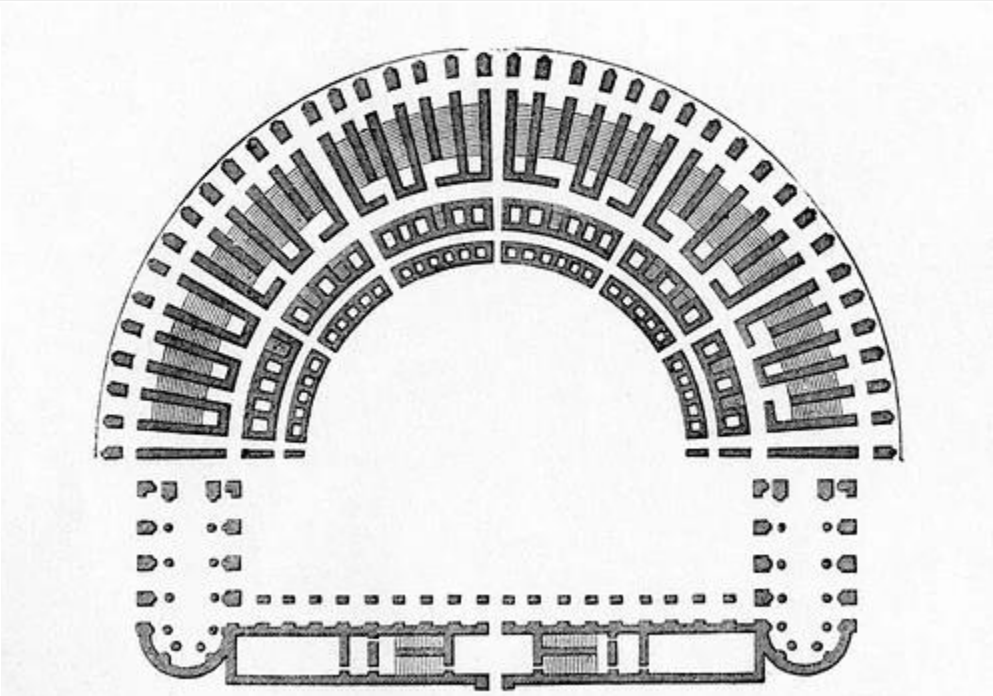


8. Theatre of Marcellus at Rome.



Waves of energy transfer through a "liquid"

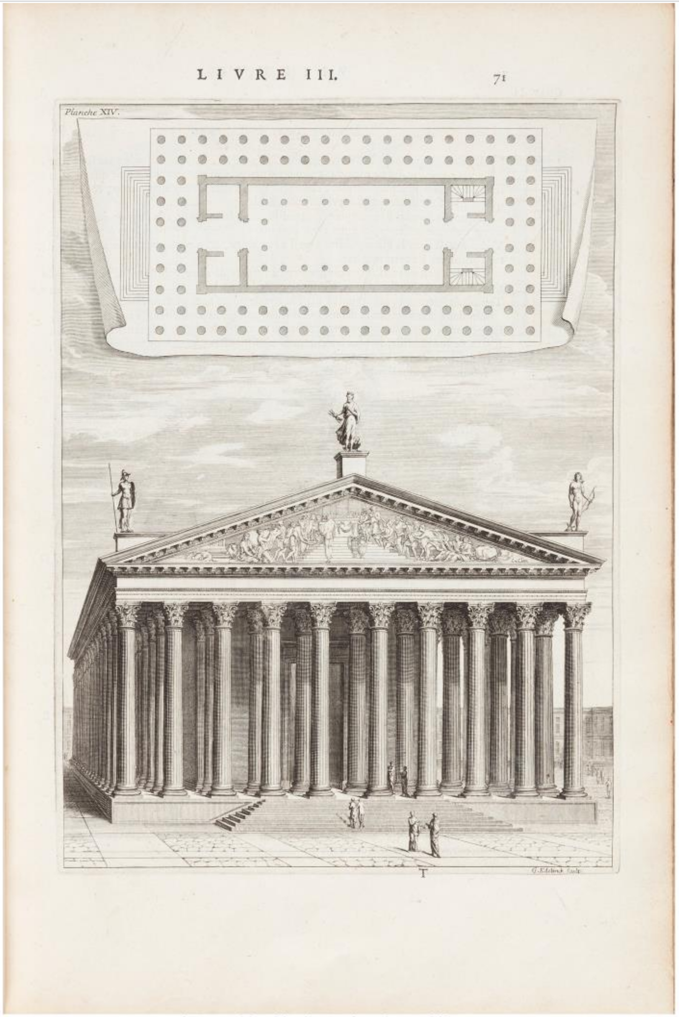




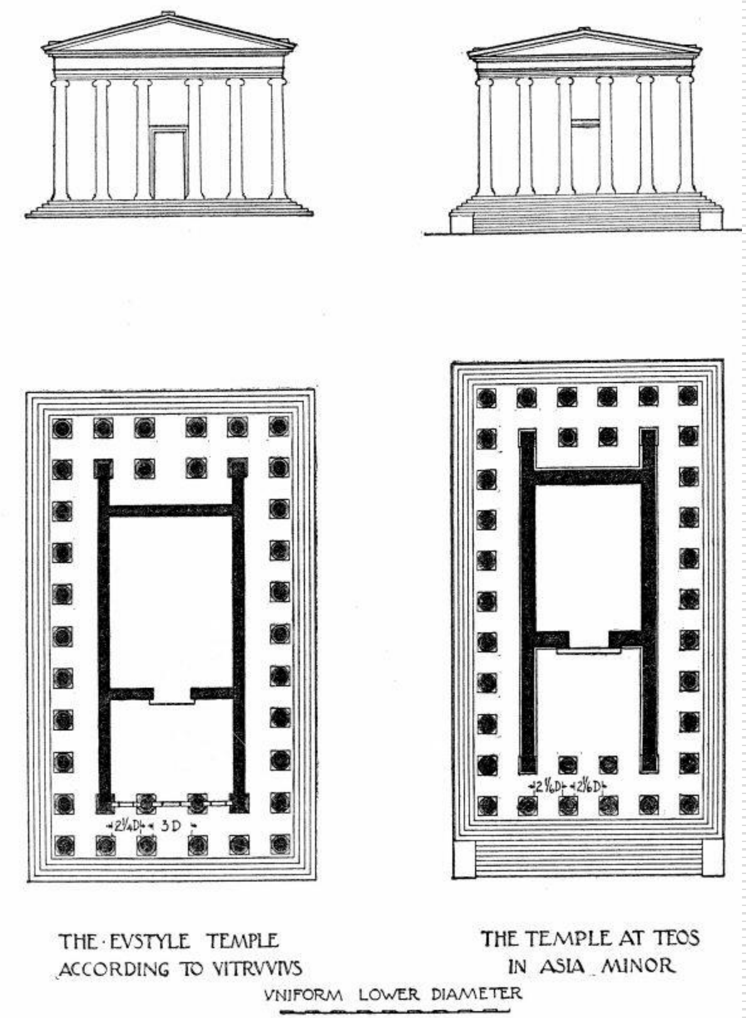
8. Theatre of Marcellus at Rome.

The physics were also applied to Temples

The temple is an “ambiance machine”

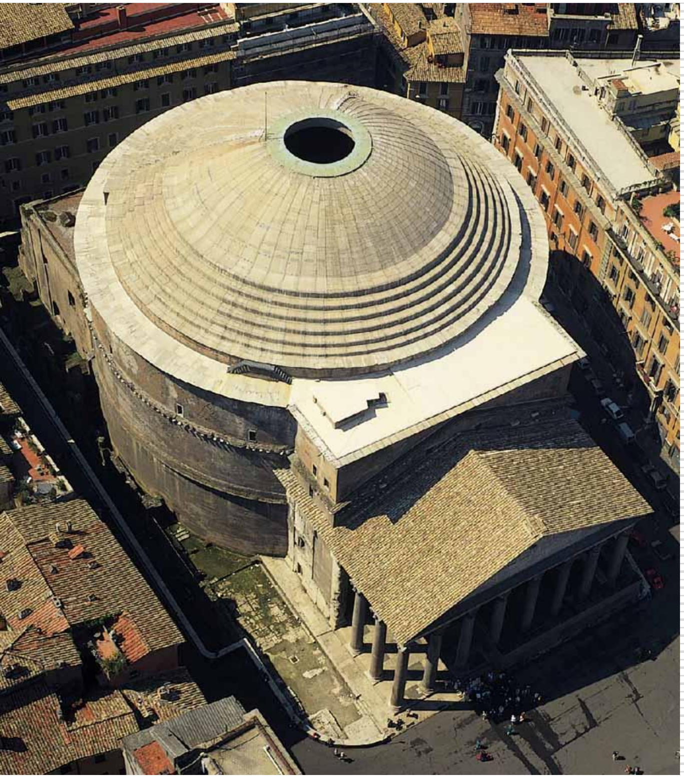


Imaged by Heritage Auctions, HA.com



Rome built sophisticated Temples – as differentiated from auditoria

The Pantheon is still a well studied engineering marvel



The Pantheon126 AD

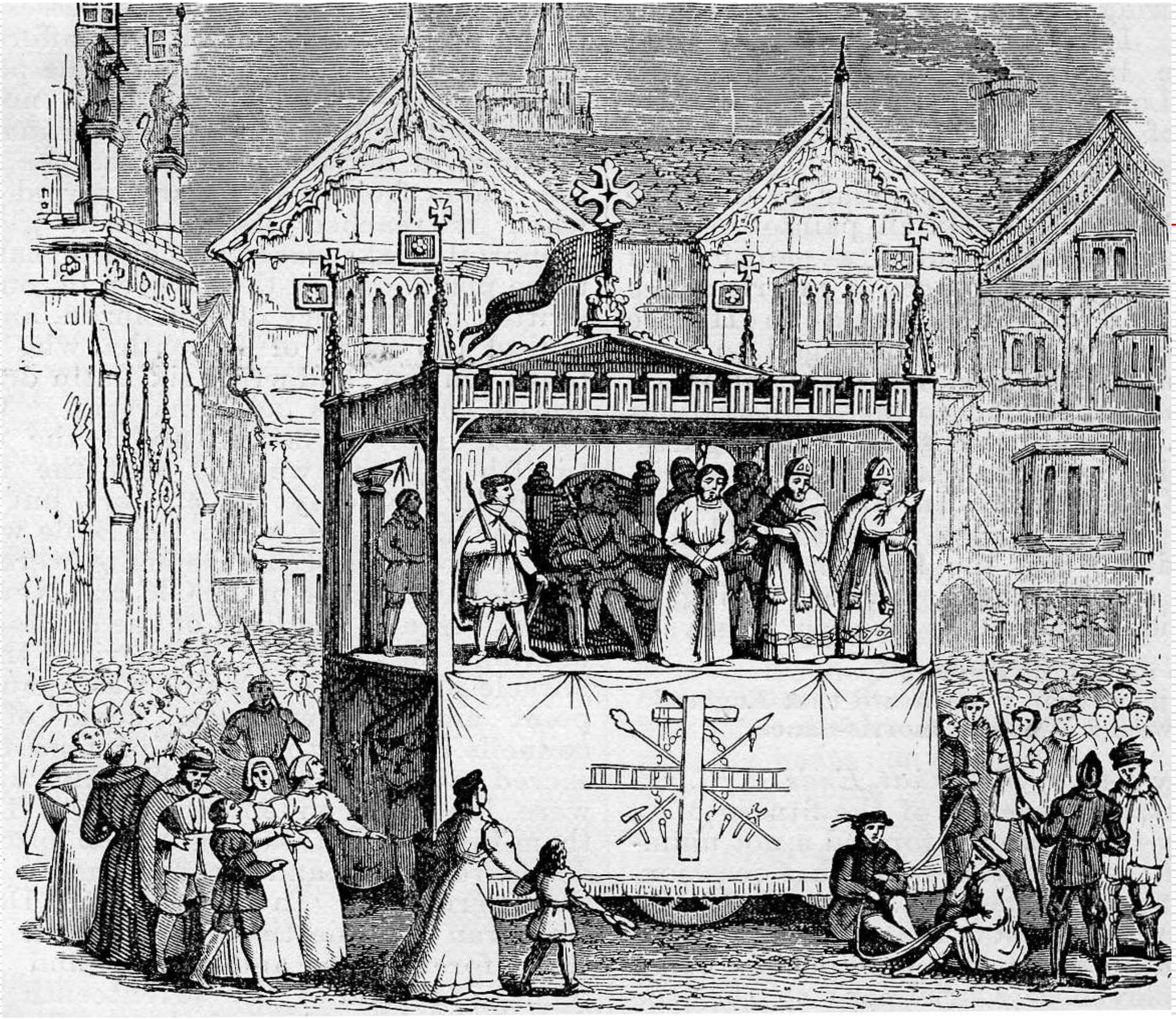
The Gothic Cathedral – Tailored for the Show

The Latin Mass was not necessarily meant to be understood

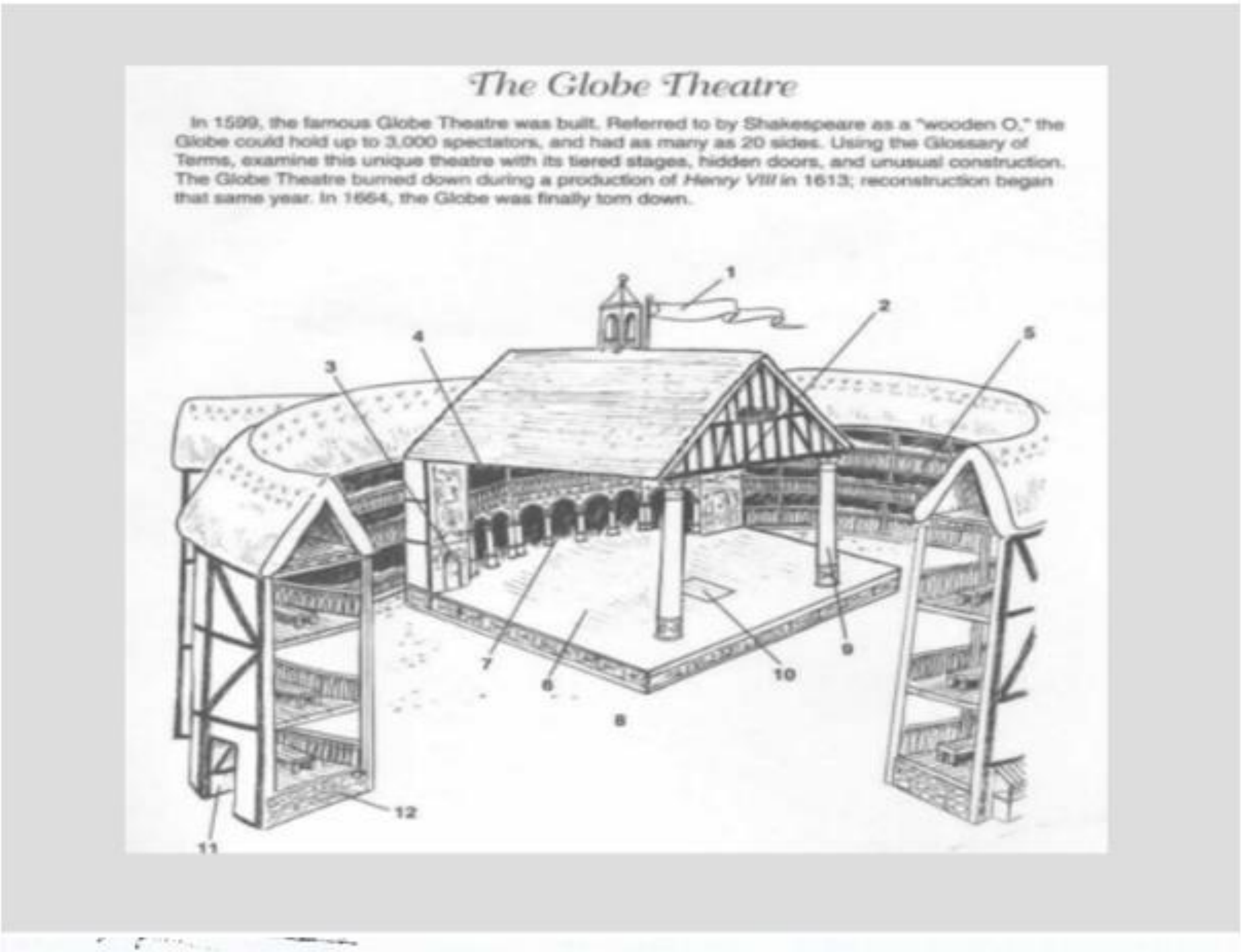
The multi second reverb time added to the mystical experience of the chanted Mass and the music

It follows that performance venues in the past were designed ***around the needs of the performance as well as an Architectural statement***

Medieval theatre didn't get the same priority as the church



By the Renaissance the lost technology was soon to be rediscovered
Intelligibility now required



Trivial Pursuit Question?

When was the first recorded published work in the “modern era” on acoustics for the purpose of understanding the propagation of sound waves?

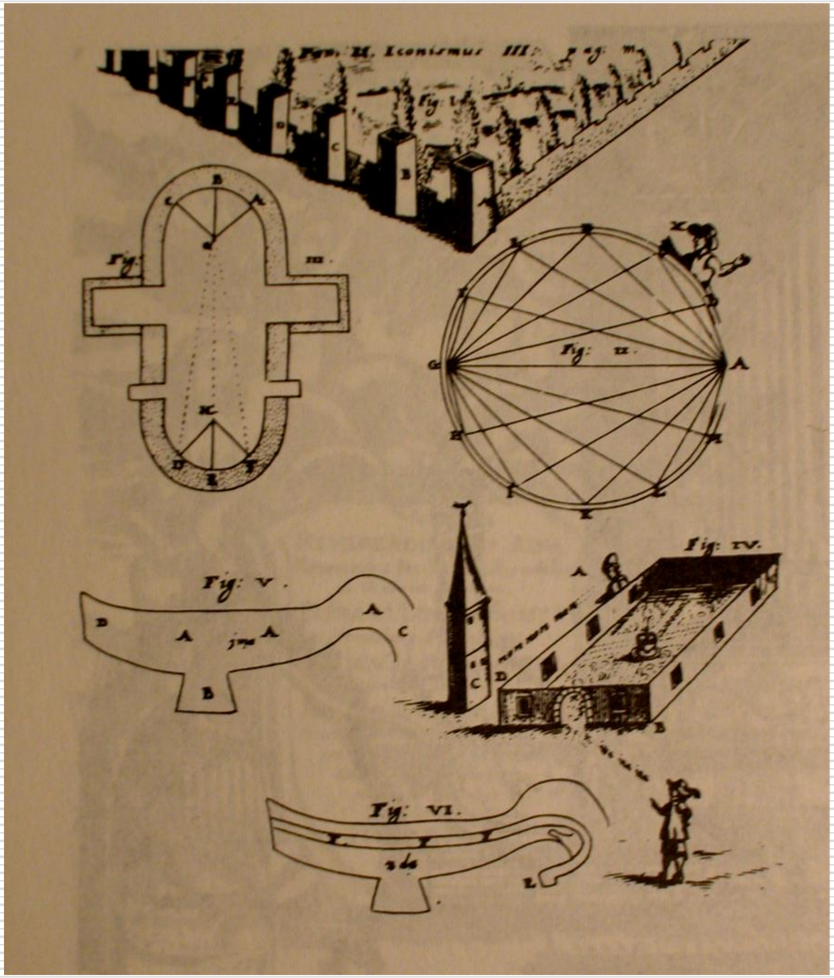
Answer: The effects of reflective surfaces

1657. The book *Magiae Universalis* by Gaspare P Schotto. (1608 – 1666) Written in Latin and published in Germany.

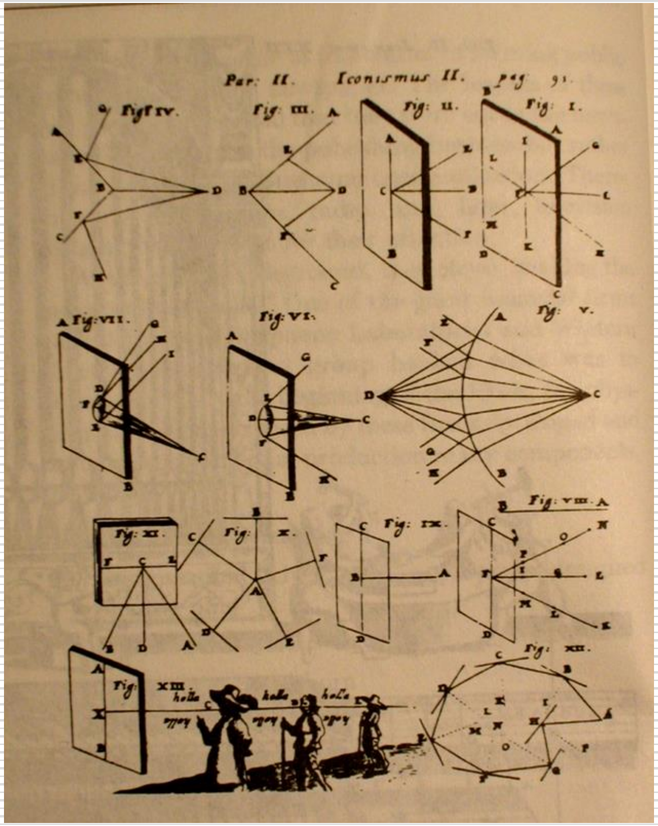
Gaspare was one of the few early researchers on acoustics whose work is still around today.

