Hands-On Training 8

# Measurement of Loudspeaker Directivity

## 1. Objective of the Hands-on Training

- Understanding the need for assessing loudspeaker directivity
- Introducing the basic theory of acoustic holography and field separation
- Applying Near Field Scanning techniques to loudspeakers
- Interpreting the results of Near Field Scanning
- Developing skills for performing practical measurements

## 2. Requirements

### 2.1. Previous Knowledge of the Participants

It is recommended to do the Klippel Training #2 "*Vibration and Radiation Behavior of Loudspeaker's Membrane"* before starting this training.

### 2.2. Minimum Requirements

The objectives of the hands-on training can be accomplished by using the results of the measurement provided in a Klippel database (.kdbx) dispensing with a complete setup of the KLIPPEL measurement hardware. The data may be viewed by downloading the measurement software dB-Lab from www.klippel.de/training and installing them on a Windows PC.

### 2.3. Optional Requirements

If the participants have access to a KLIPPEL Analyzer System, we recommend to perform some additional measurements on loudspeakers provided by instructor or by the participants. In order to perform these measurements, you will also need the following additional software and hardware components:

- Klippel Robotics
- KLIPPEL Analyzer (DA2 or KA3)
- Near Field Scanner
- Amplifier
- Microphone

## 3. The Training Process

- Read the following theory to refresh your knowledge required for the training.
- Watch the demo video to learn about the practical aspects of the measurement.
- Answer the preparatory questions to check your understanding.
- Follow the instructions to interpret the results in the database and answer the multiple-choice questions off-line.
- Check your knowledge by submitting your responses to the anonymous evaluation system at www.klippel.de/training.
- Receive an email containing a Certificate with high distinction, distinction or credit (depending on your performance).
- Perform some optional measurements on transducers if the hardware is available.

# 4. Introduction

Nowadays, there is an increasing demand for better quality of the reproduced sound in home entertainment, virtual reality, mobile, automotive, professional and other applications. This training focuses on the direct sound radiated by the audio device into the 3D space into different directions (angles) and distances (near and far field). This part gives a short introduction of the theoretical basis and the practical methods required to measure the sound pressure distribution as shown in the Figure 4.1.

At low frequencies the driver can be simulated by using a lumped parameter model. The electrical stimulus (voltage $u$ and current $i$) at the terminals generates the electro-dynamical force $F_{\text{coil}}$. This force drives the mechanical elements represented by moving mass $M_{\text{ms}}$, stiffness of the suspension system $K_{\text{ms}}$ and mechanical resistance $R_{\text{ms}}$ and generates the voice coil velocity $v_{\text{coil}}$. An electrical stimulus (voltage $u$) and a mechanical signal (velocity $v$) describe the transfer of an audio signal in a one dimensional signal path to the diaphragm (Training 1).

At higher frequencies, a model comprising a multitude of distributed parameters and state variables is required to describe the modal vibration of the diaphragm and suspension system (Training 2).

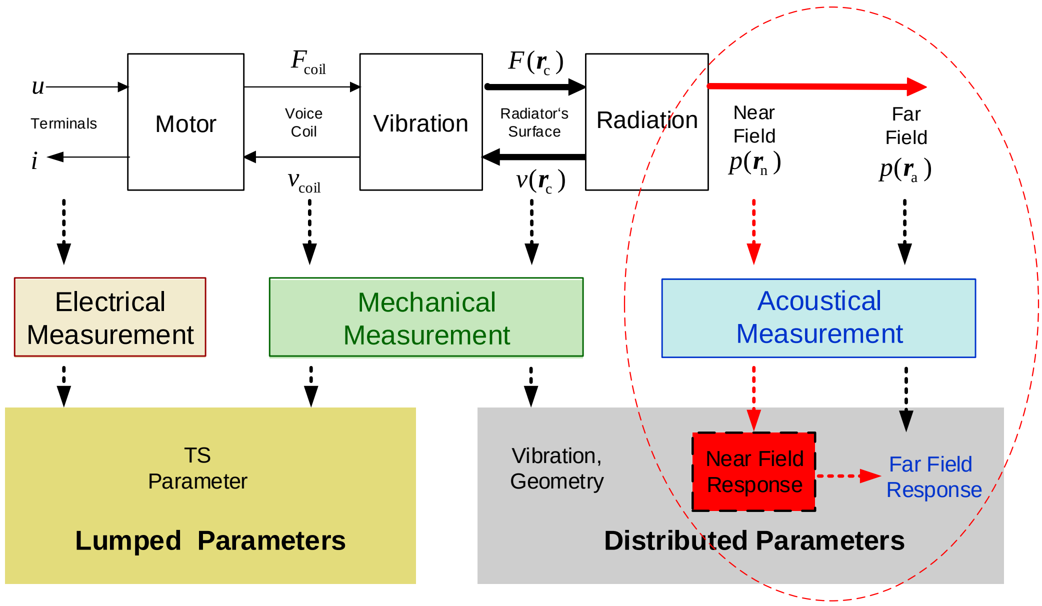



*Figure 4.1 Measurement Process of Loudspeakers [1]*

The velocity $v(r_c)$ in the normal direction at any point $r_c$ on the radiating surface generates the sound pressure values $p(r_n)$ and $p(r_a)$ in the near field and far field respectively. This tutorial focuses on the acoustical measurement in near field and far field, as shown in the Figure 4.1.

## 4.1. Direct Sound Radiated by the Loudspeaker

Sound is an oscillation of the medium (e.g. air) that generates variation in the pressure propagating as waves away from its source. The speed of sound $c$ depends on the static air pressure, density (or humidity and temperature) of the medium. The sound pressure, which is the difference between the instantaneous value of the total pressure and the static pressure, is much easier to measure than density or temperature fluctuations [3]. An important characteristic is the sound pressure level, denoted by $L_p$ or SPL and defined by:

$$L_p = 20\log_{10}\left(\frac{p}{p_0}\right)\text{dB} \tag{1}$$

with the reference pressure $p_0 = 20\mu\text{Pa}$.

The homogeneous wave equation *(2)* describes the relationship between temporal and local derivative of the sound pressure in a medium [4].

$$\Delta p = \frac{1}{c^2}\frac{\partial^2 p}{\partial t^2} \tag{2}$$

According to equation *(2)* the sound wave is propagated into different directions with the speed of sound $c$.
At low frequencies, a closed box loudspeaker system can be approximated by a point source (monopole) that generates spherical waves under free field condition that fulfil the one-dimensional wave equation in spherical coordinates:

$$\frac{\partial^2 (pr)}{\partial r^2} = \frac{1}{c^2}\frac{\partial^2 (pr)}{\partial t^2} \tag{3}$$

The product of distance $r$ and pressure $p$ in Equation *(3)* propagates with the speed of sound $c$ away from the sound source. Thus, the amplitude of the sound pressure decays inversely with distance $r$:

$$|p(r)| \propto \frac{1}{r} \tag{4}$$

### 4.2. Far-field Conditions

This so called $1/r$-law is also valid for a vented-box loudspeaker system or any other sound source with an arbitrary shape if the observation point is in the far field of the source where the distance $r$ is larger than a critical value $r_{\mathrm{far}}$. This critical value $r_{\mathrm{far}}$ depends on the wavelength $\lambda$ of a sinusoidal sound component and the largest geometrical dimension $l$ of the sound source:

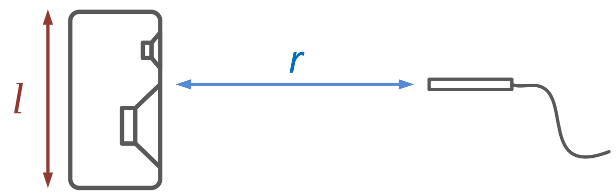
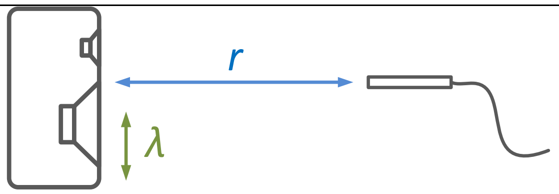
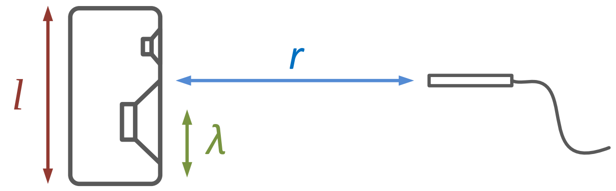
| | | |
|---|---|---|
|   | Condition 1: $\frac{r_{\mathrm{far}}}{l} \gg 1$ | (5) |
|   | Condition 2: $\frac{r_{\mathrm{far}}}{\lambda} \gg 1$ | (6) |
|   | Condition 3: $\frac{r_{\mathrm{far}}}{l} \gg \frac{l}{\lambda}$ | (7) |

*Figure 4.2 Far Field Conditions*

Condition 1 requires that the size of loudspeaker is small compared to the distance $r$. Condition 2 requires that the distance is larger than the wavelength which is a critical condition at low frequencies. Condition 3 is a critical condition for large loudspeaker systems used at the high frequencies such as line arrays, sound bars and panels. [8]

For a typical loudspeaker as used in home application the linear relationship ($1/r$-law) between sound pressure $p$ and distance $r$ begins at distance $r > 2$ m. At closer distance, the sound pressure $p$ rises at a much higher slope due to the near field properties generated by the size of the diaphragm and the number of transducers used in the system.

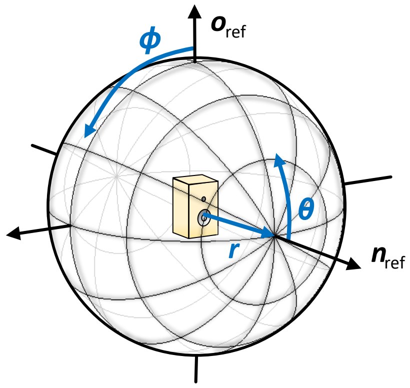



*Figure 4.3 Coordinate System*

If the complex sound pressure $\underline{P}(f, r_1, \theta, \phi)$ is given at distance $r_1$ in the direction represented by polar angle $\theta$ and azimuthal angle $\phi$, the sound pressure at a different distance $r_2$ can be calculated by

$$\underline{P}(f, r_2, \phi, \theta) = \underline{P}(f, r_1, \phi, \theta) \frac{r_1}{r_2} e^{-jk(r_2 - r_1)} \tag{8}$$

using the wave number $k = 2\pi f / c$.

In the far field the sound pressure $p$ and particle velocity $v$ are in phase and the specific sound impedance is a real value $Z_a \approx \rho c$. There is no reactive component [6] and the sound power $\Pi$ propagated by the sound wave can be calculated as the integral over the intensity $I = pv$ over the spherical surface $A$ as follows:

$$\Pi = \oint pv \, \mathrm{d}A \tag{9}$$

The Sound Power Level, denoted by $L_\Pi$, is given in decibel (dB) by:

$$L_\Pi = 10 \log_{10} \left( \frac{P}{P_{\mathrm{ref}}} W \right) \tag{10}$$

using the reference value $P_{\mathrm{ref}} = 10^{-12} W$.

The sound radiation behavior is defined under free field conditions where the influence of the sound reflections, diffraction and standing waves caused by boundaries (e.g. room walls) does not exist.

# 5. Conventional Measurement Technique

## 5.1. Measurement under anechoic condition

Loudspeakers and microphones are measured in anechoic chambers, where all boundaries are covered with an absorbing material (full space) to reduce room reflections and to approximately provide free field condition (above a cut-off frequency), as illustrated in Figure 5.1.

