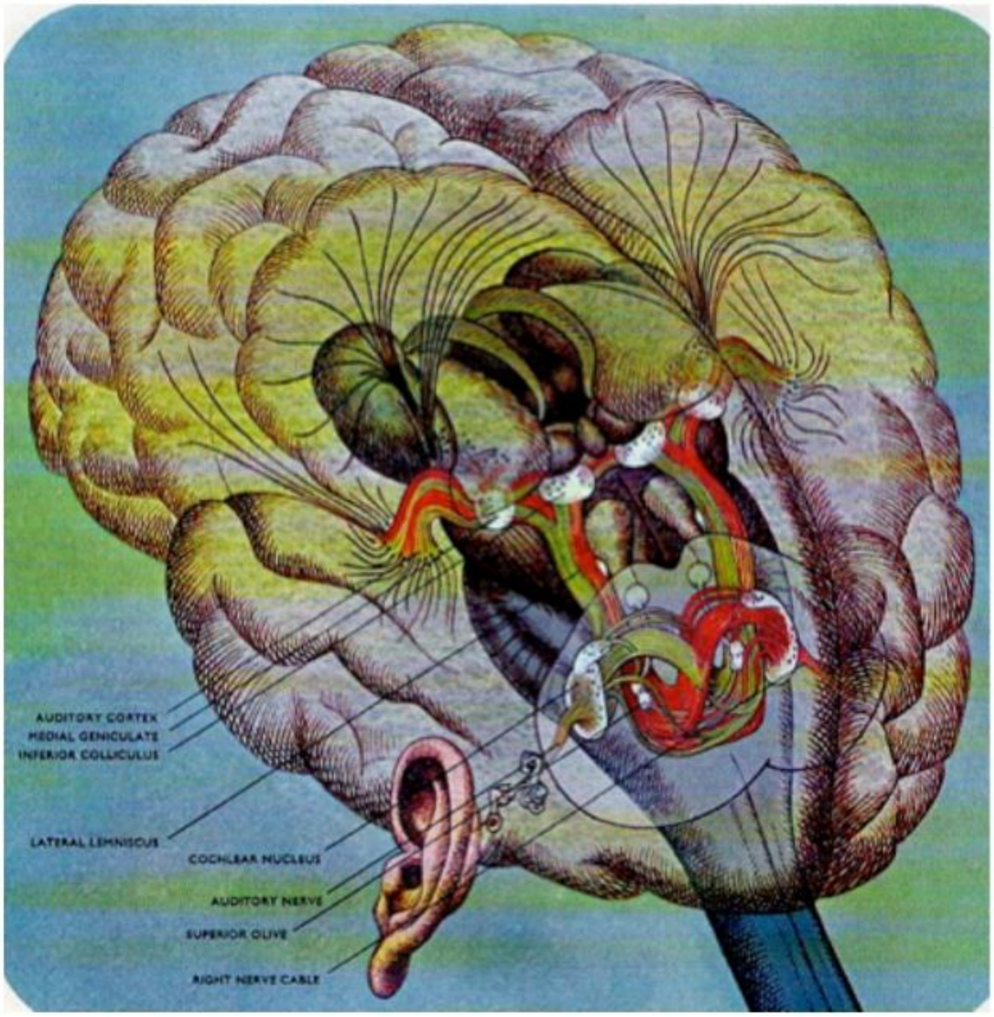


A novel Paradigm in Studio acoustics.

Waves in an ocean of air, transforming into waves in an ocean of neurons in the inner universe of our brain. **Art implies intuition and mastery. Science can aid in the development of both. But what role does luck play? Were the grand masters simply lucky? Is it luck or skill that allows an artist to appeal to a broad audience? It is in fact a combination of both. Today's room acoustics, like many arts, is an opinion-dominated field, one that is influenced by "conscious auto suggestions" as much by history as by technology.**

(Here is an graphic presentation of what's created as an alternative reality, in the **inner universe of the brain,**https://www.youtube.com/watch?v=fGPBfbeAh5w)

INTRODUCTION

The audio-video media landscape is in a major transition. The legacy recording industry controlled by the major record labels corporations are diminishing. The musician Prince started a protest against this dominance, that revolutionized the music biz**. This important change happened when the usual way of thinking about or doing something is replaced by a new and different way. The new music movement is called the indie music. The musician society began to abandon their record labels and start their own independent labels. This affects the major recording studios, by shifting the big studios to an independent project, budget based private studios. Everyone, from A to B-list musician to songwriter, producers and mixing engineers, has home based studios, some are simple, others are extreme complete ground-up builds. Small recording and control rooms become the new norm with problems following with small acoustic spaces. Most of the studio CAPEX was invested in "mixing in a box" computer programs and Digital audio workstations. This failure to invest in an acceptable acoustic treatment. Unfortunately this impacted the sound quality. Often a "MixFIX" mastering studio with an experienced sound mixer, has to do the finale mix, in a control room with a professional sound monitoring. A podcasting social media studio market has emerged producing programs on a substandard level. Gaming industry is gaining interest, setting immersive audio in focus with object panning of higher order B format discrete synthetic effects. On traditional organic music, immersive audio is presented as a 3D frontal projection i.e. stage performance and a 360 degree reflex sound field envelopment. Based on Alan Blumleins research and patent, the advanced 3D Wave field synthesizes are dedicated to big public arenas applying 1,000 to 10,000 loudspeakers. In order to do the correct placement and panning of a direct sound source all multichannel format require a well diffused acoustic space without discrete reflexions. (ghost loudspeaker).**

I Like to highlight a real acoustic test that are taken from the "Book of nature".

The test take place in a forest in Sweden, with many bare trunks. https://www.youtube.com/watch?v=QK1zeBOZgdU. This is a test performed in a forest showed in this picture. (the comments in Swedish is by my college Lennart Nilsson).The reflected sound is very dense and smooth, (Schroeder curve). The test show a very dominating lateral reflexion. The vertical reflexions are almost zero. The reason is that the forest has almost free field acoustics at the treetops.This test shows how nature is creating diffuse sounds. This is a diffuse reflective sound that has been in nature as long as a forest existed. The reverberation pattern is like Wiener Musikverein Konzerthaus in Vienna. (See Manfred Schroeder paper).The strong opinionated studio designer generation in Europe, are arguing and don't believe that reflex field spread in the time domaine exist in nature. Lennart Nilsson is the acoustician demonstrating this acoustic effect on his video.

