ACOUSTIC BASICS

Acoustic Basics Sound – How does it work?

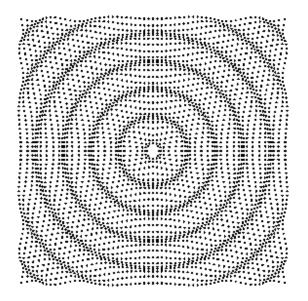
We will have a little closer look to the following topics:

- Propagation of sound
- Speed of sound
- Frequency and wavelength
- → Sound pressure Level (SPL)
- → Human Hearing window
- Weighting of Sound Pressure Level (SPL)
- → Masking Effects
- Localization of Sound Sources
- → Interference Effects



Acoustic Basics Wave Propagation & Particle Movement

- Sound waves in a medium are generated by vibration of an object
- In the air, sound travels by rarefraction and compression of air molecules
- → The amount of energy that is transported in a certain medium describes the intensity of sound. The intensity of sound is defined as sound energy per unit area





Acoustic Basics Propagation of Sound in air – The Inverse Square Law

→ The common rule:

By doubling the radius of a point source the surface increases to the fourfold, according to this the energy density will be reduced to $\frac{1}{4}$, which results in a decrease of -6 dB.

To be exactly - The above mentioned rule is only valid for a single sound source in the free field and in case of bigger distances additional influences take place, but in general this is the common way for rough calculations.

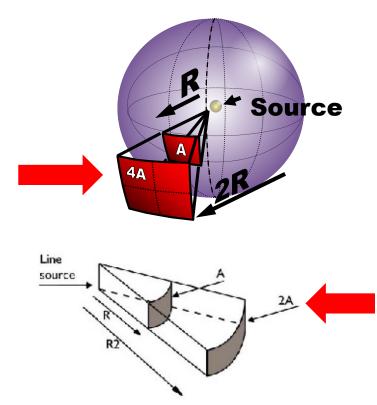
Additional influences are referred to environmental conditions like humidity, wind and temperature.



Acoustic Basics Differences in propagation of an ideal Point- and Line Source

Point Source:

For each doubling of the radius energy density spreads to -> 4A = **-6 dB**



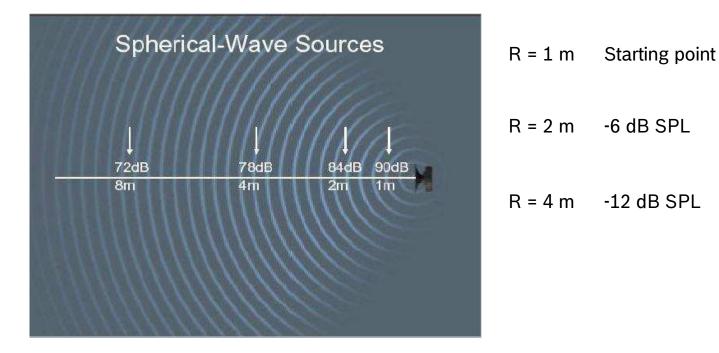
► Line Source:

For each doubling of the radius energy density spreads to -> 2A = **-3 dB**



Acoustic Basics Propagation of Sound

Example to the decrease of the sound pressure level referring to the distance of the point source.





Acoustic Basics

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dB SPL	75	69	63	57	51	45	39	33	27	21
Meter	1	2	4	8	16	32	64	128	256	512





Acoustic Basics Propagation of Sound

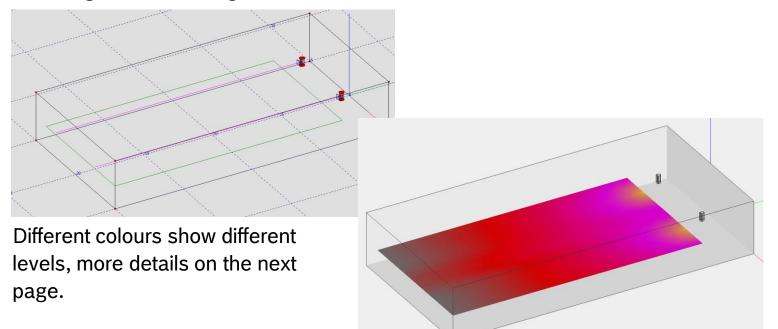
- → Why could that be interesting if it comes to a sound system?
- → Following situation:

A speech system with two loudspeaker has to be setup in a room and should be not too loud in the front rows, but of course understandable in the last rows

→ What are the possibilities?

