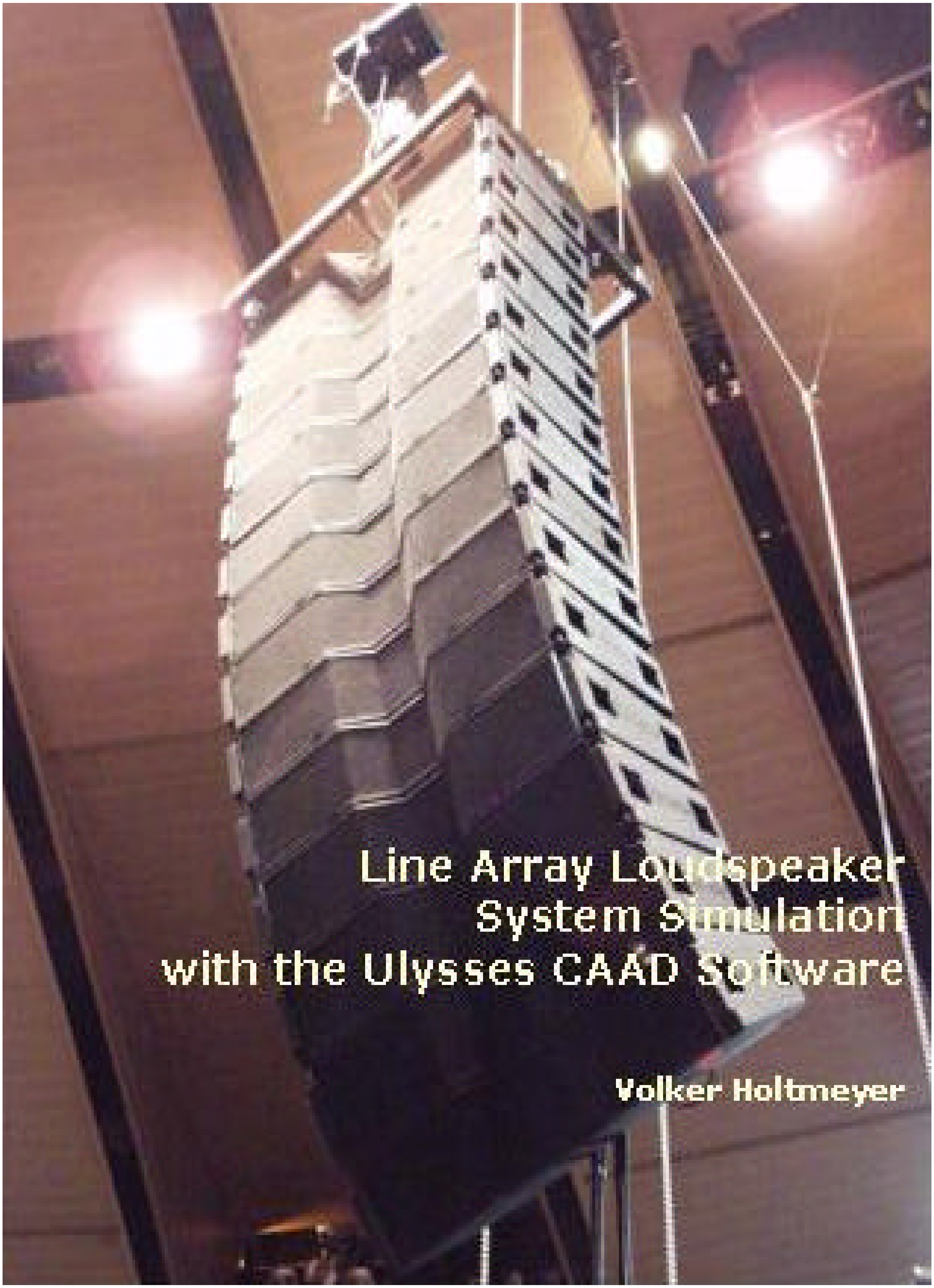

Line Array Loudspeaker
System Simulation
with the Ulysses CAAD Software
Volker Holtmeyer

Contents

1 Introduction

For high-quality sound reinforcement sufficient sound pressure levels (SPL) and even frequency response are essential. These conditions must be met equally well at all points of the area to be exposed to sonic radiation. Since single loudspeakers quickly reach their limits they are usually grouped together in clusters. When using conventional loudspeakers that radiate sound spherically, however, destructive interferences have an adverse effect on their performance.