I have experienced the same effect independent of the test performed by Lennart Nilsson. I can **guarantee** that this exist it is for real, (no faked news). I experienced this during the 70's, in an Scandinavian forests. **It has to be experienced to understand.**

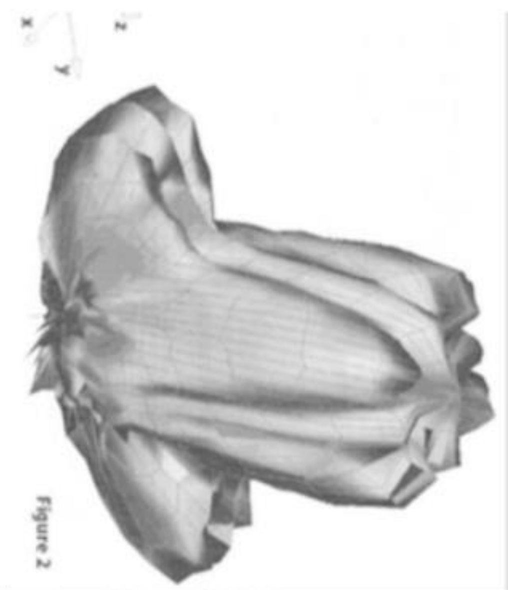
Diffusion

Sound can be reflected in a diffuse manner. The total reflections are fragmented into many reflections having less intensity, which are scattered over a wide angle, creating a uniform sound field. Diffusion can be created in a variety of ways, most often by introducing surfaces having irregularities in the form of angled planes or convex surfaces sized at least as large as the wavelength being diffused. Three-dimensional surfaces such as ornamentations, columns and statuary serve as diffusing elements and were integral to the acoustics of 17th, 18th, and 19th century performance spaces. The depth of the diffusing undulations must be at least one-tenth the wavelength being diffused. However, it is possible, if attempting to create a relatively low-frequency diffuser (for example, the octave below middle C, which has a wavelength of 2.7 meters [9 feet]). For this reason, in some concert halls, there are macro as well as micro diffusive elements to accommodate diffusion in different frequency (and therefore wavelength) ranges. Most common diffusers work well between 400 Hz and 8 kHz.

The Schroeder diffusor

Next generation of acoustic spaces and control rooms are based on diffusors. Often called reflection phase gratings, these panels scatter sound waves. A Schroeder diffuser is a structure comprising a number of wells of different, carefully chosen depths. As a sound-wave strikes the irregular surface, instead of bouncing off it like a mirror, it shattered out of each well at a slightly different angle. The frequencies at which such structures operate as diffusers depend upon their dimensions. For example, the lower limit is that frequency where the deepest well is a quarter-wavelength. The result is a richer, livelier sound with an enhanced sense of space. Listeners claim that the panels seem to make the walls disappear. Diffusors makes a small room perceived as the air of a great hall. The secret lies in the varying depths of a panel's wells. With depths based on specific sequences of numbers rooted in number theory, the wells scatter a broad range of frequencies evenly over a wide angle. The scientist who pioneered the ideas and responsible for this development is Manfred R. Schroeder, a University Professor of Speech and Acoustics at the University of Goettingen, Germany. In the 1970s, Schroeder and two collaborators undertook a major acoustical study of more than 20 famous concert halls. Schroeder discovered that number theory can be used to determine the ideal depth of the notches, resulting in an acoustic grating that's analogous to diffraction gratings used to scatter light. The operating range of a single diffuser is limited to about four octaves because, **if the deepest well is deeper than about fifteen times its width, it begins to behave as an absorber**. The well depths are most commonly given by: where **d** is the depth of the diffuser, **h** is the well number, **N** is the prime number on which the sequence is based, and **L** is the wavelength of the lowest operating frequency. Mathematical analysis shows that for such an arrangement, the spectrum of energies scattered into different directions is essentially flat, meaning that roughly equal amounts of energy go in all directions. **The graph No6** is showing a model diffusor designed by Peter D'Antonio of RPG Diffusor.

GRAPH No6



The Wing Family

The **Wing** modules consists of different length 1/2 wavelengths diffusor. (Open in both ends). The delay lines are coupling to each other globally to form a unique near field dense broadband reflex pattern stretched and optimized in the **time domain**. This is a differentiator from the traditional **Schroeder diffusor** geometric technology that can create ghost speaker in the near-field

The full story http://diffusor.com/PDF/170515_Resolution_Mag.pdf

This led us to some pragmatic tests. It is crucial to understand some elementary acoustic situations when discrete waves propagate in different acoustic spaces.

Test no 1. The Purpose is to define and listen to a direct sound in a free field situation. (sound propagation is 344m/Sek.) A free field is a large open field without any trees and buildings. If a transient sound is generated in this field, just one sound is generated. It is a direct sound without any repetitions from any "ghost loudspeaker". This simple test can be performed, by using two pieces of a 3 feet long wood plank and clapping this pieces together. This generating the transient sound that can be a test method used in the upcoming tests.

Test no 2. This is the test that Joseph Henry conducted as the first psychoacoustic experiment 1854. Repeating this test is an important experience for any unexperienced in acoustics. In this test the same large open field are used. A reflective wall has to be in the field with a size many wavelengths larger that the test signal, (100 m2 recommended), in the test the distance to the wall is 13,7m. By combining the incident direct sound and specular reflective sound ray, The total distance is 2x13,7 m = 27,5 meter, this is 80ms delay. This test shows that when a transient sound hit a reflective wall in 90 degree angle, it bounce back most energy. This is perceived as a distinct ECO.