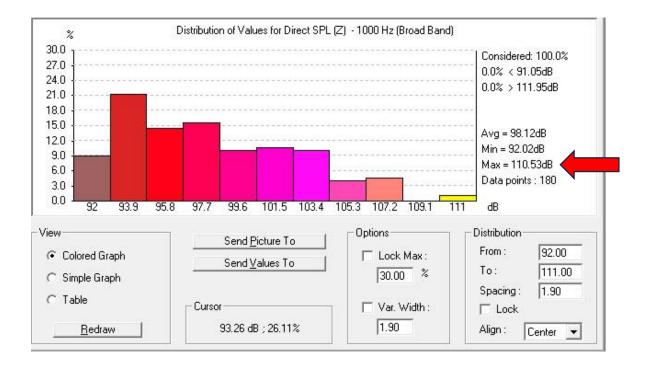


 Setup with standard loudspeaker stands in a height of 1,8m, loudspeakers are faced straight forward. Room size: 20 by 10 meters, room height 3,5m, Listening area 1,2m height (seated audience)





To make the results more clear the different SPL levels abroad the room are listed beyond, including the minimum and maximum levels:

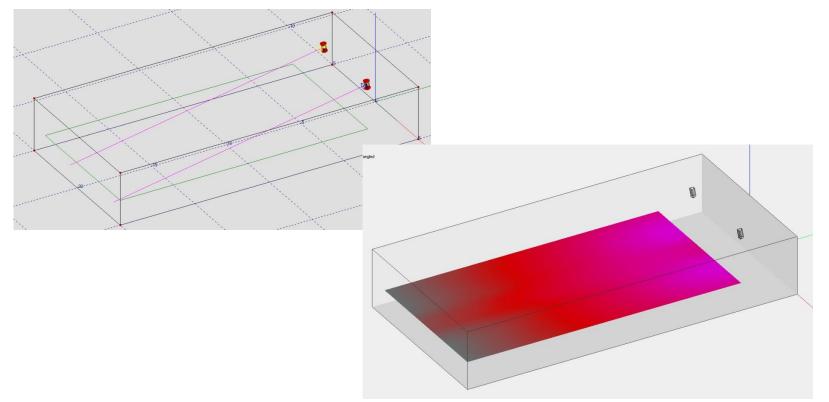


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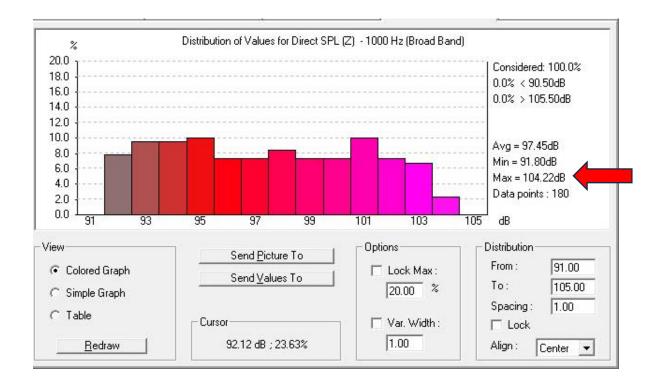


The same loudspeakers, placement and room dimensions as before, but now with the loudspeakers mounted in a height of 3m and angled.





→ As before the different SPL levels abroad the room are listed beyond:



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Acoustic Basics Examples / Conclusion

- With the background of sound propagation in mind you can avoid misplacement of loudspeakers and improve situations, even if there are limited tools to work with.
- → The example show the level improvement (there are some other important values which have to be taken into account for placement) especially for the first rows, but it is also visible, that it is not a smooth coverage over the whole area. It is not a perfect solution, but a first step to get into the right direction.



Acoustic Basics Speed of Sound

The thought still haunts in mind, that the best propagation of sound happens through the air...The fpeoples ollowing chart shows the Speed of Sound in Various Media @ 15 ° Celsius

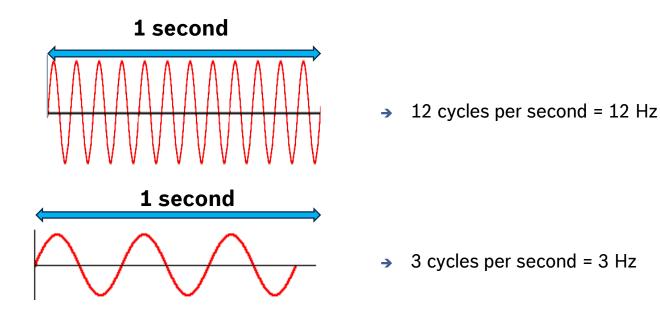
Air	341 m/s
Pure Water	1440 m/s
Saltwater	1500 m/s
Paper	2100 m/s
Iron	3400 m/s
Steel	5050 m/s
Aluminum	5200 m/s
Titanium	6100 m/s

The speed of sound always refers to the properties of the certain medium, it is not influenced by frequencies or amplitudes of the sound.