To avoid destructive cancellation line array loudspeaker systems are used which line up individual modules vertically. A line array generates a coherent wavefront without destructive interferences due to a short distance between individual drivers and by dedicated wave guides for the high frequency range. To a great extent such an alignment disperses sound cylindrically – in other words it acts as a line source. This results in a long-throw tight pattern projected vertically which enables controlled dispersion across predetermined areas.

The Ulysses CAAD software[1] has been designed originally to simulate loudspeakers which may be described as point sources only. The objective of this paper is the examination to which extent line array loudspeaker systems can be simulated with the Ulysses software as well.

[1] CAAD: **C**omputer **A**ided **A**coustic **D**esign

2 Fundamentals

2.1 Wave Dispersion

Identical conditions for wave dispersion in all directions exist if a sender (a vibrating system dispersing waves) located in an infinitely extended medium is at rest relative to that medium. Possible reflections at the boundaries have no retroactive effects on the sender. In such a case there are three distinct shapes of wave dispersion depending on sender shape which are of particular interest:

1. the plane wave (plane source)
2. the cylindrical wave (line source)
3. the spherical wave (point source)

2.1.1 The Plane Wave

The source of a plane wave is by definition a plane, i.e. the radiating area must be regarded as an infinitely extended and uniformly vibrating area. All geometrical locations of the plane wave with identical phase relations are parallel planes relative to the primary plane. The direction of the dispersion is vertical to the wave. Therefore the plane wave is a one-dimensional wave by nature.

A specific feature of the plane wave is its uniform energy density. The sound pressure level remains constant independent of distance and position. There is no level drop with increasing distance.

2.1.2 The Cylindrical Wave

Because of the vertically uniform dispersion characteristics of a line-shaped sender vibrating planes with identical phase are cylindrical planes. The sender is located in their center. With increasing distance from the center energy density decreases.

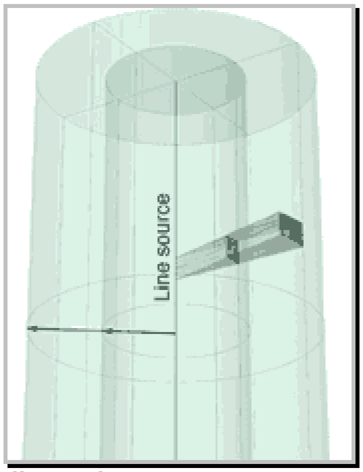


Illustration 2.1
Model of a line source

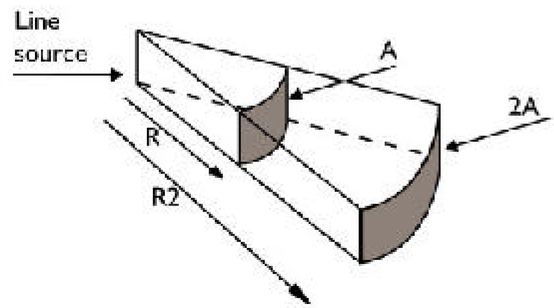


Illustration 2.2
Dispersion characteristics of a cylindrical wave

Doubling the distance from the sender disperses the total energy to double the area. This reduces energy density by half (-3 dB) and the sound pressure level by a factor of 0.707 (-3 dB).

2.1.3 The Spherical Wave

A wave originating from a point source is dispersed in a uniformly isotropic medium as a spherical wave, i.e. the planes with identical phase are spherical planes concentric to the sender with equal distance from each other.