Test No 3. In the same location with the open field and a reflective wall. When the distance to the wall is reduced to 8,5m, no distinct eco can be heard. This is an incident direct sound and specular reflective sound ray, with a total distance of 2x8,5m=17m. This distance is a 50ms delay between the direct sound and the reflective sound ray, this time difference is in the of fusion zoon of the direct and reflective sound. Known as the **"precedence effect". Fusion** is when a direct sound and reflexion sound field are merging to be perceived as merged sound. (See graph 1)

Test No 4. This test is to find out when two sounds no longer can perceived as separate. When the distance to the wall is 0.7m/2mS the two sounds perceived as a single sound. The 2ms to 50 ms are the limits of the precedence effect

Here is an Video as a summary of precedence effect, test 1 to 4.
https://www.youtube.com/watch?v=xBcPxuQMGzE

In test 1 to 4 discrete transient sound are defined as a direct sound, and the reflected sound are delayed compared to the direct sound. This reflected sound are perceived as a separate ECO if delayed 80 mS after the direct sound. When the reflected sound are closer than 50mS the direct sound and the reflected sound begin to **fuse** in to a mixed sound. If the spectral content in the reflex sound have a long envelope it is merged to a long continuous sound. If the envelope on the second sound are short as shown on **(Graph no1)** it begin to repeat in a short time frame as in Table I, showing MDD minimum Detectable Delay. Table II, show MTD Maximum tolerable delay, the delay timeframe is from 20mS to 80mS, during that timeframe a second sound begin to impact the mix of direct sound and delayed sound, loosing its distinction and speech begin to be unintelligible. The COX effect, are demonstrated this in **Test No 5.**

Test No 5. John Charles Cox an acoustician at the University of COLORADO. Drum lines was the center of his master of thesis. Mentored by Prof. Robert Ashley at the university of Colorado. Cox study was about perception of drum lines echoes, to demonstrate behavior of short repetitive sound in the precedence time frame (2ms to 50ms). This is an important study for perception of specular reflexion sound rays in a early reflexion sound field. This discrete sound ray echoes in an acoustic space is defined as a **"ghost loudspeaker"**. **Why**

are ecoes so mporan e rs ne o my rs page o my ess saes. coes are e most serious problem in architectural acoustics today. They garble announcements and ruin entertainment events. And profoundly inhibit the ability of entertainers to perform. Echoes are an extremely grave problem today. His study is called the **COX effect**. This is demonstrated in this graphics. That are a summary of Precedence effect (1854) and Haas effect (1949). The COX effect is different view on a Sonic effect of signal delay and in an interval. From 2ms to (MDD) is the minimum detectable delay and after (MTD) is maximum tolerable delay. In **Graph No 1**. you see the waveform of **drum taps** and **wooden plank** used in the experiments. The repetition rate was quarter notes (500mS/note), quarter notes triplets (333mS/note) and 32nd notes (63mS/note). This is the fastest a percussionist ever play. The minimum detectible delay is the equivalent of what Beranek called the “initial time delay gap” which is the time between the arrival of the sound from stage and the arrival of the first reflexion at a given seat in the audience. It is 20mS for symphonic music, which is confirmed by MDD **table 1**. MTD is a delay at which drum taps, representative of music, become hopelessly confused and speech becomes unintelligible. It is found this to vary from 20mS for a very fat drum taps to 84 mS for slow ones.**See tab II,**

Graph No1.

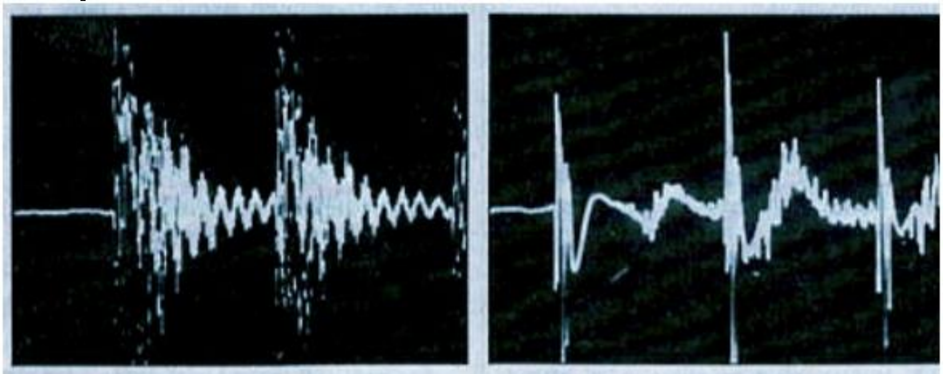


Fig. 3—Drum-tap waveforms.

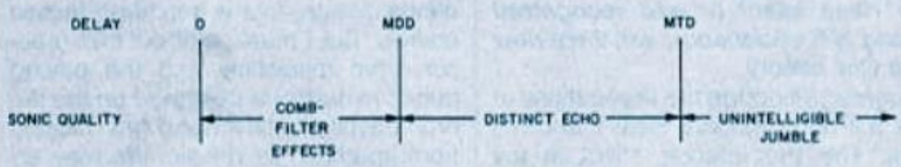


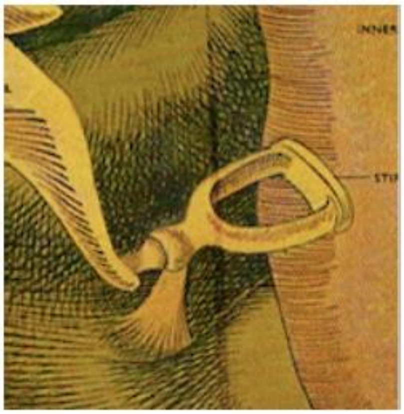
Fig. 4—Sonic effects of signal delay, showing minimum detectable delay (MDD) and maximum tolerable delay (MTD).