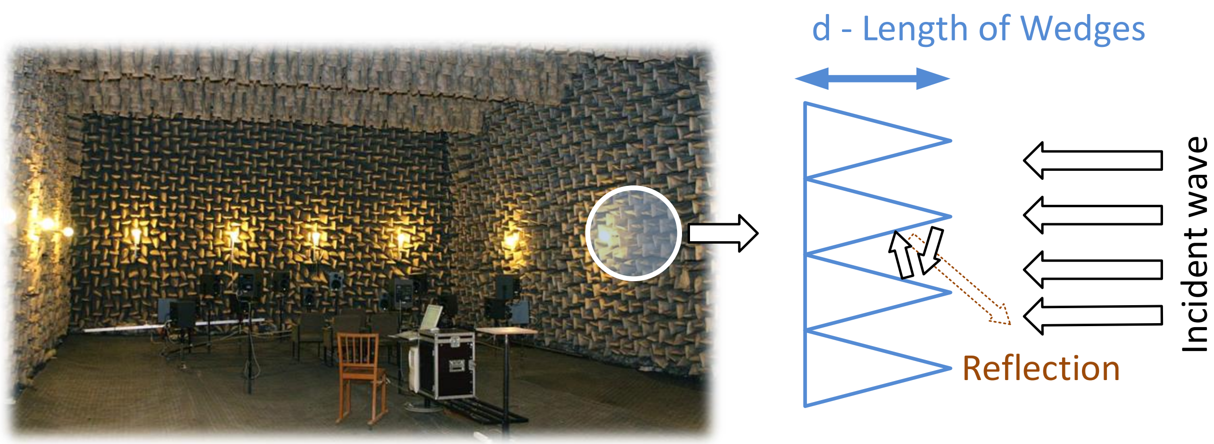



*Figure 5.1 Anechoic Room (Technical University Dresden)*

An interesting alternative is the half space anechoic room, where the floor is intentionally made of hard material to generate a reflective surface and to simplify the movement and placement of loudspeakers, turntables and microphones. An air condition system controls the temperature and humidity of air within the room.

The right-hand side of Figure 5.1 shows a sectional view of absorber using rock wool or other highly porous material placed with pyramidal shape on the walls. This configuration provides sufficient absorption at high frequencies where the wavelength is much smaller than the length $d$ of the pyramidal elements. At low frequencies for $d < \lambda/4$, the reflections from the room walls superimpose with the direct sound and generate errors in the measured transfer function $H(f)$.

### 5.1.1. Measurement Setup

The sound pressure $p(t, r, \phi, \theta))$ is measured under free field condition in the far field at defined distance $r > r_{\text{far}}$, polar angle $\theta$ and azimuthal angle $\phi$ in accordance with Figure 4.3 while exciting the loudspeaker with a broadband stimulus $u(t)$ generated by the analyzer and a power amplifier in the frequency band of interest.

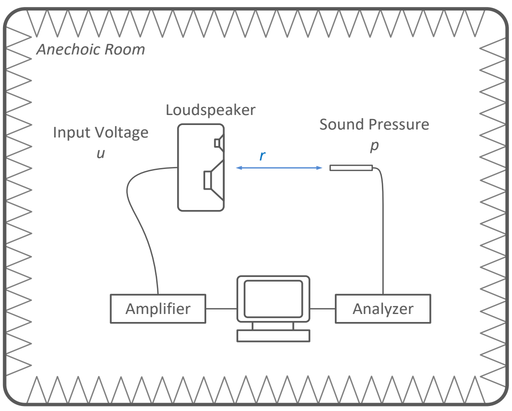



*Figure 5.2 On Axis Measurement*

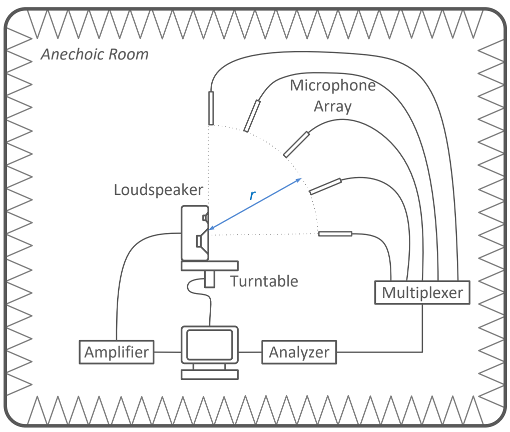



*Figure 5.3 Directivity at Multiple Points*

Figure 5.2 shows the measurement setup of assessing the radiated sound pressure $p(t, r, \theta = 0^\circ)$ at the reference axis (also called "On-Axis response") to describe the sound pressure output in the direction that is most relevant for the particular application. The loudspeaker under test and the microphone are placed at a fixed position. Figure 5.3 depicts the multi-point measurement using a turntable or other robotics for turning the loudspeaker around polar angle $\theta$ or azimuthal angle $\phi$ while using a microphone array for measuring the sound pressure versus the orthogonal angle. The spacing of the microphones and the increments of the turntable movements determine the number of measurement points and the angular resolution of the directivity data. The measurement over the complete sphere ($4\pi$) requires a compromise between the angular resolution and the number of measurement points:

| **Angular resolution over the complete sphere ($4\pi$)** | **Number of measurement points** |
|---|---|
| 5 degree | 2592 |
| 2 degree | 16200 |
| 1 degree | 64800 |

Most conventional measurements provide no higher angular resolution than 2 degree and assume symmetry in one or two-planes to reduce the measurement time.

### 5.1.2. Directional Transfer Function

This chapter gives an overview on the post-processing to calculate the linear transfer function from the loudspeaker input voltage $u(t)$ to the sound pressure output $p(t, r, \phi, \theta)$ at the microphone position.

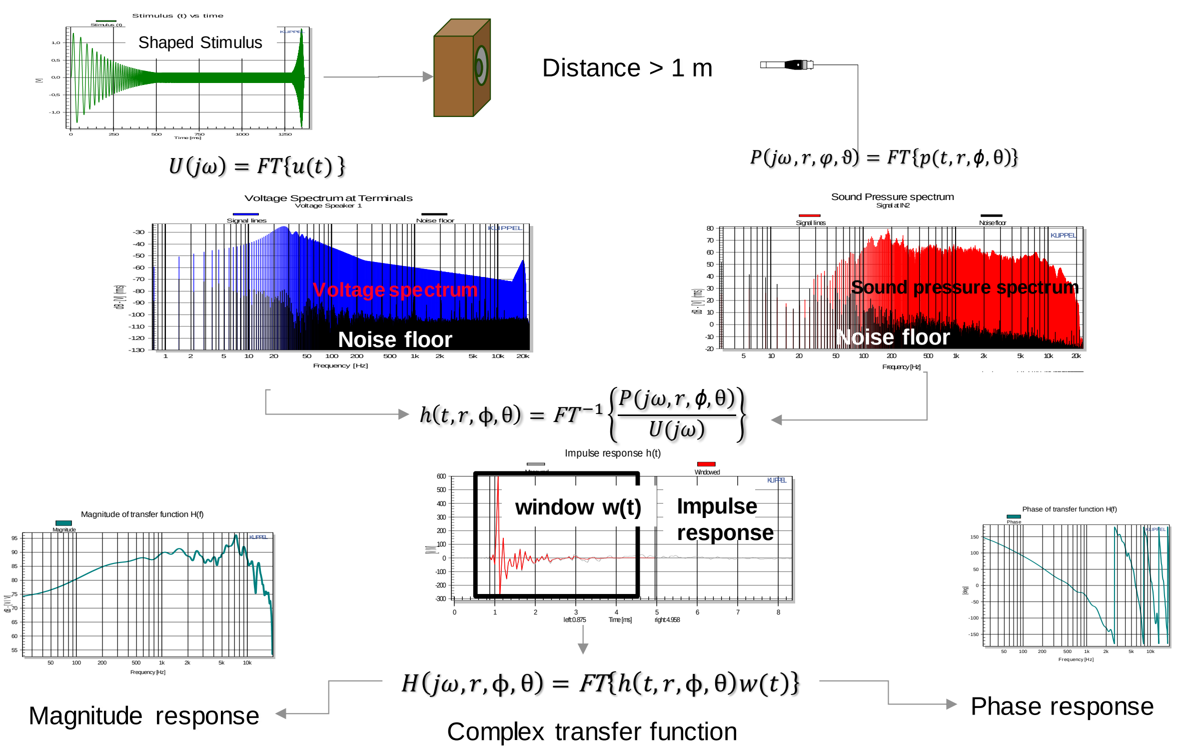



*Figure 5.4 Overview on the Calculation of the Complex Transfer Function [10]*

The first step is the calculation of the complex spectrum $U(j\omega)$of the input signal $u(t)$ generated by the analyzer and the sound pressure spectrum $P(j\omega, r, \phi, \theta)$measured by the microphone. Sufficient signal to noise ratio (SNR) is required to measure the transfer function

accuracy. Figure 5.4 shows an example of insufficient SNR in the sound pressure signal for frequencies below 30 Hz. The SNR can be improved by applying higher amplitude shaping to the stimulus at low frequencies. Alternatively, averaging of the measured sound pressure signal can be applied if a deterministic stimulus such as a sinusoidal chirp is used.
The second step is the calculation of the impulse response $h(t, r, \phi, \theta)$ defined as

$$h(t, r\ \phi, \theta) = FT^{-1}\left\{\frac{P(\mathrm{j}\omega, r, \phi, \theta)}{U(\mathrm{j}\omega)}\right\} \tag{11}$$

based on the inverse Fourier transform applied to the ratio of complex output and input spectra. A window function $w(t)$ may be applied to the impulse response $h(t, r, \varphi, \theta)$ in order to separate the direct sound from room reflections and to suppress measurement noise. The Fourier transform applied to the windowed impulse response gives the directional transfer function:

$$H(\mathrm{j}\omega, r, \phi, \theta) = FT\{h(t, r, \phi, \theta)w(t)\} \tag{12}$$

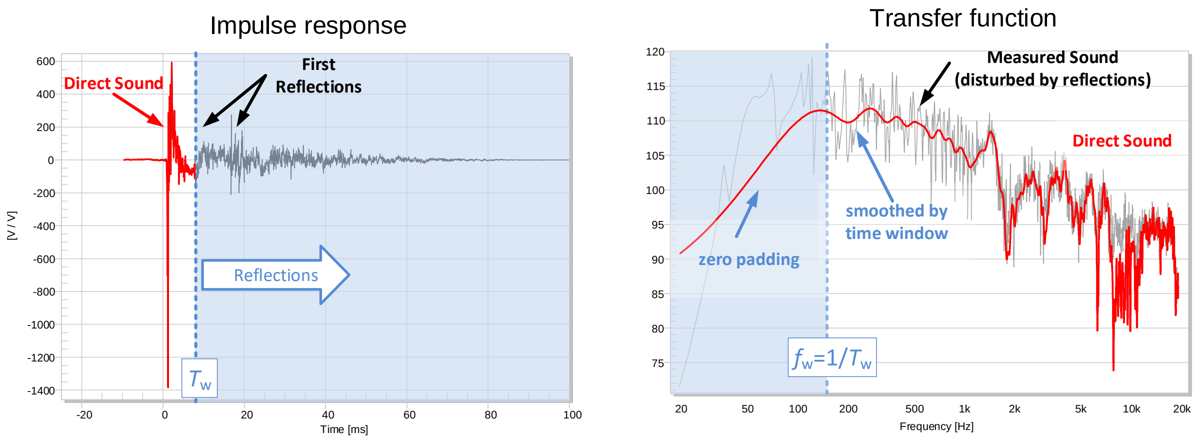



*Figure 5.5 Windowing of the Impulse Response*

Figure 5.5 illustrates time windowing of the impulse response $h(t, r, \phi, \theta)$ to suppress room reflections while preserving the direct sound at higher frequencies $f > f_{\mathrm{w}}$. This critical cut-frequency $f_{\mathrm{w}} = 1/T_{\mathrm{w}}$ depends on the effective length $T_{\mathrm{w}}$ of the time window $w(t)$. The window in Figure 5.5 has a flat top region generating an effective window length $T_{\mathrm{w}} = 7$ ms and reduces the true spectral resolution. At low frequencies below $f_{\mathrm{w}} = 150$ Hz the measurement points in the frequency response are generated by zero padding and are not real measurement data.

