


Fundamentals of Electroacoustics	 <p>prolight+sound mediasystems Fachmesse für Medientechnik und Systemintegration Frankfurt am Main, 24.02.-25.04.2006</p>
PLS mediasystems 2006, Frankfurt am Main	
Speaker: Volker Löwer, IFBcon	

Fundamentals of electroacoustics

Is it really possible to calculate acoustics?

How does the acoustics of a room affect the sound of the loudspeakers?

Speaker: Volker Löwer, IFBcon

Foreword

Acoustics in a scientific sense is a very wide-ranging discipline. It includes everything from technical acoustics, which, for example, deals with the vibration analysis of buildings and motor vehicles, the sound proofing of walls etc, to the psychology of hearing, which is concerned, amongst other things, with noise as a nuisance factor.

There is one particular branch of 'acoustics' that is the subject of much talk and discussion, the 'acoustics' at a concert, the 'acoustics' of rooms, what makes good and bad acoustics. It is often maintained that 'acoustics' is not tangible, since the acoustic experience, that is to say the hearing experience, is a subjective process, that evades objective criteria. Consequently, there is also a wide body of opinion that argues that it is likewise not possible to 'calculate acoustics', and that doing so is more or less a product of chance which experts, i.e. acousticians, play around with, together with others and then have a look at what results.


Clearly, the hearing experience is a subjective process and is subject, therefore, to many subjective influences. All our experiences of the outside world are assimilated through our organs of sense and so they all must be understood as subjective experiences.

Now it would be a foolhardy venture to want to calculate or predetermine subjective experiences and perceptions, although disciplines do exist that do deal with such in a wider sense.

Let us concentrate here on the transmission of sound events, with or without electroacoustic amplification, that is to say, on the branch of 'acoustics' which is frequently the subject of such heated debate.

Without doubt, the subjective attitude and perception of the listener does play an important role here. However, for this perception to even happen at all, a transmission of the acoustic event to the listener location in a physical and / or physics sense must first take place.

If you believe in the causal interrelation of cause and effect in our world, the quality and the scope of this transmission therefore provides the base, onto which is superimposed the subjective perception of the listener. This implies in other words: subjective components only come into play in relation to an event, once this event can actually be perceived and sensed.

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Accordingly, the acoustic result or experience is not a mystery; it is always the consequence of the plans and designs, the actions and deeds of those who configure the rooms, install electroacoustic systems, the actors who perform the programme and the subjective components of the listener.

There is a long tradition of scientific research, representative models and mathematical models concerning the transmission and quality of sound events. These permit us to describe the quality of the sound events in terms of physics. Of course, these descriptions are subject to certain limitations; we are using models that describe interrelations between cause and effect.

What is more, all models are limited in the so far as they are only valid within certain boundary conditions. These models are never perfect, since they simplify the complexity of the natural world to enable us to at least have some grasp of what is happening. However, this factor is prevalent in all parts of our lives and the technical world. Structural engineers, mathematicians, doctors, sociologists, educationalists, economists and ecologists also use representative models to describe interrelations and for planning and decision-making.


For the practical application and success of this approach it is important that the models provide sufficiently accurate information in order to be able to make evaluations and decisions about the corresponding options with respect to design and implementation.

For quite some time the acoustic models for room and electro-acoustical transmission have provided results that are quite usable. In fact, they are in part also so developed that it is possible to describe interrelations between certain elements of the subjectively perceived hearing experience and the objective parameters of acoustic transmission quality. What is more, there are now software products that are able to take over what are in part very complex mathematical calculations of acoustic interrelations and to some extent even represent them audibly.

Unexpected and undesired results frequently have the following causes:

- The existing models have not been applied at all
- The input parameters for the models are too inexact or incorrect
- The results from the models are incorrectly interpreted
- No account is taken of the model's viability range
- The implementation diverges greatly from plan
- Compromises are made in favour of other properties

The transmission of sound events using electro-acoustic amplification frequently comprises the task of intelligibly transmitting speech. It is relatively simple to check whether something has been understood in an acoustic sense or not, even though understanding is a very complex process, which, alongside the transmission of the sound event in physics terms, does incorporate implicitly a large element of the subjective experience level of the listener, hearing physiology and psychology.

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Overview

The quality of the sound event at the listener location is determined by the properties of the room, i.e. the room acoustics, the properties of the sound source, their spatial relationship to each and the position in which the listener receives the sound. If these properties are known, it is possible to determine the quality of the sound event at the listener locations and / or calculate certain fundamental parameters at the listening location.

The design of electro-acoustic transmission facilities to improve the quality of the sound event at the listener location (or to widen the listening circle) is, therefore, heavily dependent on room acoustical conditions.

In an ideal world the acoustic of a room should also be designed with the use of a sound system in mind.

It is not possible to effectively design a sound system to improve the quality of the sound event at the listener location without knowledge of the room acoustic conditions.

The intention here is to illustrate a few fundamental interrelations and dependencies using the example of the acoustic quality criterion of speech intelligibility.

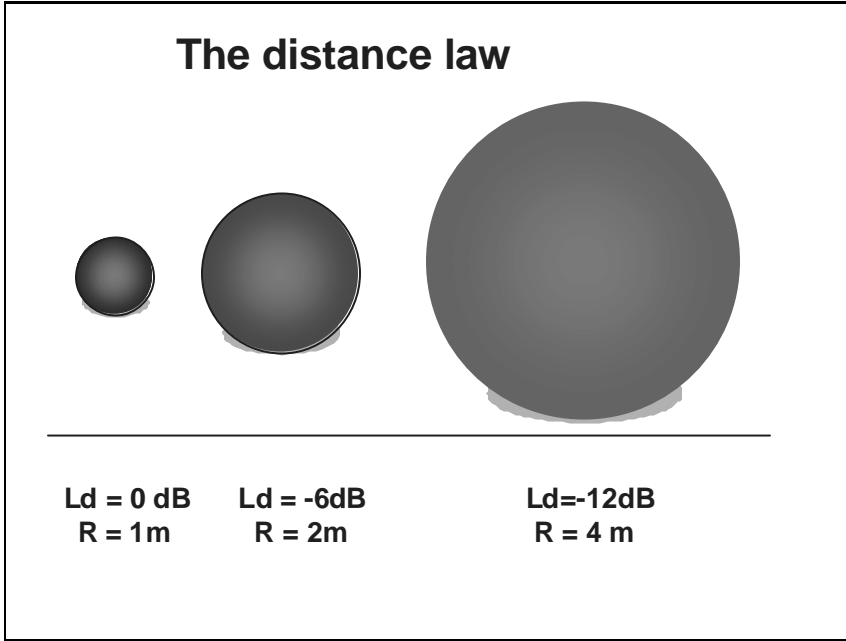
Due to the available framework of this event, the following representations and interrelations contain a number of simplifications that are not specifically denoted. You can find more advanced and detailed studies on this subject in relevant specialist literature.

Free-field

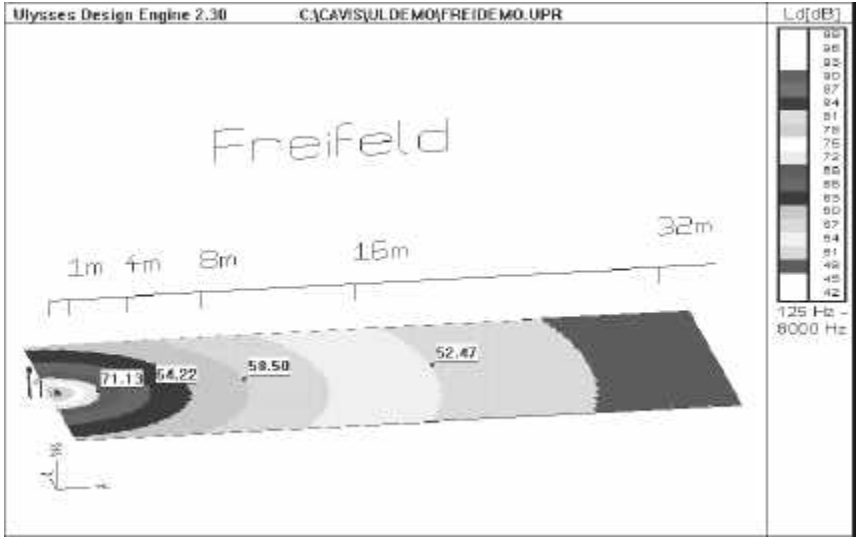
In acoustics so-called free-field conditions are understood as being an environment where sound propagation is undisturbed (without the effects of spatial restrictions). You can imagine this through the analogy, for example, of a person speaking on a large, open space.