Table 1. Minimum detectable delay (MDD)
for delayed and non-delayed signal of equal level

Minimum Detectable Delay, mS	Tap signal interval, mS
15,8	63
16,8	83
20,0	125
20,0	167
20.0	250

Table II Maximum tolerable delay (MTD) for delayed and non delayed signal of equal level

Maximum Tolerable Delay, mS
20
25
44
56
84



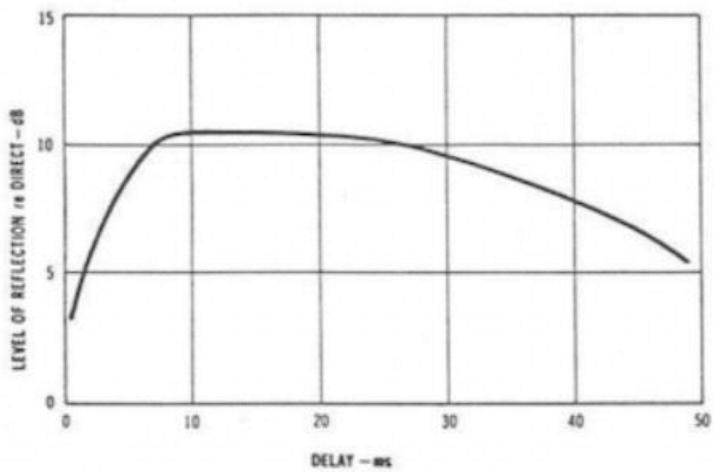
Pict 1. Stapedius muscle is holding and controlling the movement of the stapes

Dynamic response of our machinery of hearing. The stapedius muscle are controlled by the electrochemical signal from the cochlea by the tonotopic organization. Here is an video information. https://www.youtube.com/watch?v=kfwdkaBMKAU A system of delay stations (processors) in the two ways communication system from ear providing junctions where the brain can direct partial or complete elimination of sound signals that have no immediate importance. The sound level limiter (stapedius muscle) are processed in this tonotopic organisation. If a high level sound entering the ear canal, the signal from cochlea over the 30,000 cords to the brain where it is processed and send in return a control signal to the stapedius muscle that will be tense and reducing the stapes signal to cochlea. This is similar to a electronic dynamic compressor that are controlled from the tonotopic organisation. Unfortunately the attack time is slow and the first part of the transient sound are passing. This strong puls will reduce the hearing temporary but will return slowly to normal hearing again. If levels over the **threshold of pain** (130 dB) over an extended period of time, like a rock concert, this can harm the hear cell in cochlea that will die and create a permanent hearing loss. The stapedius muscle is shown in **picture 1**.It is a electro chemical muscle that are controlling and limiting the stapes movement. .To learn more about the the machinery of hearing, this is a basic pragmatic video presentation. https://www.youtube.com/watch?v=J96ZAZDgiLQ. Xxx. https://www.youtube.com/watch?v=PeTriGTENoc

The direct sound is the **first wave front** and masking later sound as **Haas effect graph No 2.** is showing. If a **second sound** has higher level than 10dB at 10mS or later delay as per the precedence effect curve. The **direct sound** are loosing its masking effect and the precedence effect are collapsing.

Graph 2

Haas effect:



An acoustic or electronic delay can be used to mask the apparent direction of a sound source. A sound arriving at the listener's ears within a window of 10 to 30 msec. after another, will be interpreted as a reflection. The direction of the earlier sound will be taken as the point of origin, even if the delayed sound is as much as 10 dB louder.

Acoustic recording spaces. In small room acoustics the reflex field is strong and in a recording space it is crucial to control the reflex sound field. Discrete reflexes are devastating to sound quality as the sound rays are comb filtering with the direct sound.This is perceived as a hollow and muddy sound, (From 0 s to MDD see graph 2). This is impacting the **reverberant radius** (Sphere) that show the difference in level between the direct sound and the reflective sound field. When the **reverberant radius** has a small diameter the difference between the direct sound and the reflexion sound field is small. When a microphone is positioned at a music instrument in a acoustic space with a small **reverberant radius**, The reflex field has a strong coloration on the sound that the microphone are recording. The sound will be dull, (like a heavy fog) and will be difficult to mix. **The microphone with the smallest reverberant radius will determine the total quality of the recording.** (This can be experienced on many live video recordings). The problem with non diffused room reflex field will be a serious problem when immersive audio, and multichannel reproduction will be a commodity. We are now in a situation where social media are domination this media space of VR and object oriented multichannel Cinema and Gaming program. The 22+2 channel 8K Broadcast standard and multi channel (WFS) Wave Field Synthesizes, Dolby atmos and MPEG-H alliance, are setting a new demand on mix studio acoustic spaces and monitor listening spaces. This is almost impossible to do a good mix in small room acoustic spaces, that have a small **reverberant radius. (**Small diameter Spheres) this creates serious sound source separation and problem concerning panning of sound objects.

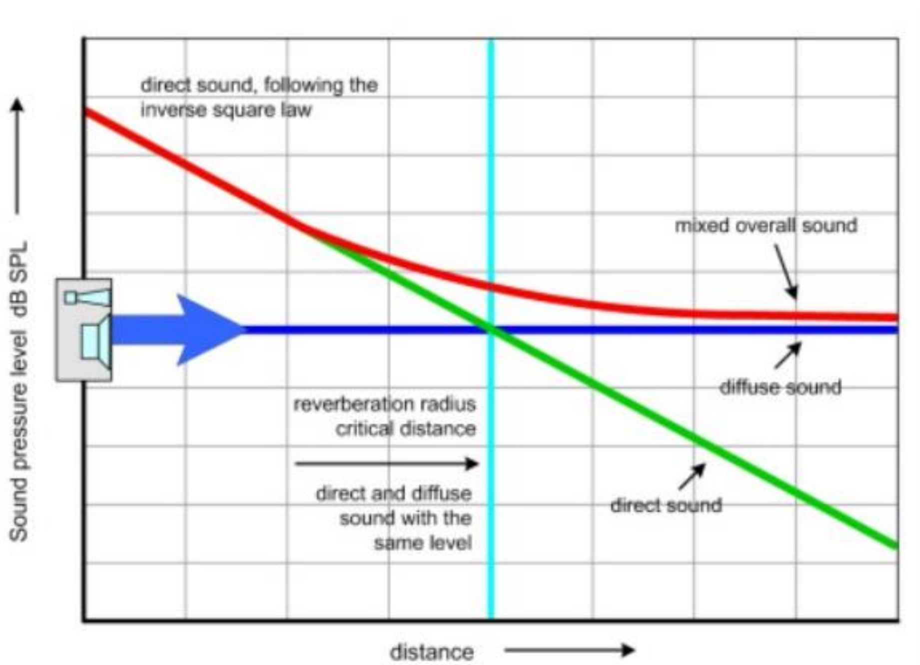
In a case of a big **Reverberant radius** the microphone has a dominant direct sound and the separation of music instrument is good. This will simplify the mixing of each individual instrument if they are well separated and not effected by the reflex field. A golden role for separation of music instrument is to have -20dB lower reflex field than the direct sound. This role is easy to achieve when the reflex field are diffuse and spread in the time domain like in a room where the **wing** diffusor are installed. If there is strong discrete reflexes the Critical Distance is dependent on what **direction** the discrete reflex is coming from. This can create a smaller critical distans and the separation of acoustical instrument can be reduced. Separation of organic sound sources is a problem in today's project studios, if the **Critical Distance or reverberant radius** are small. Music recorded under this conditions will create problems for the FiXMIX mastering studio sound mixer. He has to use all his experience and skills in his "mixer in a box" and use a toolbox of plugins to compensate for a substandard quality recording.