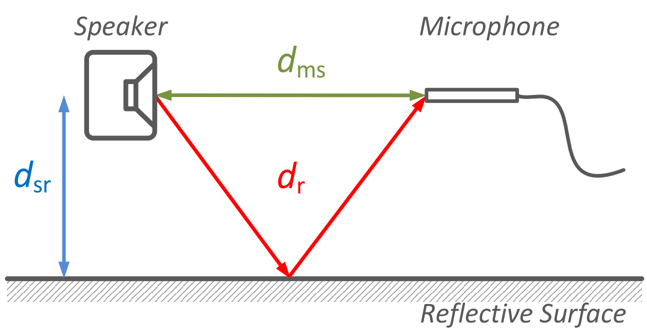



*Figure 5.6 Sound Reflected on the nearest Boundary*

Distance of first reflection:

$$d_{\mathrm{r}} = 2\sqrt{\left(\frac{1}{2}d_{\mathrm{ms}}\right)^2 + (d_{\mathrm{sr}})^2} \tag{13}$$

$d_{ms}$ distance microphone to speaker
$d_{sr}$ distance speaker to room wall

The maximum length of window $T_{\mathrm{w}}$ should be shorter than reflection free time $T_{\max}$ defined as:

$$T_w < T_{\max} = \frac{d_{\mathrm{r}} - d_{\mathrm{ms}}}{c} \tag{14}$$

## 5.2. Directional Far-Field Characteristics

The term *directivity* has been introduced to describe the distribution of the direct sound on a sphere depending on the polar angle $\theta$ and azimuthal angle $\phi$ at a particular distance $r$ in far field.

### 5.2.1. Directional Factor and Gain

The ratio between the complex sound pressure value $\underline{P}(f,r,\theta,\phi)$ at any azimuthal angle $\phi$ and polar angle $\theta$ to the sound pressure value $\underline{P}(f,r,\theta_r,\phi_r)$ defined by reference angles $\phi_r$ and $\theta_r$measured in the far-field of the DUT gives the directional factor

$$\underline{\Gamma}(f,\theta,\phi) = \frac{\underline{P}(f,r,\theta,\phi)}{\underline{P}(f,r,\theta_r,\phi_r)} = \frac{\underline{H}(f,r,\theta,\phi)}{\underline{H}(f,r,\theta_r,\phi_r)} \tag{15}$$

And the directional gain in dB

$$D(f,\theta,\phi) = 20\log\left|\underline{\Gamma}(f,\theta,\phi)\right| \tag{16}$$

In most applications the reference angles $\phi_r = 0°$ and $\theta_r = 0°$ describe the on-axis response measured on the reference axis perpendicular to the diaphragm of the loudspeaker. [17]

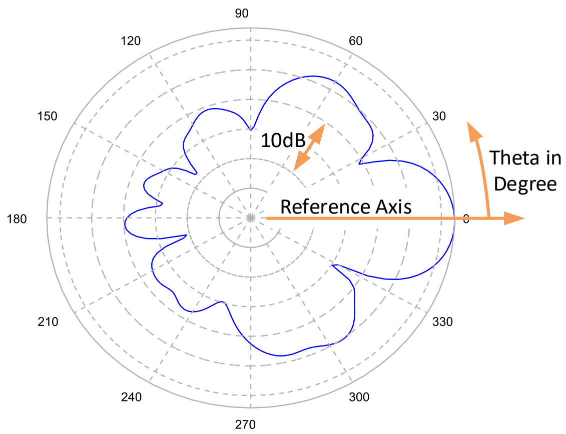



*Figure 5.7 Polar plot*

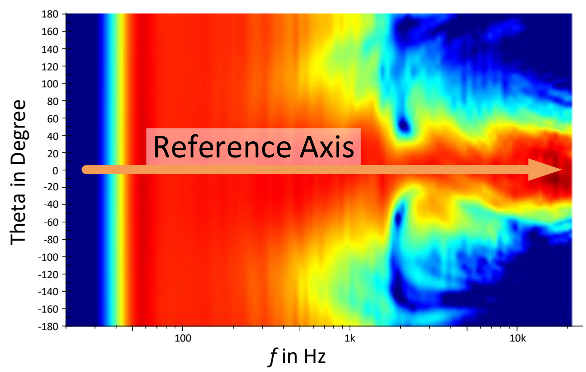



*Figure 5.8 Contour plot*

The directional gain is useful to visualize the beaming of the loudspeaker and the decay of the sound pressure level versus angle $\theta$ as a polar, contour or balloon plot.

### 5.2.2. Directivity Factor and Index

By integrating the squared directional factor over the unit sphere the directivity factor $Q(f)$ can be calculated as

$$Q(f) = \frac{4\pi}{\int_\Omega |\Gamma(f,\theta,\phi)|^2 \, d\Omega} \tag{17}$$

giving the directivity index in dB:

$$DI(f) = 10\log_{10} Q(f) \tag{18}$$

A loudspeaker operated in a sealed box has a directivity index $DI \approx 0$ dB at low frequencies, which corresponds to an omnidirectional radiation pattern generating the same sound pressure level at the same distance $r$ for any polar angle $\theta$ and azimuthal angle $\phi$. At higher frequencies, the loudspeaker generates a lower sound pressure on the rear side than on the front side which increases the directivity index.

### 5.3. Limitations of the Far Field Measurements

The precision of the far field measurement setup depends on the following conditions:

1. Defined microphone placement
2. Defined positioning of the DUT (while turning heavy devices)
3. Sound reflection from turntable
4. Early reflections on room walls
5. Temporal and local variation of air properties (temperature, humidity) versus the propagation path affect the phase response (2° Kelvin deviation generates 90° phase error in 5 m distance at 5 kHz)
6. Air convection (wind)
7. Influence of ambient noise
8. Sufficient angular resolution

An anechoic room usually provides good wind protection and sufficient suppression of ambient noise. However, room modes can generate 1-6 dB errors at low frequencies and the temperature can cause significant phase errors at high frequencies. Furthermore, the conventional measurement technique cannot check those requirements and detect errors automatically. Since there is usually no redundancy in the measured data set, it is difficult to identify faulty measurements and exclude those points from the post-processing.

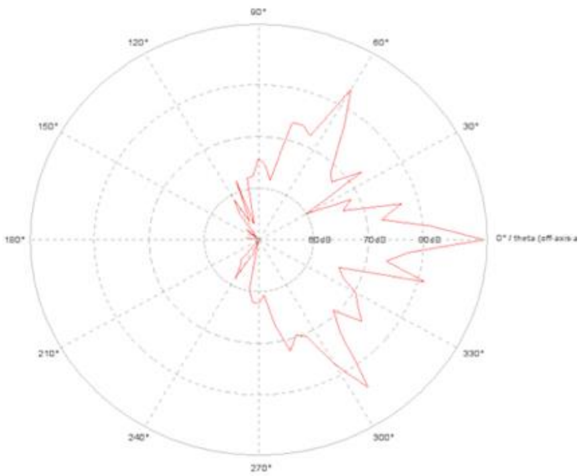


*Figure 5.9 5° Angular Resolution*

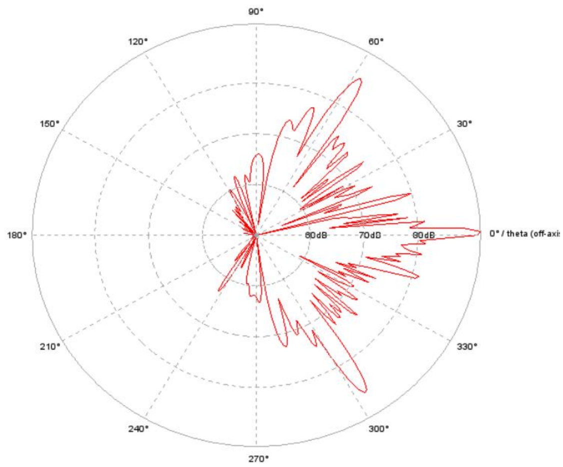



*Figure 5.10 1° Angular Resolution*

Interpolation of the phase and amplitude between the measurement points will introduce a significant error if the angular resolution in the measurements cannot cope with directional complexity of the loudspeaker. For example, the polar plot shown in Figure 5.9 measured with 5° angular resolution cannot represent the narrow lobes coming up in the measurement with 1° resolution as shown in Figure 5.10. Some of the peaks and dips in the polar plot are completely missed.

# 6. Near Field Measurement

## 6.1. Motivation

There are multiple reasons why measurements in the near field are relevant and beneficial although the simple extrapolation based on the $1/r$-law is not applicable in general.

- The near field properties of the DUT determine the sound pressure at the listener's ear in studio monitors, laptops, tablets, smart-phones and other portable audio devices.
- Sound bars, professional line arrays and other large loudspeakers have to be measured in the near field, because the far field distance $r_{\text{far}}$ cannot be realized in available anechoic rooms.
- The measurement in the near field provides a good signal-to-noise ratio, which reduces the influence of ambient noise, the generation of nonlinear distortion and dispenses from excessive averaging (time saving).
- The amplitude of direct sound is much greater than room reflections. The direct sound arrives much earlier at the microphone than the first reflections from room boundaries. This provides good conditions for simulating free field conditions by gating techniques or windowing of the impulse response.
- There is only a minimal influence from air properties.

## 6.2. Holographic Near Field Measurement

The holographic near field measurement exploits the general three-dimensional wave equation *(2)* expressed in spherical coordinates as given in [15]. The solution of this equation can be described by a spherical wave expansion

$$\begin{aligned} P(j\omega,r,\theta,\phi) &= \sum_{n=0}^{N}\sum_{m=-n}^{n} c_{n,m}^{out}(j\omega)h_n^{(2)}(kr)Y_n^m(\theta,\phi) \\ &+ \sum_{n=0}^{N}\sum_{m=-n}^{n} c_{n,m}^{in}(j\omega)h_n^{(1)}(kr)Y_n^m(\theta,\phi) \end{aligned} \tag{19}$$

which is valid in the region where no boundaries and sound sources are found as shown in Figure 6.1.

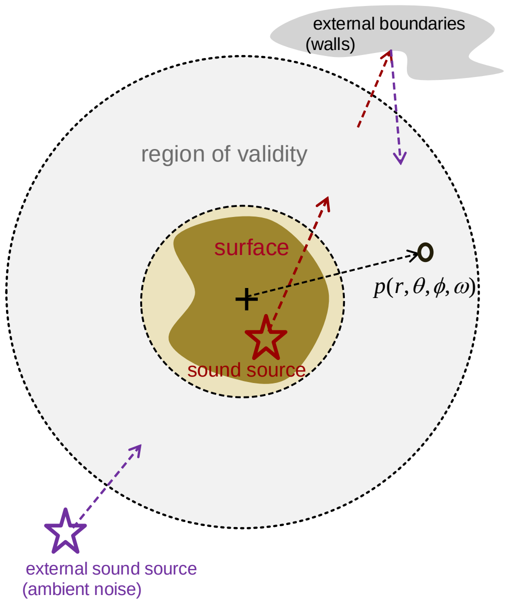



*Figure 6.1 Spherical wave expansion*

The first expansion term containing the Hankel function of the second kind $h_n^{(2)}(kr)$ represents the outgoing waves. The second term containing the Hankel function of the first kind $h_n^{(1)}(kr)$

represents the incoming waves. The Hankel functions describe the dependency of the radial coordinate $r$. For large distances $r > r_{\mathrm{far}}$ the absolute value of the Hankel functions of any order n decay inversely with $r$ corresponding with the $1/r$-law. The spherical harmonics $Y_n^m(\theta, \phi)$ found on both terms describe the dependency on azimuthal angle $\phi$ and polar angle $\theta$.

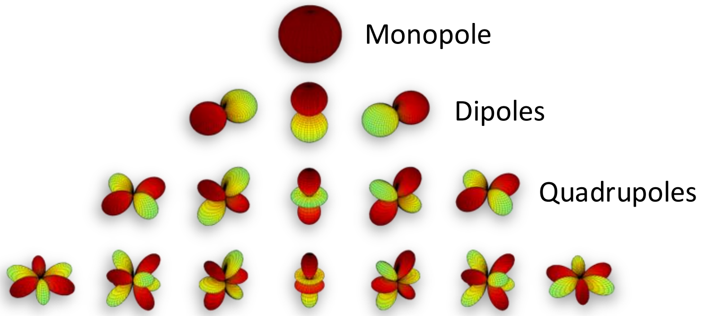



*Figure 6.2 Spherical harmonics (real parts)*

The spherical harmonic $Y_n^m(\theta, \phi) = 1/\sqrt{4\pi}$ represents a monopole with an omnidirectional radiation behavior as shown in Figure 6.2. The first order spherical harmonics $Y_1^{-1}(\theta, \phi)$, $Y_1^0(\theta, \phi)$ and $Y_1^1(\theta, \phi)$, can be used to represent angular directivity of a dipole with any orientation in the spherical coordinate system. The spherical harmonics are orthonormal and provide a complete set of basic functions for the spherical wave expansion with the Hankel function of order $n$.