If you know the properties of the sound source (speaker or loudspeaker) and the distance to the listener location concerned, you can use the sound propagation velocity to determine the time delay and the distance law to determine the sound pressure level.

The sound waves from a sound source propagate at a constant velocity (340m/s at 20°C, or approx. 1m / 3ms) in all directions. As the distance increases, so the intensity of the sound event diminishes, since, as the distance increases, the sound energy is spread out over an every greater area. The spherical area around a source increases by four times its size each time the radius is doubled, the intensity decreasing to $\frac{1}{4}$ or by 6 dB, when expressed in terms of sound level.



Therefore, each time the distance doubles the sound pressure level reduces by 6 dB (e.g. in 1m 75dB SPL, in 2m 69dB SPL, in 4m 63 dB SPL...), the further you move away from the sound source, the quieter it gets.



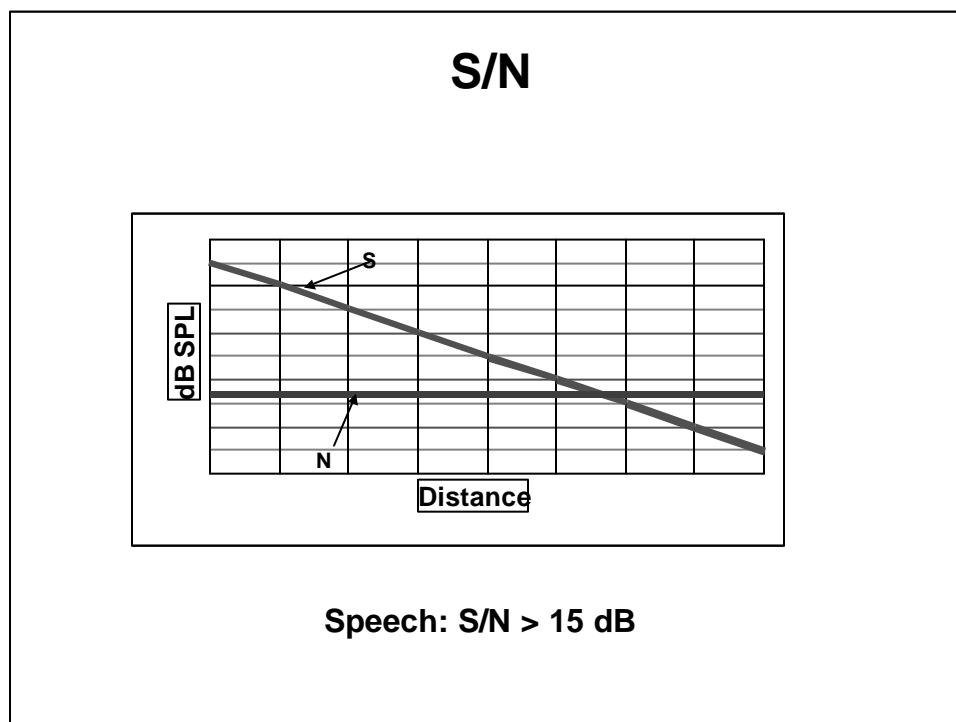
That also accords with what we experience every day. If you look at (hear) the sound pressure level (sound volume) of the signal at different distances, you discover that in our example a listener at a distance of 32 m still receives 45 dB SPL; in fact, even at a distance of around 500 m it is still 21 dB SPL.

Since the hearing threshold stands at 0 dB SPL, it must be possible for the signal of the speaker to still be heard and understood at a listener location 500 or 1,000 metres away. However, we know from our experience that this is not generally the case.

Signals / noise

Our environment is full of surrounding noises (noise) that impair the quality of the acoustic transmission at the listener location.

As the signal weakens with increasing distance, it is progressively obscured by the noise. Although the signal is still there and can be heard, from a certain distance the noise has such a high level, that it is no longer possible to recognise and understand the signal.




Investigations have shown that speech can be almost fully understood when the sound level of the signal at the listener location is around 15 dB greater than the level of the noise signal ($S/N > 15$ dB). This sound level differential implies that the modulation of normal speech is totally above the noise level.

As the interference noise increasingly obscures the speech modulation, intelligibility likewise decreases. Taking to its extreme, under certain circumstances (talking in a disco or next to a jet engine) it is no longer possible to understand the signal at all. The specification of the sound level differential S/N is certainly sufficient to describe one element of the acoustic quality at the listener location, e.g. also the intelligibility.

Accordingly, it is quite fundamental to always achieve a sufficiently high sound level in relation to noise. This applies to open-air transmissions as much as it does to those inside, whether they use electro-acoustic systems or not.

As a result, it is easy to understand that the highest quality transmission is worth nothing, when it is buried by noise and cannot, therefore, be perceived and understood.

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What is room acoustics?

Room acoustics is concerned with the structural design of rooms from the perspective of their acoustic properties. How does a particular room transport and distribute the energy from acoustic events.

People often talk generally about good or bad acoustics. This evaluation does not stand up to interpretation as such, since certain acoustic properties are suitable for one particular application, but may be unfavourable for another purpose. So, for example, the long reverberation time of a church is appropriate for organ music, but less so for a conference.

As a consequence, the question as to the quality of room acoustical properties can only be answered in relation to the intended use.

The room acoustical properties are particularly important when the intention is to use a sound system, since the loudspeakers, together with the room acoustics, create a new acoustic situation throughout.

The mechanics of room acoustics

The intention here is to deal only with the basic mechanics of propagating acoustic energy in rooms.

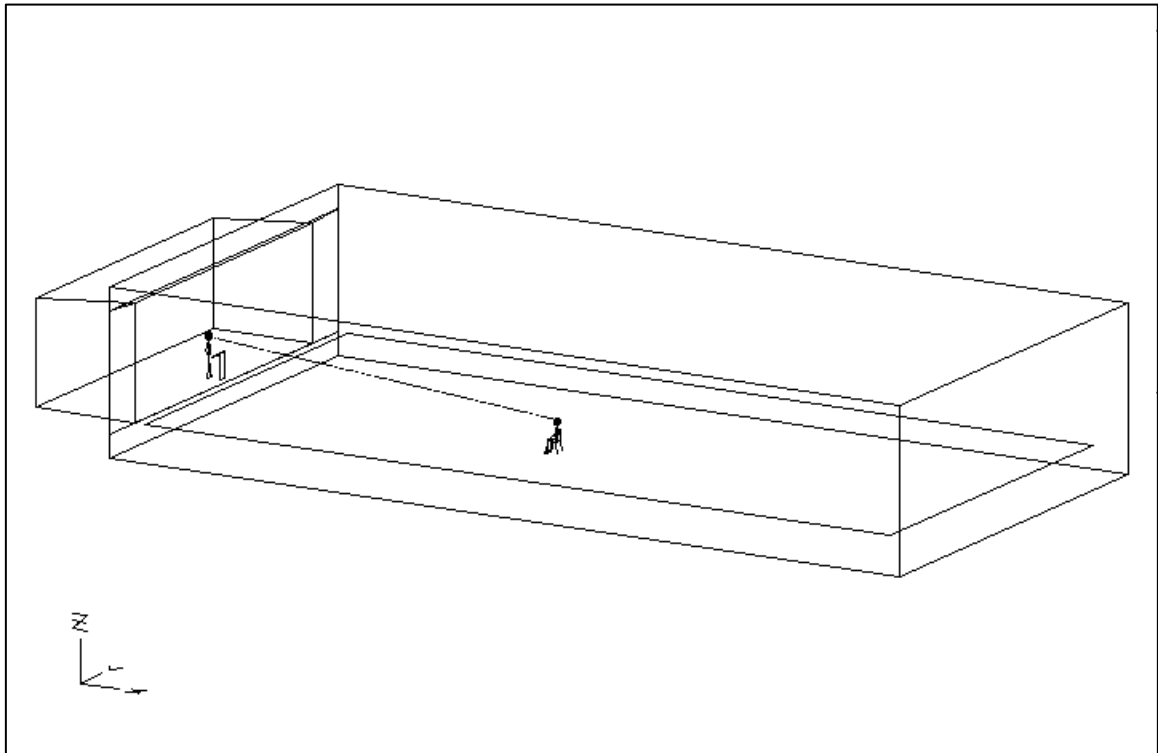
The element of acoustic energy that travels directly from the source to the listener is called direct sound.