He was the first recorded person since the early Greeks to consider sound behaviour and distribution as a largely geometric problem.

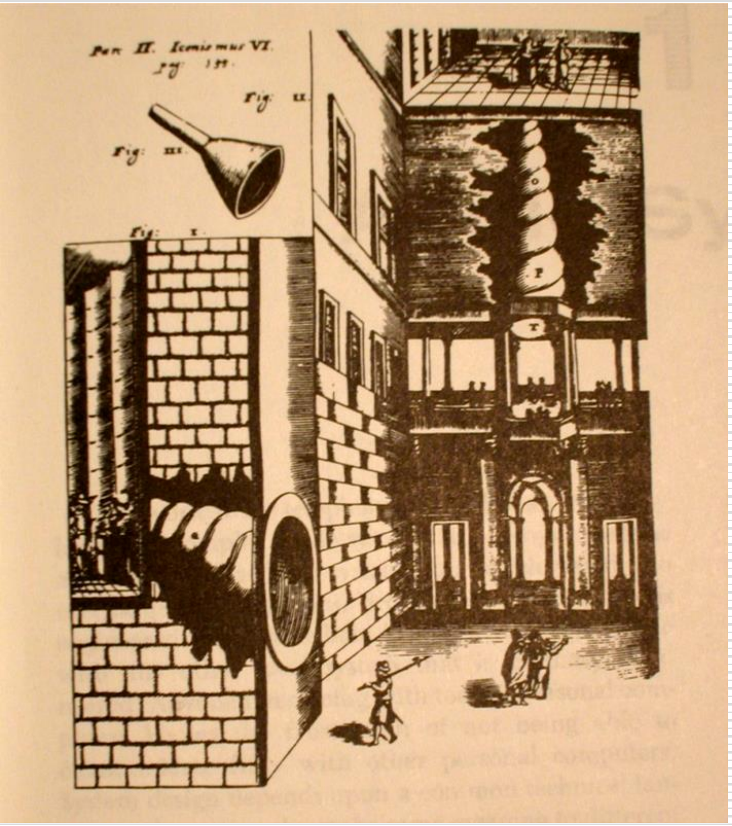
Illustration of reflections and sound wave behaviour within a semicircular reflective environment such as under a dome or within a parabolic stone wall.



Geometry and the horn

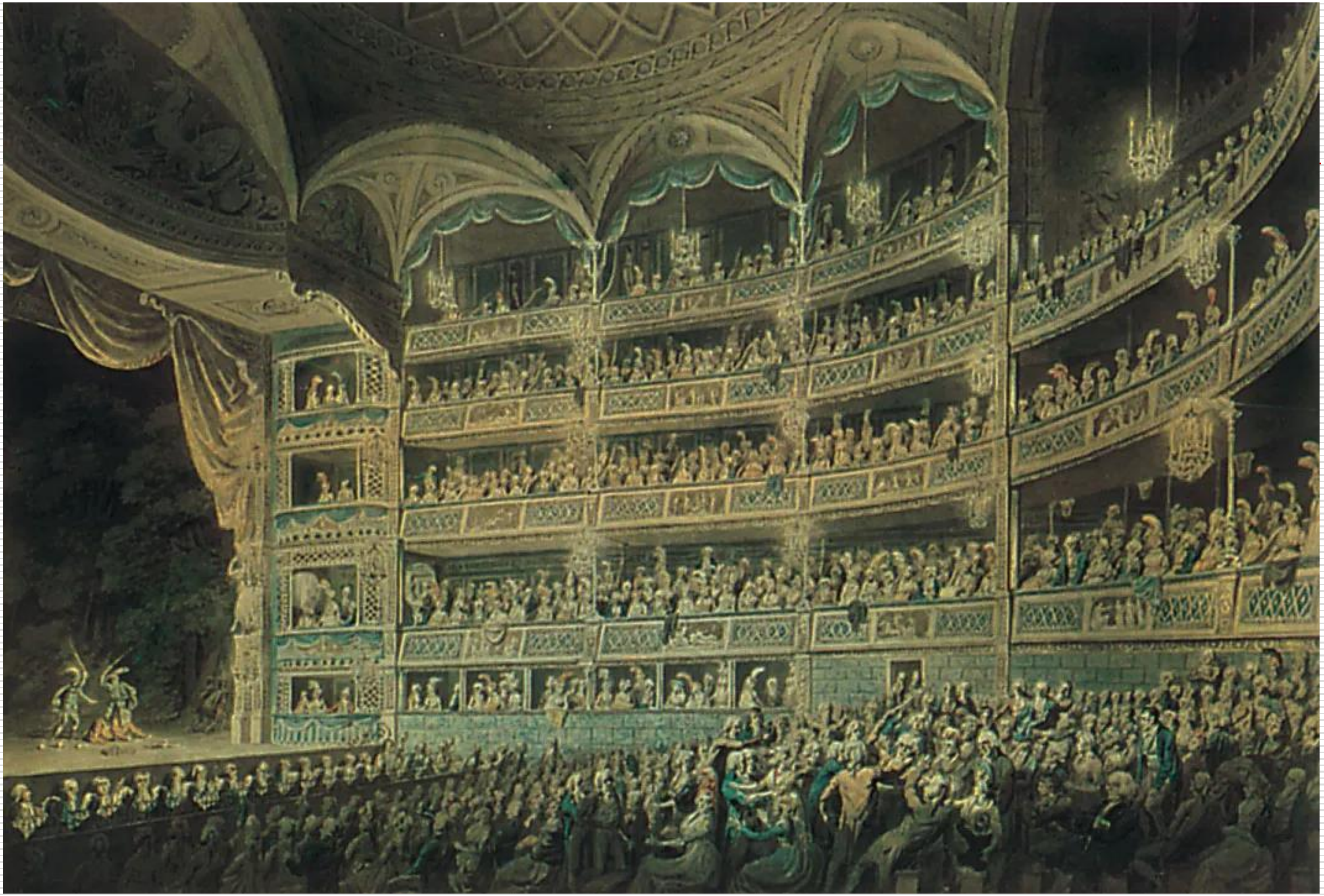


A graphic representation of the rules regarding reflective sound expressed as a geometric problem.



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

The late Renaissance the lost technology was refined to a new level - Intelligibility to the masses



The Teatro Olimpico - 1595 (Italy Olympic Theatre)
At least 3,000 years on from the Greek theatre

1890 Post Renascence but still Classically influenced (Still no Audio System)

The last 100 years – The demands of democracy

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.

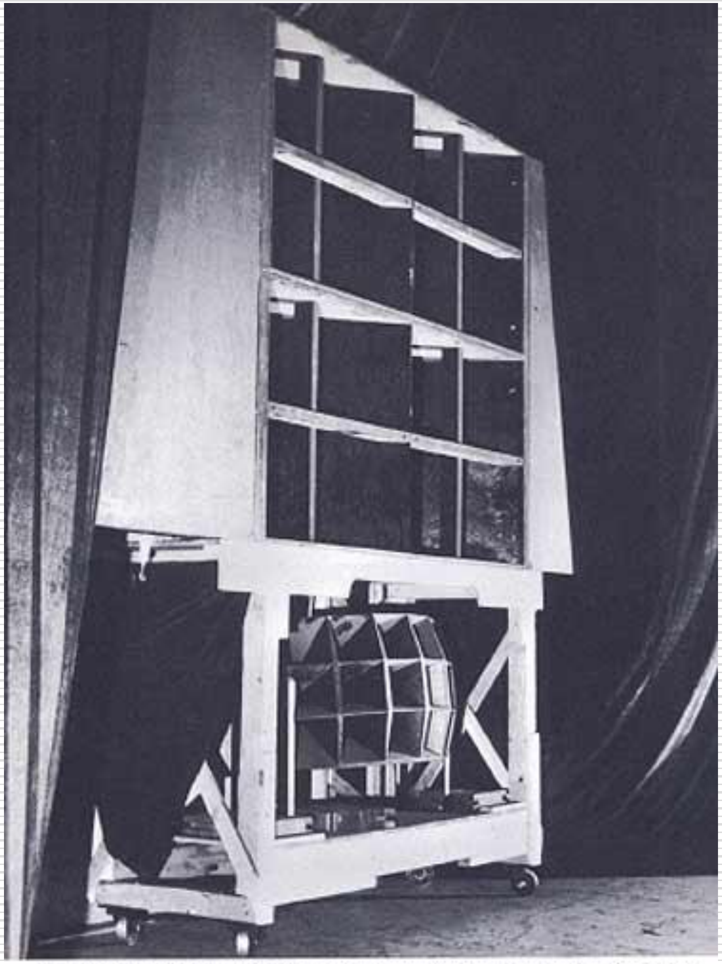
"If the first attempt in 1915 was primitive, by 1920 the convention had become a 'big gig'

The last 100 years

Cinema was the mother on invention

The main rush of developments came in the late 20's from the motion picture industry, eager to make the silent movies talk with synchronised pre recorded sound cover an auditorium full of people

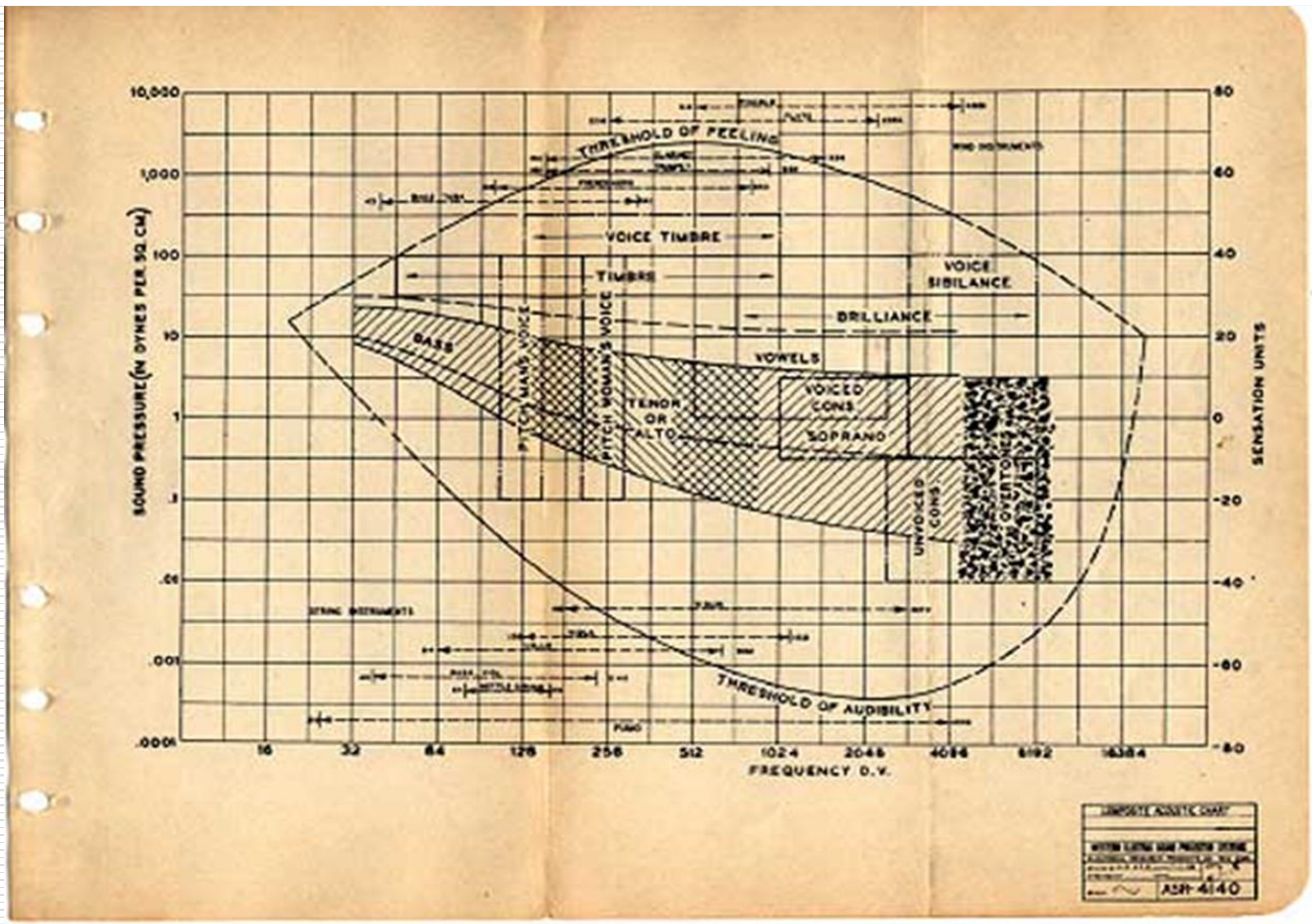
Amplifier power was limited so efficiency was the name of the game with horn loading



And this battery of loudspeakers behind the screen doesn't look much like Edison's horn.

The last 100 years

And as with all engineering exercises, measurement and prediction soon followed. Ok as a start but… Sensation units?



Buildings and Auditoria today?

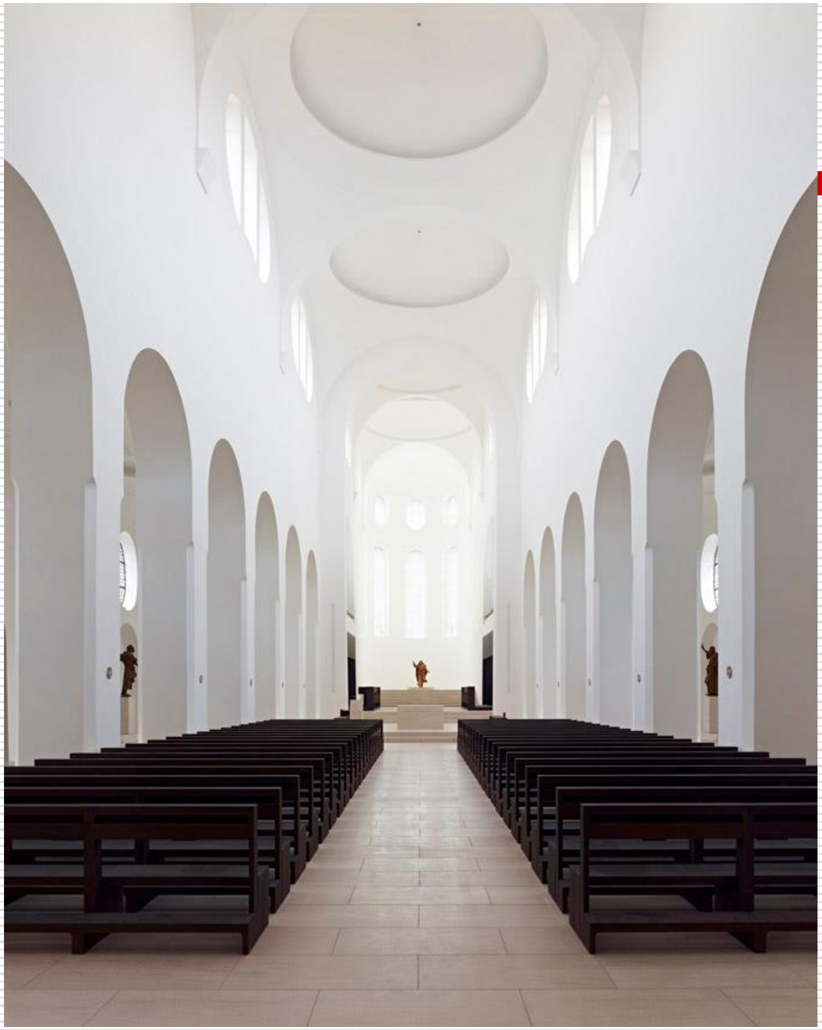
Once understood in ancient times, the science of speech intelligibility was now an "unknown unknown"

The architectural statement now is the priority

"We know where they [Iraq's WMD] are"

The real enemy of intelligibility

Reverberant fields and RT 60



The purpose of the audio system

Is to duplicate the live or recorded signal to the listeners in the listening area. To make the space

"A place of hearing"

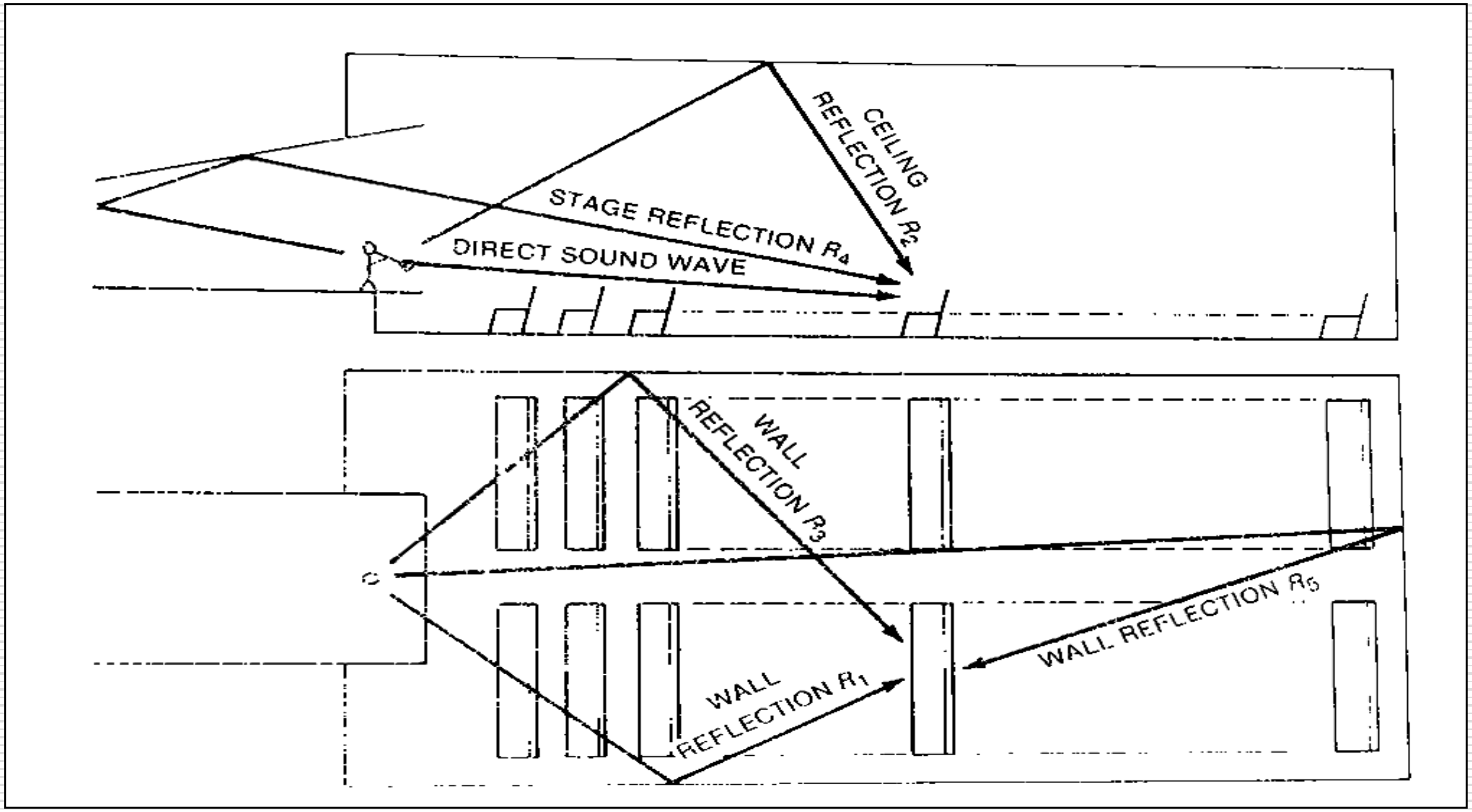
Buildings and Auditoria today?