The coefficients $c_{n,m}^{out}(\omega)$ and $c_{n,m}^{in}(\omega)$ weight each basic function and may be interpreted as the spherical wave spectrum analogically to the coefficients of the Fourier series.

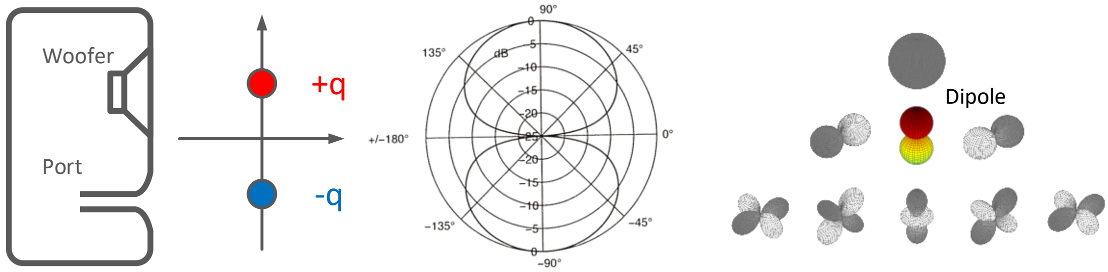



*Figure 6.3 Spherical Wave Expansion of a Vented-Box System below port resonance*

For example, a vented-box system as shown in Figure 6.3 behaves below the port resonance like a dipole where the port and the diaphragm generate the same amplitude but opposite sign of the volume velocity $q$. The polar plot of the directional gain would generate two lobes in opposite direction separated by a null that is perpendicular to the line between the two monopoles. If the port and woofer are placed in the z-axis with polar angle $\theta = 0°$, the spherical harmonic $Y_1^0(\theta, \phi)$ corresponds to the directivity of the vented box system and a single coefficient $c_{1,0}^{out}(\omega)$ represents the radiation behavior.

This example shows that the maximum order $N$ of the wave expansion and the number of coefficients depend on

- the directional complexity of the loudspeaker under test, which rises usually with frequency
- the location of the expansion point (origin of the internal coordinate system) in the acoustical center
- the orientation of the loudspeaker in the spherical coordinate system.

## 6.3. Directional Transfer Function

The directional transfer function of the loudspeaker between input voltage $u(t)$ and sound pressure output $p(t)$ at the measurement point $\boldsymbol{r}$ under free field condition, can be expressed by considering only the direct sound $p_{\mathrm{Dir}}(t)$ in Equation (19):

$$H(\mathrm{j}\omega, r, \theta, \phi) = \frac{Pp_{\mathrm{Dir}}(\mathrm{j}\omega, r, \theta, \phi)}{U(\mathrm{j}\omega)} = \sum_{n=0}^{N}\sum_{m=-n}^{n} C_{mn}(\mathrm{j}\omega)\, h_n^{(2)}(kr) Y_n^m(\theta,\phi) \qquad (20)$$

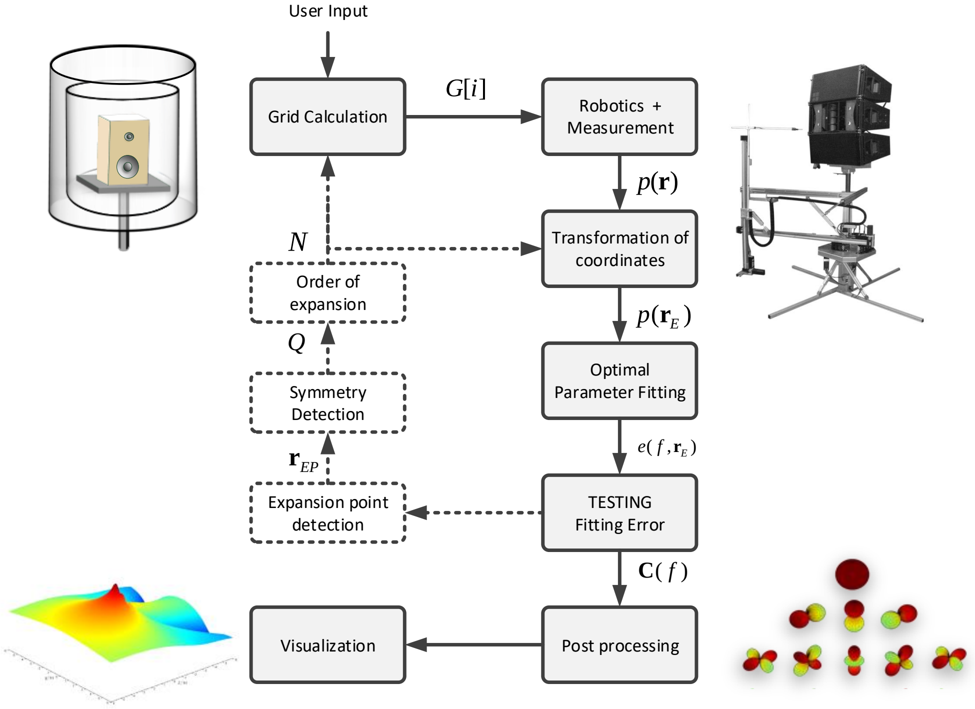



*Figure 6.4 Practical measurement procedure*

## 6.4. Practical Measurement Procedure

Figure 6.4 illustrates the measurement procedure which comprises the following steps:

1. The device under test (loudspeaker) is placed on the robotics as shown in Figure 6.5.
2. The microphone is moved manually to the upper corner of the device under test and to the tweeter to measure important geometrical information which are used for calculating the initial scanning grid $G[i=0]$. The scanning grid contains measurement points distributed on an inner and an outer surface which is a requirement to separate the incoming and outgoing sound waves (field separation) at lower frequencies.
3. The measurement determines the transfer function between the input signal $u(t)$ and the sound pressure $p(t,\boldsymbol{r})$ at any point $\boldsymbol{r}$ on the scanning grid $G[i]$.
4. The holographic processing starts with the transformation of the data $p(t,\boldsymbol{r})$ in the measured coordinates into an internal coordinates system $p(t,\boldsymbol{r}_{\mathrm{E}})$ that has its origin close to the acoustical center for high frequencies (tweeter position) and the coordinates are aligned to the orientation of the speaker.
5. The coefficients $c_{n,m}^{out}(\omega)$ and $c_{n,m}^{in}(\omega)$ in Equation *(19)* are estimated by minimizing the mean squared error $e(f)$ between the modelled and measured sound pressure on the scanning grid.
6. The final coefficients $C_{mn}(\mathrm{j}\omega)$ in Equation *(20)* representing the direct sound radiated by the loudspeaker are calculated based on the field separation technique.

7. In an optional iteration additional measurement points can be acquired based on the position of the expansion point $r_{EP}$ to improve the angular resolution and the maximum order of the expansion $N$.
8. The sound pressure level or the directional transfer function $H(\mathrm{j}\omega, r, \theta, \phi)$ can be calculated at any measurement point outside the outer scanning surface. Post-processing provides additional characteristics such as directional gain, directivity index and sound power.

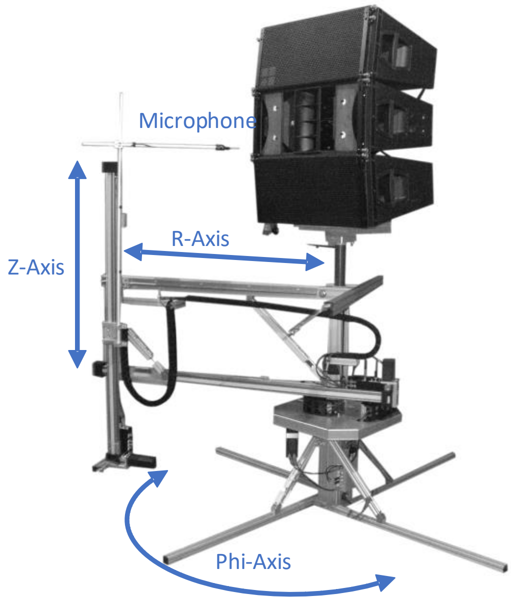



*Figure 6.5 Near Field Scanner*

## 6.5. Checking the Accuracy of the Measurement

The accuracy of the directional transfer function can be easily checked by assessing the fitting error

$$e(f) = \frac{\sum |H(f,\boldsymbol{r}) - H_{\mathrm{meas}}(f,\boldsymbol{r})|^2}{\sum |H_{\mathrm{meas}}(f,\boldsymbol{r})|^2} \cdot 100\% \tag{21}$$

that compares the modeled response $H(f,\boldsymbol{r})$ and the measured response $H_{\mathrm{meas}}(f,\boldsymbol{r})$ at all measurement points $\boldsymbol{r}$ on the scanning grid $G[i]$.

The calculation of the error $e(f)$ requires that there is some redundancy in the measured data set, that means the number $M$ of measurement points is larger than the number of unknown coefficients in the wave expansion $J$.

$$M \geq J = (N+1)^2 \tag{22}$$

The error may be caused by the following factors:

- Maximum order $N$ of the wave expansion is too low to describe the directional complexity of the loudspeaker
- Ambient noise or a poor signal-to-noise ratio (SNR) of the microphone corrupt the measured data
- The position of the loudspeaker or the microphone have been changed during the scanning process

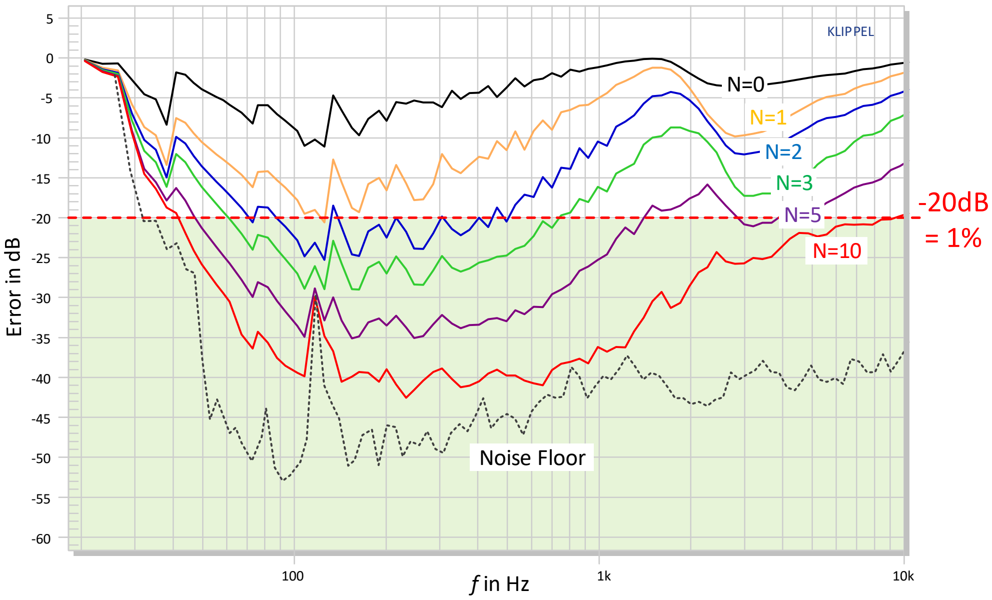



*Figure 6.6 Fitting Error as a function of the maximum order of the expansion N*

For example, Figure 6.6 shows the fitting error versus frequency for the wave expansion with varying maximum order $N$. The closed-box loudspeaker system cannot be described with sufficient accuracy by using a single monopole ($N = 0$). Considering a monopole together with dipoles ($N = 1$) reduces the error between 50 Hz and 500 Hz to -10 dB. Increasing the maximum order $N = 2$ and considering quadrupoles the error can be reduced to -20 dB. This means that only 1 % of the radiated power cannot be explained by the model. Above 500 Hz the loudspeaker starts beaming and additional side lobes increase the directional complexity that require a maximum order of $N > 10$ to reduce the fitting error below -20 dB at 10 kHz.
Increasing the order $N$ will not reduce the fitting error below 30 Hz because the error is caused by ambient noise. The grey dashed line shows a relative sound pressure level measured with no excitation signal on the loudspeaker under test.