Direct sound, as that element which has the shortest distance to cover and which is always the first to reach its target. Direct sound is free of the influences of room acoustics and corresponds to the conditions as already described for the free-field.

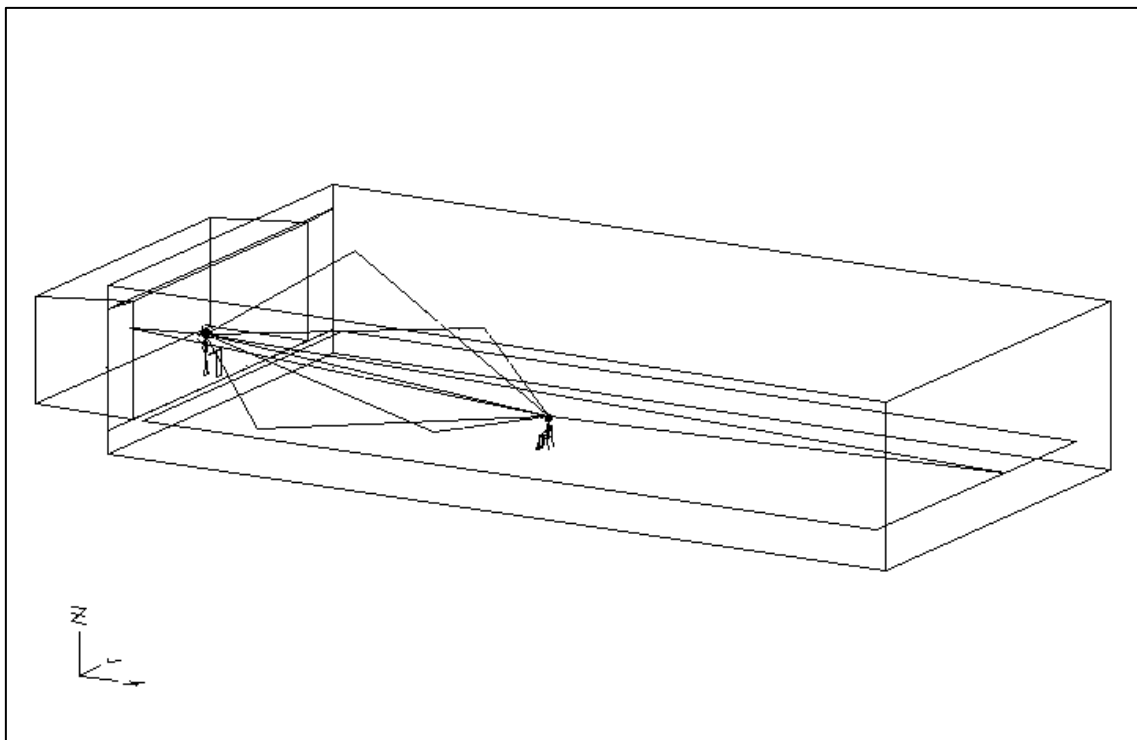
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Depending on the size and shape of the room, the sound at some point comes up against restrictions, walls, the floor, the ceiling. At this point the sound energy can be thrown back (reflection), taken in (absorption) or let through (transmission). The amount that is absorbed, reflected or transmitted depends on the acoustic properties of the boundary surface.



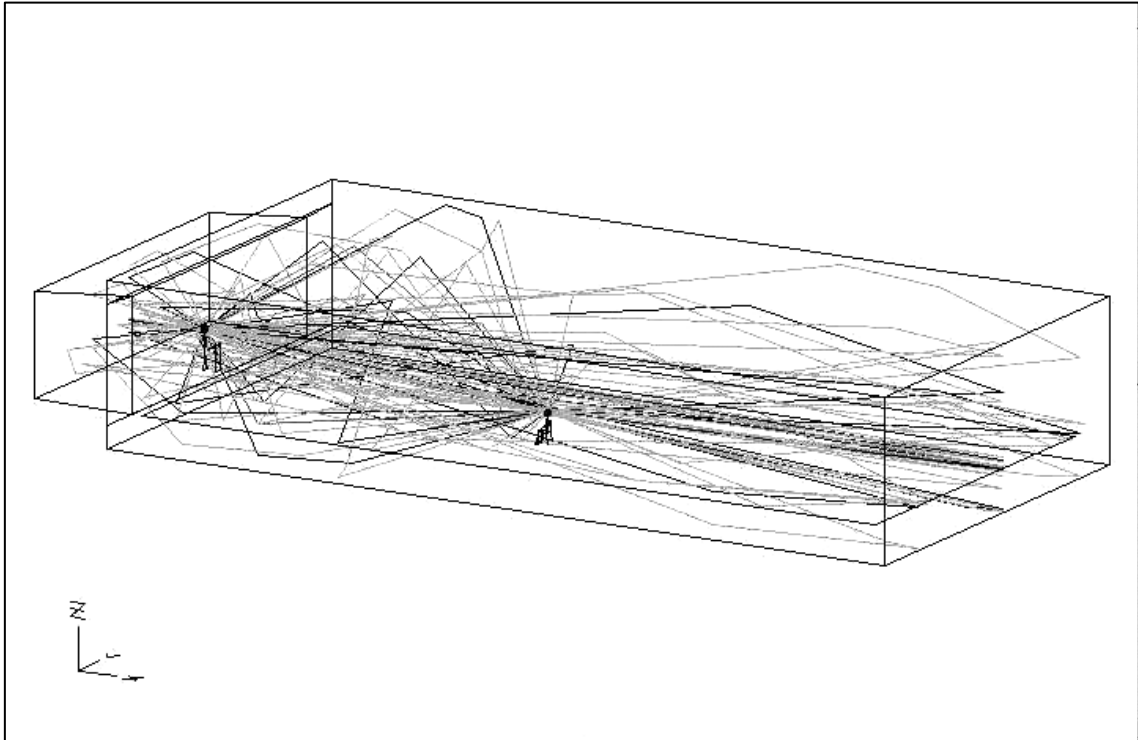
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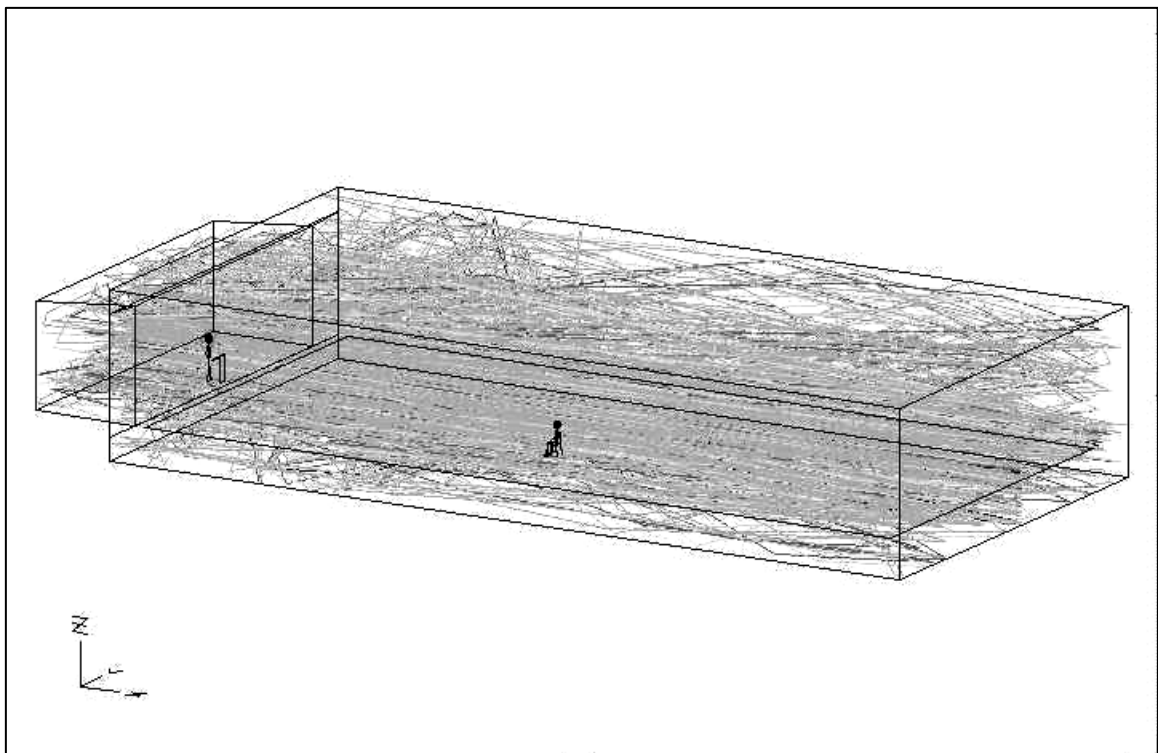
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Due to the fact that the acoustic energy is to a greater or lesser extent reflected back into the room by the room restrictions, a multiplicity of reflections occur.