The architectural statement is the priority

Things are probably not going to change

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

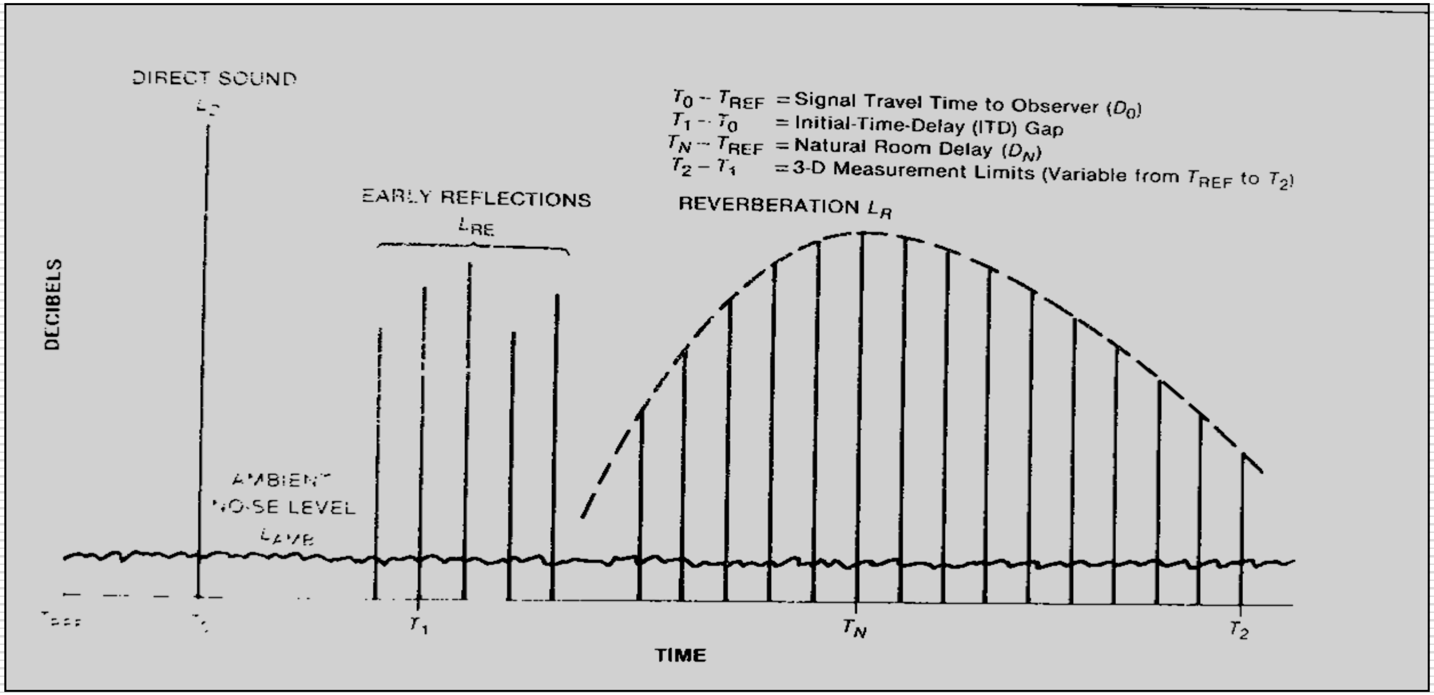
Echo (first reflection) **and reverberation** get confused when in fact they are related. Lets see what happens when we generate some sound in a small hall.



The Reverberant Field

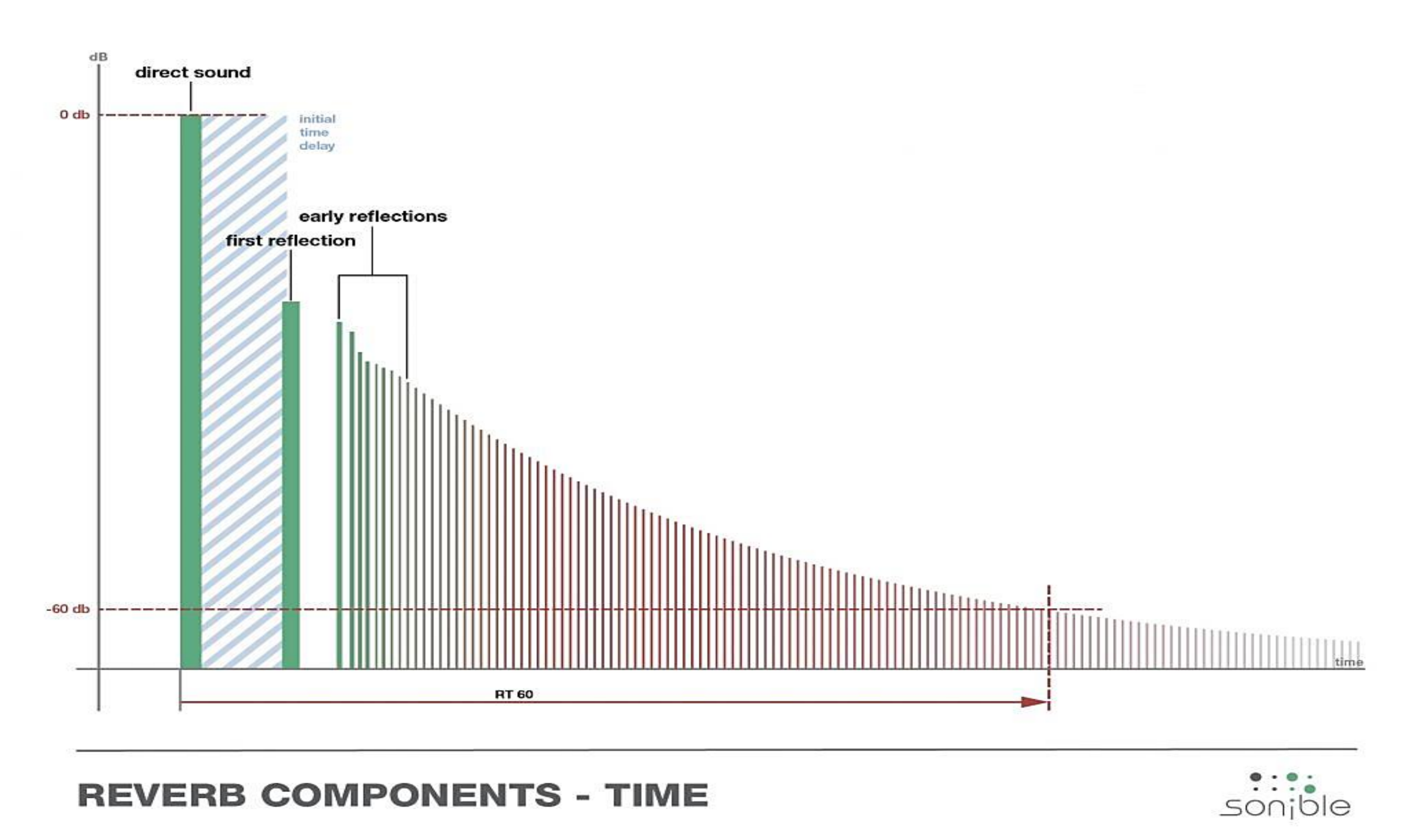
The reflections collapse into a random field of micro echoes in a ‘big swooch’ as the energy dissipates through out the room.

RT60 is the time that it takes the reverberant sound to decay 60 dB.

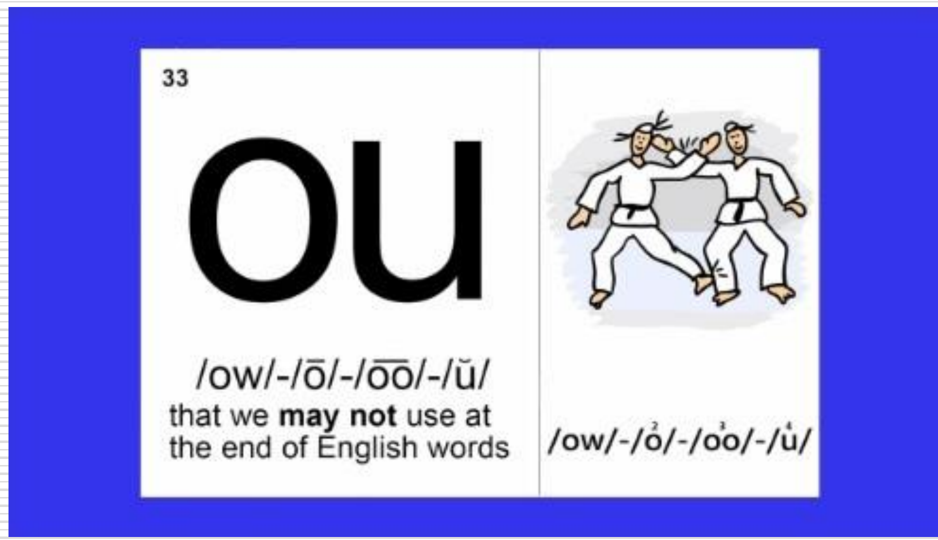


Early Reflections and Acoustic Energy

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

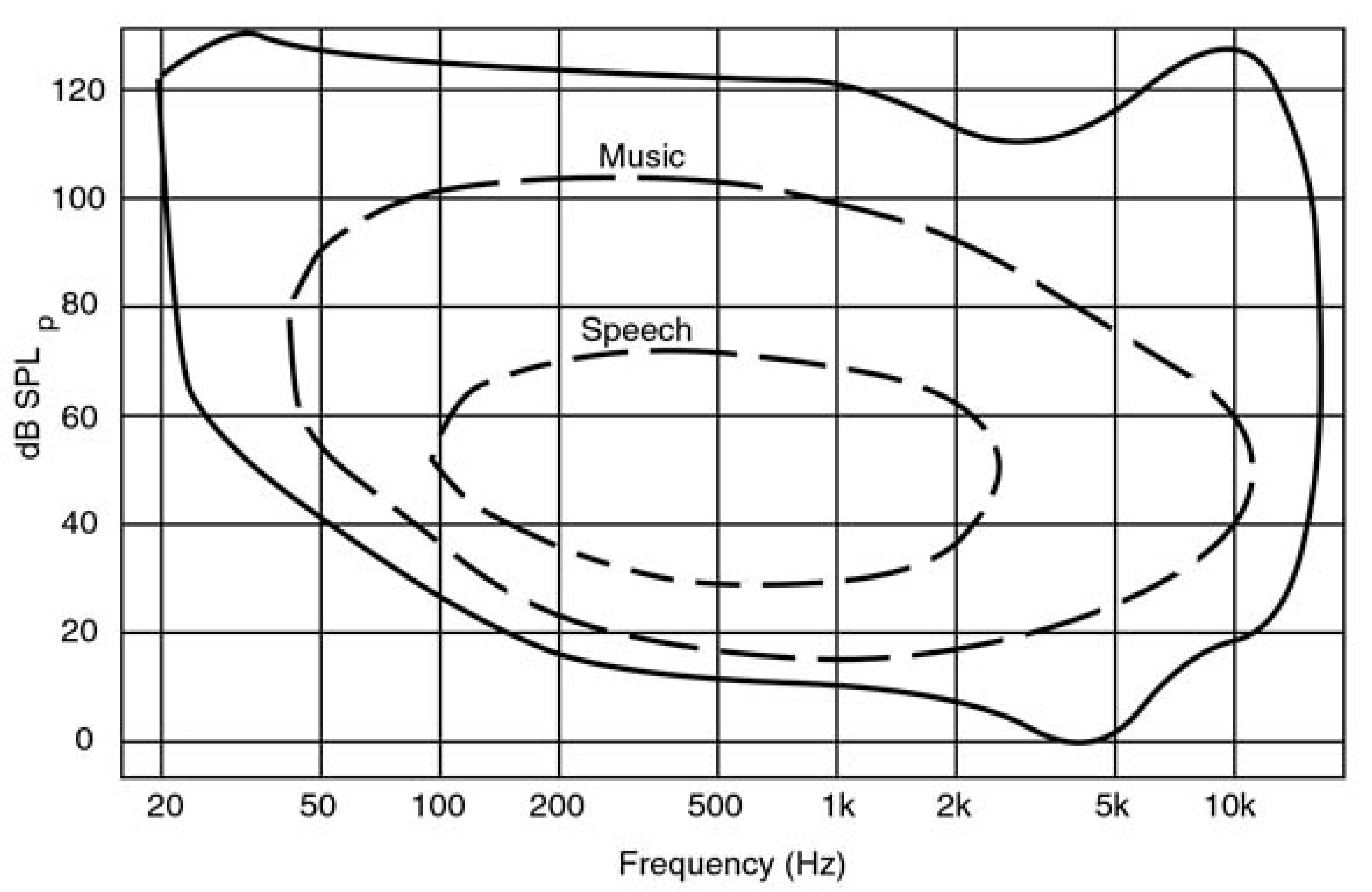


Influence of Reverberation on speech

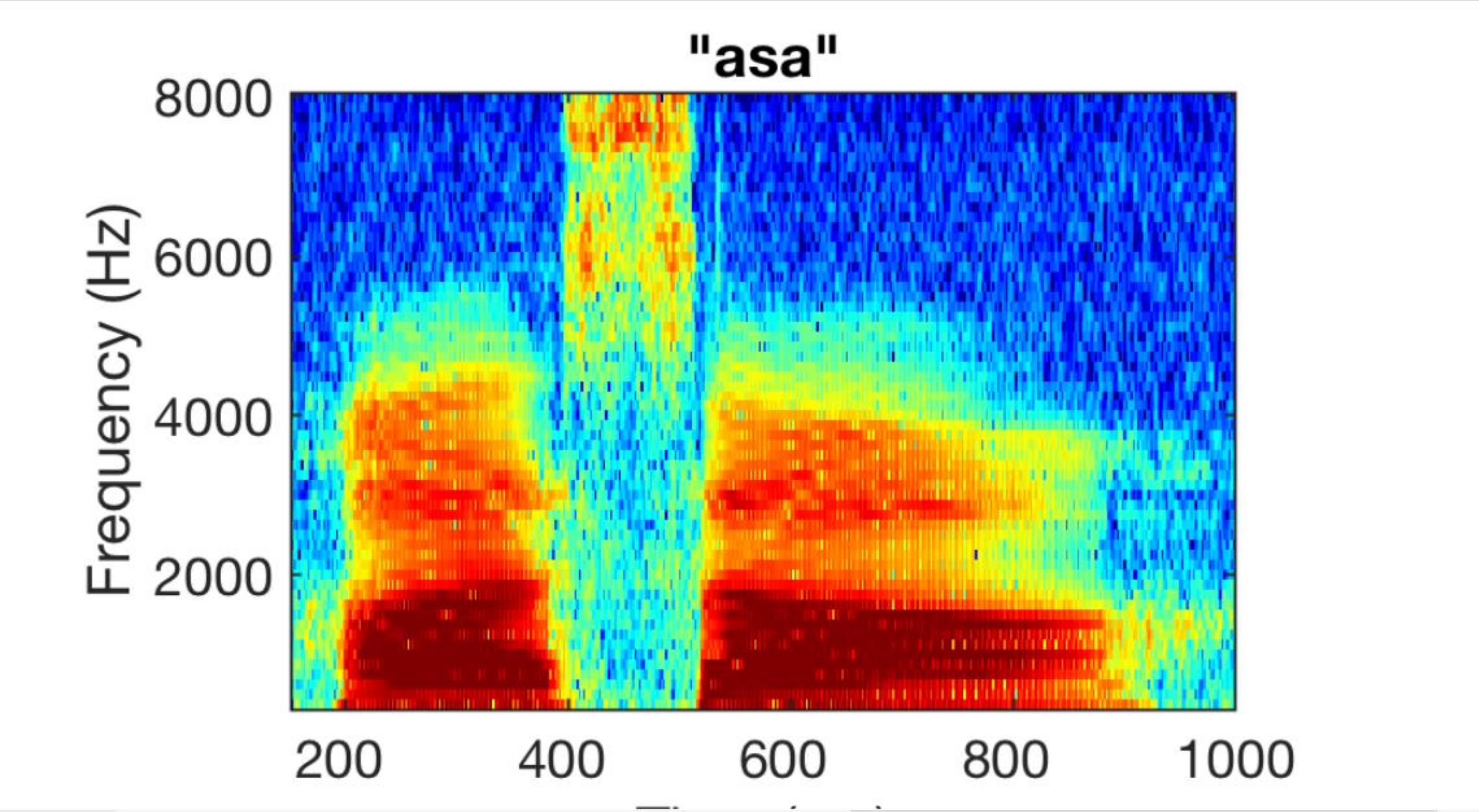


Influence of Reverberation on speech

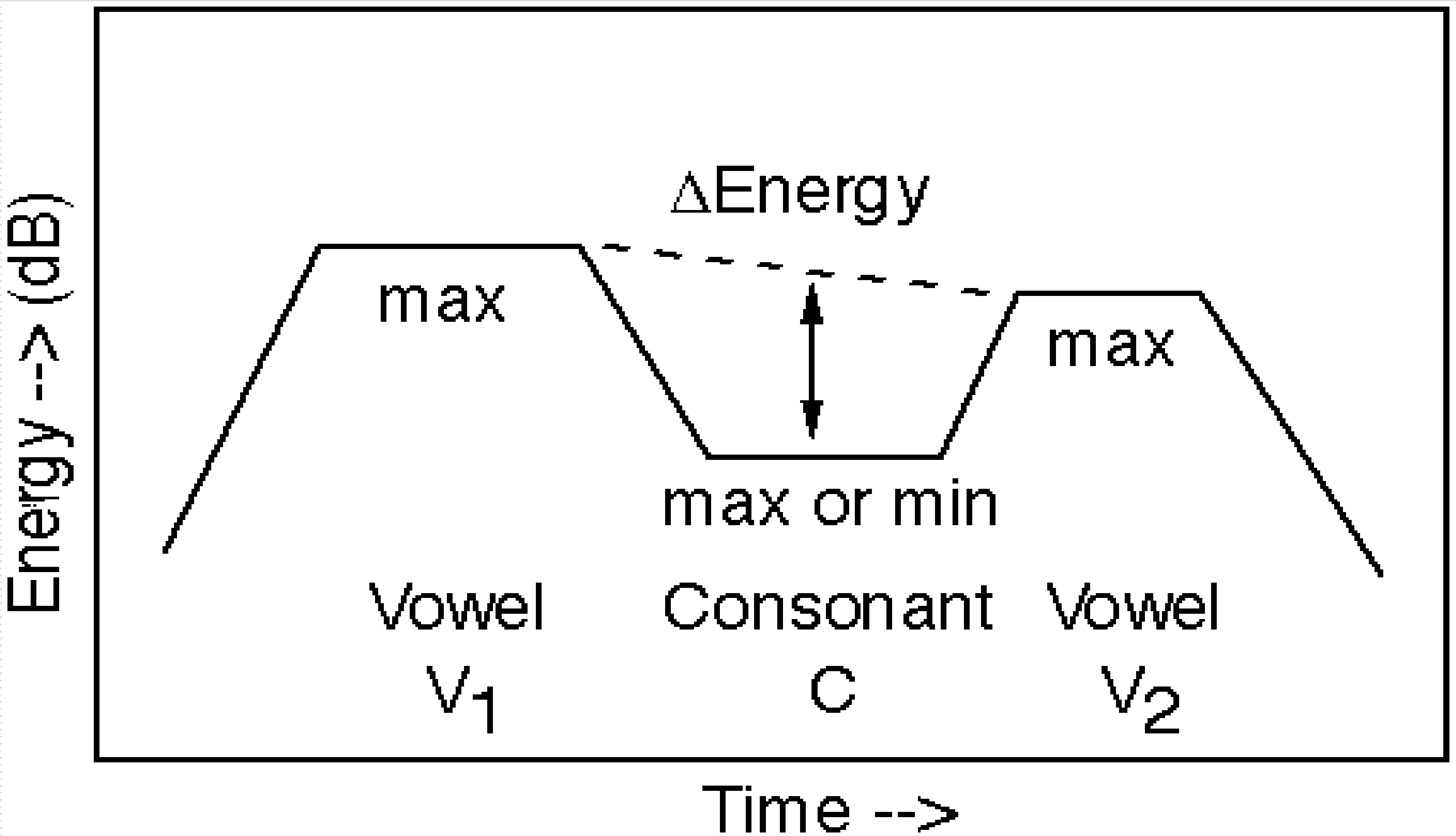
(And the energy levels)