Acoustic Basics Frequency and wavelength

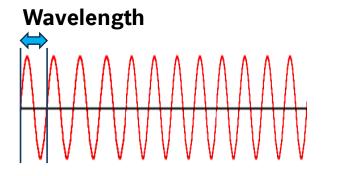
- The number of cycles per unit is called **frequency**, commonly used cycles per second.
- → Many cycles = high frequency, less cycles = low frequency



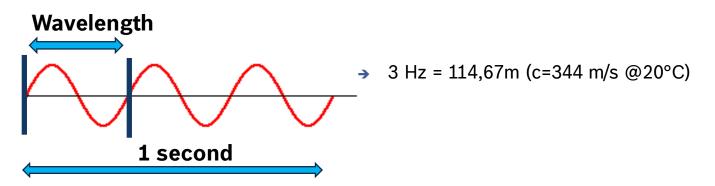


Acoustic Basics Frequency and wavelength

→ The wavelength describes the length of one cycle



→ 12Hz = 28,67m (c=344 m/s @20°C)





Acoustic Basics Frequency and wavelength

range of frequencies is given in tabular form below c = speed of sound in air Frequency (Hz) Wavelength (ft) <u>Wavelength (m)</u> [m/s] 20 56.5017.20 (344 m/s @20°C) 31.5 35.87 10.92 40 28.25 8.60 50 22.60 6.88 f = frequency [Hz] 63 17.94 5.46 80 14.30 4.30 100 11.30 3.44 125 9.04 2.75 160 7.06 2.15 200 5.65 1.72 250 4.52 1.38 **Examples:** → 315 3.59 1.09 400 2.83 0.86 500 2.26 0.69 630 1.79 0.55 800 1.413 0.430 Frequency [Hz] Wavelength [m] 1,000 0.130 0.3441,250 0.904 0.275 100 3,44 \rightarrow 1,600 0.706 0.215 2,000 0.565 0.172 \rightarrow 1.000 0,344 2,500 0.452 0.138 0.359 0.109 3,150 4,000 0.283 0.086 \rightarrow 0,0344 10.000 5,000 0.226 0.0696,300 0.179 0.055 8,000 0.1410.043 12.500 0.090 0.028 16,000 0.071 0.022 20.000 0.057 0.017

The relationship between wavelength and frequency for a



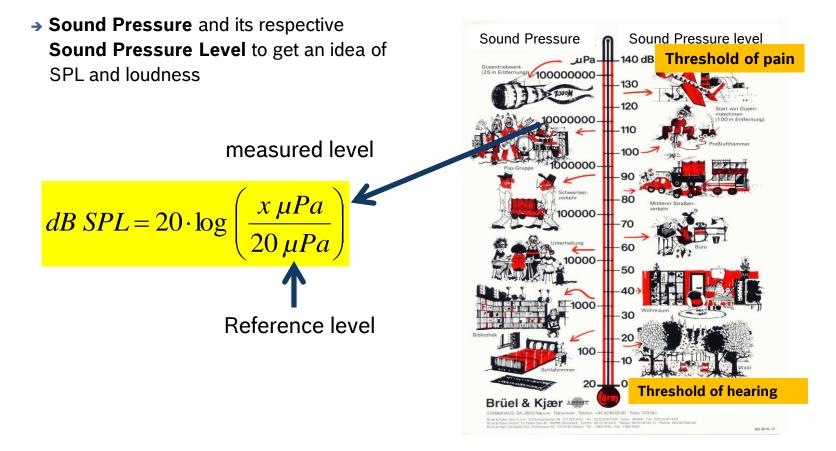
Acoustic Basics Sound Pressure Level (SPL)

- An audible sound pressure is a tiny modulation of the atmospheric pressure on earth.
- The sound pressure level is a relative quantitiy referring to a e.g. measured sound pressure and the reference or fixed sound pressure. The reference sound pressure is usually the human threshold of hearing.
- → The reference of sound pressure level is 20 μ Pa. That is the smallest audible change in the atmospheric pressure.