Repeated reflections and further propagation eventually result in the remaining sound energy being spread almost homogenously throughout the room.



The sound field that thus emerges is called a reverberation. The reverberation time denotes how quickly the homogenously distributed sound energy is absorbed by the boundary surfaces.

The spatial, temporal and frequency-specific composition of direct sound reflections and reverberation can be used to specify a room's most important acoustic properties.

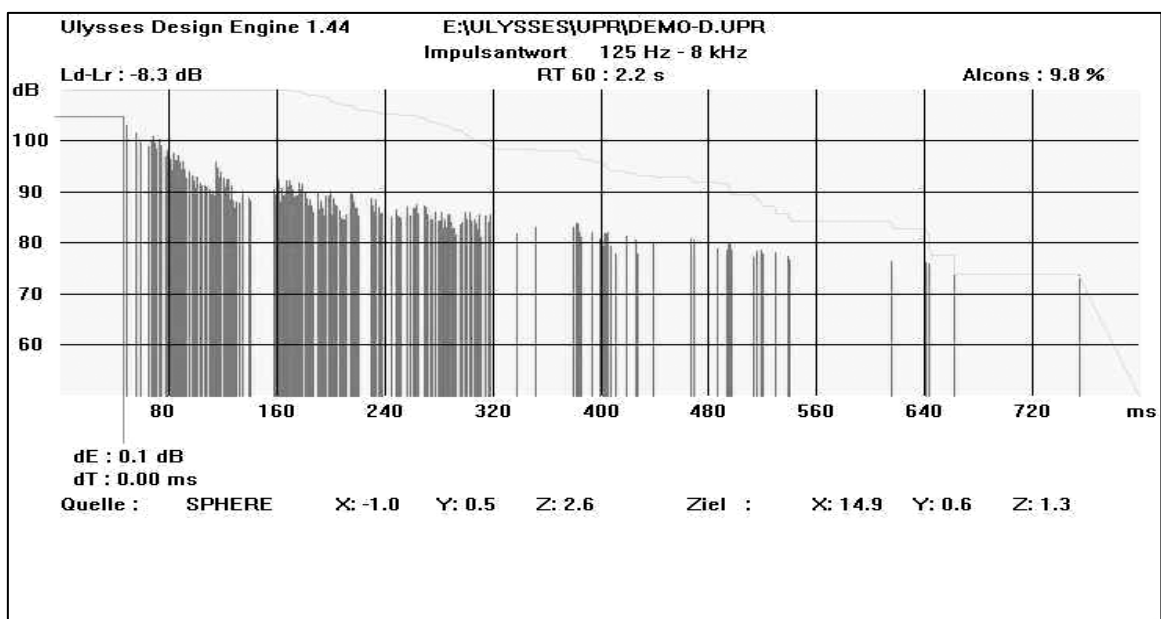
Direct sound


If you know the properties of the sound source and the distance to the listener location concerned, you can (just as with the free-field) use the sound propagation velocity to determine the delay and the distance law to determine the sound pressure level (**direct sound field**)

Reflections

All reflections reach the listener with a time delay in respect of direct sound and in a normal case, i.e. when the room boundaries do not produce any focussing effects, also with less energy.

If you know the position, distance and reflection or absorption properties of the surfaces that border the room, it is likewise possible to determine the delays, directions and sound pressure level of the individual reflections for the respective listener location. The temporal representation of an impulse is also called an impulse response or more generally referred to as a reflectogram. It applies only to the respective receiver location.



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The fact that the same acoustic event reaches the ear at different times can have both positive as well as negative effects.

It depends on the sound level conditions and the time delay of their arrival at the listener location whether for the listener they merge into one event (useful reflection) or are perceived as separate events (echo).

A fundamental task of room acoustics is to create useful reflections and at the same time to eliminate or minimise detrimental reflections.

Of course, these conditions should also provide for the use of a sound system.

In this context, special account must be taken of the fact that natural sources (speakers, instruments) radiate from other places in the room than do the loudspeaker systems used.

These different source configurations also produce different reflection structures that accordingly also contain other useful and detrimental reflections. Unfortunately, this fact is a factor which is frequently overlooked in many projects.

Reverberation

The reverberation time T_{60} is one of the most frequently used parameters of room acoustics. It describes the time after which a sound event in room has decreased to a millionth (-60 dB) of its initial energy. Similarly, this equates to a reduction in the sound pressure of 60 dB.

The reverberation time can be determined from the volume of the room and the sound absorption properties of all the boundary surfaces (see also Wallace. C. Sabine).

The correlation of calculation and reality is conditional upon a homogenous distribution of the reverberation energy in the room (statistical model, diffuse reverberation field), which in turn presupposes a uniform distribution of the absorbent surfaces in the room.

The reverberation time alone is not entirely appropriate to a simple and direct evaluation of a room's acoustic quality. In any case, it is necessary at least to also take the intended use and the volume of the room into consideration. In general it can be said that it is possible to tolerate longer reverberation times in larger rooms than in smaller rooms. Furthermore, in rooms with the same volume longer reverberation times are beneficial for music events and short times for performances involving the spoken word.

As a consequence, it cannot be argued necessarily that a room with a reverberation time of 2.0 s has a 'worse acoustics' than another room that has 1.2 s.