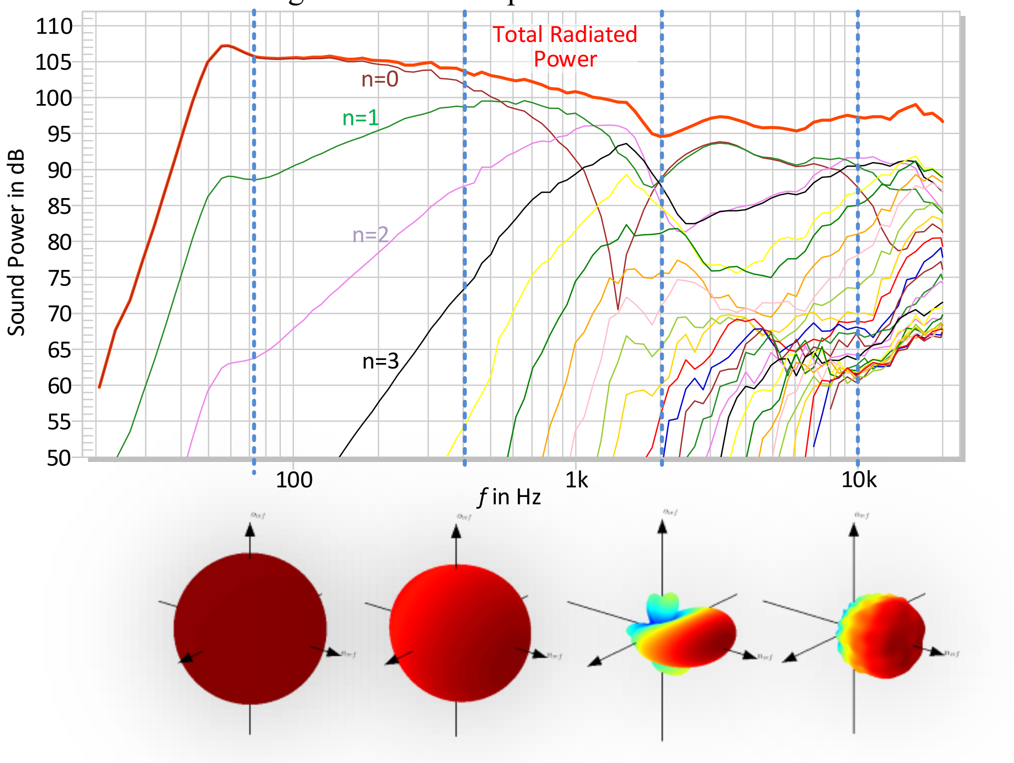



*Figure 6.7 Sound Power contributed by spherical waves*

Figure 6.7 shows the contribution of the spherical waves of order $n$ to the total sound power radiated from the loudspeaker under test into the far field. At 70 Hz the monopole ($n = 0$) dominates the total power which corresponds to the closed-box design and the fact that the wavelength is much larger than the geometrical dimension of the diaphragm and the box. However, the dipole ($n = 1$) provides ~20 dB less power, which can be explained by the fact that the acoustical center of a sealed box system is about 5 cm in front of the diaphragm of the woofer and not identical with the origin of the wave expansion which is placed close to the tweeter. The quadrupoles ($n = 2$) are ~40 dB lower but rising rapidly with frequency and exceed the monopole at ~1 kHz. While the higher-order waves ($n > 5$) provide a small contribution below 5 kHz they come more and more important at higher frequencies.

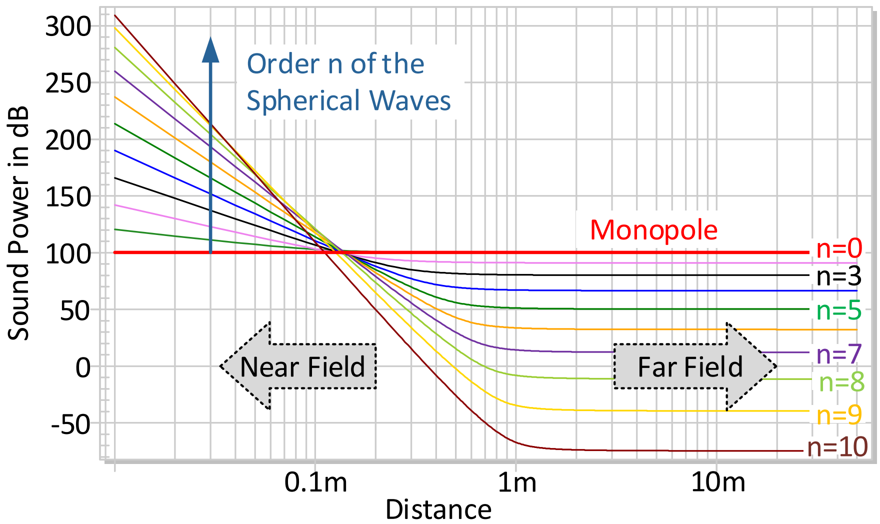



*Figure 6.8 Apparent Sound Power versus Distance r*

Figure 6.8 shows the apparent sound power of the $n^{th}$-order waves versus the distance $r$ at 1 kHz. The apparent power describes also the near field ($r < 1$ m) where the sound pressure and velocity are not in phase. There the apparent power is inversely related to the distance $r$ corresponding to the $n^{th}$-order Hankel functions $h_n^{(2)}$(kr) and $h_n^{(1)}$(kr) in Equation *(19)*. At high order $n$ the apparent power rises with high slope to significant values which are much higher than real power radiated into the far field.

The estimated values between the surface of the loudspeaker enclosure and the scanning area (0.2 m < $r$ < 0.4 m) are less accurate than the extrapolated values outside the scanning surface. The values that are predicted inside the loudspeaker system ($r$ < 0.2 m) have to be considered as virtual values generated by wave expansion neglecting any boundary and have no practical value.

The apparent power becomes constant for a distance $r > r_{far}$ which is at 1 kHz approximately 1 m. Here, the apparent power becomes identical with real power propagated into the far field because pressure and particle velocity are in phase. At this frequency the wave expansion truncated after $N = 5$ can sufficiently describe the radiation of the loudspeaker while the other higher-order waves ($n > 5$) propagate negligible power.

## 6.6. Field Separation

Windowing of the impulse response according to Equation *(12)* is a simple and reliable method to separate the direct sound from the room reflections at high frequencies. The effective window length $T_w$, limited by reflection free time $T_{max}$ as defined by Equation *(14)*, provides sufficient resolution to capture the direct sound at high frequencies and is usually to short measure the direct sound with sufficient resolution at low frequencies.

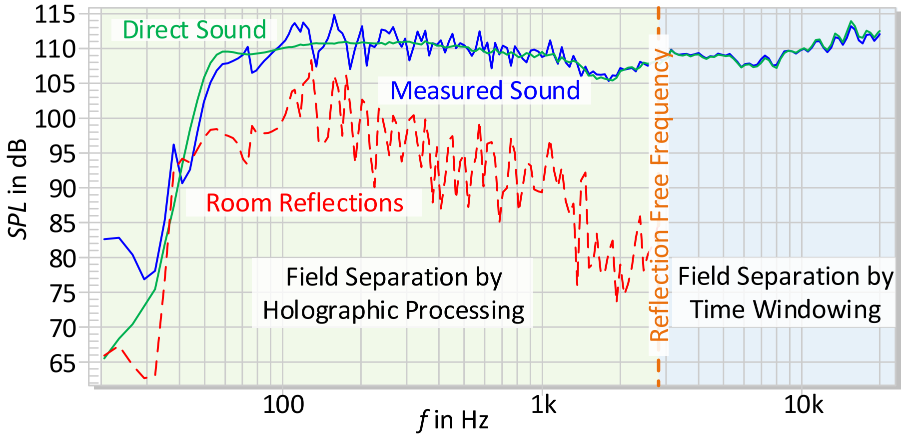



*Figure 6.9 Results of the sound separation*

The wave expansion according to Equation *(19)* provides an alternative to separate direct sound from the room reflections at low frequencies. This method requires a low order ($N < 10$) of spherical harmonics due to the large wavelength in this frequency band ($f < 3$ kHz). Figure 6.9 shows the amplitude response of a loudspeaker system between the input and the total sound pressure measured at a particular microphone position. This curve reveals peaks and dips in the amplitude response below 500 Hz which are caused by the interferences of room modes and early reflections. The sound separation provides the amplitude response of the direct sound that corresponds with the expected theoretical behavior. The second term in Eq. (19) comprising the coefficient scaling the Hankel function $h_n^{(1)}(kr)$ represents the incoming sound generated by the sound reflections at the room boundaries. Figure 6.9 also shows the generated amplitude response of this incoming sound part with peaks and dips at particular frequencies which depend on the size and shape of the room, the acoustical properties of the reflecting surfaces and the position of the speaker and microphone. If the level difference between the direct sound and the room reflections is small, the measured sound will be affected. The holographic wave separation is not required at higher frequencies ( $f > 3$ kHz) where the time windowing of the impulse response can be applied.

## 7. Preparatory Questions

Check your theoretical knowledge before you start the regular training. Answer the questions by selecting all correct responses (sometimes, there will be more than one).

**QUESTION 1:** Why is the measurement of the directivity important for assessing loudspeakers? (section 4)

- □ **MC a:** The directivity of the loudspeaker describes the amplitude and phase of direct sound radiated into a particular direction.
- □ **MC b:** The directivity of the loudspeaker describes the properties of the loudspeaker which are important to model the interactions with the acoustical environment such as early and late room reflections and standing waves.
- □ **MC c:** The directivity of the loudspeaker describes the maximum output of the loudspeaker limited by transducer nonlinearities and thermal overload.

**QUESTION 2:** What are the differences between far field and near field of a complex sound source? (section 4)

- □ **MC a:** The sound pressure and particle velocity are in phase in the far field.
- □ **MC b:** The total sound power will not change with the distance $r$ from the source in far field.
- □ **MC c:** The total sound power will decay by 6 dB per octave for doubling the distance $r$ in far field.
- □ **MC d:** The sound power in the near field has a reactive component.
- □ **MC e:** The sound power in the far field has a reactive component.

**QUESTION 3:** How to ensure far field measurement conditions? (section 4.2)

- □ **MC a:** Distance $r$ between sound source and measurement point should be larger than the largest dimension $l$ of the radiating surface.
- □ **MC b:** Distance $r$ is smaller than the largest dimension $l$ of the radiating surface.
- □ **MC c:** Distance $r$ is larger than the wavelength $\lambda$ of the lowest spectral component measured.
- □ **MC d:** Distance $r$ is smaller than the wavelength $\lambda$ of the lowest spectral component measured.
- □ **MC e:** The ratio $r/l$ between distance $r$ and geometrical dimension $l$ should be larger than the ratio $l/\lambda$ between geometrical dimension $l$ and wavelength $\lambda$.
- □ **MC f:** The distance $r$ should be large enough to ensure that the sound pressure is 6 dB lower than in the near field of the source.

**QUESTION 4:** Does the $1/r$-law depend on frequency? (section 4.2)

- □ **MC a:** Only high frequency components decrease inversely with distance $r$ in far field (with $r > r_{far}$).
- □ **MC b:** All frequency components decrease inversely with distance $r$ in far field (with $r > r_{far}$).
- □ **MC c:** The critical distance $r_{far}(f)$ is usually a function of frequency $f$.
- □ **MC d:** The critical distance $r_{far}$ is a constant which only depends on the size of the loudspeaker but is independent of the frequency $f$.

**QUESTION 5:** Why is the directivity factor defined in the far field of the source? (section 5.2)

□ **MC a:** The directivity factor is independent of the distance $r$ if the measurement is performed in the far field ($r > r_{\text{far}}$).

□ **MC b:** The measurement of the sound pressure output is performed at a larger distance from the source (far field condition) to avoid that ambient noise corrupts the measurement results.

□ **MC c:** Any positioning error of the microphone and loudspeaker has a smaller impact on the measured magnitude response at a larger distance ($r > r_{\text{far}}$).

□ **MC d:** Any positioning error of the microphone and loudspeaker has a smaller impact on the measured phase response at a larger distance ($r > r_{\text{far}}$).

□ **MC e:** The measurement of the sound pressure output is performed at a larger distance from the source (far field condition) because the direct sound can be more easily separated from room reflections by windowing the impulse response.

**QUESTION 6:** Why is an anechoic room beneficial for loudspeaker measurements? (section 5.1)

□ **MC a:** Air movement (wind) can be avoided.