Sound Pressure is measured in µPa, Sound Pressure Level in dB SPL



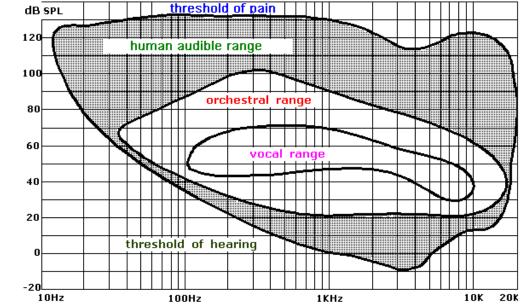
Acoustic Basics Sound Pressure Level (SPL)





Acoustic Basics The Human Hearing Window

- → The human hearing range describes the frequency range which can be heard by humans, which is commonly known between 20Hz and 20kHz. Nevertheless it depends on the individual and the age, where a decline of high frequencies is normal.
- The graphic shows the "standard" human hearing range referring to levels over frequencies with examples for vocals and music.





Acoustic Basics Weighting of Sound Pressure Level (SPL)

- To adapt measurements of sound to the human hearing window, different filters were developed. All these filters were optimized for a special range in the decibel scale -> A-Filter for small and e.g. D-Filter for high Sound pressure Levels.
- A-weighted SPL is adapted to human hearing window and is expressed in dB (A). It is the most common filter type if it comes to SPL measurements.
- C-Weighting is almost linear in the hearing range and is expressed in dB(C)

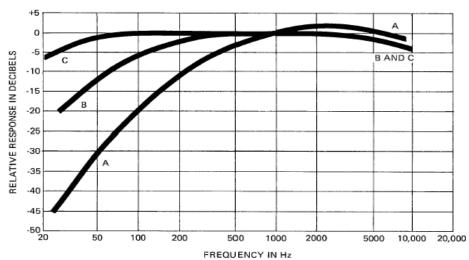


Figure 2-3. Frequency responses for SLM weighting characteristics



Acoustic Basics Frequency, SPL and Loudness

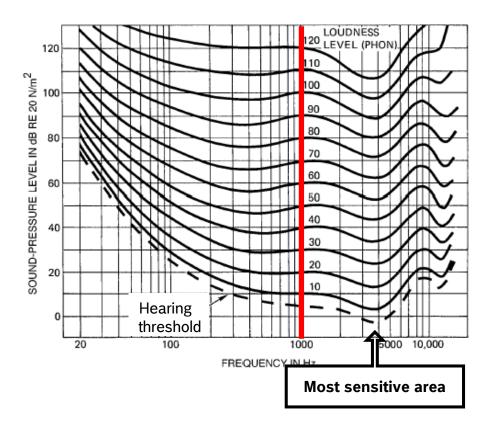
- → As we have seen in the "human hearing window" graphic the perception of loudness depends on frequency, i.e. that two different tones with the same level would not appear with the same loudness.
- → Loudness is a subjective value describing the strength of the human ears perception of a tone or sound. A unit of loudness, the **phon** has been established. The value in phon of a certain "noise" is equal to the value in decibels of a 1000Hz tone, which is judged by an audience to be or appear equally loud.
- → The loudness reference point is 1000Hz, where SPL and PHON curve overlap.



Acoustic Basics Frequency, SPL and Loudness

- Reference line in red at -1000Hz
- It is visible, that our ears -> sensible most are 2000 between and 5000Hz, i.e. that even very small changes in SPL are already noticeable, quite contrary to the low end.

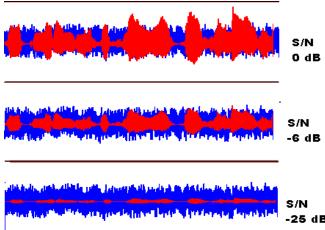
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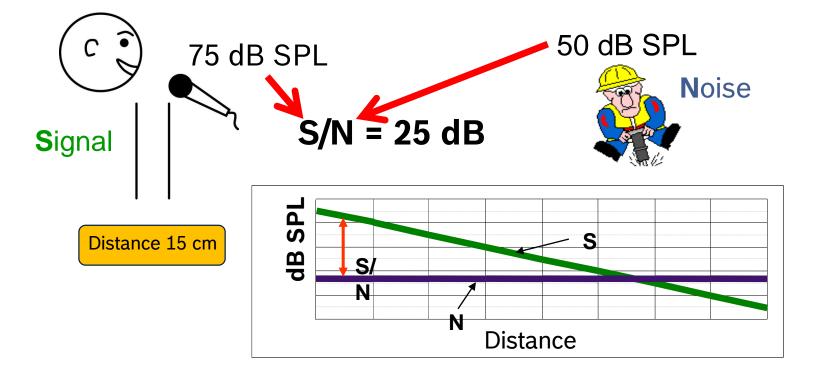
Acoustic Basics Masking Effects – Level Masking

- Level Masking describes the effect which happens if a background or environmental noise becomes louder or about the same level as the "useful" signal. The consequence of this is the loss of information of the useful signal.
- → A critical point is reached, if the noise level is about 6 dB beyond the signal.
- With a difference of 25 dB between signal and noise we loose the perception of the lower signal.