What is more, reverberation time is a parameter that dependent on frequency, since the room boundaries likewise have frequency-dependent absorption properties. For this reason, a reverberation frequency response should also always be stated,

calculated or measured. In general, the objective is for the reverberation frequency response not to contain excessive non-linearities.

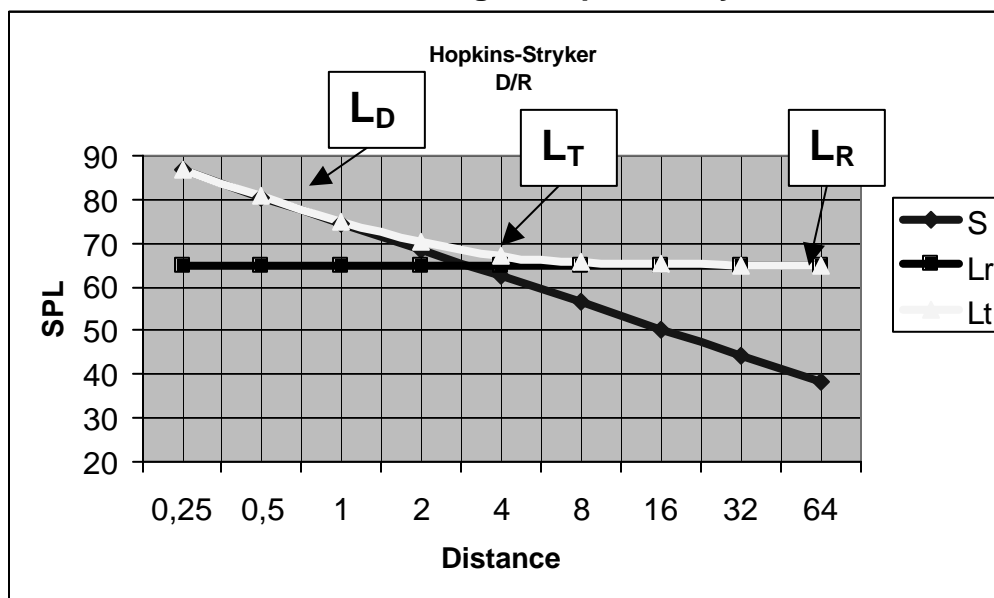
Finally, it must also be taken into account that at different events the audience absorbs sound at different rates. In this case it is necessary to ensure that appropriate acoustic conditions can be expected even at a minimum determined occupancy.

Critical Distance

There is a parameter that is much more meaningful in terms of evaluating the reverberation properties of a room, the reverberation radius. The reverberation radius specifies the distance from a spherically radiating sound source in a room, where the size of the direct sound field is equal to that of the reverberation field.

If you now plot the direct sound path and the reverberation level against the distance, both curves will intersect at the critical distance. The reverberation curve (reverberant sound level) shifts upwards for longer reverberation times and downwards for shorter reverberation times (doubling of the reverberation time +3dB, halving -3dB). The critical distance can be determined from the volume of the room V and the reverberation time T_{60} .

Sound level in a room according to Hopkins-Stryker



L_D : Direct sound level, diminishes by 6 dB each time the distance doubles

L_R : Reverberant sound level, based on room volume and absorption surface

L_T : Total, sum of L_D and L_R

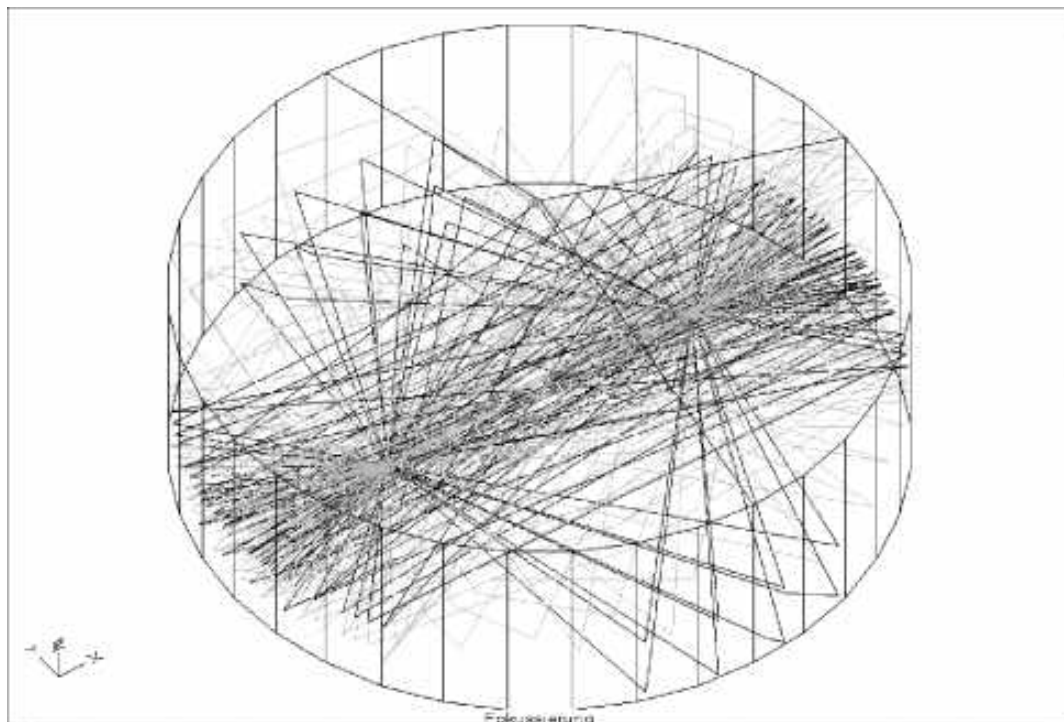
This graph makes it possible to establish the ratio between **direct sound level** and **reverberant sound level**, as well as the **total sound level** as the sum of these levels for different distances from the sound source. This ratio represents a fundamental factor for the intelligibility and clarity of the acoustic event.

The specification of the difference in level between direct sound and reverberant sound (D / R) gives us the second important criterion with which to describe intelligibility.


Observing how the sound fields develop likewise reveals the full complexity of the problems associated with the reverberation sound field. On the one hand, the reverberation sound field helps to provide the further away listener with a volume increase with respect to the free-field. On the other hand, if the ratio of the reverberation sound field to direct sound is too great, it adversely affects intelligibility and clarity.

Special phenomena

Often real rooms throw up special phenomena. Indeed, the previously mentioned echo-effects, for example, multiple, separately perceived reverberations or concentrations of sound energy in a particular area of a room (e.g. in the middle of a reverberant, cylindrical room) are often the cause of complaints about the 'acoustics'.



In fact, these phenomena are frequently interpreted as 'too long a reverberation time'.

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As a result of this misinterpretation additional sound absorption material is brought into the room to reduce the reverberation sound field. However, in many cases the criticised effects are now even more pronounced. In very unfortunate cases additional disruptive effects will even be heard, that were previously completely concealed by the reverberant sound.

Consequently, when minimising disruptive effects, it is imperative to always determine the exact causes and then also focus on combating them. A disruptive rear wall reflection, for example, which is perceptible on the stage as an echo, can likewise only be eliminated in terms of room acoustics by measures applied to the back wall. In this case, although additional absorption on the side walls will reduce reverberation time, it simultaneously makes the reverberation more clearly audible.

These effects can be observed both with and without electroacoustic systems. All the same, you need to be aware that different surfaces of the room can cause critical reverberations and concentrations depending on where in the room the sound source is located (natural speaker, loudspeaker).

For further information you need to refer to the relevant specialist literature.

Interrelations

As already mentioned, the S/N and D/R ratios are the two principal mechanisms that determine the intelligibility and clarity of an acoustic transmission.

On the one hand, you must also ensure that the signal is as strong as possible compared to noise, or, if possible, minimise the noise.


This is the purpose of a sound system, e.g. at an open air event, when the sound volume of the natural source is not sufficient for listener locations further away.