Control-Room acoustic space

If we look into the situation in the control room space, a small critical distance is devastating to the monitor sound quality. The reason is often to strong reflex sound field in the control-room that muddy the sound. This can be checked by having a high quality headphone as an reference. The strong reflex field in the listening environment is the reason why most studios have very short distance to the Aura tone-NS10 minimonitors. The reason is to increase the direct sound that will improve the stereo image. When you listen to big horn type main monitor system as in a Westlake type control-room, the stereo image is different compared to headphone listening. The phantom images are probably wandering between the horn monitors. I recommend using the **LEDR** signal to listen in your big monitor system. Possibly, you will be surprised, as I was. A critical listening space has to be "neutral," without added "coloration," then the early reflection levels should be at or below the threshold for image shift. (**Between line A and B in graph No 8)** The threshold for image shift is the level at which a sound image appears to move from its actual location. (**Between line B och C In graph 8**). Achieving these low reflection levels in a studio control room requires treatment of all surfaces involved, to achieve a **transparent acoustic space**. **Critical Distance/Reverberant Radius** are shown In **graph No3**,

Graph No3

Reverberation Radius/Critical distance formula=



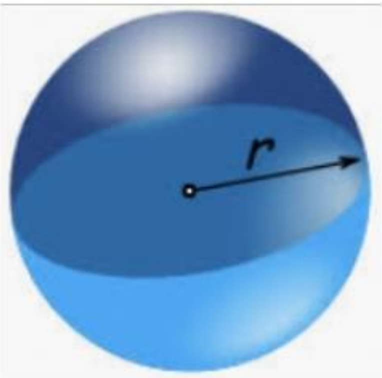
Reverberation Radius/Critical distance formula =

$$d_c \approx 0{,}057 \sqrt{\frac{V}{RT}} \quad [\,m\,]$$

What is the relation between reverberant radius, critical distance and transparent acoustic spaces.

A New generation of strong Opinionated studio business activity has emerged. This information is about basic **Natural phenomena** found in the book of nature, not generally known, many have read the book but don't understand, this often create **misbelieves**. Defining factors in confined acoustic spaces is the relations between direct sound and reflex sound field. Precedence effect, Haas effect and COX effect is explained in the previous text. Other important factors in recording spaces and listening spaces are Reverberant radius and Critical distance. Explained in a pragmatic way, direct sound source are embedded in a reflex sound field in an enclosed space . When the direct sound field and the reflex sound field have the same sound level a reverberant radius is created, it is comparable to a sphere (**see picture 2**). At the center of this sphere the direct sound are dominating, outside the sphere the reflex field are dominating. Factors that determine the radius of the sphere is the direct sound source directivity, it can vary between omni or cardioid characteristic. A dominas of discrete reflexions in the reflex field, will reduce the radius of the sphere. If the reflex field are more diffuse the sphere has a bigger radius. If a microphone are placed in the sphere, the closer to the center the more the direct signal will dominate. When the reflex field contain strong discrete reflexions the microphone must be positioned closer the the direct sound .i.e. the sphere has a smaller diameter with a discrete reflexions in the reflex field. The opposite happen when the reflex sound field are more diffused,(the sphere get bigger) and the ratio between the direct sound and reflex sound field are improved. With a larger diameter sphere, the separation of the instrument are increasing. With the **wing** diffusor the reflex field are spread out in the time domain, and the reflex sound are covering a broder time window. The reflexes will be lower in amplitude and more diffuse in both geometric domain and time domain. The spreading in the time domain of the reflex field, will increase the Reverberant radius/critical distance, this will further separate the levels between the direct sound and the reflex sound. What is preferred in a recording acoustic situation is to have a clean recording without an acoustic fog that are masking the direct sound. The same situation is in a listening acoustic space. A big critical distance will give a reasonable listening distance to the monitor loudspeakers, resulting in a wider sweet spot and a stable stereo picture. If a substandard monitor loudspeakers are used in a listening space with a small critical distance. The listening has to be in a near field situation, in order to pan the the instrument in the calculated way.

Picture 2



Without any acoustic treatment in a small room it is inevitable to have a dominating reflex sound with strong discrete reflexes. A well diffused sound field helping the direct sound to penetrate into the diffuse sound. A direct sound can easy be carried through the reflex sound if it is spread in the time domain. This test demonstrate https://www.youtube.com/watch?

v=BXova8pRNEU&feature=youtu.be , that a diffus and a time shifted reflex sound field like in a forest is not masking the first wavefront, in this test a record voice. The sound source are moved to different listening distance starting with 10 m/ 30foot changed to 15m/ 45foot and finally to 20m/60 feet. The diffuse forest reflex- field show that the first wavefront is not masked . The voice quality are intact but will be lover in level with increased distance between the listener and the direct sound source. This is a proof of whats happen in a confined acoustic space where the wing diffusors are used. When you have a spread-out reflexion as shown in this forest, you can find out the clarity of the voice when the direct sound is moved further into the forest. The speech is still easy to hear even if the direct sound is moved 10-20m from the listener. This is what I call **transparent acoustic space**.

How can this be created in a small room space? The function of the **wing** is to spread the early sound and diffuse sound in the time domain. The result is a acoustic space mimicking the test in the forest. This is a revelation to musicians experiencing all the inner detail in their music instruments.