Acoustic Basics Masking Effects – Level Masking Signal to Noise Ratio - Example

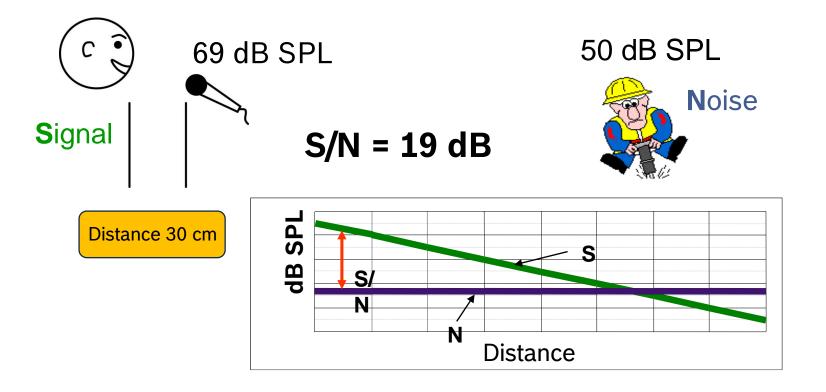


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Acoustic Basics Masking Effects – Level Masking Signal to Noise Ratio - Example

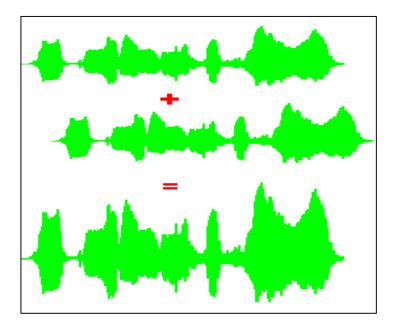






Acoustic Basics Masking Effects – Time Masking

The brain integrates multiple identical signals of different arrival times into the first arrival signal up to dT < 30ms, depending on the frequency spectrum of the signal and the individual, i.e. we recognize only one signal, even if it comes from different sources or reflections.</p>





Acoustic Basics Localization of Sound Sources

→ Our brain has the ability to identify sound sources around us, but how?

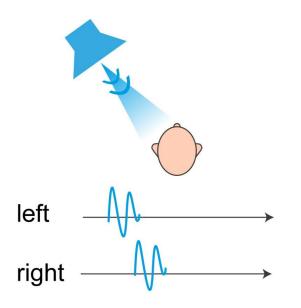
Our outer ear is formed somehow like a shell and facing forward, from this point of view it is logical that we can hear things better – and with different sounding - from the front than from the back. This different sound profile helps us to the vertical localization. It is not as accurate as the horizontal one, but it works even well for frequencies above ~4000Hz.

The second point, the horizontal detection refers to the position of our ears. Depending on the position of the sound source there is a small level and time difference between the received signals and that is taken from our brain to "triangulate", i.e. to localize the source.



Acoustic Basics Localization of Sound Sources

→ Very high detection precision of differential arrival times between left and right ear. We can localize sources up to a few (~2) degrees.

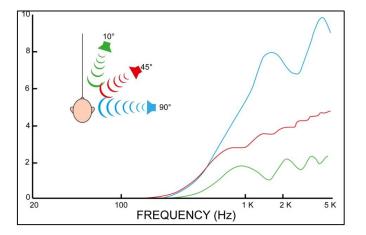


Path Difference -> Arrival Time

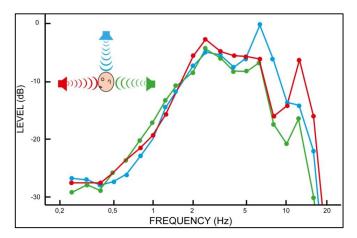


Acoustic Basics Localization of Sound Sources

Horizontal:
High resolution above 1 kHz



 Vertical: Approximate differenciation above 4 kHz





Acoustic Basics Localization of Sound Sources – Haas Effect

The Haas effect is a psycho acoustic effect related to the human hearing / localization of sound. In easy words, if two or more identical sounds from different sound sources arrive at our ears we will localize the sound from the first arriving signal.