□ **MC b:** Inhomogeneous temperature distribution can be reduced by using an air condition system.

□ **MC c:** The microphone can be placed very close to the loudspeaker while maintaining far-field conditions.

□ **MC d:** External ambient noise can be reduced by using thick walls and a floating foundation.

□ **MC e:** A full space anechoic rooms provides perfect free field conditions.

**QUESTION 7:** Are loudspeaker measurements affected by standing wave (modes) and sound reflections in an anechoic room? (section 5.1)

□ **MC a:** Yes, the absorbing material placed on the walls cannot damp the sound reflections at low frequencies, which may affect the sound pressure response measured in the far field of the loudspeaker.

□ **MC b:** No, if all reflecting boundaries are covered with absorbing material (the thickness is not critical) the measurement room will have perfect anechoic properties.

□ **MC c:** If the microphone is placed in the near field of the loudspeaker, close to the diaphragm and the distance to the walls is significantly large, than the reflective sound may be negligible compared to the direct sound.

**QUESTION 8:** Why are characteristics that describe the near field properties important for assessing audio devices? (section 6.1)

□ **MC a:** The near field properties of studio monitors are relevant for sound engineer who have their listening position close to the speakers.

□ **MC b:** Laptops, tablets, PC multi-media audio, smart phones and other personal audio equipment are used in the near field.

□ **MC c:** The near field properties reveal the dominant nonlinearities inherent in the loudspeakers.

**QUESTION 9:** Under which condition can the far field characteristics of a loudspeaker be derived from near field data? (section 6.2)

□ **MC a:** When the loudspeaker is a transducer that is mounted in a sealed enclosure, it generates an omnidirectional radiation characteristic (monopole) at low frequencies. In this case, the 1/r-law is valid for the far field and also for the near field.

□ **MC b:** The sound pressure in the near field has to be scanned at multiple points on a scanning surface around the loudspeaker with sufficient angular resolution. The measured sound pressure data is modelled by superposition of scaled basic functions that are the solution of the wave equation and describe the propagation of sound into the far field (holographic measurement).

□ **MC c:** Valid far field data can be calculated from near field measurements by windowing the impulse response.

□ **MC d:** Accurate far field data can only be derived from a single sound pressure measurement if the microphone is located in the acoustical center of the source.

**QUESTION 10:** What affects the angular resolution of the measured directivity determined by near field scanning technique and holographic wave expansion? (section 0)

□ **MC a:** The angular density of the scanning grid, corresponding to the number and placement of the measurement points, determines the angular resolution of the measured directivity.

□ **MC b:** The holographic processing performs an angular interpolation between the measured samples based on the basic functions that fulfil the wave equation. Thus the angular resolution of the measured directivity may be higher than the angular density of the measurement points on the scanning surface. This interpolation is only correct if the order of the wave expansion is high enough to describe the sound field at the investigated frequency without aliasing.

□ **MC c:** The maximum order $N$ of the wave expansion determines the angular resolution of measured directivity.

□ **MC d:** The orientation of the loudspeaker during the scanning process determines the angular resolution of the measured directivity data.

**QUESTION 11:** What are the results of the holographic measurement? (section 0)

□ **MC a:** The frequency response of the complex coefficients $c_{n,m}^{out}(\omega)$ weighting the spherical harmonics $Y_n^m(\theta,\phi)$ and Hankel function $h_n^{(2)}(kr)$.

□ **MC b:** The amplitude and phase of the directional transfer function $H(\mathrm{j}\omega, r, \theta, \phi)$ describing the relationship between loudspeaker input voltage and sound pressure of the outgoing wave at a point in the far field.

□ **MC c:** The basic functions such as the spherical harmonics $Y_n^m(\theta,\phi)$ and the Hankel functions.

□ **MC d:** The total sound pressure and the sound pressure generated by an outgoing wave at any point $r$ in the 3D space outside the scanning surface.

□ **MC e:** The total sound pressure extrapolated at any point $r$ in the 3D space between loudspeaker surface and the scanning surface.

**QUESTION 12:** Why is the extrapolation of the sound pressure to an observation point in the far field possible based on a wave expansion measured in the near field? (section 6.2)

- □ **MC a:** The extrapolation is possible, because the Hankel functions $h_n^{(1)}(kr)$ describe the relationship between sound pressure versus distance r for an incoming wave.
- □ **MC b:** Spherical harmonics $Y_n^m(\theta,\phi)$ describe the sound pressure versus distance $r$.
- □ **MC c:** The basic function, which is composed of Hankel function $h_n^{(2)}(kr)$ and spherical harmonics $Y_n^m(\theta,\phi)$, tells the relationship between sound pressure versus distance and angles for outgoing waves.
- □ **MC d:** There are no additional sound sources and boundaries in the space between the scanning surface in the near field and observation point in the far field.

**QUESTION 13:** How can the accuracy of the holographic measurement results be checked? (section 6.5)

- □ **MC a:** The sound pressure is scanned at multiple layers where measurement points are located at similar direction ($\varphi_1 \approx \varphi_2$, $\theta_1 \approx \theta_2$) but at different distances ($r_1 \neq r_2$) from the source. The holographic processing calculates the fitting error between measured and modelled sound pressure and uses the redundancy in the input data to evaluate the discrepancy in the modelling.
- □ **MC b:** The measurement is repeated by using the identical scanning process (position and number of the measurement points) and the agreement between the two independent holographic measurements is evaluated.
- □ **MC c:** The standard deviation of the sound pressure is calculated over all measurement points and compared with the permissible threshold.

**QUESTION 14:** Does a very low fitting error ensure an accurate measurement? (section 6.5)

- □ **MC a:** Yes, a low fitting error means the constructed sound pattern perfectly matches the real sound pattern
- □ **MC b:** No, a good fitting error means that the measured data agrees with the modeled data. If the position and orientation of the loudspeaker is not well defined or has been changed during the scanning process the measured data will be misinterpreted and have no practical value.
- □ **MC c:** No, if all measurement points are placed on a single scanning surface there is no redundancy of the data versus distance $r$ and the measurement noise will be interpreted as directional information. Therefore, measurement on to interlaced scanning surfaces is useful for checking the accuracy of the measurement even if separation of incoming and outgoing waves is not required in this frequency range.

**QUESTION 15:** Which are the theoretical reasons for an unacceptable fitting error? (section 6.5)

- □ **MC a:** The maximum order $N$ of expansion is not high enough to model the sound field generated by the loudspeaker.
- □ **MC b:** Ambient noise corrupts the measured data.
- □ **MC c:** The loudspeaker generates plane or cylindrical waves which cannot be fitted by spherical waves.

**QUESTION 16:** How can the direct sound be separated from room reflections? (section 6.6)

- □ **MC a:** Applying windowing to the measured impulse response, extracting the first part of the impulse response which corresponds to the direct sound and attenuating the late part of the impulse response which corresponds to the room reflections.
- □ **MC b:** Scanning the sound pressure on two interlaced surfaces around the sound source and performing an expansion of the measured sound pressure in outgoing and incoming sound waves.
- □ **MC c:** Scanning the sound pressure on a single surface around the sound source and performing a holographic processing of the data.

**QUESTION 17:** What are the benefits of using a holographic scanner in an anechoic room? (section 0)

- □ **MC a:** The ambient noise caused by external sound sources (e.g. traffic, production noise) can be reduced.
- □ **MC b:** The measurement time can be reduced by skipping a double scan required for sound separation, because the room influence at low frequencies can be neglected in the near field where the direct sound is dominant.
- □ **MC c:** Multiple holographic scanning systems can only be operated in an anechoic room.

**QUESTION 18:** Which restrictions affect the separation of direct sound and room reflections at high frequencies by windowing the impulse response? (section 0)

- □ **MC a:** The loudspeaker should be placed in the middle of the room to generate the maximum distance to the room boundaries that gives the largest time difference between the direct sound radiated by the loudspeaker and the room reflections arriving at the loudspeaker diaphragm. The position of the microphone is not critical.
- □ **MC b:** The microphone should be placed in the middle of the room to generate the maximum distance to the room boundaries and the largest delay of the room reflections. The position of the loudspeaker is not critical.
- □ **MC c:** The distance $d_{ms}$ between the microphone and speaker and the minimum distance $d_{sr}$ between the speaker and the room boundaries (walls, ceiling, ground) determines the reflection free time $T_{max}$ between first arrival of the direct sound and room reflections. This time $T_{max}$ determines the width $T_w$ of the window and the spectral resolution $\Delta f = 1/T_w$ of the measured transfer function.

**QUESTION 19:** How can the number of measurement points in the scanning process be reduced while maintaining sufficient angular resolution and accuracy? (section 6.4)

- □ **MC a:** The maximum order N of the expansion can be reduced if the fitting error does not exceed a critical limit (-20 dB) in the frequency range of interest.
- □ **MC b:** The scan can be performed on a single layer if the transfer function is assessed at higher frequencies where the sound separation can be performed by windowing of the impulse response.
- □ **MC c:** Improving the signal to noise ratio of the measurement by increasing the amplitude of the stimulus.

**QUESTION 20:** How can the direct sound be separated from the room reflections? (section 6.4)

□ **MC a:** Holographic field separation can be applied to the sound pressure measured on two interlaced scanning surfaces with different distance $r_1 \neq r_2$ from the expansion point.

□ **MC b:** Holographic field separation cannot be used at low frequencies where the wavelength is larger than the size of the device under test.

□ **MC c:** Windowing of the impulse response can be used to suppress the room reflections at high frequencies where the reflection free time $T_{max}$ is larger than the window length $T_w = 1/\Delta f$ providing the requested frequency resolution $\Delta f$.

□ **MC d:** Windowing of the impulse response can be used to separate the direct sound at low frequencies where the reflection free time $T_{max}$ is much smaller than the window length $T_w = 1/\Delta f$ providing the requested frequency resolution $\Delta f$.

□ **MC e:** Windowing of the impulse response requires scanning of the sound pressure on two interlaced scanning surfaces with different distance $r_1 \neq r_2$ from the expansion point.

**QUESTION 21:** Which of the following statements about time windowing are correct? (sections 0 and 0)

□ **MC a:** Time windowing can be used for the entire frequency range to separate incoming and outgoing waves.

□ **MC b:** Time windowing is used for the high frequency range because it saves time and is easy to implement.

□ **MC c:** Windowing of the impulse response will not affect the spectral resolution of the transfer function at low frequencies if the truncated impulse response is extended by zero padding before applying the FFT giving a high spectral resolution.

□ **MC d:** The measurement of the sound pressure on one scanning surface is sufficient to separate the direct sound from the room reflections by time windowing.

**QUESTION 22:** Why does the near field scanner move the microphone, but keep the sound source at the fixed position? (section 0)

□ **MC a:** A turntable cannot be used for moving the loudspeaker, because the holographic processing requires scanning data which correspond to a constant interaction between loudspeaker and acoustical boundaries (e.g. walls).

□ **MC b:** Moving heavy loudspeakers at high accuracy puts high demands on the positioning system. A microphone can be moved at a higher speed, higher accuracy and at lower cost.

□ **MC c:** Traditional techniques cause artefacts in the measured transfer function due to sound reflections at the turntables while the near field scanning minimizes those errors by placing the device under test on a small post. All other parts of the robotics are outside the scanning surface and can be separated by sound separation.

□ **MC d:** To minimize the cost of the 3D scanner required for measuring the sound field generated by the loudspeaker in three dimensions (spherical coordinates). There are no acoustical reasons for moving the microphone.

## 8. Interpretation of measurement results (no hardware required)

Step 1: View the demo video *Measurement of Loudspeaker Directivity* provided at www.klippel.de/training/ to see the practical measurement of loudspeakers

Step 2: Install the KLIPPEL *R&D* software *dB-Lab* on your computer and download the database corresponding to this training.

Step 3: Start *dB-Lab* by clicking on the file <>

**Advice: It is recommended to do the following exercises offline and to note the answers of the multiple choice questions on a paper!**

### *Basic Sound Transmission*

In *dB-Lab*, firstly open the folder NFS Tutorial. And then open the object ***1. Near Field vs. Far Field***. Double click on *Device Introduction (line array)* to see the line array loudspeaker in the coordinate system. Double click the 4 objects ***1a*** to ***1d*** that are showing polar plots measured at different distances $r = 2$, 4, 8 and 16 m.