Concerning absorption material

As a general guide, it is not advisable to concentrate large amounts of sound absorbing material on one surface only, particularly where that surface is distant from a group of listeners. In order to create a totally acoustic dead room, sound absorbing material needs to be distributed over both the wall and ceiling surfaces, (anechoic rooms)**.See graph No4**. In a rectangular space, for example, it is not a good design practice to concentrate sound absorbing on two parallel surfaces or on two pairs of parallel surfaces. This simply reduces reflections coming from the absorptive surfaces and may result in an echo by enhancing the audibility of the reflected sound from the remaining pair (or pairs) of reflective room surfaces. The reflections from the absorptive surfaces are reduced, resulting in an increase in the amplitude of the remaining reflections. Reverberation is **directly proportional** to room volume, **inversely proportional** to the surface area and **inversely proportional** to the amount of sound absorbing material. It is possible to reduce reverberation by the following means: adding sound absorbing material, reducing room volume or increasing surface area. Common porous absorbers include carpet, draperies, spray-applied cellulose, aerated plaster, fibrous mineral wool and glass fiber, open-cell foam, and felted or cast porous ceiling tile. Generally, all of these materials allow air to flow into a cellular structure where sound energy is converted to heat. Porous absorbers are the most commonly used sound absorbing materials. Thickness plays an important role in sound absorption by porous materials. Fabric applied directly to a hard, massive substrate such as plaster or gypsum board does not make an efficient sound absorber due to the very thin layer of fiber. Thicker materials generally provide more bass sound absorption or damping.

We have come to a point to look at the pragmatic part of studio design and its benefits and drawbacks.

Lets start with the history about **NON Environment acoustic spaces**. Non Environmental control rooms are based on a free field concept experienced in **test No1**. As a student at Lassen Jordan in Denmark, I learned about the term **Non Environment studios**. Jordan was demanded to build radio and TV studios for the Reichskanzler of Germany during the second war. It was important for the German propaganda to have modern studios in the occupied countries. Lassen Jordan as an pioneer, designed the first **non environment studios** for northern Europe. It was like a **cookie cutter** system. All studios and control-room build at that time used these cookie cutter. I have worked in this **non environment control-room** in

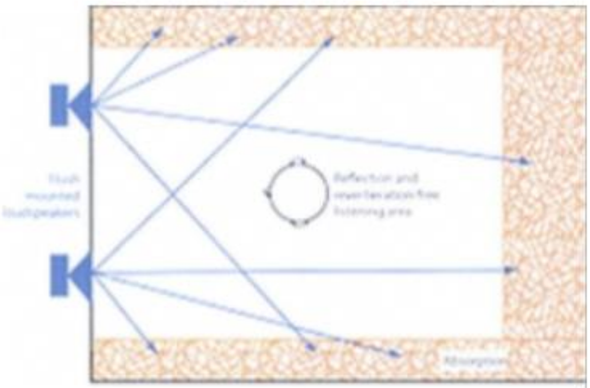
Sweden over 15 years. I have been there done that. In 1940,s there was no electrical reinforced bass instrument and consequently the low frequency content was not prioritized. Studios at that time ignored the wave acoustic frequency range 20Hz to 200Hz. This situation changed in the 60’s when the electric bass instrument became common in studios. The low frequency instrument was muffled in the studio because of strong standing waves in the "Wave acoustic” frequencies.

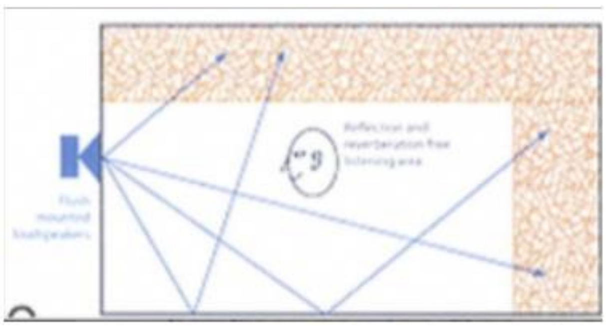
Tom Hidley worked at JBL at that time. He become aware of the problem in LA studios. He found out that the problem was to many standing waves in the control-room. He tested a low frequency loudspeaker at JBL and played it on the roof, mimicking a free field situation. And **voil'a** the problem was solved. He designed a bass trap, that reduced the reverb time in **"wave acoustics"** to very low values. He used plywood panels and mineral wool hanging from the ceiling on all walls. This traps was protruded almost 1 meter in the acoustic spaces. Not practical in small rooms. The reverb time was set to between 0,25 s to 0,35 seconds broadband. It was a part of this famous **cookie cutter “Westlake studio design"**. Twenty-five years later, the **next generation studio designer** found Lassen Jordans and Tom Hidley’s **cookie cutters**, and reinvented a non environment control room novel to the second generation of **non environmental** studio designers.

A problem with non environment acoustic spaces is what Rupert Neve “the grandfather of pro audio” called **SHEEN**. A high frequency sound gives a **shimmer** and **air** to the top of music recording, (10kHz -80kHz) I recommend reading the explanation by Rupert Neve https://www.youtube.com/watch?v=AGt0KXW_T1Y&list=RDAGt0KXW_T1Y&index=1, David E Blackmer, https://earthworksaudio.com/wp-content/uploads/2012/07/The-world-beyond-20kHz.pdf , George Messenburgers comments on http://www.sanken-mic.com/en/report/reports.cfm?top=1&id=7 and the new Swedish microphone **Ehrlund EHR-T** that can record the **SHEEN** in an **transparent acoustic space**. Listen to demo **test 7.** www.youtube.com/watch?v=IcPPp0HNg84&feature=youtu.be xxx www.youtube.com/watch?v=OqDSxlZBIYs&feature=youtu.be

The non environment control room acoustic space will kill the **SHEEN** and the sound lack that “**shimme**r" on cymbals and “**air"** on the string section. It is easy to test this in a anechoic chamber, Our machinery of hearing in the Inner universe are very uncomfortable when interacted with an absorption environment. **Graph No 4** Showing the non environmental Control room with **no** reflexion from the walls except the reflective front wall. Other problem I have encountered when designing surround control room, when the rear loudspeaker are direkted to the hard front wall. There will be a discrete reflex (ghost loudspeaker) from the front wall, that create discrete reflexions that create comb filtering and image shifting in the area between line B and C in graph 8 This will be a drawback for non-environment acoustic spaces for surround and immersive audio.That makes **non environmental** out of date and **obsolete**.