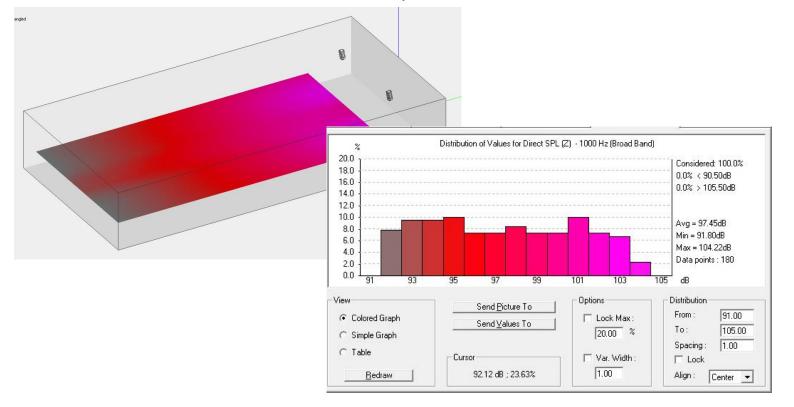


Acoustic Basics Localization of Sound Sources – Haas Effect

- → The most common application of this effect is to build so called "Delay Lines". A delay line or a distributed loudspeaker solution is used in a lot of different applications to reach a smooth coverage over a bigger area, under balcony, long rooms,...
- → The goal of a well done delay line is to extend main systems and make the sound more even over a dedicated area and and that is a very important factor, to keep the localization to the main loudspeaker system, the stage, screen,...in music or concert sound applications.
- Now we come back to our first example, which was improved, but still not perfect regarding the sound distribution.



→ Just to remember, the results of the first improvement:



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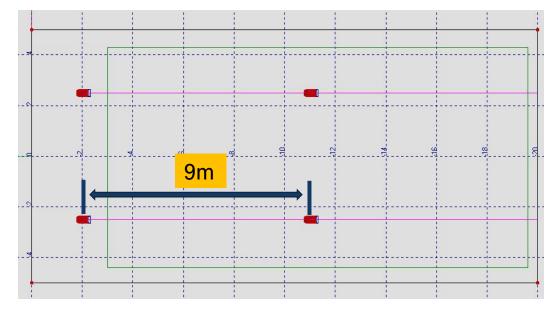


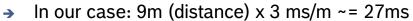
- → The next step, all general values the same, but now with two more speakers of the same type, placed at 11m, delayed by 27ms that's the "minimum" delay time in our case (will be explained later on).
- But why 27ms, how do we calculate it? Is it all which has to be taken into account when we are setting up a delay line?

No, there are still more things, but in the beginning I will only mention a part of it. A major condition in this case is that the used loudspeakers are all the same and mounted in the same axis. The basic calculation method is:

Delay time ~= Distance of loudspeakers [m] x ~3 [ms/m] -> speed of sound (344 m/s @ 20°C)







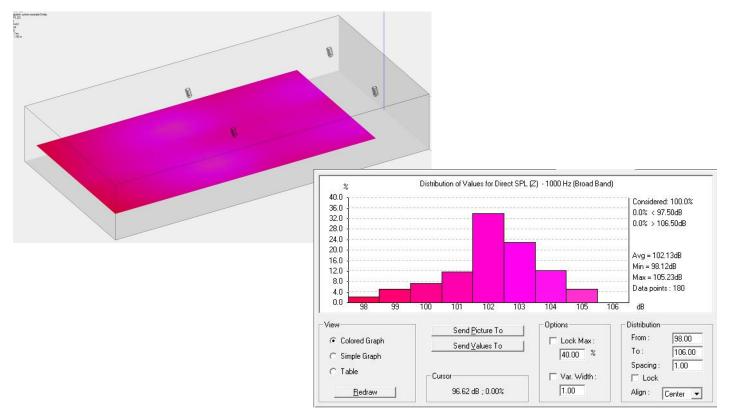


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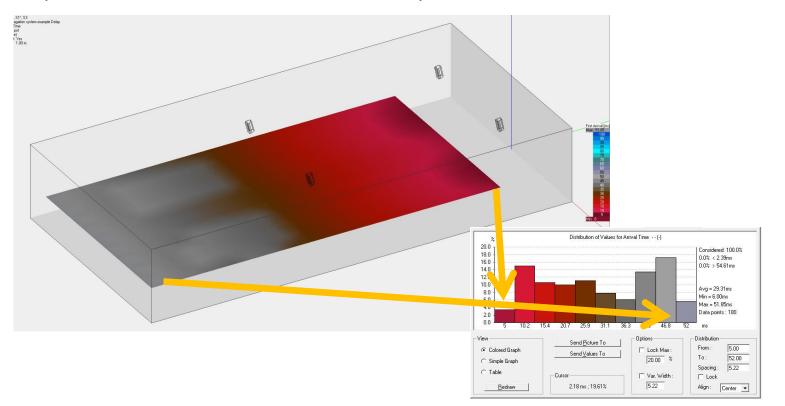