In this case intelligibility is simply achieved by generating a stronger signal.

Limitations of this include the power rating and effectiveness of the loudspeaker systems as well as the system's maximum acoustic amplification before feedback occurs.

Of course, in rooms the first objective is likewise to ensure that the signal is of sufficient strength compared to the prevalent level of noise. Similarly, the effectiveness of the loudspeaker systems and the system's maximum acoustic amplification limits the possibilities.

However, in rooms you must also take into consideration that you require a satisfactory ratio of direct sound to reverberant sound, in order to ensure sufficient intelligibility. Simply increasing the radiated acoustic energy (increase the volume of the source) or adding to the number of sources (more loudspeakers) does indeed produce more direct sound at the listener locations, however the reverberant sound level increases to the same extent, since the reverberant energy comes, of course, from the sum of the total acoustic energy radiated from sources.

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This clearly demonstrates this time the problem is totally different from that involving signal / noise; direct sound and reverberant sound are, in contrast to signal and noise, interdependent signals.

Consequently, the only option to improve intelligibility is to resolve the D/R ratio through either a greater proportion of direct sound at the listener location or, alternatively, less reverberant sound. One possible solution is to look at the directional characteristics of the loudspeakers, i.e. their directivity properties.

These properties describe how a loudspeaker system radiates the acoustic energy in the form of direct sound into the different parts of the room. A system that radiates acoustic energy at the same level in all directions is called an omni-directional (spherical) transducer, while a system that radiates energy especially in one particular area of the room is known as a (uni)directional transducer.

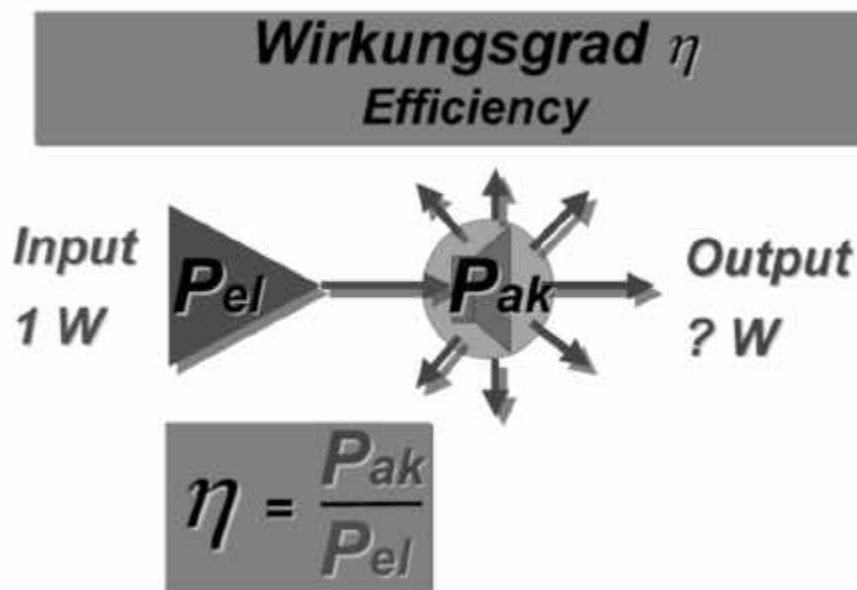
The advantage of such a (uni)directional transducer compared to an omni-directional one is that it produces a greater direct sound field in the main radiation axis with exactly the same radiated energy. However, since the reverberant sound field is determined by the total radiated energy and the sound absorption surface area of the room, the direct sound increases in the main radiation axis of the (uni)directional loudspeaker, while the reverberant sound level remains the same. The result is an improved D/R ratio for the listener locations in the main radiation axis and, accordingly, also improved intelligibility in these locations.

For sound system design this means that it is possible to improve the intelligibility of the transmission by using appropriately configured (uni)directional loudspeakers.

Loudspeakers, a few parameters

Efficiency

The efficiency η describes the ratio of radiated output (in the form of sound) to the electrical power supplied. Consequently, it states how much acoustic wattage we get from the amplifier for the electrical wattage supplied.

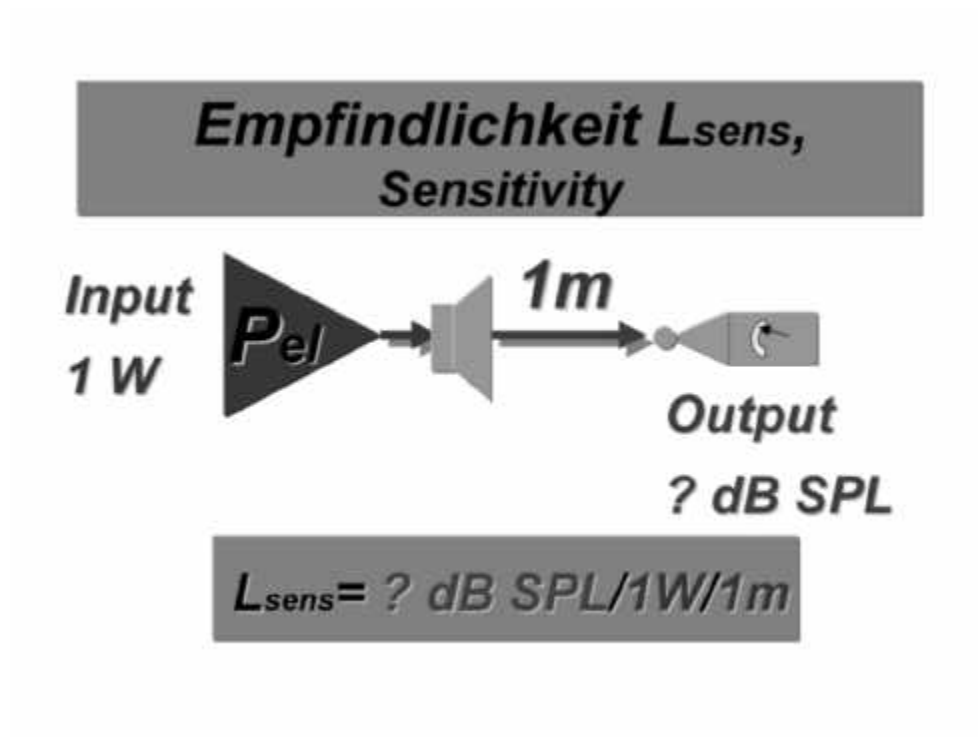


Typical values are between $\eta = 0.1\text{-}1\%$ for standard hifi loudspeakers and up to $\eta = 25\text{-}30\%$ for professional high-end systems.

Sensitivity Lsens

The sensitivity describes the sound pressure level (SPL) which is generated by the loudspeaker at a specific distance on its main radiation axis, when we supply a specific electrical input.

It states, therefore, how much sound pressure level we get at a specific distance along the main axis for the electrical wattage supplied from the amplifier.