Graph No4





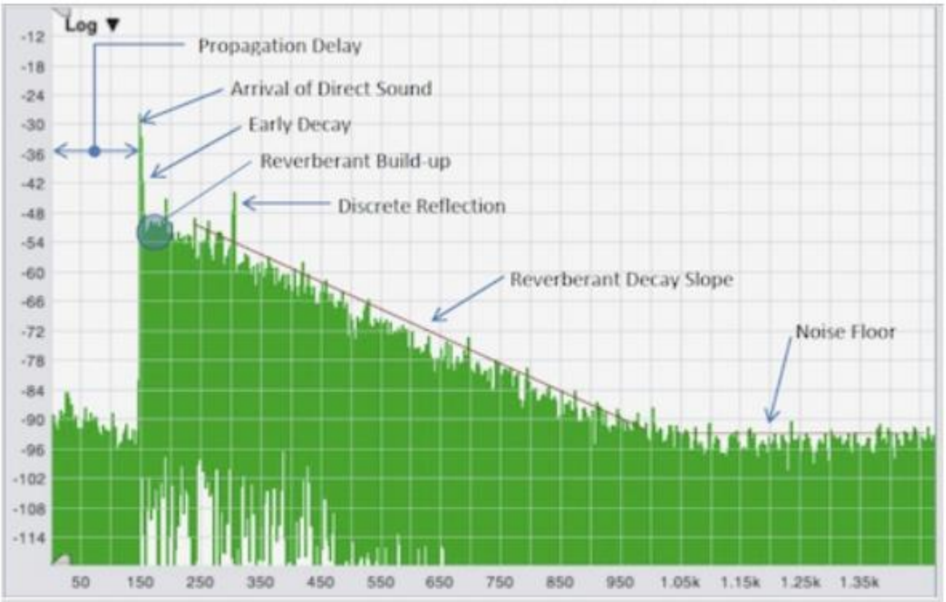
Richard Heiser, a talent NASA physician mathematician and audio guru, invented **TEF** technology resulting in a novel visual presentation of direct sound and reflected sound in 3D **T**ime **E**nergy and **F**requency. A time energy curve is shown in graph 7. **EF** display show the direct sound **ITG** (initial time gap), followed by early reflex field. Some specular reflexion at 150mS ,12 dB down are shown i.e.(Ghost loudspeaker).

LEDE concept

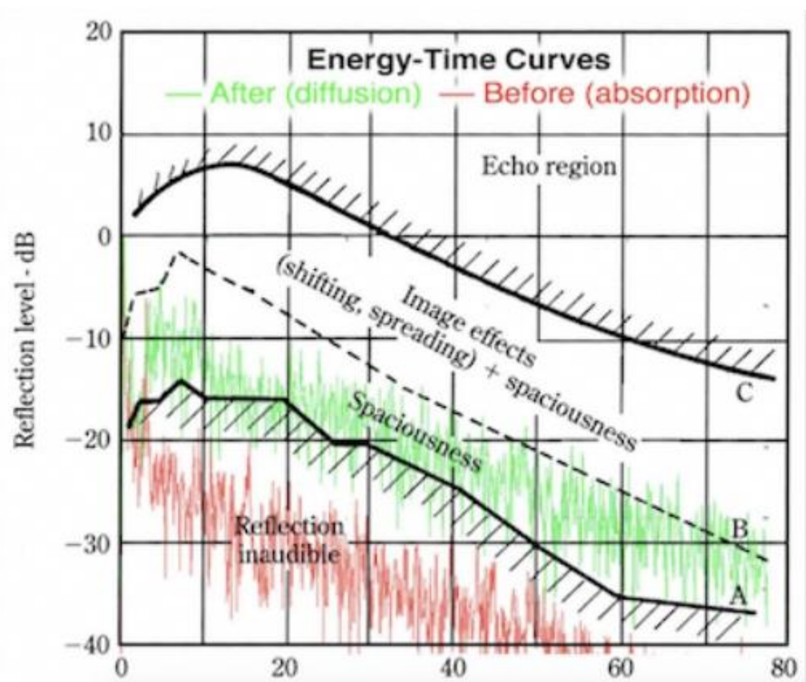
A new generation of **cookie cutter control room** was invented by Don and Carolyn Davis in their Synergetic Audio Concept. A collaboration of acoustic and studio designers. They called the studio **LEDE** (Live End Dead End) acoustic space. The most brilliant researchers in the audio industry shared their knowledge to the studio colleagues.This attracted the studio and acoustic designers to meet where knowledge and experience was shared in a unique way. This has never happened before. The Besserwisser studio designers was silent. The concept of **LEDE** was to create a non environment acoustic space in front of the listening space. The monitor loudspeaker was imbedded in the front wall to reduce the edge diffraction from the loudspeaker corner. See the energy time curve **graph No8** the non environment part of the room is represented in **red** under the **A** line. The rear half of the room has a diffuse acoustic space created with Schroeder diffusors, designed by Manfred Schroeder The diffuse sound field **Graph No 8** marked here in **green** between the **A** line and **B** line are the spaciousness area. The area on **graph No 8** between **B** line and **C** line is a area that image effects like **shifting or spreading** develops.

The effects of these **Syn-Aud-Con** classes was that the studio industry took a giant step ahead, powered by a common interest to advance the studio industry. Break through acoustic events were released during those classes. **Paddy Rodgers PhD** work **PINNA EFFECT** was demonstrated for the first time outside Western University by **Doug Jones**. I was one of the**The Syn-Aud-Con** attendees. I was fortunate to be a part of this studio design community, that became one of the certified studio designers on the world market. Hundreds **"state of the art studios"** with novel designs resulted from this **golden age** in studio technology. As a consequence of my participation a Don Davis certified LEDE studio was build in Sweden at Swedish Radio, a first in Scandinavia.

Graph No 7.



Graph No 8.