→ Result with the above mentioned values (please compare with previous result):



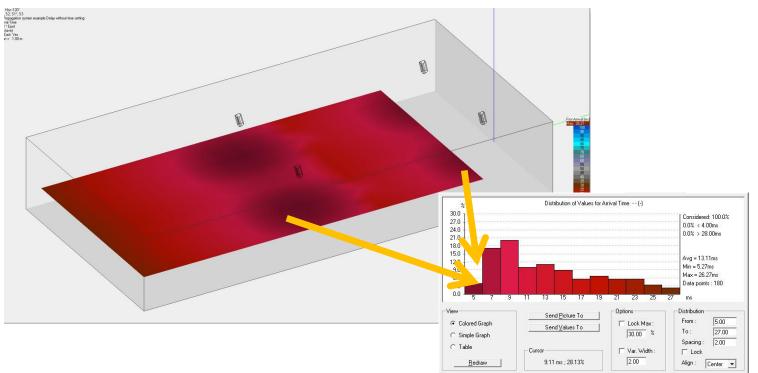


→ Following picture shows "arrival time" in milliseconds of the sound, i.e. this picture shows the "travel time" from the loudspeaker to the audience area.





If we would <u>not use</u> a delay time for the second row of loudspeakers it would look like that.





What is the result, if no time alignment is done especially in bigger rooms with delay times of ~>50ms?

Echoes will appear, it would be harder to understand speech and music also won't sound clear and transparent anymore.

Very important for delay applications:

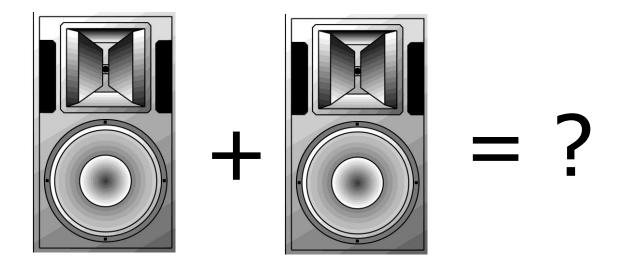
Loudspeakers have to face in the same direction, it is not possible to make a reasonable delay system "around the corner". A delay time alignment will fit anyhow exact to one certain point, but with all speakers facing in one direction you get bigger "acceptable" areas.



Acoustic Basics Interference Effects

 \rightarrow In general:

Interference effects are local and dependant on frequency, time and level ratios of different sound sources at receive point.

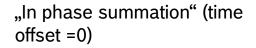




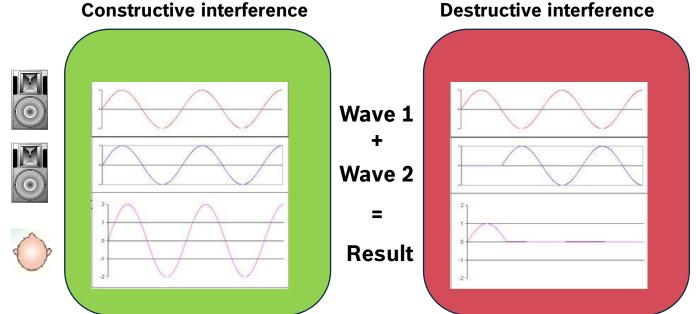
- When two or more different sources are present at the same time and in the same medium they will interact with each other and the result is a new wave or sound. The interaction of these waves is called interference effect.
- → Interference effects can be constructive and destructive.
- → Audible examples of this effect are
 - room modes or so called standing waves, which are depending on construction and size of a room, changeable only with modifying room acoustic.
 - Stearable loudspeaker arrays, which are created by loudspeaker manufacturer to create a so called beam forming, that means the sound energy is faced into a dedicated direction.