It is easily seen that a spherical wave distributes energy density to increasingly larger areas, i.e. it decreases by $1/r^2$. This results in a decrease of wave amplitude by $1/r$. This decrease of energy density is called inverse square root law.

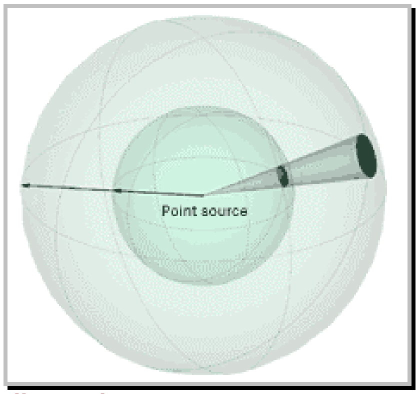


Illustration 2.3
Model of a point source

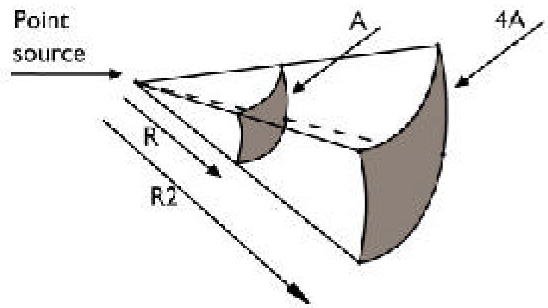


Illustration 2.4
Dispersion characteristics of a spherical wave

In other words doubling the distance from the sender quadruples the dispersion area which reduces energy density to a quarter (-6 dB) and the sound pressure level to half (-6 dB).

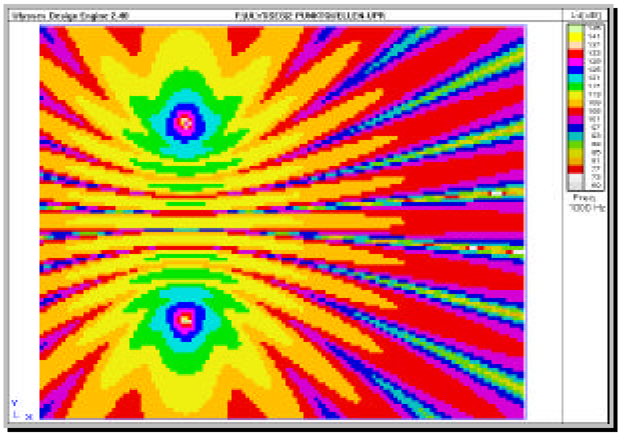
2.2 Interference

Like vibrations waves can interfere. The amount of interference is depending on the amplitude of the individual waves and the phase difference at the interfering point, i.e. it varies in space and time.

Waves with identical length (respectively frequency) dispersed from two senders with a time-constant phase difference generate consistent areas where both waves have an equal phase difference, i.e. they generate interference patterns in a room.

The curves connecting all points with equal phase difference are confocal hyperboles (focal points are the point sources), since these are mathematically defined as geometric locations of all points with equally differing distance from two reference points (focal points).

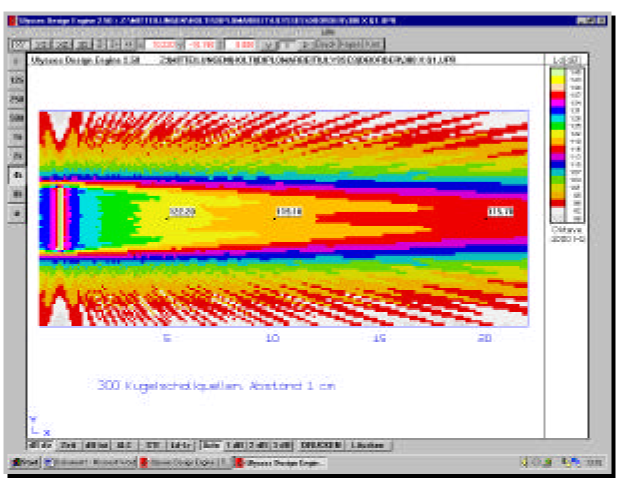
At all points of the listening area where the two waves arrive with a phase difference of 2π or multiples thereof vibration is increased whereas at all points with a phase difference of π or uneven multiples thereof vibration is canceled.