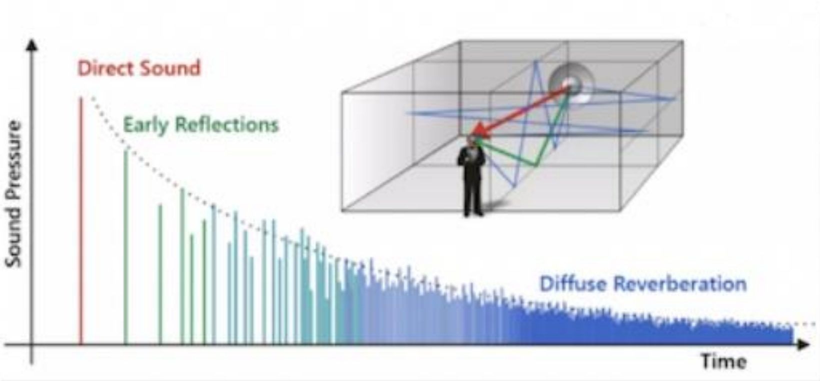
Small Room acoustics and large room acoustics

There is a distinct difference of small room and large room acoustics. Any room can have a reflex sound pattern starting with simple sound rays. The ***mean free path*** is defined as the average distance a ray of sound travels before it encounters an obstacle and reflects. An approximate value for the mean free path, due to Sabine, is explained by this formula l m, (λth) = 4V/ S. The approximate MFP is a function of room volume, i.e. a larger room has a larger MFP and a smaller room has a smaller MFP. This relationship can be understood from the Sabine formula. Each MFP is like a **delay-line** for the sound ray.The result of this is that a dense broadband reflex pattern is stretched in a large room in the **time domain**. It is easy for any person with normal hearing and a perception to identify a large room from a small room. In a small room the MFP is shorter and the delay-line has a shorter delay. This creates a condensed and compressed early reflexion field. This effects the reverberation radius /critical distance in a way that the sphere becomes smaller and effects the separation of direct sound and reflex sound in a negative way.

A small acoustic space results in a recorded sound with a greater proportion of reflex sound that creates a colored room sound, i.e. (Comb-filtering). Graph 7 shows the initial early reflexion in a rectangular room. The Energy/time curve shows the early reflexion as function of time. In a small room the early reflexion field is compressed in the time domain, resulting in a compact sound field with little acoustic transparency. In a large room the early sound field is stretched in the time domain and a more transparent acoustic is achieved. I believe most recording engineers and musicians have experienced these facts. What is the solution to this problem? In a small room the early reflexion field has to be spread out in the time domain. This makes the reflex field more like a large room, where early reflexion field is spread out in time-domain. The test performed by Lennart Nilsson in a forest with spread out tree trunks is demonstrated in a situation where the reflex field is spread out in the time domain. It is easy to compare the similarity with a large room acoustic space. To find this out yourself, please repeat this test in a silent forest near you. This helps you to better understand the forest test.

I have earlier in my text explained a critical distance or Reverberant radius has to be as big circle as possible to have as big separation between different instrument. The rule of thumb is to have -20 dB different levels in the critical distance. In small spaces this is crucial as a strong small room reflex field has a severe coloration of the sound. This is the **Achilles' heel** for small room acoustics. If a insufficient critical distance is present and recorded, it is almost impossible to eradicate the muddy room color from the recorded audio. Even if there is a digital plug ins, that reduce this muddy reflex sound, (Unfortunately it is a penalty of distorsion in the process, you have to choice between pest or cholera), the only common sense and intelligent way to solve this problem is to install the wing diffusor in the recording space. **This creates a transparent acoustic recording**, where it is easy to separate the music instruments. This is important because a clean recording with a dominating direct sound, will help the mixing process and the positioning of the instrument in the frontal stereo projection. This makes your work much easier and you will be known for high quality recordings.

Demo test No 7,
is a perceptual proof of the wing diffusor and its creation of a transparent acoustic space in a Studio environment. This test has to be performed in a professional monitoring system or on a headphone manufactured by Stax or comparable high end.
Headphone.www.youtube.com/watch?v=IcPPp0HNg84&feature=youtu.be xxx
www.youtube.com/watch?v=OqDSxlZBIYs&feature=youtu.be



Let us find out what has happened in the **control room acoustic space**. A strong reflex sound field will interact with the direct sound from the loudspeaker. The critical distance is as important in the listening situation, where the music mix has to be evaluated. If the direct sound is muffled by the small room reflex field, it will be difficult to hear the sound mix and perform a correct position of the music instrument. Discrete reflexes in the listening acoustic space are creating **Ghost loudspeakers.** This will reduce a clear and well defined front projection with a wide stage, depth and height. In this situation t**he wing diffusor** will be instrumental in creating an accurate studio monitoring situation. There is an control-room test signal **LEDR (Listening Environment Diagnostic Recording)** designed to guide and help the **machinery of hearing** to evaluate the accuracy of listening situation. This test signal is developed at Western University by **Paddy Rodgers** when working on a PhD on the pinna effect. Unfortunately she passed away and **Doug Jones** continued her work. He recorded her test signal on a audio tape. It is called LEDR, Listening Environment Diagnostic Recording, a test to subjectively evaluate the accuracy of stereo image reproduction. I had the opportunity to test a big horn loaded monitor in a listening situation in some "world class" recording studios In Los Angeles, that are known for mixing million selling records. The result was surprising, the human **"machinery of hearing"** is one of the hidden wonder of the world. The reproduction was not following the test signal imaging reproduction. Some fantom sources were reproduced in the wrong position. In one test the phantom source was way up in the ceiling. This tells me that it was a strong discrete reflex sound ray (ghost loudspeaker) from that position.The **LEDR** signal has a controlled phantom source moving in a predefined pattern. If this movement pattern are altered there is a discrete reflex (Ghost loudspeaker) that distort the sound movement. No electronic instrument is necessary, just two good ears and a trained perception.   This LEDR test signal is now possible to download from internet.

To improve the frontal image reproduction accuracy in a control-room acoustic space, a **Wing diffusor** will create that **transparent acoustic space** that makes it pleasant to work in an audio recording and mix situation.

This is why I call, The wing a new PARADIGM in acoustics.

Hans Bristell