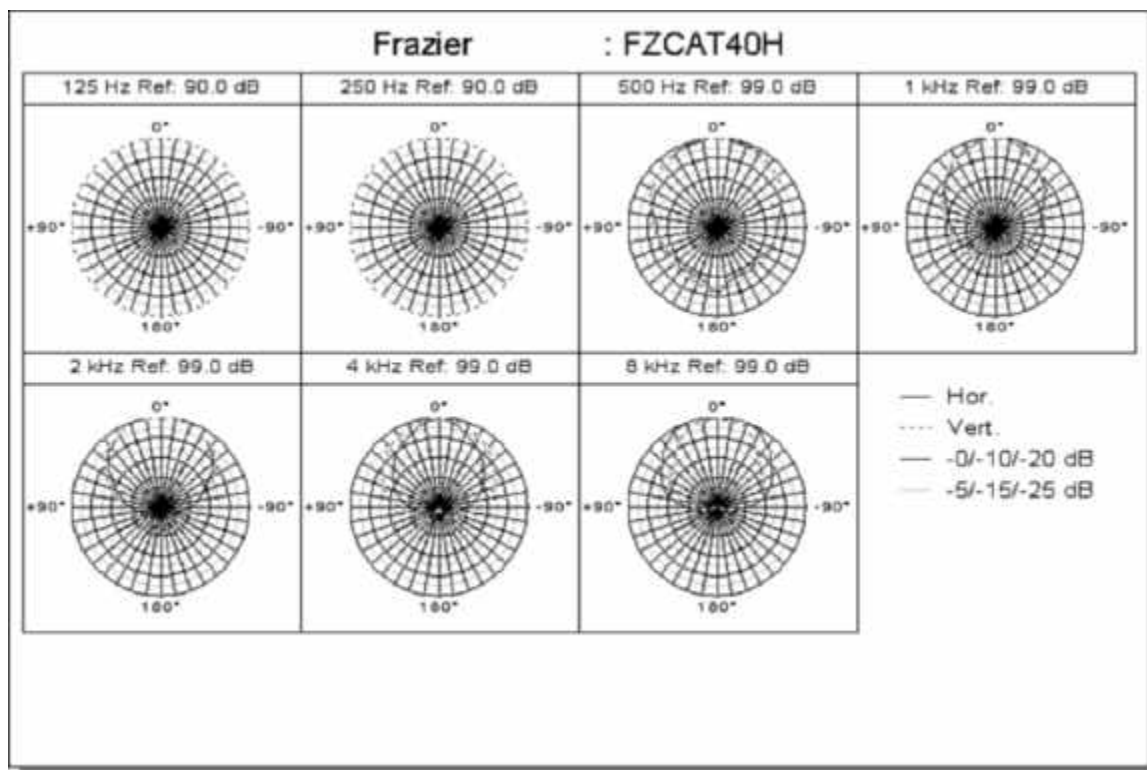


Typical values are between $= 88 \text{ dB SPL/1W/1m}$ for standard hifi loudspeakers and up to $L_{sens} = 117 \text{ dB SPL/1W/1m}$ for professional high-end systems.

Coverage Angle

The coverage angle denotes the angle at which the sound pressure level does not exceed a certain tolerance range. It states, therefore, the angle around the main radiation axis in which the sound pressure is radiated for the most part homogeneously (mostly $\pm 3\text{dB}$ or $\leq 6\text{dB}$). Generally, specifications are given for the horizontal and the vertical plane, e.g. $90^\circ \times 60^\circ$ stands for a 90 degree horizontal and a 60 degree vertical coverage angle. This specification provides no indication as to the sound pressure level or the energy generated outside this area.

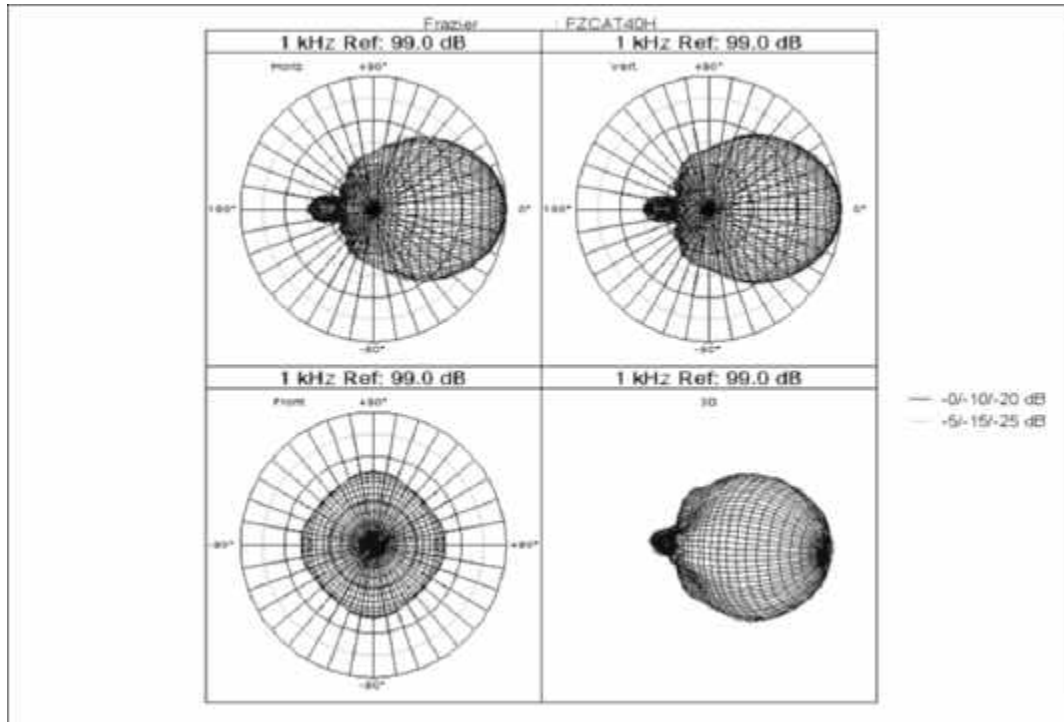
Polar diagrams



This information does not indicate the sound pressure level or energy generated outside this area.

Balloons

This requires a 3-dimensional representation, the so-called balloon. The balloon indicates the radiation in all directions.



Different balloons can be presented for the different frequency ranges, which demonstrate how the directional characteristics change with frequency.

Directivity Factor Q, Directivity Index DI

If a system radiates acoustic energy in all directions at the same sound level, you talk about an omni-directional (spherical) transducer. This has a directivity factor of $Q=1$ or $DI=0\text{dB}$.

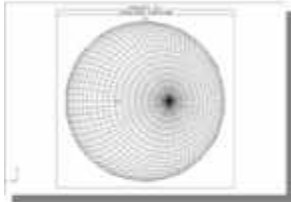
If a system radiates the same amount of energy as the previously considered spherical transducer, but is focussed on a particular area of the room, we speak of a (uni) directional transducer.

This results, therefore, in increased energy and sound pressure level in that particular area ($Q>1$, $DI>0\text{dB}$), while in the other less supplied areas the energy and, therefore, the sound pressure level falls ($Q<1$, $DI<0\text{dB}$).

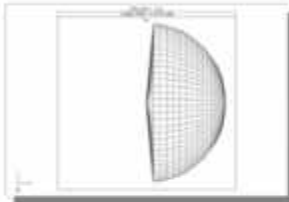
As a rule, directivity data are provided for the main radiation axis; other radiation directions have other directivity data, because they also have other sound pressure levels.