**QUESTION 23:** Watch and compare the 4 polar plots at 4.9 kHz and explain the differences.

- □ **MC a:** The graphs in red ($\phi = 0°$) showing the vertical sound pattern are much more beaming to the reference axis ($\theta = 0$) and has much more side lobes than the blue horizontal pattern ($\phi = 90°$).
- □ **MC b:** The graphs in red ($\phi = 0°$) showing the vertical sound pattern changes significantly with increasing distance $r$ while the blue horizontal pattern ($\phi = 90°$) is almost constant.
- □ **MC c:** Multiple transducers placed vertically in the line array generate a significant near field effect.
- □ **MC d:** The far field of this loudspeaker starts at a distance of 4 meter.

Double click on the object ***1e. SPL decay (1/r Law)***, which is showing the difference of the SPL responses when doubling the distance.

**QUESTION 24:** Which of the following statements are correct?

- □ **MC a:** The measurement at low frequencies ($f < 1$ kHz) is already performed under far field condition for a distance $r \geq 2$ meters, because the SPL decreases by approximately 6 dB per doubling the distance.
- □ **MC b:** All the measurements are not performed under far field condition, because the SPL decreases not by approximately 3 dB for doubling the distance r.
- □ **MC c:** The mismatch of the SPL difference per doubling the distance to the $1/r$-law at high frequencies is caused by external noise during the measurement.
- □ **MC d:** The mismatch of the SPL difference per doubling the distance to the $1/r$-law at high frequencies is caused by the near field effect.

**QUESTION 25:** Which is the most important far field condition for the line array operated at high frequencies?

□ **MC a:** Condition 1: $\frac{r_{\mathrm{far}}}{l} \gg 1$, the far field begins when the distance is much larger than the size of the box.

□ **MC b:** Condition 2: $\frac{r_{\mathrm{far}}}{\lambda} \gg 1$, the far field begins when the distance is much larger than the wavelength.

□ **MC c:** Condition 3: $\frac{r_{\mathrm{far}}}{l} \gg \frac{l}{\lambda}$, the far field starts when the $\frac{r_{\mathrm{far}}}{l^2}$ is much larger than the reciprocal of the wavelength.

Now open the object ***2. Room Mode*** showing the measurement on a two-way speaker using a vented box system and horn compression driver. View the operations ***2a. Near field SPL front side*** and ***2b***. ***Near field SPL rear side***, which show the SPL response at same distance on-axis at front and rear side of the loudspeaker. The red curve labeled ***Measured*** shows the SPL directly measured by the microphone. The blue curve labeled ***Radiated*** shows the direct sound separated from the room reflections (dashed blue curve) by the field separation technology.

**QUESTION 26:** Explain the difference of the SPL response at the rear and front side of the loudspeaker?

□ **MC a:** The wave length is large at low-frequencies ($f < 1$ kHz) and the radiated sound is less influenced by diffraction at the loudspeaker box.

□ **MC b:** Voice coil inductance increases the electrical input impedance at higher frequencies and reduces the input current, displacement and radiated sound pressure.

□ **MC c:** With rising frequency the wave length becomes shorter and the geometry of the loudspeaker box reduces the sound pressure level on the rear side.

Select the curve ***Room Reflections*** in the window ***Near Field SPL Response*** in operation ***2b***, copy this curve into the clip board (by using right mouse button) and paste this graph in the corresponding window of ***2a***. Compare the SPL representing the room reflections on the rear and back side of the loudspeaker.

**QUESTION 27:** Why are the fluctuations in the SPL ***measured*** at the rear side (2b) much stronger than in the SPL measured at the front?

□ **MC a:** The room reflections approaching the rear side of the loudspeaker are much stronger than the room reflections approaching the front side.

□ **MC b:** The room reflections have almost the same SPL at the rear and front side. But the room reflections have a much higher impact on the measured response on the rear side because the direct sound (fitted) radiated by the loudspeaker has a much lower SPL there.

□ **MC c:** The SNR is much lower at the rear side than that at the front side.

## *Loudspeaker Analysis*

Open the object ***3. Sound Power & On-axis SPL response*** showing the measurement results of a studio monitor.
Inspect the window ***Far Field SPL Response*** in operation ***3a On-axis SPL and Power response*** that shows the SPL (red curve) at 10 m distance on axis and the ***Radiated Sound Power*** (dashed green curve) at a reduced level (-30 dB). Compare the shape of the two curves.

**QUESTION 28:** Why is the Sound Power response decreasing significantly above 200 Hz while the SPL Response on the reference axis ($\theta = 0°$) stays approximately constant?

- □ **MC a:** The loudspeaker starts becomes more directional at higher frequencies and starts beaming to the front side. Since the active loudspeaker system is equalized to an approximately constant on-axis response, the sound power decreases at high frequencies.
- □ **MC b:** The loudspeaker behaves at high frequency like a monopole.
- □ **MC c:** The sound power increases to lower frequencies because the loudspeaker is more directional at low frequencies.

The holographic measurement of a line array, sound bar or other loudspeaker system using multiple transducers, a wave expansion with a large maximum order $N > 30$ and a large number of measurement points $J > 1000$ are required to fit the sound pressure in the near field at sufficient accuracy. An interesting alternative is the superposition principle where each transducer in the loudspeaker system is measured separately and the total system is modeled by the superposition of the sound fields generated by each transducer.
Optionally, a linear control system can be applied to each transducer for beam shaping and steering the directivity pattern. This approach requires less measurement points than the holographic measurement of the sound field generated by the complete loudspeaker system.

Open the object ***4. Crossover*** that shows further measurements on the Studio monitor which uses a tweeter and a woofer in a two-way configuration with a crossover at 2.1 kHz. The window ***Far field SPL*** response in the first operation ***4a*** shows the SPL response of the complete two-way system where woofer and tweeter are excited by the stimulus. The operation ***4b*** shows the SPL responses of the woofer and tweeter where two independent scans where only one transducer is excited during each scanning process. The red curve shows total SPL response calculated by adding the complex transfer functions of the woofer and tweeter.
The operation ***4c*** shows the difference between the SPL response measured on the total system and the SPL response calculated by the superposition of the tweeter and woofer measurements.

**QUESTION 29:** Which benefits brings the superposition of SPL measured by multiple scans?

- □ **MC a:** The scanning grid can be optimized according to the position of the transducer excited during the individual test. This reduces the number of measurement points required to fit the measured sound field by the wave expansion.
- □ **MC b:** The origin of coordinate system used in the spherical wave expansion can be placed in the acoustical center of the particular transducer excited during each scanning process. The coordinates of the individual expansion points shall be considered in the superposition of the sound fields.
- □ **MC c:** Performing multiple scans improves the signal to noise ratio because the multiple SPL curves are added.

Open the object ***6. Angular Resolution*** and inspect the results of a near field scanning (***6a***) and a conventional far field measurement (***6b***) applied to the same line array. The operation ***6a (NFS)*** shows the extrapolated SPL in the far field (r = 10 m) as a polar plot in the horizontal plane ($\phi = 90°$) and vertical plane ($\phi = 0°$) with $180° < \theta < 180°$ based on 2500 measurement points placed close to the speaker. The operation ***6b. Polar plots (*****conventional)** shows the result of the direct measurement of the SPL in the far field ($r = 10$ m) using a turntable with 5 degree increments over the azimuthal and polar angles generating approximately the same number of measurement points as used in the near field scanning.
Compare the vertical polar plot ($\phi = 0°$) shown as red graph with $-180° < \theta < 180°$ in the operations 6a and 6b.

**QUESTION 30:** What causes the difference between the vertical polar plots (red graphs) provided by the two measurements?

□ **MC a:** The polar plot provided by the near field scanner (NFS) provides a much higher angular resolution than the density of the measurement points on the scanning grid, because the polar pattern is reconstructed between the measured points on the grid by using the solution of the wave equation.

□ **MC b:** The polar plot provided by the conventional far field measurements provides an angular resolution which is identical with the angular increments of the measurement points generated by the turntables.

□ **MC c:** The higher complexity of the curve shape in the polar plot generated by the near field scanner is influenced by background noise.

Compare the horizontal polar plot ($\phi = 90°$) shown as blue graph in the two measurements ***6a (NFS)*** and ***6b (conventional)***.

**QUESTION 31:** What causes the lower complexity of the blue curves in the horizontal polar plot ($\phi = 90°$)?

□ **MC a:** The angular resolution provided by both measurement techniques (NFS and conventional) is too small to represent the horizontal directivity.

□ **MC b:** The blue graph shows the horizontal polar plot of the loudspeaker, which is not complex in directivity, because all tweeters used in the loudspeaker are placed on line rectangular to the horizontal plane. A smaller number of measurement points generating less angular resolution would be sufficient to represent the directivity in this plane.

□ **MC c:** The horizontal directivity of the complete line array corresponds approximately with the horizontal directivity of each tweeter used in the line array. Thus, the directional properties of a single tweeter determine the horizontal directivity of the line array.

## *Basic Loudspeaker Knowledge*

Select the object ***7. Driver in vented or sealed box or in free air*** and view the operation **7. *Device Introduction*** where a woofer in free air and mounted in a vented Demo-box is shown. The port of the Demo-box can be sealed by a plug to generate a sealed box. Each device is measured by near field scanning and the results are presented in the following operations:
View the operation ***7a. Sound Power Comparison*** and inspect the three diagrams showing the far field SPL response on axis in 10 m distance, directivity index and the radiated sound power. Compare the curves of ***Far Field SPL Response (10m front ON-Axis)*** of ***Vented***, ***Sealed***, and ***Driver in free air***.

**QUESTION 32:** Why does the ***Driver in free air*** (green curve) generate a much lower SPL response at low frequencies than the driver mounted in a ***vented*** and ***sealed enclosure***?

- □ **MC a:** Acoustic cancellation takes place between the volume velocity generated on the front and rear side of the driver in free air that reduces the SPL in the far field.
- □ **MC b:** The driver operated in free air generates less cone displacement because there is no additional stiffness provided by the air in the box. The lower displacement generates also a lower SPL in the far field.

Inspect ***Far Field SPL Response (10m front ON-Axis)*** of the three devices at approximately 150 Hz.

**QUESTION 33:** Why does the vented box loudspeaker system generate a peak in the SPL frequency response at 150 Hz which is not found in the sealed box system?

- □ **MC a:** There is a standing wave in the sealed enclosure which increases the acoustical impedance at this frequency and reduces the acoustical output.
- □ **MC b:** The air mass in the port and the acoustical compliance of the air in the box build up a resonator which increases the acoustical output at the port resonance.
- □ **MC c:** This peak is caused by a room mode increasing the sound pressure at the microphone position.

Inspect the diagram ***Directivity Index*** and compare the curves of the three devices at low frequencies.

**QUESTION 34:** Why does the driver in free air generate a larger directivity index than the driver operated in an enclosure?

- □ **MC a:** At low frequencies, the driver can be regarded as a dipole, so it is more directional than a monopole.
- □ **MC b:** At low frequencies, the driver in free air can be regarded as a monopole, so the directivity is higher.
- □ **MC c:** At low frequencies, the driver in free air can be regarded as a quadrupole, so the directivity is higher.

Inspect the diagram ***Radiated Sound Power*** and compare the curves of the three devices below 50 Hz.

**QUESTION 35:** Why does the driver in the sealed enclosure radiate more sound power than the driver in the vented enclosure?

- □ **MC a:** The acoustic cancellation between the volume velocities generated by the diaphragm and the port reduces the sound power. Therefore, the SPL frequency response decreases with a higher slope to lower frequencies than at the sealed box.
- □ **MC b:** The enclosed air in the sealed box provides additional acoustical stiffness which increases the resonance frequency. A higher resonance frequency generates more acoustical output at low frequencies.

Inspect the operations ***7b-7g*** showing a balloon plot and the total sound power and contribution of order $N$ for the driver in free air, mounted in the sealed and vented enclosure.

**QUESTION 36:** Which spherical wave generates the largest contribution to the total sound power at 200 Hz?

- □ **MC a:** Driver in free air: The spherical waves of order $N = 1$ (dipoles) radiated by the driver in free air generate the dominant contribution.
- □ **MC b:** Sealed box: The spherical wave of order $N = 0$ (monopole) radiated by the driver mounted in a sealed enclosure generates the dominant contribution.
- □ **MC c:** Vented box: The wave of order $N = 0$ (monopole) radiated by the vented box system generates the dominant contribution.
- □ **MC d:** Vented box: The vented box system behaves as a dipole at low frequencies. Thus, the spherical wave of order $N = 1$ generated by a dipoles provides the dominant contribution to the total power in the far field.