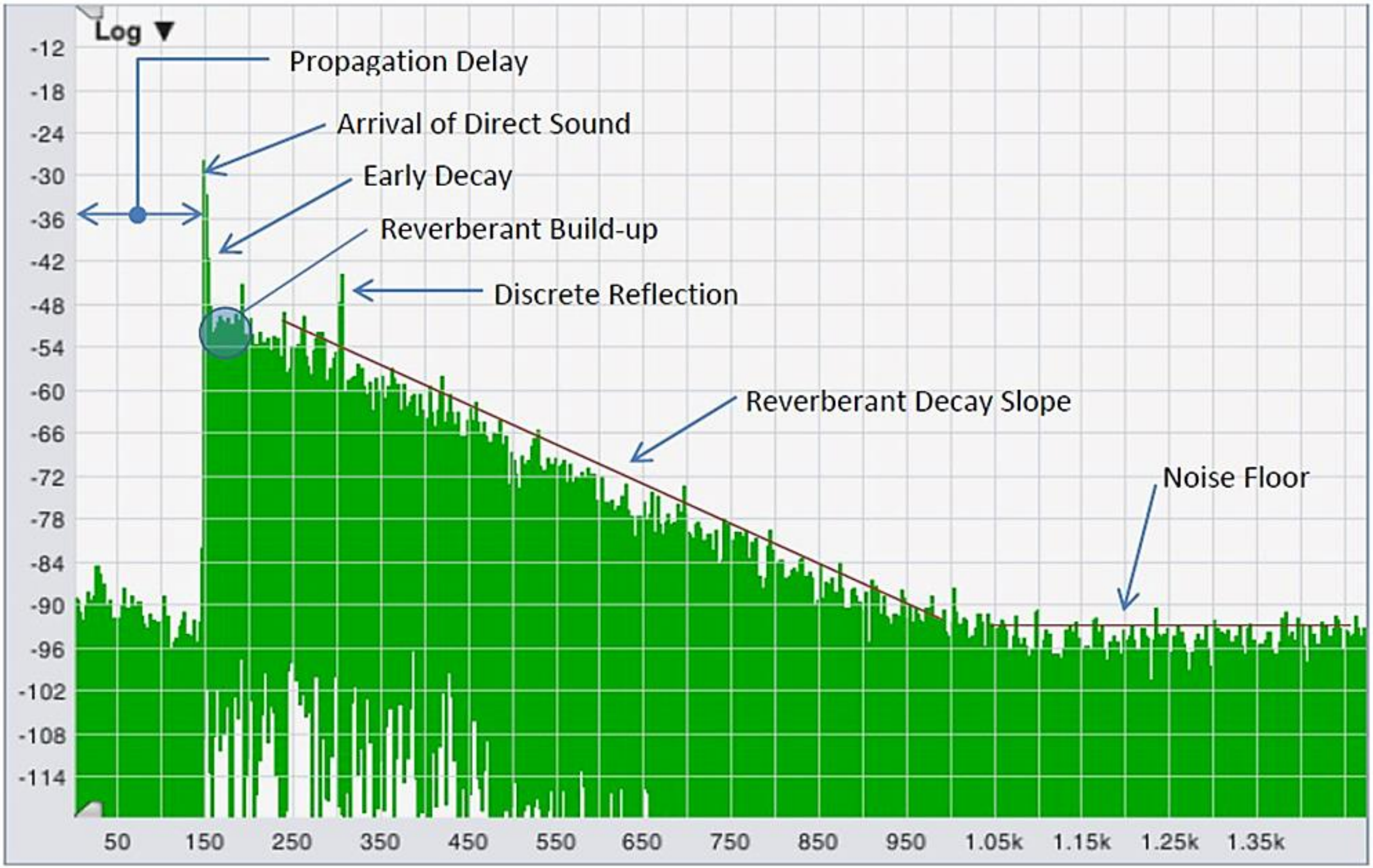
Influence of Reverberation on speech



Influence of Reverberation on speech



Influence of Reverberation on speech



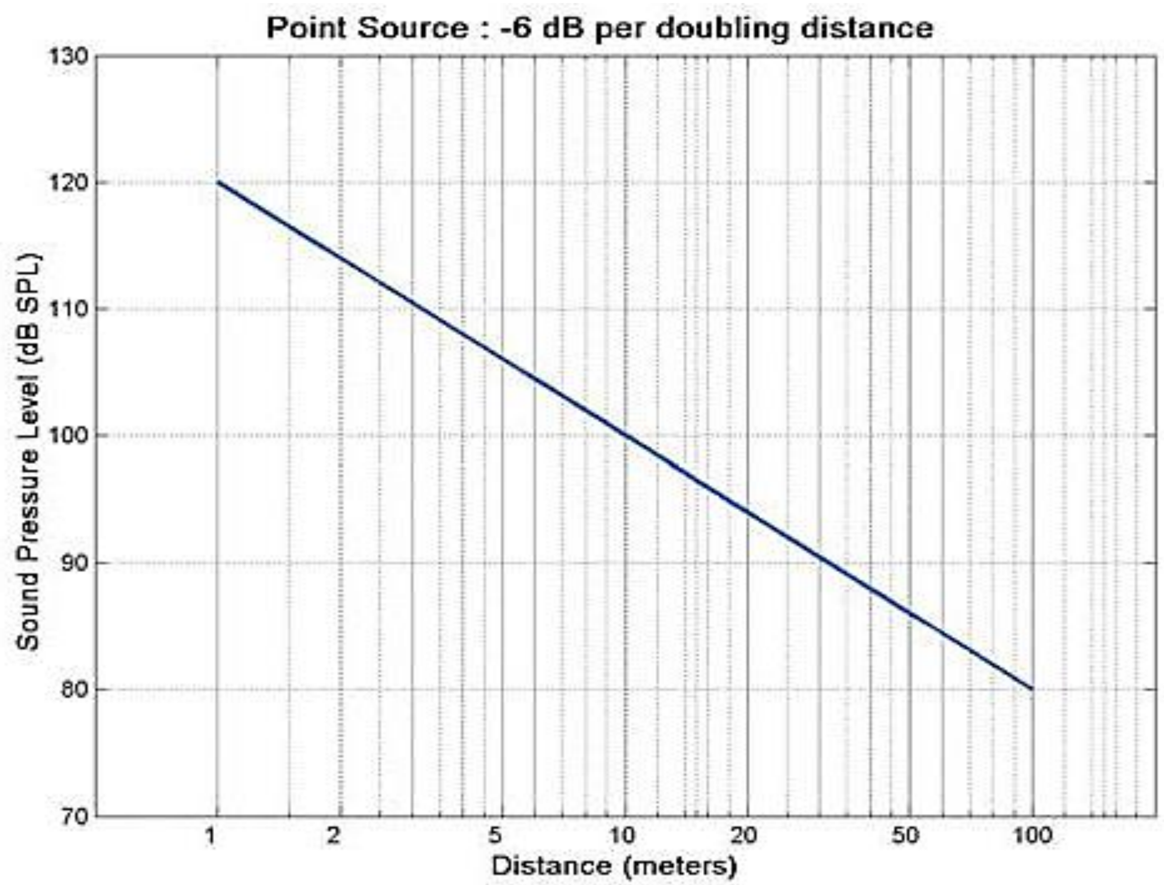
The sound system- Is loud enough.. enough?

“Loud” isn’t loud for long

Commencing with a sound wave SPL of 120 dB. Even a single box potentially capable of 135dB peaks, at 40-50 meters we will have a big drop in level

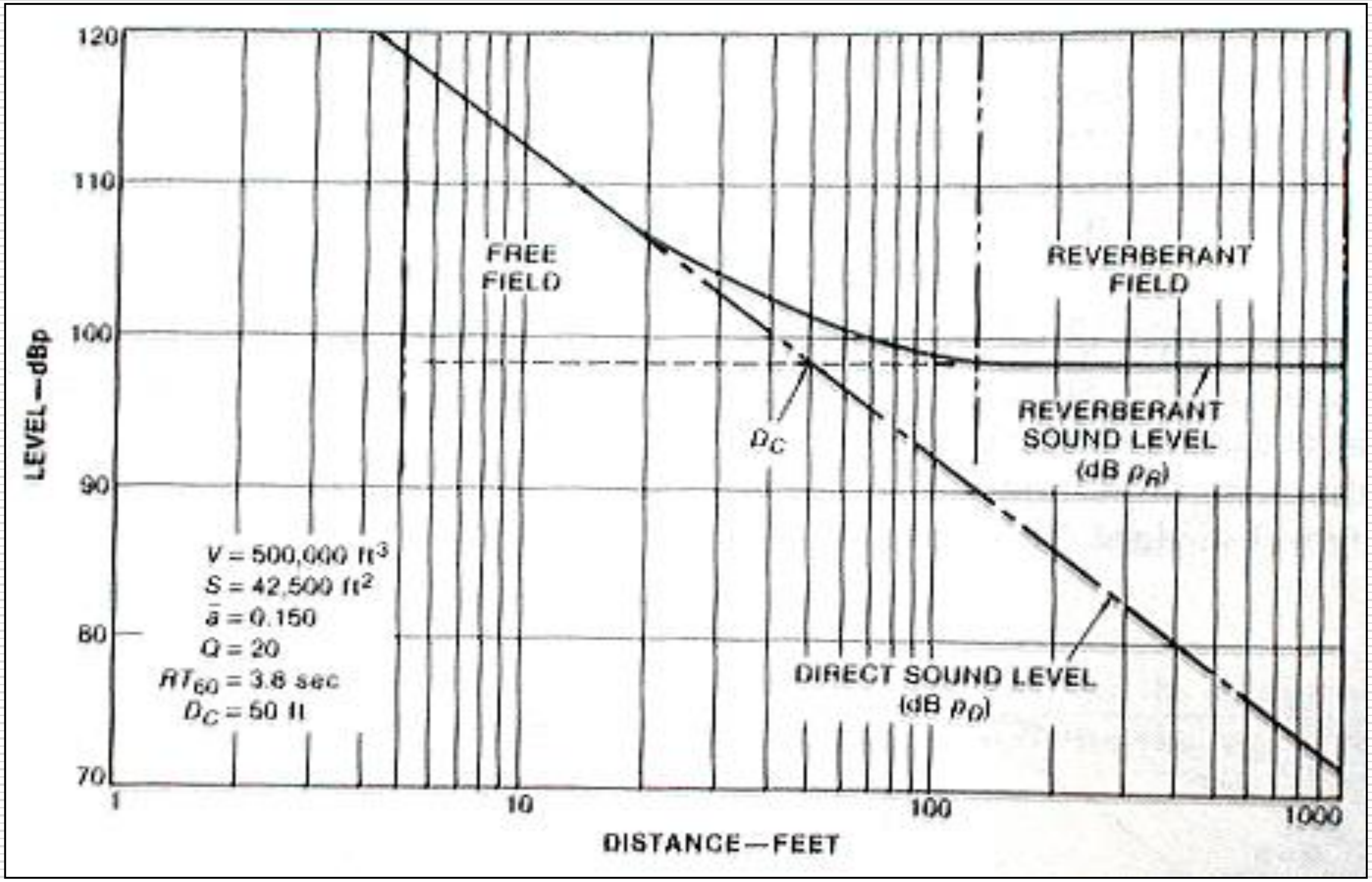
At 50 meters a person can speak louder than the PA.

So by 50 meters, a boost is needed



PA coverage - 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



The inverse square law

It is a ratio that a change of one unit more or less in quantity, will double the result.

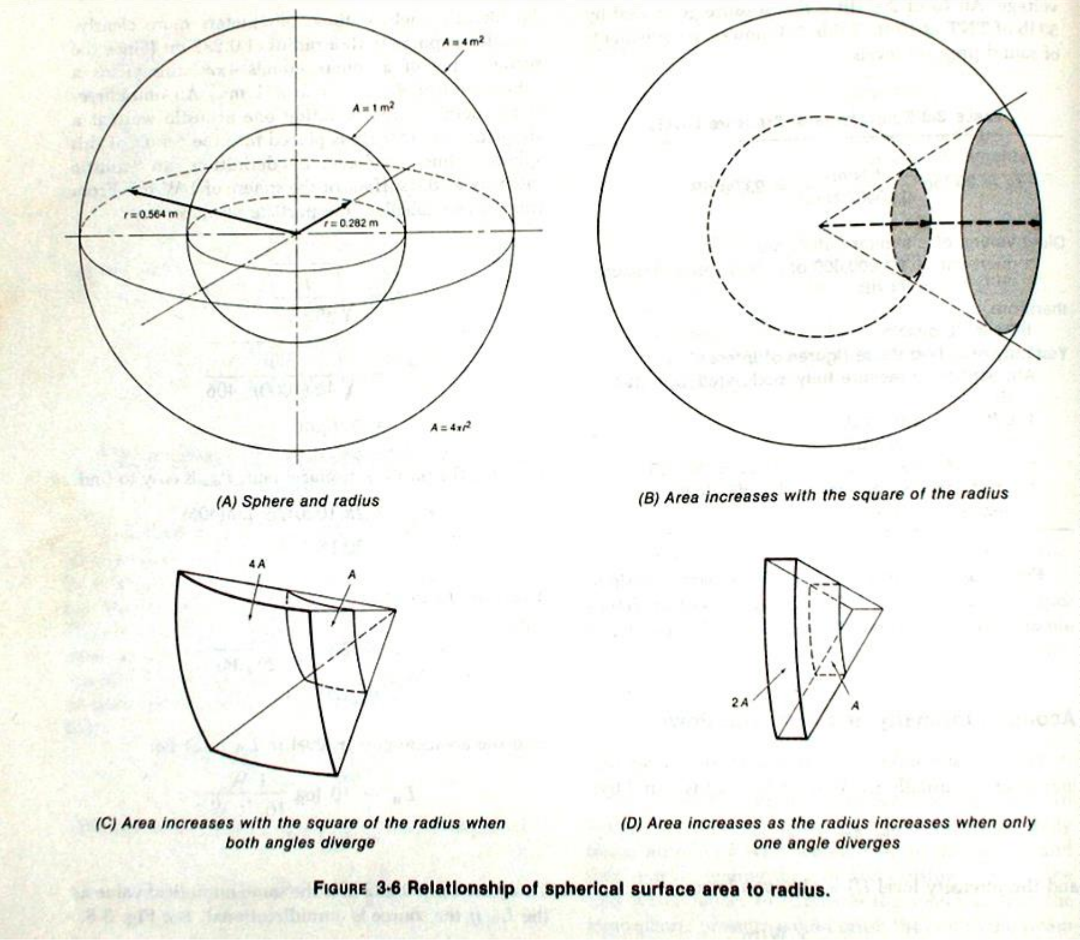


FIGURE 3-6 Relationship of spherical surface area to radius.

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.

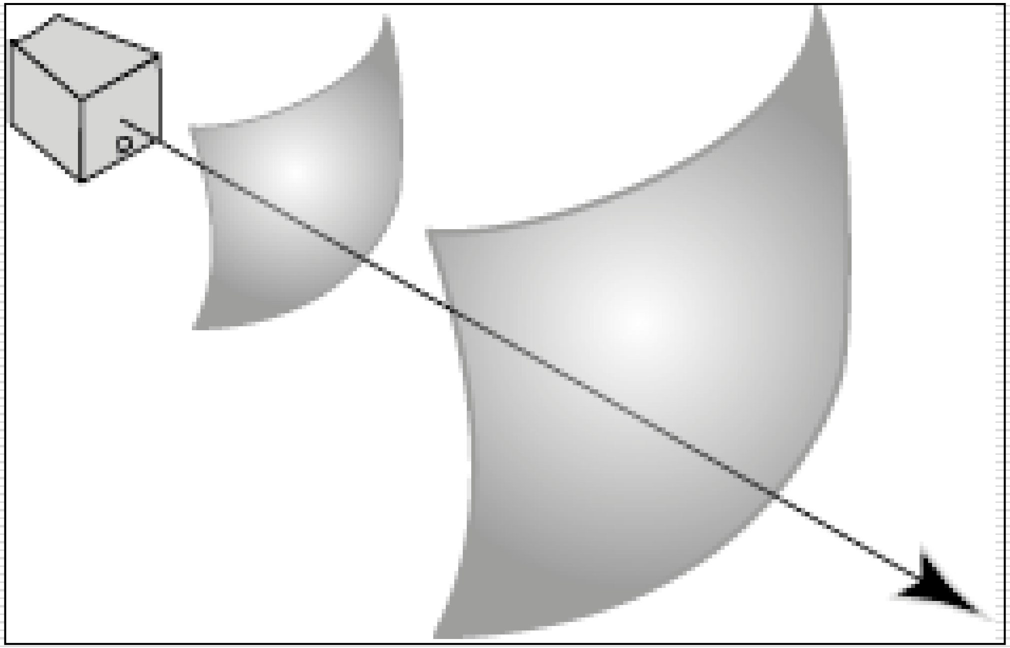
At 2 metres the light intensity will be one quarter the light level at 1 metre.

Getting coverage over longer distances

A system engineer in charge of installing a concert PA system must understand how sound travels through air

Sound is a vibration that travels through the air at about 1,100 ft (366.7 meters) per second.

Good coverage is an energy distribution problem and to solve you need to understand wave behaviour



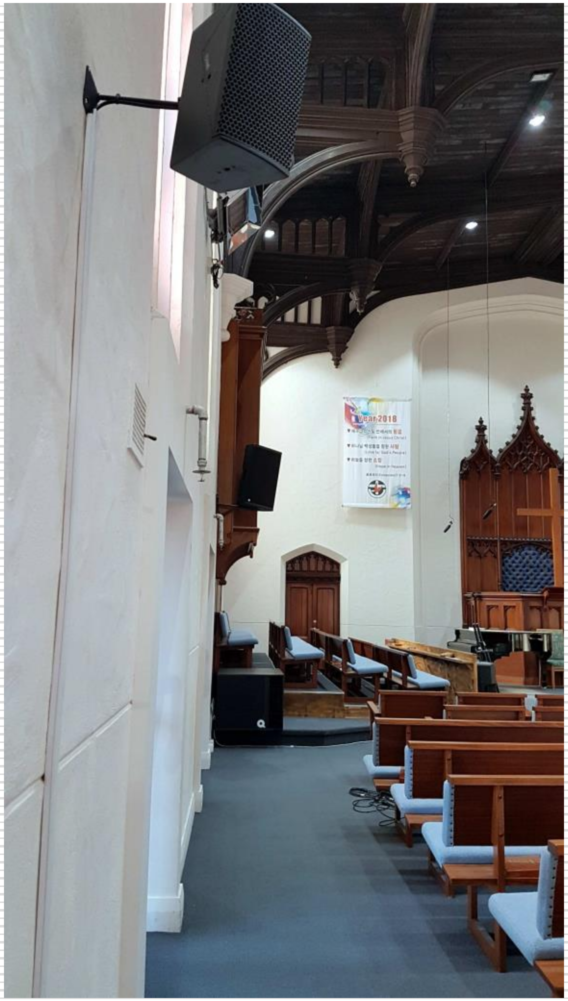
Additional loudspeakers to "boost" dB

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

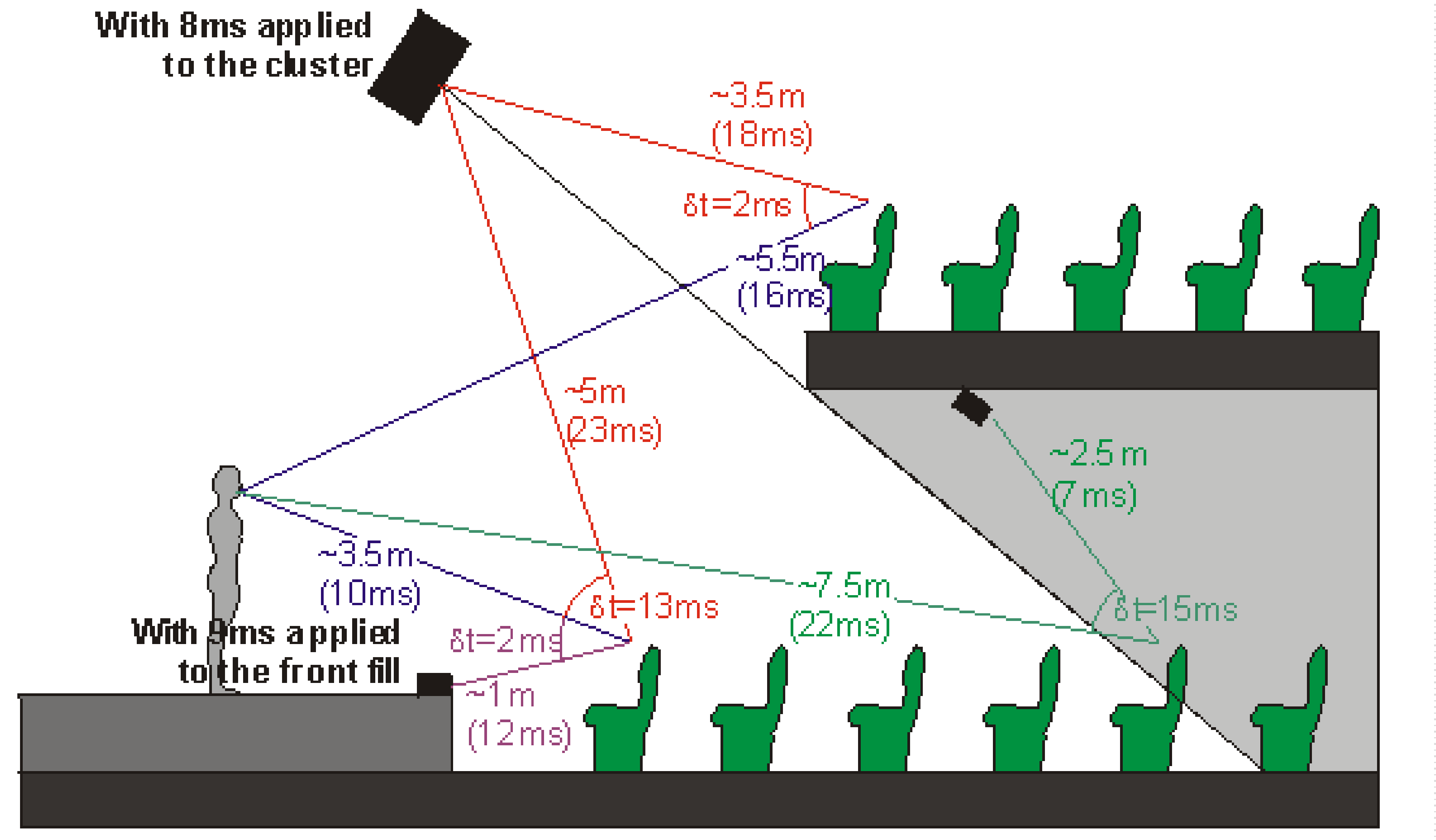
Pictured is a front of house supported by a delayed system behind the mixing position.