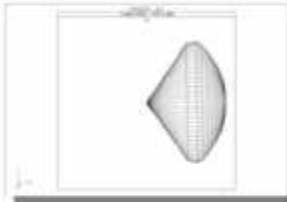
Bündelung/ Directivity Q , D_l



$$Q=1$$
$$D_l=0 \text{ dB}$$



$$Q=2$$
$$D_l=3 \text{ dB}$$



$$Q=4$$
$$D_l=6 \text{ dB}$$

$$D_l = 10 \log Q$$

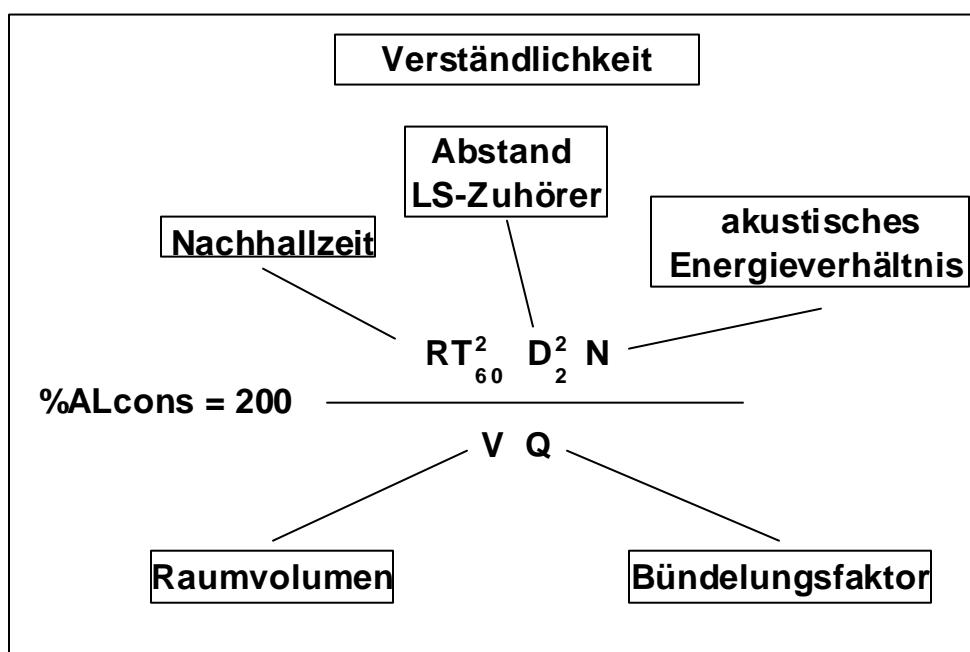
The sum of all directivity factors of a system over a whole surface is always 1 (in mathematical terms the surface integral of the directivity Q over the total spherical area).

This implies that increasing the directivity in one area of the room always results in the diminution of the directivity in another (put simply: if you get more at the front, you therefore invariably get relatively less at the back).

Intelligibility, clarity

For the most simple design you can use the room volume, the reverberation time, the directivity factor of the source (loudspeaker) and the distance between the source / listener to establish the so-called expected loss of consonants in % alcons (articulation loss of **con**sonants) as an indication of speech intelligibility.

Admittedly, this very simple relation (according to V.M.A. Peutz, D. Davis) is based on the assumption of an adequate S/N of a minimum of 25 dB and a statistically diffuse reverberation field without other anomalies.



Verständlichkeit= Intelligibility %ALcons

Abstand = Distance of Source to listener D^2

Nachhallzeit= Reverberation time RT_{60}

Akustisches Energieverhältnis =Acoustic energy ratio N

Raumvolumen = Room volume V

Bündelungsfaktor = Directivity factor Q

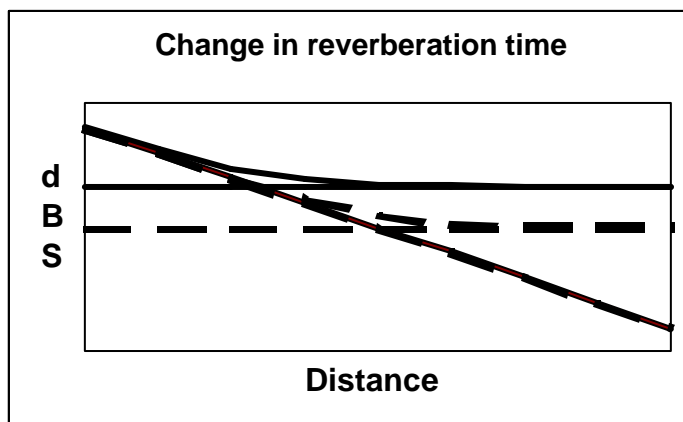
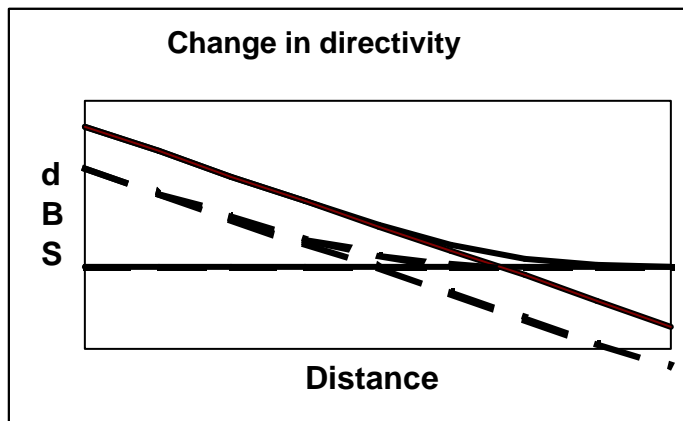
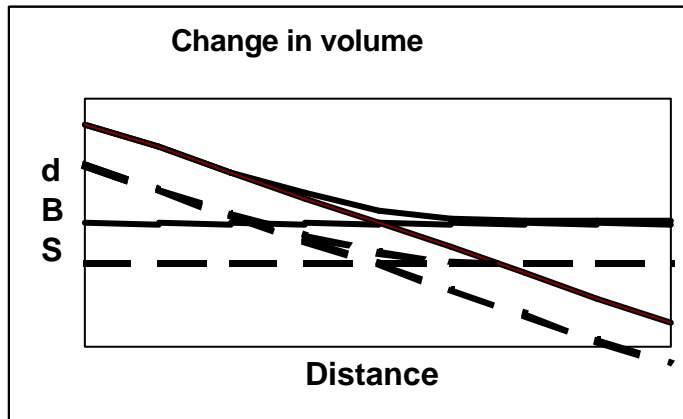
An option to improve intelligibility involves reducing the room's reverberation time, which results, of course, in a reduction of the reverberant sound level and, therefore, an improvement in the D/R ratio.


Nevertheless, reducing the reverberant sound level by 3 dB (equal to a halving of the reverberation time) necessitates doubling the room's absorbent surfaces.

In practice such design alterations not only take a lot of time and money, but usually they never get past the design stage due to architectural objections.

Changes to the different parameters

The following are diagrammatic representations of different changes to the sound field parameters.



Fundamentals of Electroacoustics	 <p>prolight+sound mediasystems</p> <p>Fachmesse für Medientechnik und Systemintegration</p> <p>Frankfurt am Main, 24.02.-25.04.2006</p>	
PLS mediasystems 2006, Frankfurt am Main		
Speaker: Volker Löwer, IFBcon		

Conclusion

The quality of an acoustic transmission is determined in essence by the properties of the room, i.e. the room acoustics, the properties of the sound sources, their spatial interrelation and the position of the listener receiving the sound.

As a consequence, the design of electro-acoustic transmission facilities is dependent to a considerable extent on the room-acoustical conditions. Ideally, room acoustics should also be designed with the use of a sound system in mind.

In addition to ensuring moderate reverberation times, the design must also take care to avoid both with and without a sound system the propagation of anomalous room acoustical effects, such as disruptive reflections, concentrations or echoes.

The principal parameters determining the quality of the acoustic transmission in terms of intelligibility are the ratio of signal to noise S/N and the ratio of direct sound to reverberant sound D/R at the listener location.

These are dependent both on the properties of the loudspeaker as well as the room acoustical situation.

In this respect the most important loudspeaker properties are their radiation characteristics. They describe how the loudspeaker spatially distributes its energy. Particular significance is given to the coverage angle and the directivity factor i.e. the extent of directivity as well as the total spatial distribution of energy, the so-called balloon.

To describe the room acoustical situation the minimum you should take into account is the reverberation time and the volume of the room as well as an adjustment for the respective use case.

In fact, suitable models do enable us to determine with sufficient accuracy certain key parameters of acoustic transmission quality.

As a result, it is likewise possible to determine quite accurately one of the bases for the subjective hearing experience, in this case understanding.

A condition of such, however, is to respect the validity range of the models used, ensure that input parameters are sufficiently exact and interpret the results of the model calculations correctly.

Given a concordant design and implementation, it is possible to expect results that on the whole do not diverge fundamentally from those gained through the model calculations.