## *Field Identification*

Select the object ***8. Maximum Expansion Order*** showing the NFS measurement applied to a ***Studio Monitor***. Inspect the following operations ***8a-8d*** showing the fitting error as a function of the maximum order $N$ with $N$  1, 5, 10 and 14.

**QUESTION 37:** What is the optimal value for the maximum order $N$ used in the wave expansion if the sound radiation shall be measured over the full audio band (20 Hz … 20 kHz)?

- □ **MC a:** The maximum order of the expansion should be larger than order 14 because the fitting error exceeds -20 dB at low frequencies.
- □ **MC b:** The maximum order of the expansion shall be set to $N = 14$ because this is a good comprise between a low fitting error ($E <$ -20 dB) and number $M$ of measurement points on the scanning grid to ensure that all unknown parameters in the wave expansion can be identified. The error below 50 Hz corresponds to the poor SNR due to the low SPL generated by the loudspeaker. The rising fitting error corresponds to the rising noise floor shown as a dashed grey curve in the diagram.
- □ **MC c:** The maximum order of the expansion shall be set to $N = 5$ because the fitting error E is smaller than -20 dB over a wide frequency band but increases at low and high frequencies due to the poor SNR shown as the dashed curve in the diagram.

Open object ***9. Time Windowing&Frequency Resolution*** and the operation ***9a. TRF transfer function (off-axis)***. Inspect the result window ***Impulse Response***. The grey curve shows the original impulse response *h*(*t*) and the red curve shows the impulse weighted by a half-Turkey window *w*(*t*) between the black cursors. You can zoom in the graph by dragging with the mouse, and undo the zoom by typing Z.

**QUESTION 38:** When does the first room reflection arrive at the measurement point?

- □ **MC a:** 1.1 ms after the direct sound
- □ **MC b:** 10.4 ms after the direct sound
- □ **MC c:** 15.7 ms after the direct sound

Move the cursors limiting the left and right side of the window and inspect the influence on the graph ***Magnitude of transfer function H(f)*** at low frequencies.

**QUESTION 39:** Where is the optimum window position?

- □ **MC a:** The time window should be as small as possible in order to separate the direct sound completely from the room reflections.
- □ **MC b:** To ensure sufficient resolution at low frequencies the time window should be as large as possible.
- □ **MC c:** The left cursor of a half window (there is no attenuation on the left window side) should be at the maximum of the impulse reponse.
- □ **MC d:** The left cursor of a half window should be just before the direct sound starts and the right cursor should provide sufficient attenuation of the first room reflections while maximizing the length of the time window to provide maximum frequency resolution.

A simulated free field response can also be generated by holographic field separation of the sound pressure data that are measured on two interlaced scanning surfaces in a non-anechoic environment.
Open the object ***10. Field Separation*** and inspect the results of the measurements ***10a with Field Separation*** and ***10b without Field Separation*** applied to a two-way loudspeaker.
Compare the diagrams ***Fitting Error*** and ***Near Field SPL Response*** of the two measurements.

**QUESTION 40:** What causes the differences in the ***Fitting Error*** and ***Near Field SPL Response*** below 2.3 kHz?

- □ **MC a:** The fitting error is significantly reduced by considering the incoming and outgoing waves.
- □ **MC b:** The residual error which cannot be explained by field separation is close to the noise floor.
- □ **MC c:** The direct sound shown as blue solid curve (***Radiated***) is much smoother at low frequencies than the total sound pressure response (***Measured***) and corresponds to the theoretical behavior of a vented box system.
- □ **MC d:** The room reflections shown as the blue dashed curve reveals the influence of the standing waves in the room. The peaks of the room reflections correspond to the peaks in the fitting error of the wave expansion without field separation.
- □ **MC e:** The field separation technique uses a larger number of measurement points which improves the signal-to-noise ratio SNR in the averaged data set.

**QUESTION 41:** Why is the field separation not used at high frequencies (above 2.3 kHz)?

- □ **MC a:** A higher density of measurement points on the scanning grid is required to model the room modes and sound reflections at higher frequencies. The shorter wavelength and the discrepancy between the shape of the wave-front of the room modes (approaching plane waves after propagation over some distance) and the spherical waves used in the model requires a significantly larger maximum order *N* in the wave expansion.
- □ **MC b:** Windowing is a simple and reliable alternative to separate the direct sound from room modes and sound reflections at high frequencies.
- □ **MC c:** Field separation technique assumes that the loudspeaker under test has an omnidirectional behavior (directivity index $\approx 0$) but most loudspeakers starts beaming at high frequencies.

## *Trouble Shootings*

Open the object ***11. TroubleShooting 1***. And inspect the operations ***11a*** and ***11b***, showing two Field Identification operations based on the same scanning data, but using a different maximum order of the expansion *N*.

**QUESTION 42:** Which operation uses a better setup for performing a fast and accurate NFS measurement?

- □ **MC a:** Operation 11b provides more accurate results, because the fitting error stays over a wide frequency range below -20 dB. The fitting error in operation 11a rises with the frequency *f* and exceeds the limit -20 dB at 7 kHz.
- □ **MC b:** Operation 11a uses a smaller maximum order $N = 5$ of the expansion and requires less measurement points than the setup used in operation 11b. If the device under test is a subwoofer, where only the measurement data below 1 kHz are required, this setup would provide sufficient accuracy in a shorter measurement time.
- □ **MC c:** Both setups are good enough for most applications because the fitting error is not a critical characteristic to evaluate holographic near field measurements.

Compare the impact of the setup on the directional characteristics shown in the operation ***11c. Comparison of the Results***.

**QUESTION 43:** How does the maximum order *N* affect the directivity data?

- □ **MC a:** The directional data and characteristics measured at low frequencies (< 1 kHz) are almost not affected.
- □ **MC b:** There are no major errors generated in the measured characteristics because the low order coefficients in the wave expansion with order $N < 5$ are almost identical in operation 11a and 11b. This is caused by the orthogonal basis functions used in the wave expansion where higher order terms may be added without changing the lower order coefficients.
- □ **MC c:** The loudspeaker under test shows a high directivity index and a complex polar pattern that requires a wave expansion of order $N > 14$ to ensure accurate data at high frequencies.
- □ **MC d:** The measurement is corrupted by ambient noise and the setup in operation 11a using the maximum order $N = 5$ reduces the impact of the noise on the measured directivity information.

Open the object ***12. TroubleShooting 2*** comprising the results of two measurements performed on the same device under test by using the same setup (stimulus, scanning grid, analysis). Inspect the operation ***12a. First measurement*** and ***12b. Second measurement*** and check the reliability of the data.

**QUESTION 44:** Which measurement is valid and provides accurate and reliable results?

□ **MC a:** The first measurement 12a provides valid results in the frequency range from 40 Hz to 10 kHz because the fitting error stays below -20 dB.

□ **MC b:** The second measurement is valid because the fitting error is below -20 dB at selected frequencies.

□ **MC c:** The second measurement 12b provides invalid results below 1.3 kHz, because the error exceed -20 dB.

□ **MC d:** Both measurements are not valid because the fitting error exceeds the limit -20 dB at low and very high frequencies.

**QUESTION 45:** What are the causes of the invalid measurement?

□ **MC a:** The position of the loudspeaker has been changed during the scanning process due to an improper clamping of the device under test on the platform.

□ **MC b:** The position of the microphone has been changed during the scanning process because the microphone was loosely clamped.

□ **MC c:** The invalid measurement is affected by ambient noise which shown as increased level of the grey dashed curve labeled ***Noise Floor***. The reduced SNR has a strong impact on the field separation technique that is used for frequencies below 1.3 kHz. The poor SNR has much less impact on the separation technique based on windowing the impulse response that is applied at higher frequencies.

Open the object ***13. TroubleShooting 3*** showing the results from a holographic measurement of a two-way loudspeaker system using a woofer at low frequencies and a tweeter at high frequencies. The data collected during scanning process have been used for different holographic processing. The operation ***13a. Expansion point tweeter*** shows the result of the holographic processing where the origin of the spherical coordinate system was placed close to the acoustical center at high frequencies which is close to the tweeter. The operation ***13b. Expansion point woofer*** shows the result of the holographic processing where the expansion point was placed close to the woofer generating the acoustical center at low frequencies.

**QUESTION 46:** Where should the expansion point be placed?

□ **MC a:** The expansion point shall be placed close to the acoustical center.

□ **MC b:** In a two-way system, the acoustical center is a function of frequency and moves from the woofer at low frequency to a point close to the tweeter at high frequencies. The tweeter generates a more complex directional pattern than the woofer and requires a higher maximum order $N$ for reducing the fitting error at higher frequencies. If the same expansion point shall be used for all frequencies, this point should be placed close to the tweeter.

□ **MC c:** The expansion point can be placed anywhere in the space between the outer surface of the loudspeaker (diaphragm, enclosure) and the scanning grid. A high fitting error is generated, if the expansion point is placed virtually inside the loudspeaker enclosure.

□ **MC d:** The expansion can be placed close to the woofer if the SPL frequency response is only measured and evaluated at low frequencies where the woofer is active.

# 9. Performing a Sound Radiation Behavior Scan (Near Field Scanner needed)

If a near field scanner is available, it is highly recommended to perform a scan on a loudspeaker. This is supposed to enhance the understanding of the features of Near Field Scanner.

## 9.1. Information to Near Field Scanner

The demo video *Measurement of Loudspeaker Directivity* gives you a direct view of the software and analysis before you start to perform your measurements yourself.

## 9.2. Performing a scan

Please see the NFS User Manual for a detailed description of the near field scanner measurement [16] .

# 10. Reference


[1] W. Klippel, *Vibration and Radiation Behavior of Loudspeaker's Membrane*, pp. 1–19, Mar. 2014.

[2] W. Klippel, *Linear Lumped Parameter Measurement*, pp. 1–19, Feb. 2013

[3] F. Jacobsen, *An Elementary Introduction to Acoustics*, pp. 1–64, Nov. 2011.

[4] S. W. Rienstra and A. Hirschberg, "An Introduction to Acoustics," pp. 1–296, Jan. 2015.

[5] Earl Geddes,GedLee LLC, "Directivity in Loudspeaker Systems," pp. 1–14, Jan. 2010.

[6] R. J. Baken and R. F. Orlikoff, *Clinical Measurement of Speech and Voice*. Cengage Learning, 2000.

[7] F. Jacobsen, *Active and reactive, conherent and incoherent sound fields*, pp. 1–15, 1988.

[8] M. Möser, *Technische Akustik*, pp. 1–552, Oct. 2015.

[9] J. G. Rodriguez, *Anechoic room mechanism*, pp. 1–131, Aug. 2011.

[10] W. Klippel, *Sound Quality of Audio Systems Part 4:Sound Radiation, Measurement and Room Interaction*," 2016, pp. 1–125.

[11] BAY SYSTEM Ltd., *Building a semi-anechoic Chamber*, pp. 1–7, Apr. 2004.

[12] D. B. KeeleJR., *Low-Frequency Loudspeaker Assessment by Nearfield Sound-pressure Measurement*, pp. 1–9, 1974.

[13] G. Weinreich and E. B. Arnold, *Method for measuring acoustic radiation fields*, pp. 1–8, Oct. 2011.

[14] C.-X. Bi, X.-Z. Chen, and J. Chen, *Sound field separation technique based on equivalent source method and its application in nearfield acoustic holography*, *J. Acoust. Soc. Am.*, vol. 123, no. 3, pp. 1472–7, 2008.

[15] E. G. Williams, *Fourier Acoustics*, pp. 1–321, 1999.

[16] Software Manual: *Near Field Scanner R&D System User's guide*, Klippel GmbH
[17] IEC (E) 60268-21: *Acoustical (Output based) Measurements*, 2015 International Electrotechnical Commission

[18] IEC 62777 Ed.1: Quality Evaluation Method for the Sound Field of Directional Loud-speaker Array, 2014 International Electrotechnical Commission
[19] CEA-2034: Standard Method of Measurement for In-Home Loudspeakers, , 2013 Consumer Electronics Association