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.

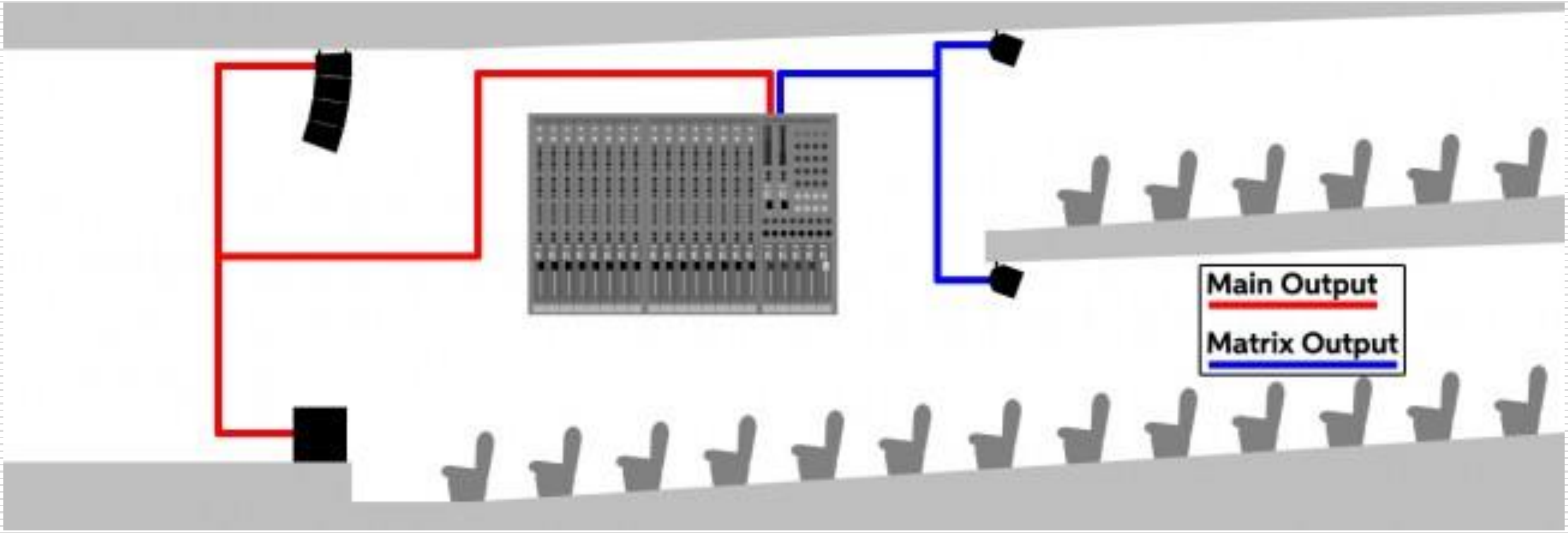


How is it done without the whole area being drowned in echoes and reflections?



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.



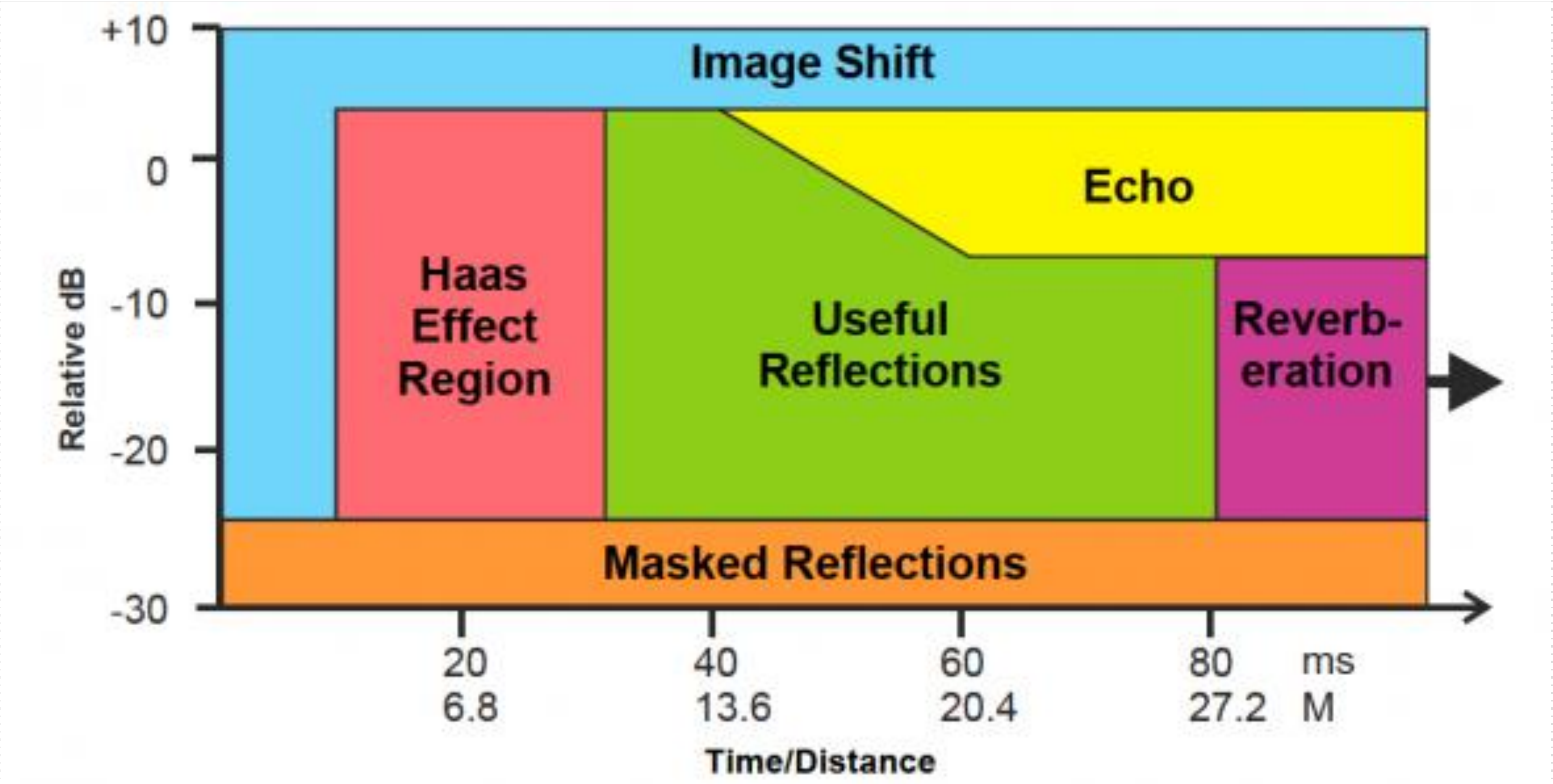
How to set the delay times

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.

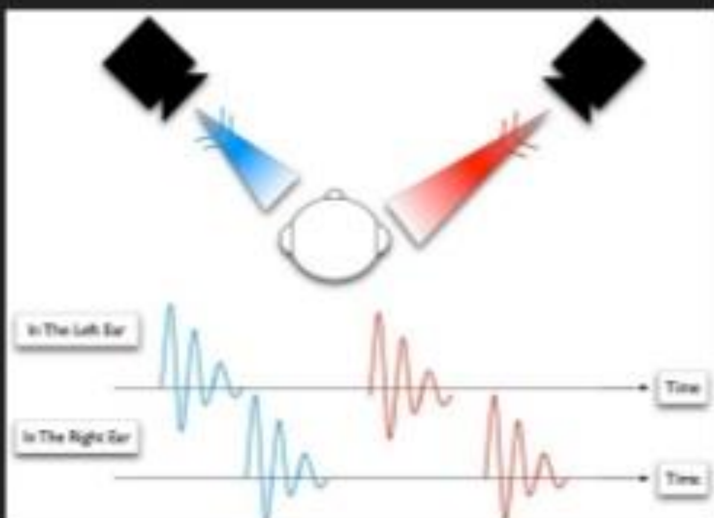
The Directional Illusion

The **HASS effect also known as the Precedence Effect**, is a psychoacoustic effect described in 1951 by Helmut Haas, who discovered that when a sound is followed by another sound with a short delay time in between them, the listener perceives a single sound from the earlier source



Applying Delay Times and Haas Effect

The Haas Effect
(or precedence effect)



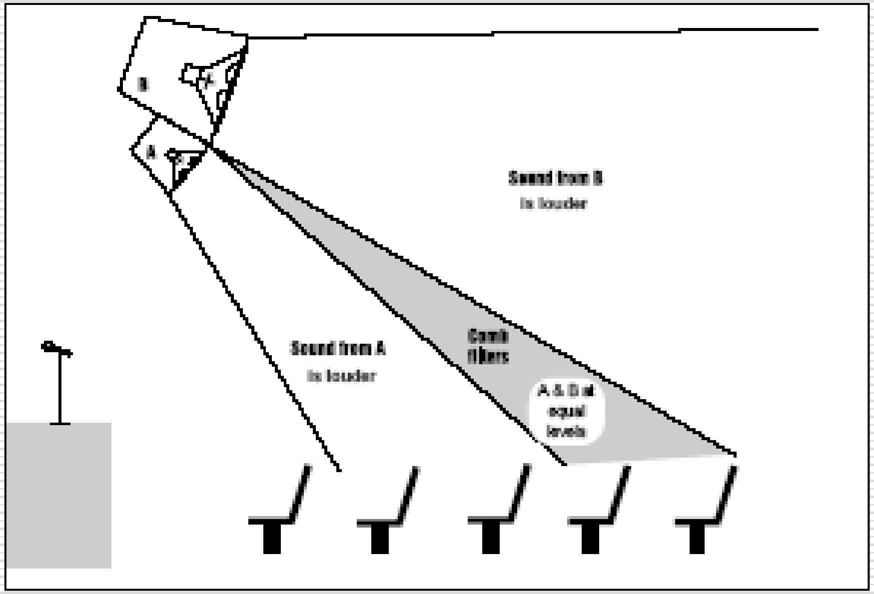
The Haas effect can be summarised as follows:

- The ear will attend to the direction of the sound that arrives first and will not attend to the reflections providing they arrive within 30 ms of the first sound.
- The reflections arriving before 30ms are fused into the perception of the first arrival. However, if they arrive after 30ms they will be perceived as echoes.

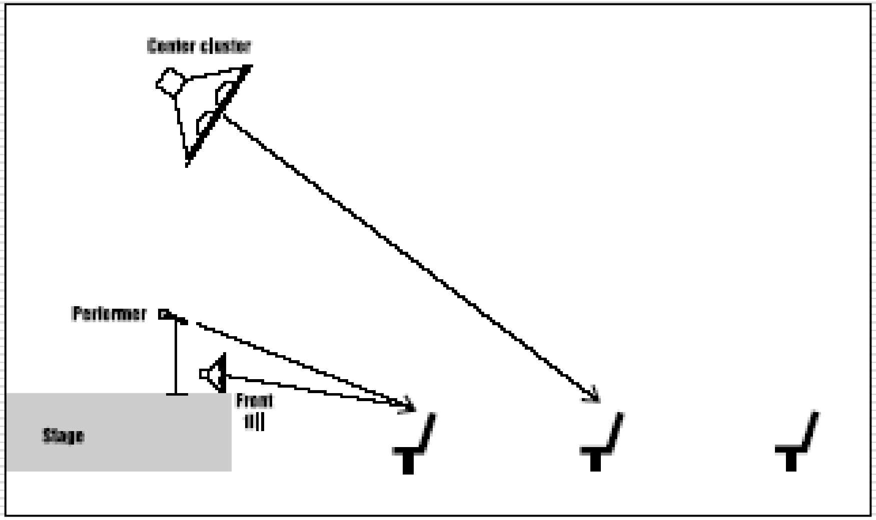
Angus, J & Howard, D (2009) Acoustics and Psychoacoustics. UK, Elsevier.

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.



The key points for setting delays

Visualise how the wave is going to propagate from the primary system and locate the delay boxes so they will be pointed in the same direction as the radiating wave.

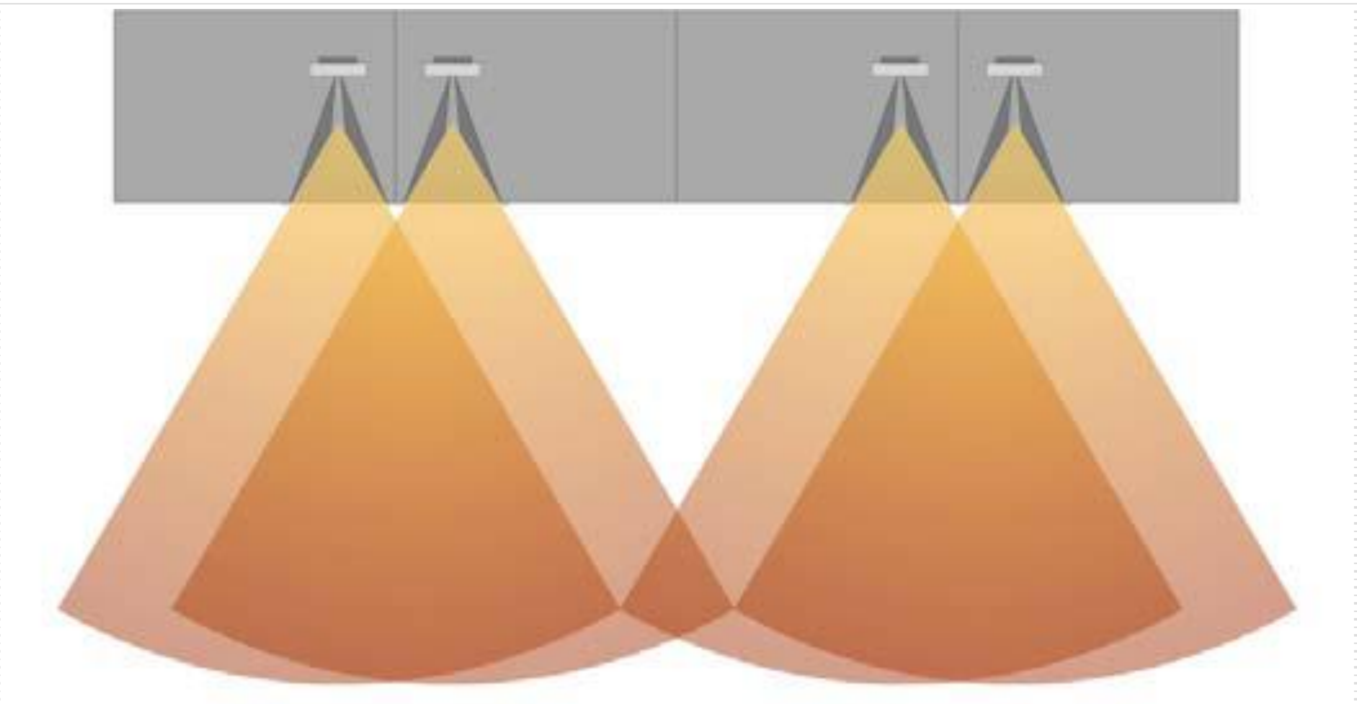
Work out the distance and add 30 ms and continue adding time until the image shifts from the delay box to the FOH. Check around the coverage area for uniform image

Delay boxes don’t have to be loud and in most cases can be high passed at 100 Hz for better intelligibility. It is the high frequencies that are the most important for intelligibility

Point Source Arrays

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.

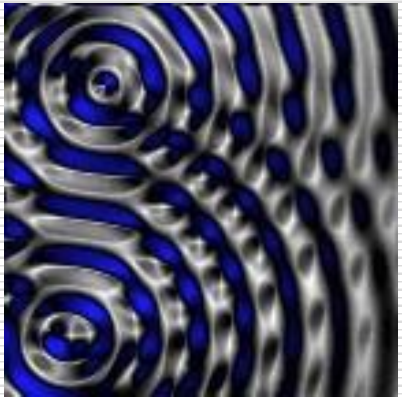


Wave energy and interference effect

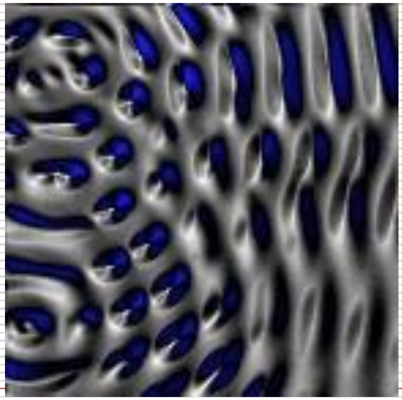
The successful audio engineer will work with the acoustic environment, not against it. To do this successfully, he will understand the behaviour of sound waves and how we hear

Practical Example

Throw a stone into a pond. Even waves spread out in all directions, their energy dissipated uniformly

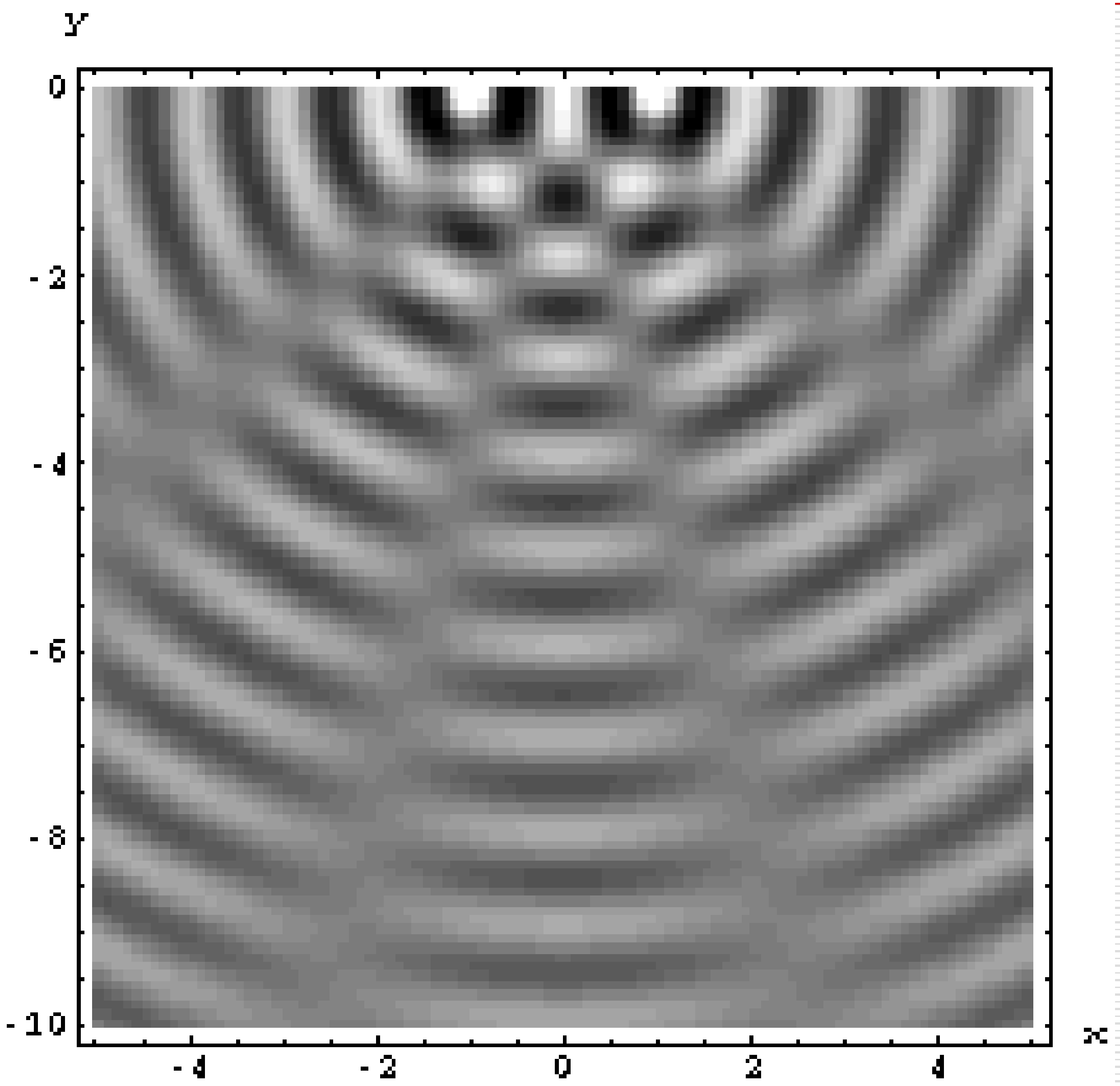


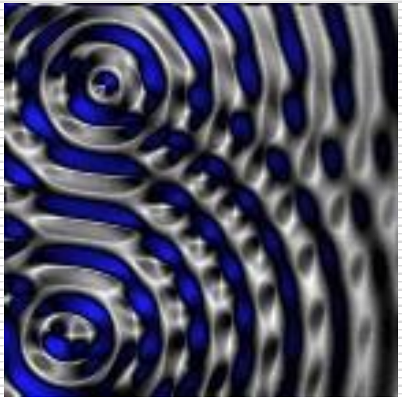
Throw a handful of gravel instead of a small stone, there will be a disturbed wave flow right from the first splash.