Interference depends on the amplitude of single waves and phase difference in the point of → overlapping



"Out of phase summation" (Phase offset 180°)



Destructive interference

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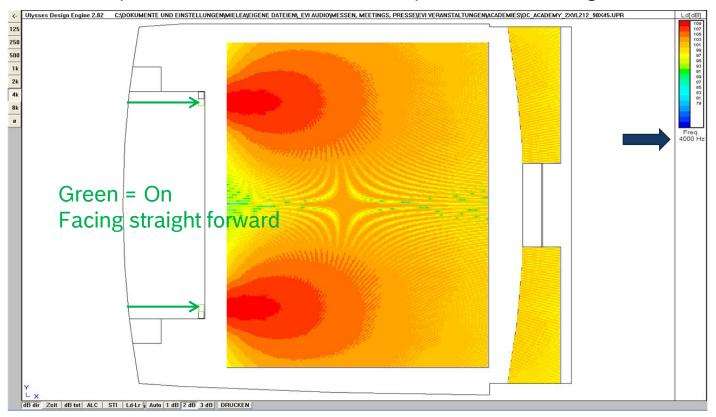


Acoustic Basics Interference - Examples

- On the following pages interference effects were made visible with an acoustic design software at a certain frequency (4000Hz) - as already mentioned, it looks different at every frequency.
- → This examples will help to avoid big misplacement of loudspeakers, because many people are still the opinion, that the easiest way to get more output in a certain direction is to add some more loudspeaker side by side to an existing system, all facing in the same direction. The result is shown on the following pages...



 \rightarrow Two loudspeakers active with a nominal horn pattern of 90 x 45 degrees.

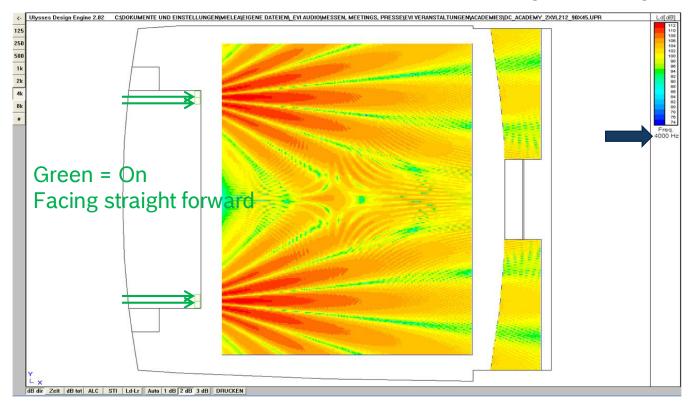


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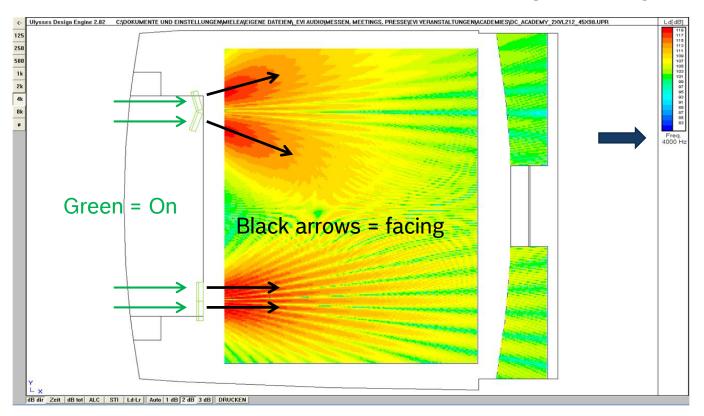


→ Four loudspeakers with a nominal horn pattern of 90 x 45 degrees, not angled.





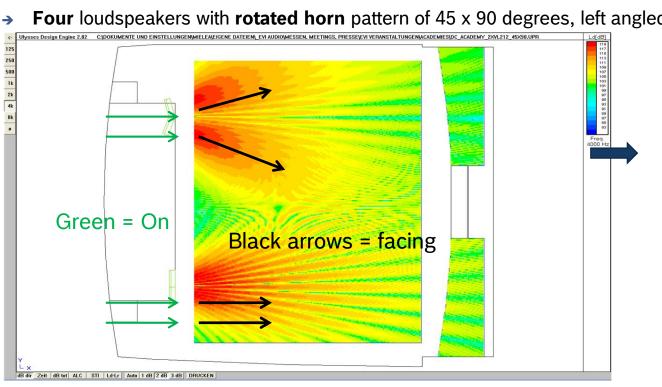
→ Four loudspeakers with rotated horn pattern of 45 x 90 degrees, left angled.







Interference Acoustic Basics



Four loudspeakers with rotated horn pattern of 45 x 90 degrees, left angled.