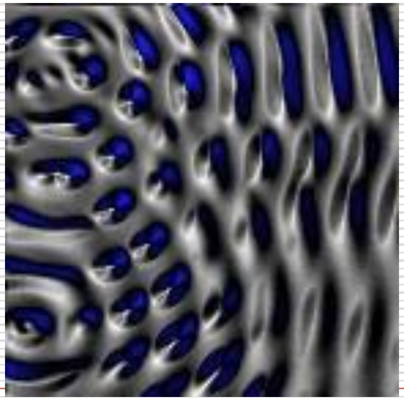


Imagine one speaker source as the small stone and an array of speakers as the hand full of gravel.

Wave energy and interference effect

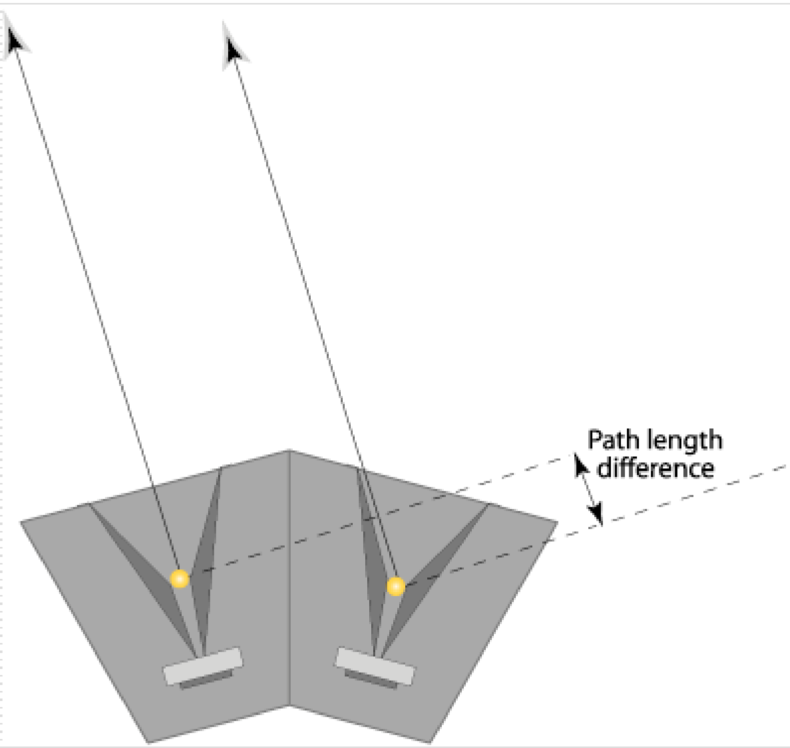


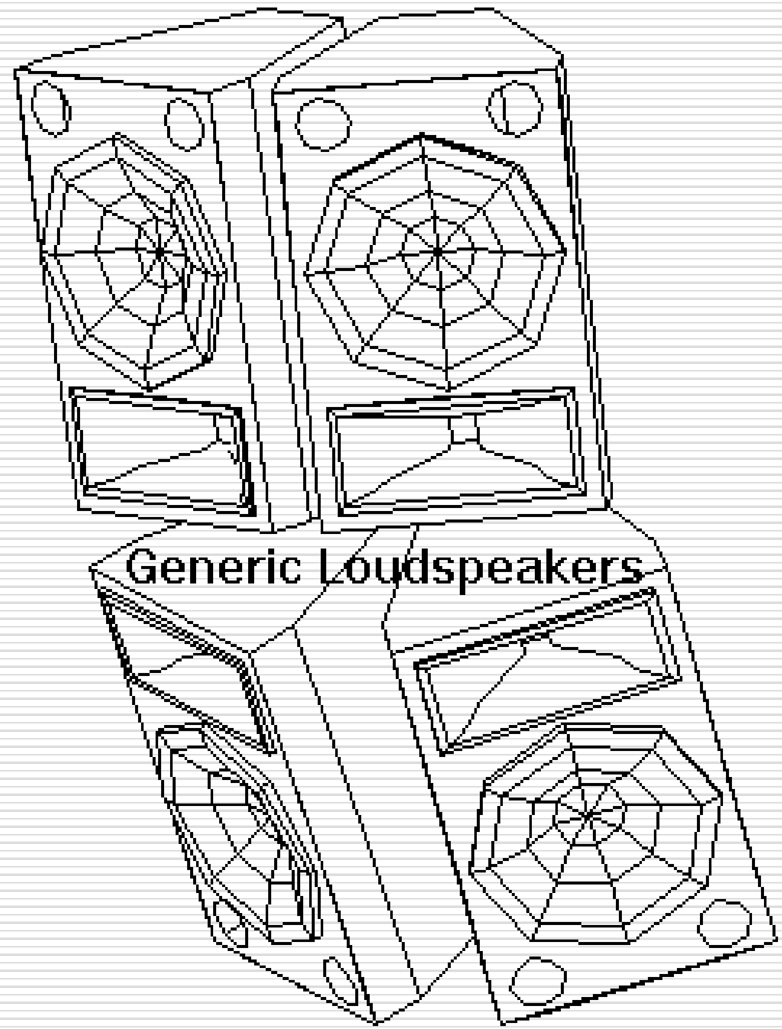




So let's add more boxes to our two box array? Maybe that will help

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

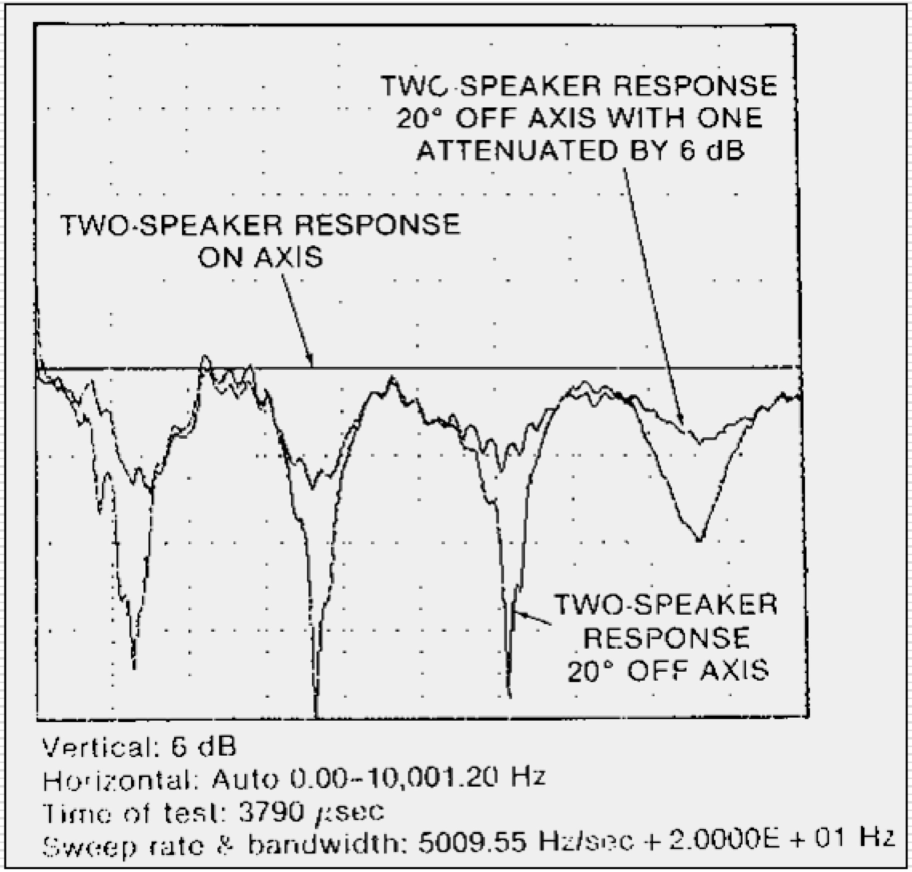




Comb Filtering - The array's enemy

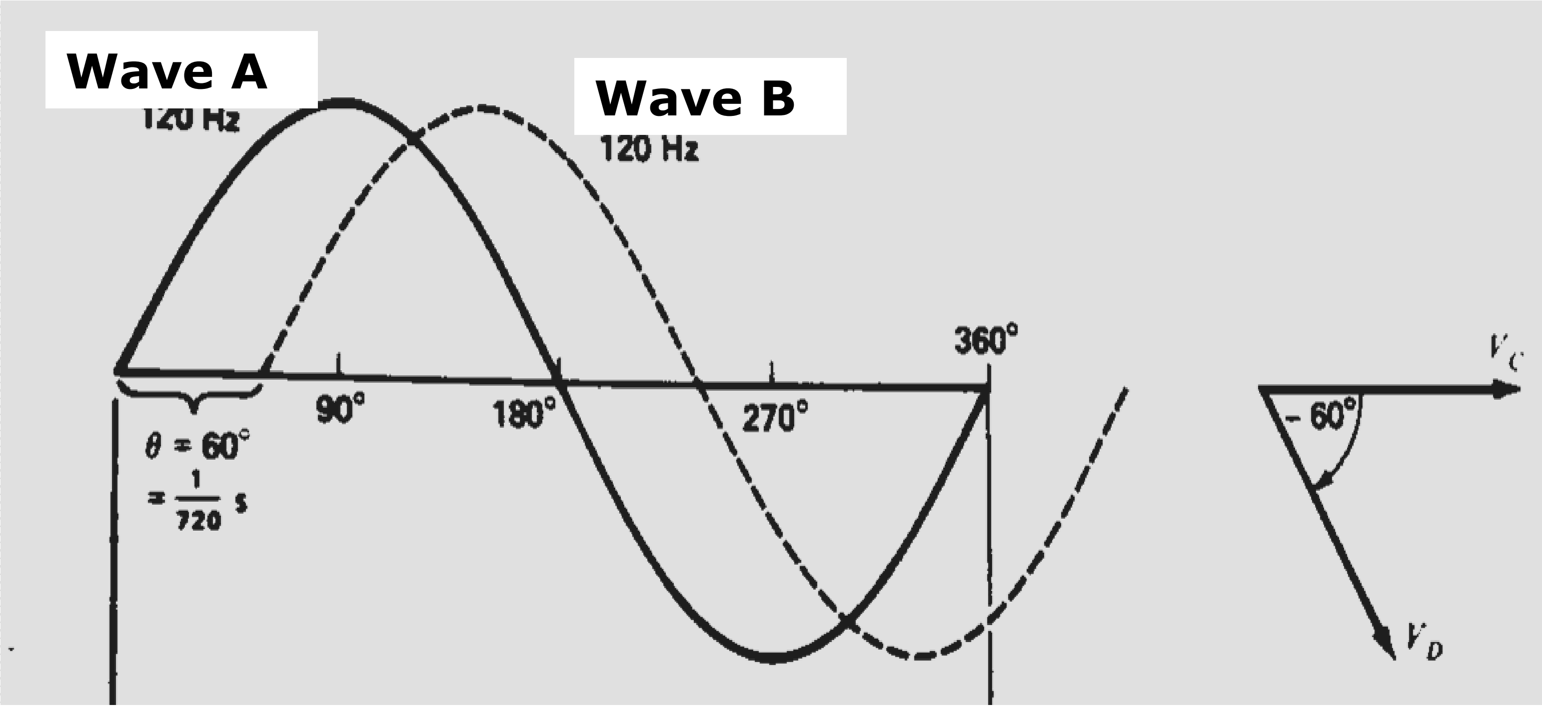
There is a practical clue for a sound guy about arraying a big PA, can you figure it out?

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.



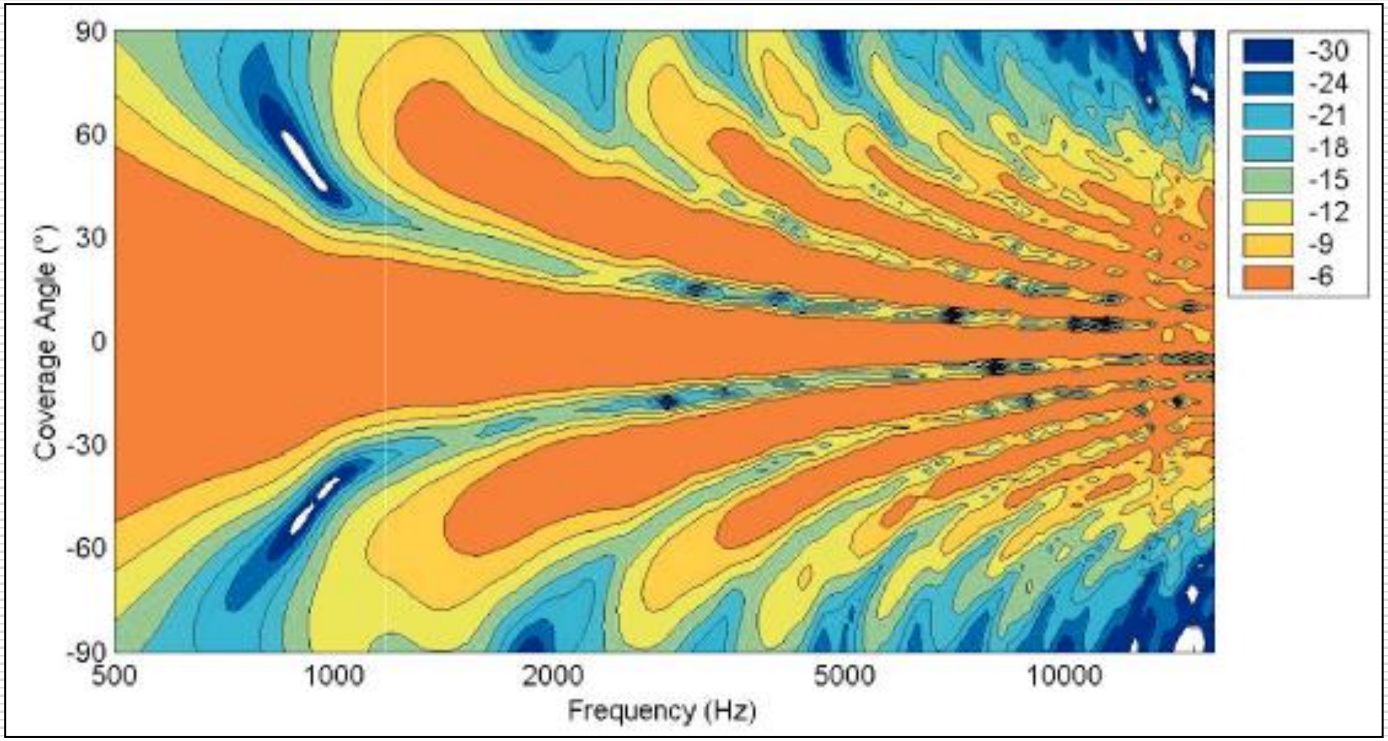
A graphic representation

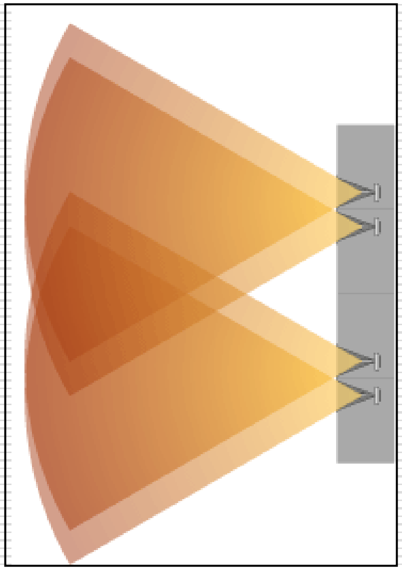
Wave A is now cutting across **Wave B** out of time sync. The slight time variation (60 degrees) will mean that there is a pressure lag between the waves and they will be working against each other. It's like a tug of war with everybody pulling at a slightly different time.



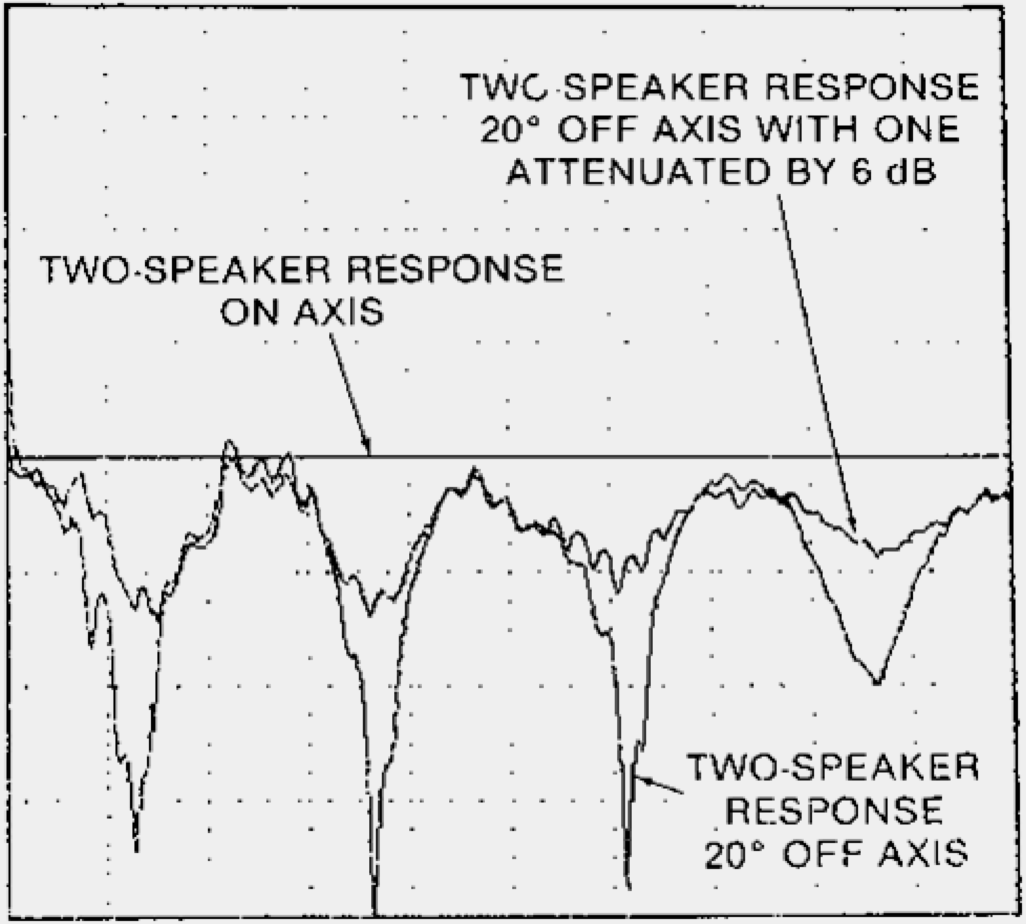
Even 2 Boxes Have Problems

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.





A controller or EQ can't fix this



Vertical: 6 dB
Horizontal: Auto 0.00–10,001.20 Hz
Time of test: 3790 μsec
Sweep rate & bandwidth: 5009.55 Hz/sec + 2.0000E + 01 Hz

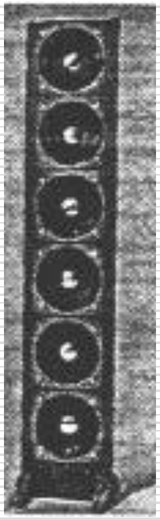
Line Arrays

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

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*.

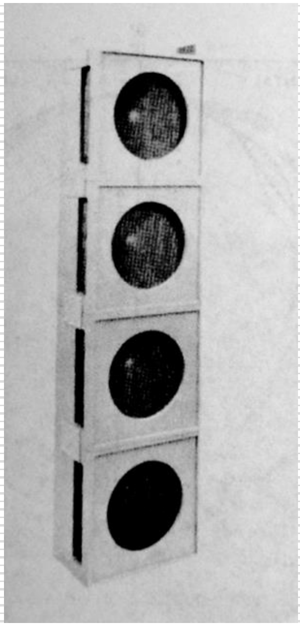


An early line source box and Harry F. Olson. He held over 100 patents.

Line array – not all are real line arrays

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**

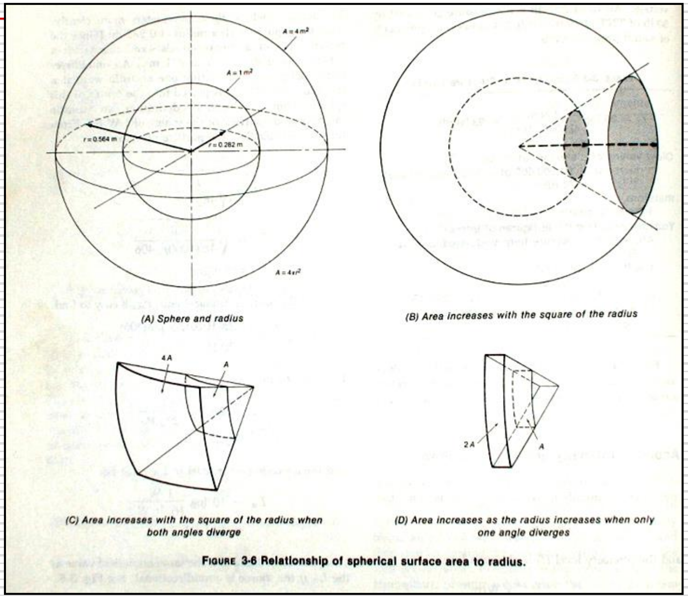


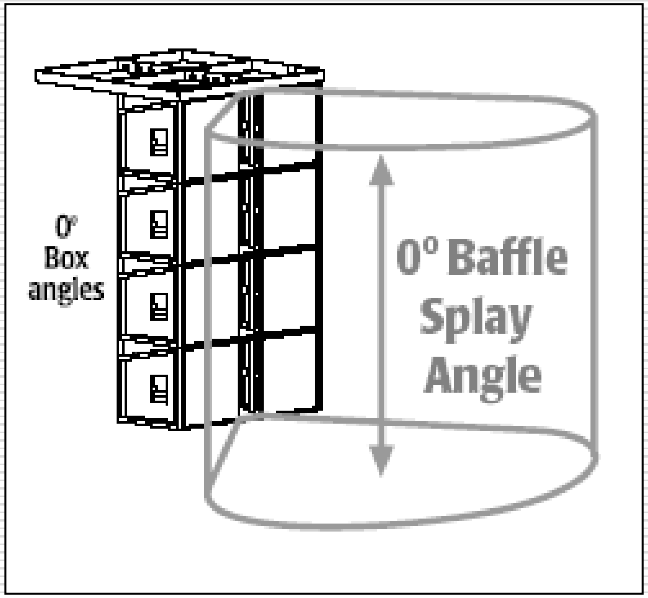
Remember, the original research and formulas are intended for line source boxes that were a long line of small speakers, all the same producing a relatively narrow band of frequencies

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**



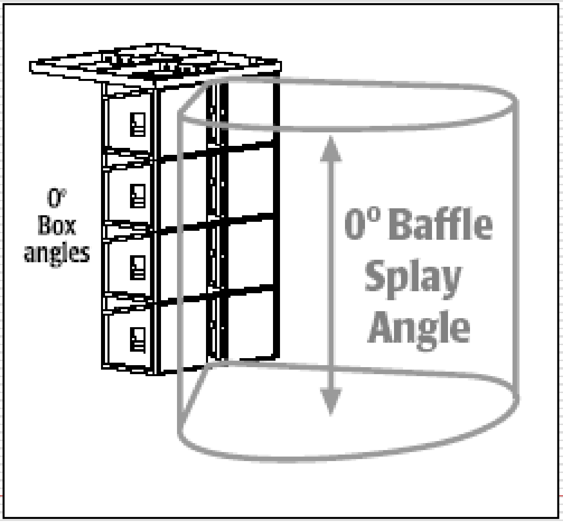


In a line source, the inverse square law no longer applies, or does it?

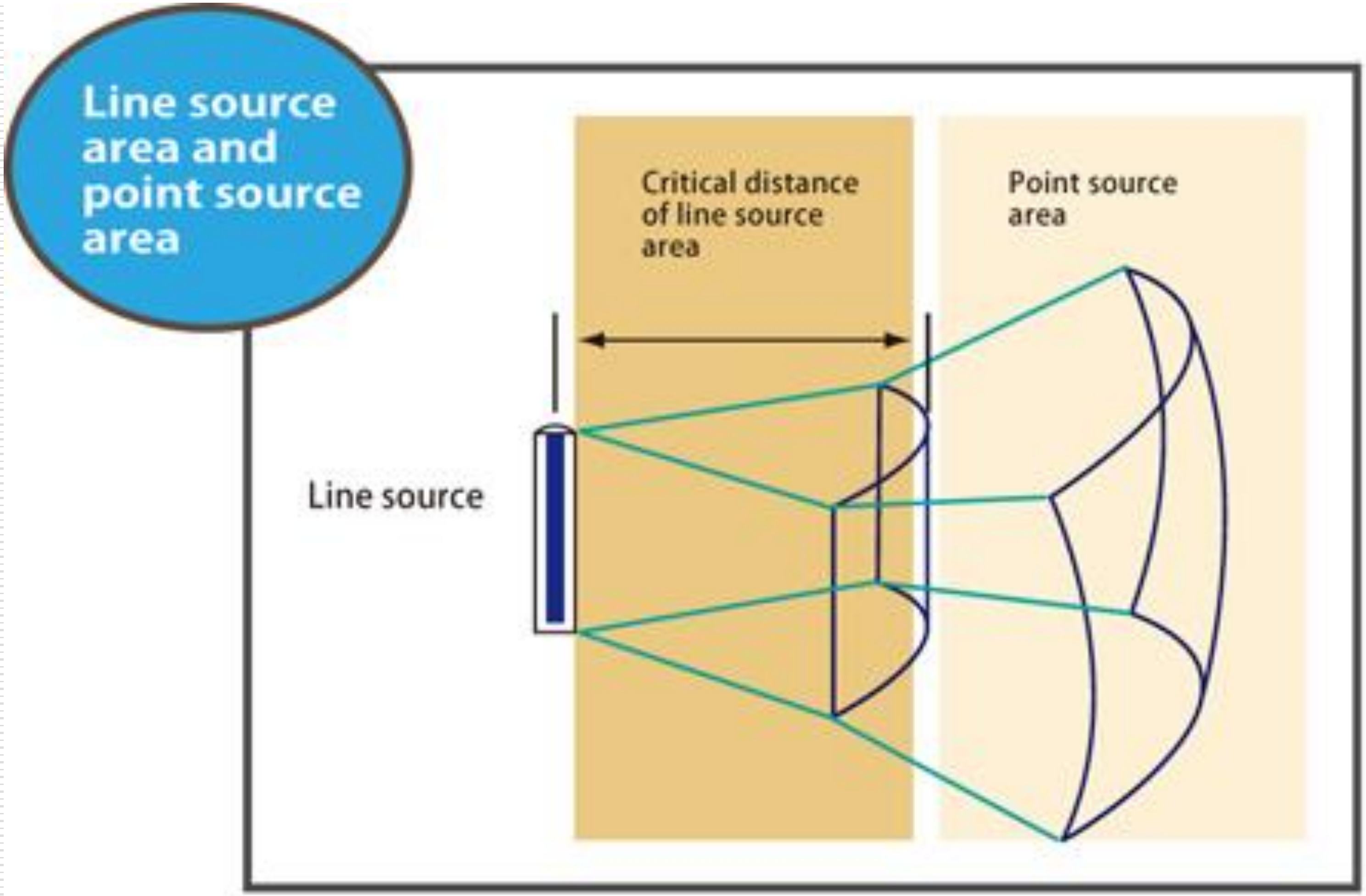
A line source of wave propagation has the characteristic of having its SPL fall off at a rate of **3 dB per doubling of distance, (as oppose to 6dB)** This is a well-known characteristic of infinite line sources

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**.*



The cylindrical wave eventually breaks down to spherical



At distance beyond the critical distance, the sound from the line source spreads not only leftward and rightward, but also upward and downward, as if it comes from a point source.

Full range line array?

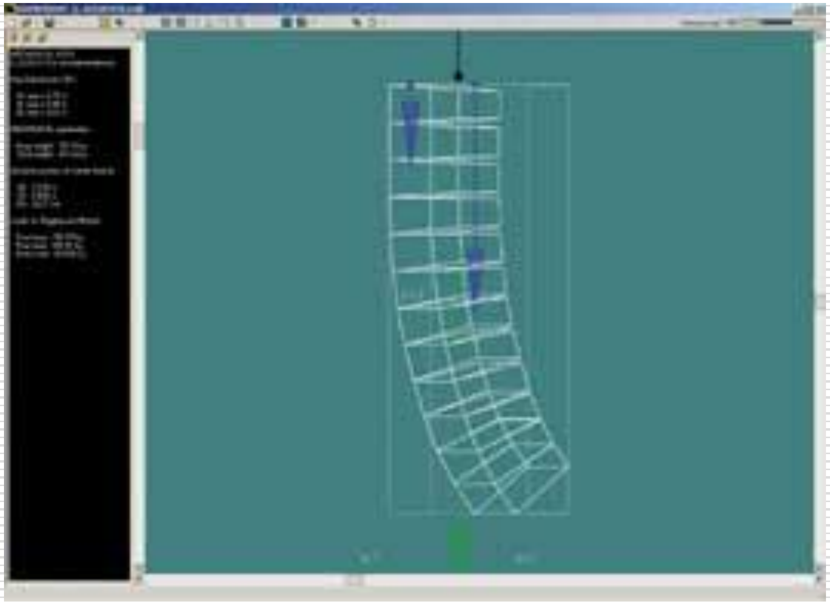
An unknown (outside of France), French company called L-Acoustics burst on to the scene with the first really practical system marketed as a line array back in the early 90's.

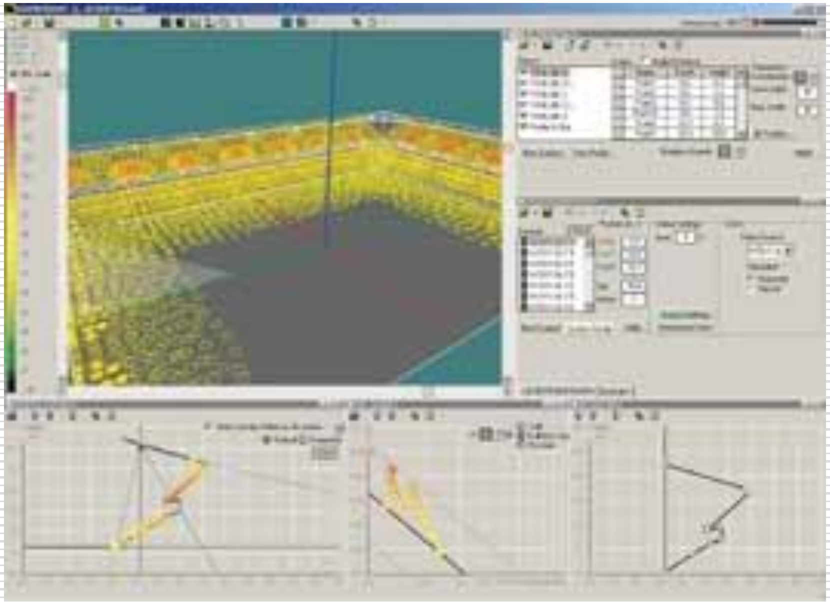
A V-DOSC line array

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.

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.

What happens to a line array in the real world?

A large outdoor space in the Middle East covered by a line array. The reflections make it unintelligible. The only fix for this is a distributed system

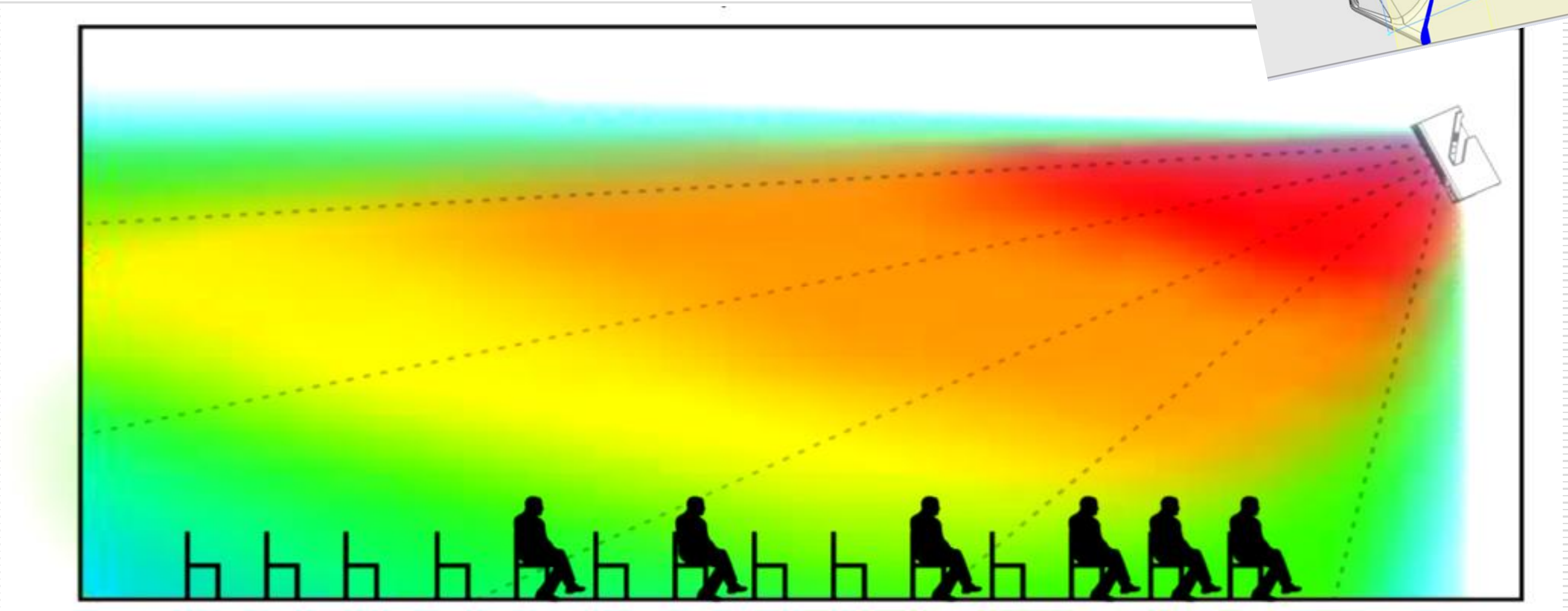
What happens to a line array in the real world?

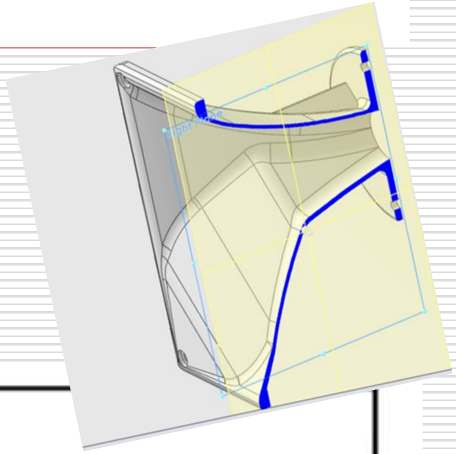
What happens to a line array in the real world?

An assortment of solutions in the one space, a distributed system of line source and others

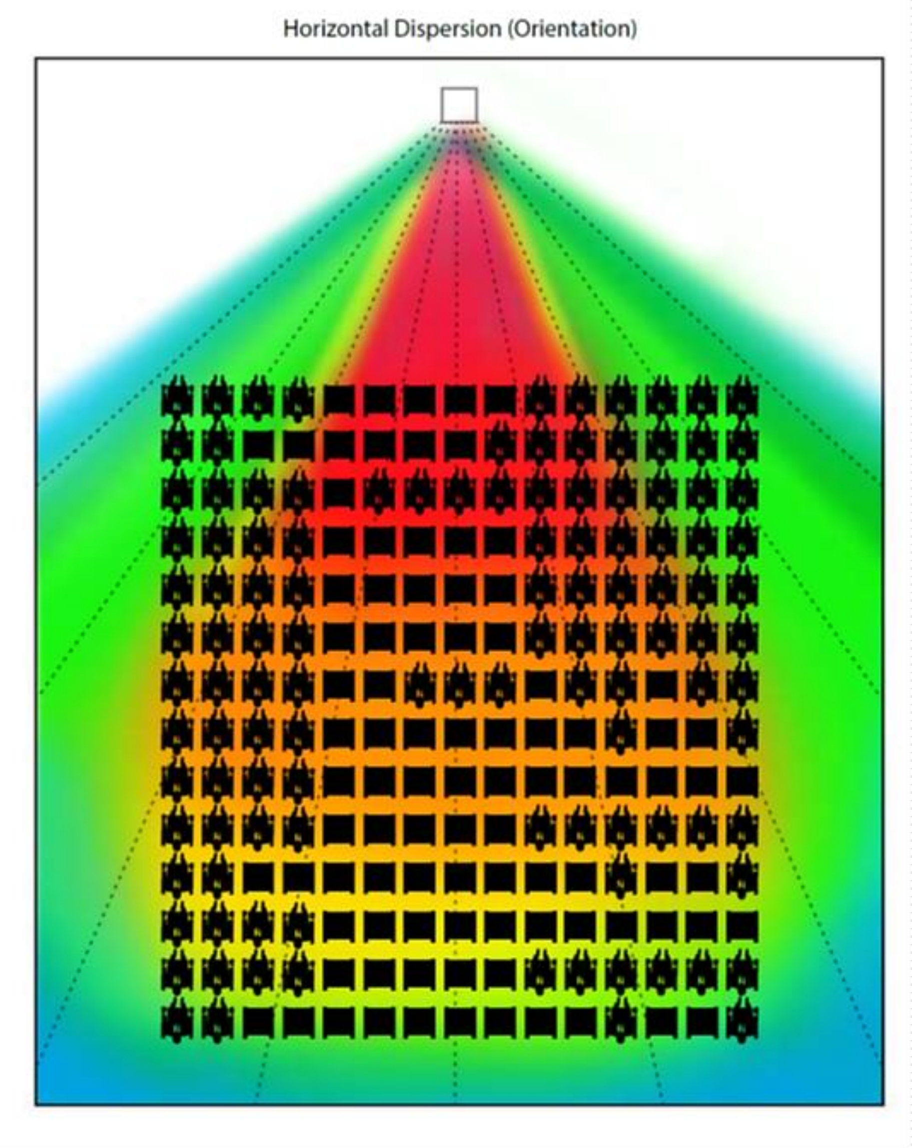
The asymmetrical projection solution – Energy is managed

The asymmetrical solution – Sound to where the people are

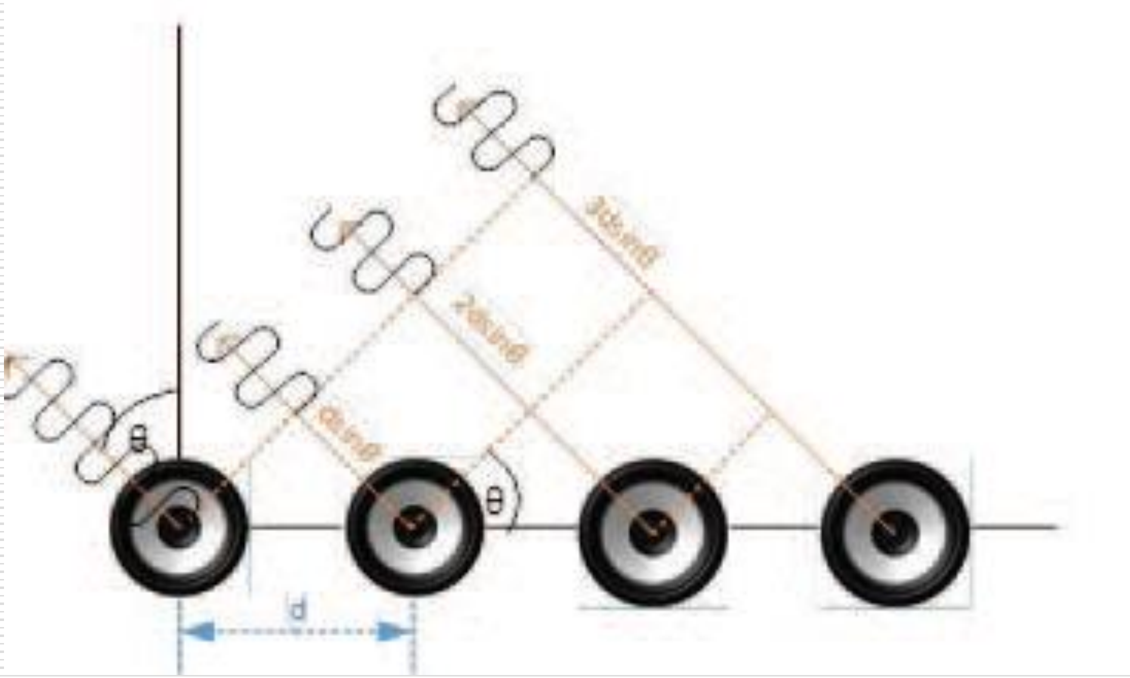




The asymmetrical solution – Sound to where the people are

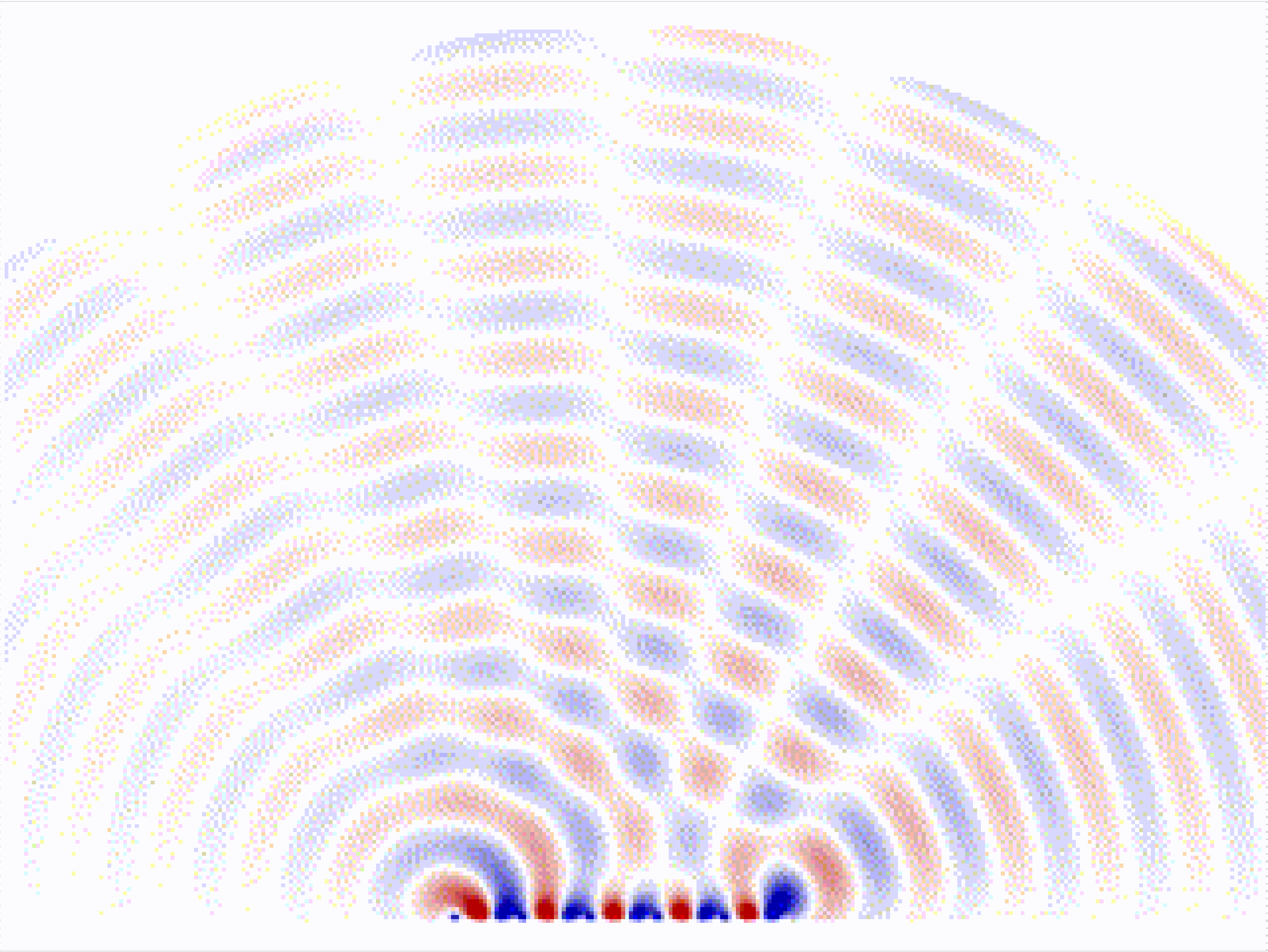


Beam Steering is another option

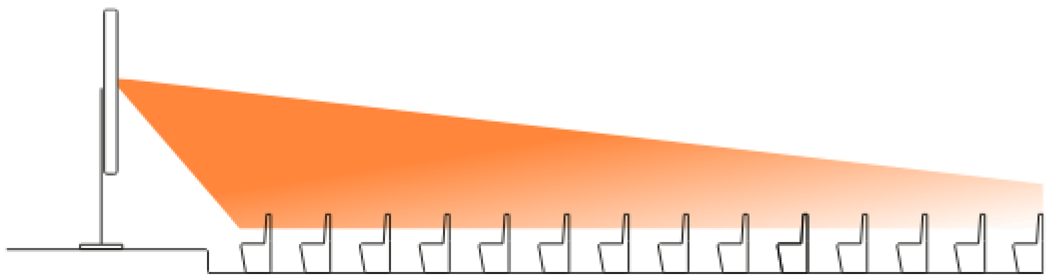


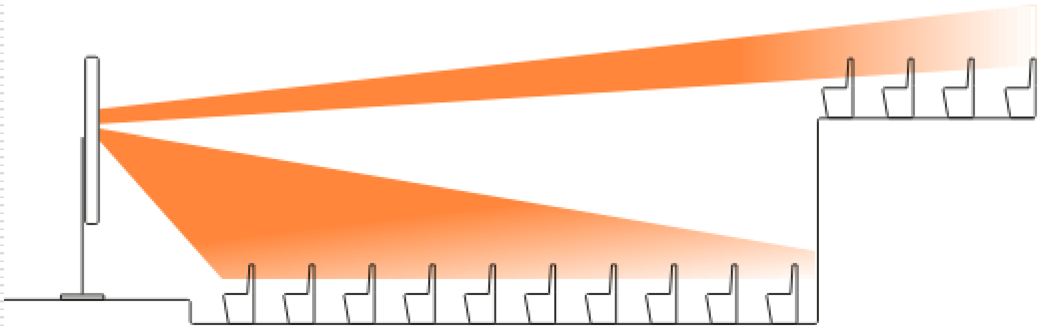
The underlining physics principle being used here is the fact that, when several sources interact together, they have constructive or destructive interference depending on the direction. We can use these interference patterns to our advantage, to choose were most of the energy is sent

Beam Steering is another option

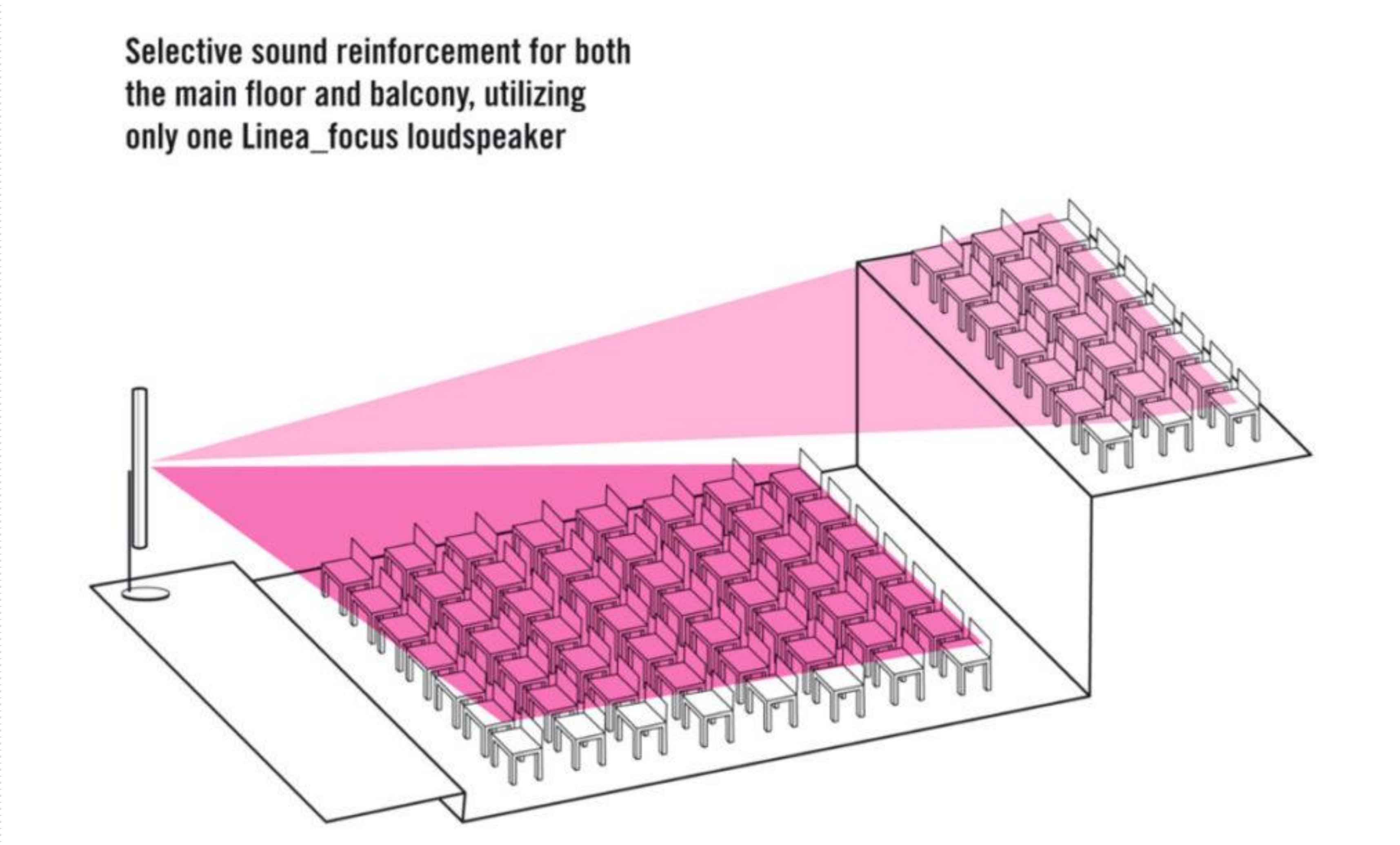


Beam Steering is another option





Beam Steering is another option



Line array, column array, point source distributed system, beam steering?

Type of Speaker	Point Source Speakers	Vertical Line Arrays	Column Speakers
Advantages	• Versatility of Size and Deployment • Pattern Coverage • Horizontal Coverage • Expense	• Preventing Drop Off With Distance • Vertical Pattern Control • Sound Pressure Level (SPL) Capability	• Aesthetics • Coverage • Consistent Sound Levels Front-to-Back • Affordability
Disadvantages	• Low-Frequency Limitations to the Pattern Control • Arraying Challenges • Throw distance	• Shallow Rooms • Height Requirement • Sightlines • Expense	• Passive Column Aiming • Volume • Bass Output • Shallow Rooms

It’s all about retaining consonant intelligibility

The acoustic energy from the speaker system must be directed to where you want it (the audience), and not to the reflective surfaces around the room. The design and location of the speakers must work with the room to provide an undistorted upper midrange where consonant intelligibility is essential.

It’s all about acoustic energy management

It’s all about retaining consonant intelligibility

This is something that can be measured and there are several systems for doing this

Each system has benefits and limitations

The Modern Venue paradox

The venues designed by the “ancients” work for a point source system (a person), but are another matter when presented with a 2 – channel (stereo) audio system

The modern venue for sound reinforcement needs to be almost acoustically inert (acoustic-ally dark), with a capacity to uniformly absorb acoustic energy across the full audio spectrum

Try telling that to an architect!

The educated will be in a job for a while

Live Testing Demystified

Safety in Numbers (% intelligibility numbers that is)

Measurement Conventions

Clarity by numbers

% Alcons, STI or C50?

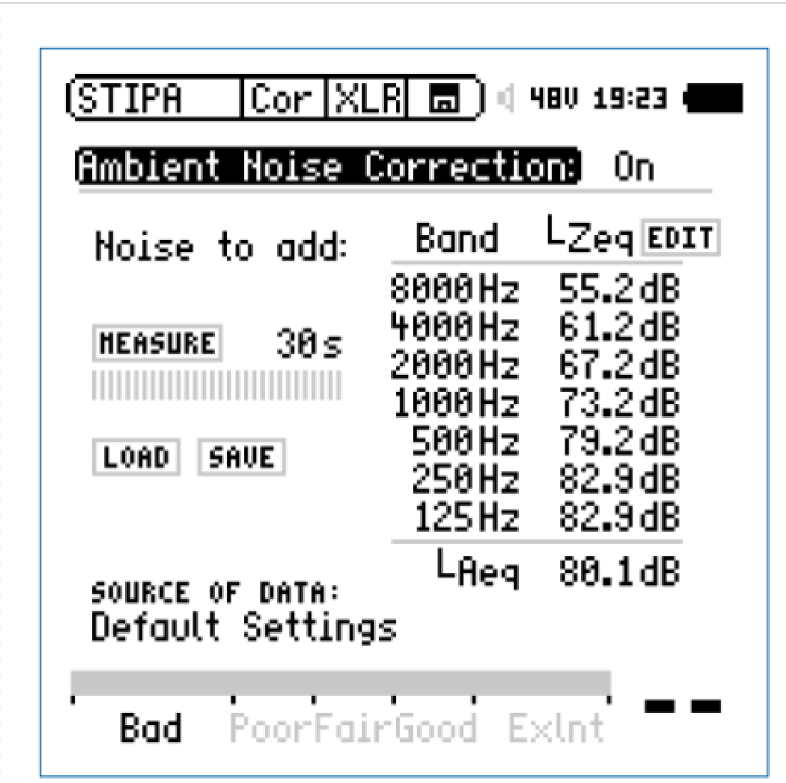
% Alcons was developed primarily as predictive technique,

STI was developed as a speech duplication measurement method

C50 is another system for determining clarity

Influence of Reverberation on speech

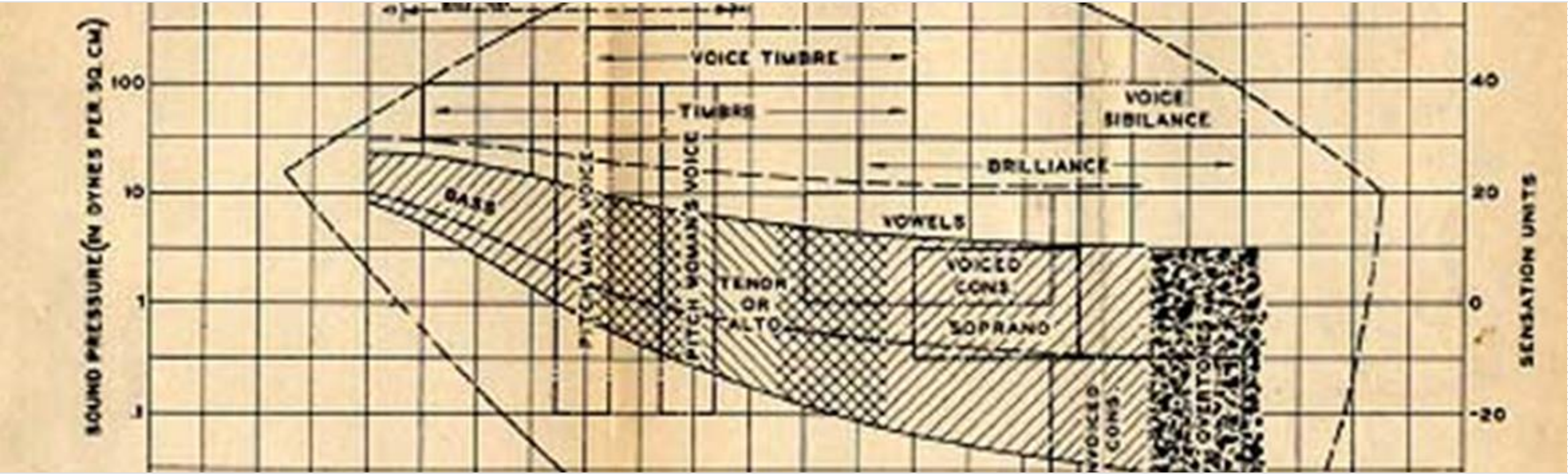
Modulation Transfer Function (MTF)



Measurement Conventions

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

Sensation Units?



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

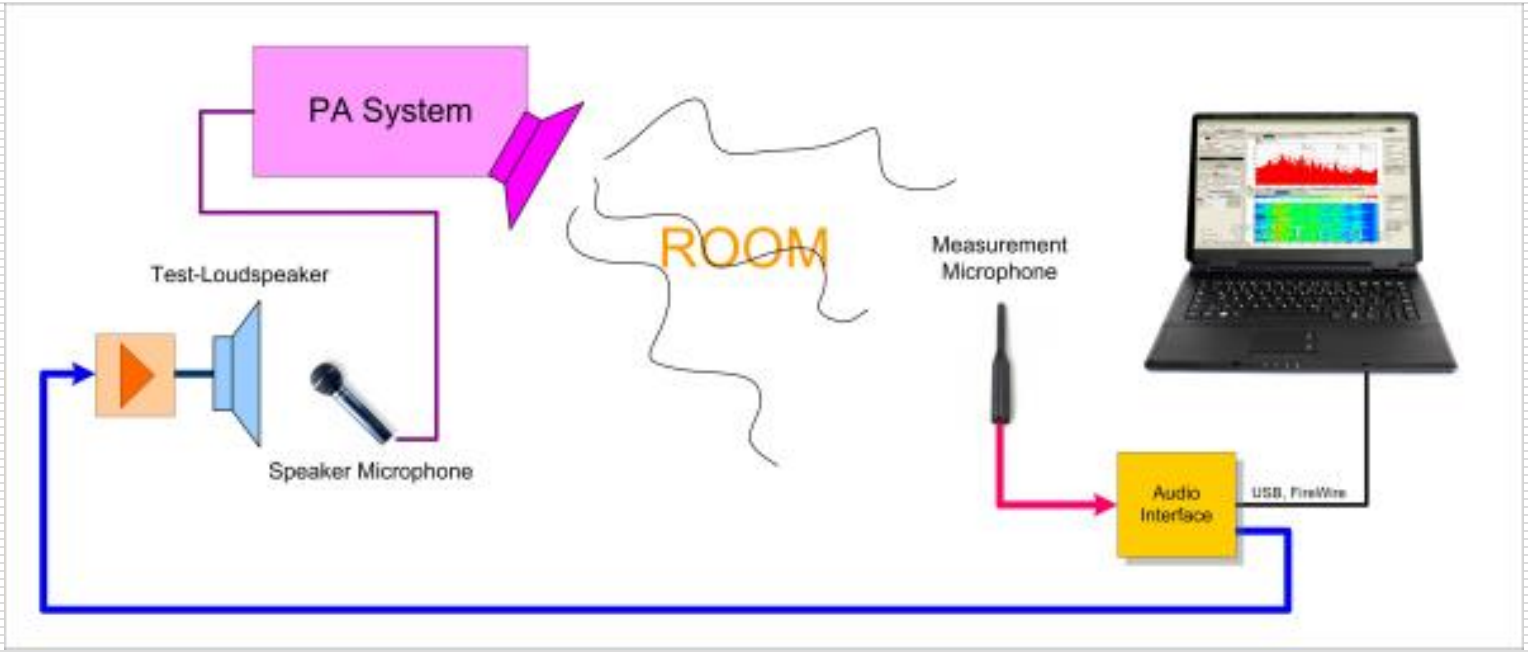
STI	0 - 0.3	0.3 - 0.45	0.45 - 0.6	0.60 - 0.75	0.75 - 1.0
	unacceptable	**poor**	**fair**	**good**	**excellent**
ALcons	**100 - 33%**	**33 - 15%**	**15 - 7%**	**7 - 3%**	**3 - 0%**

%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

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




"the show must go on!"