Ill. 2.5
The Ulysses software was used to visualise the interference pattern on the left. The spherical waves dispersed from the two point sources are characterised by equal amplitudes and zero phase difference. The frequency that was examined is f = 1000 Hz; both sources disperse a signal of equal wavelength.

Transfering the above from the two-dimensional plane into three-dimensional space results in hyperboloids of rotation as locations of maximum wave amplification or complete cancellation which share the two senders as focal points.

If wavelength is high relative to the distance of the sources from each other a unique kind of interference occurs. In this case spherical waves dispersed from an even number of point sources interfere and transform into a cylindrical wave at a certain distance from the sources. Thus the origins of a cylindrical wave can be regarded as an infinitely dense alignment of point sources.



Ill. 2.6
This illustration shows the directivity of 300 point sources lined up with a distance of 1 cm from each other.
At the examined frequency is 4 kHz the typical directivity of a line source is apparent: doubling the distance results in a level drop of 3 dB. Increasing air absorption at higher frequencies has been implemented in the calculation algorithm.

2.3 Line Array Theory

If all elements are coherently coupled a modern line array has the basic dispersion characteristics of a cylindrical wave. Because of the finite length these characteristics only apply up to a certain amount of propagation. There is a continuous transition from the nearfield with cylindrical dispersion to the farfield with spherical dispersion. This transition can be approximated by the formula

$$d_{Border} = \frac{3}{2} h^2 f \sqrt{1 - \left(\frac{1}{3hf} \right)^2}$$

in which h is array height in m and f is the frequency in kHz.

For typical frequencies and array heights the square root approximates 1. Therefore the transition is often roughly calculated by the formula

$$d_{Border} = \frac{h^2 f}{2c}$$

in which h is array height in m, f is the frequency in Hz, and c is sound velocity.

It can be seen that the transition is highly dependent on array height and frequency as shown below.

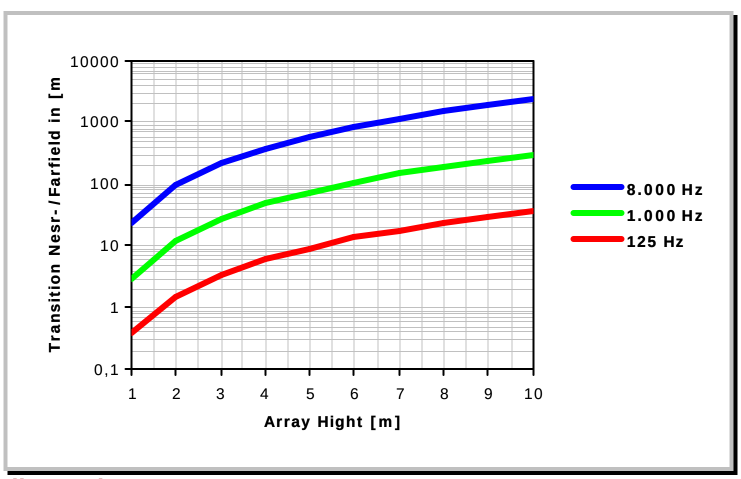


Illustration 2.7
Transition from nearfield to farfield

The nearfield, also called Fresnel zone occasionally, is characterised by the advantage that sound pressure levels are reduced by only 3 dB each time the distance doubles.

In the farfield, also called Frauenhofer zone, sound pressure levels decrease by 6 dB each time the distance doubles which is how conventional loudspeakers perform.

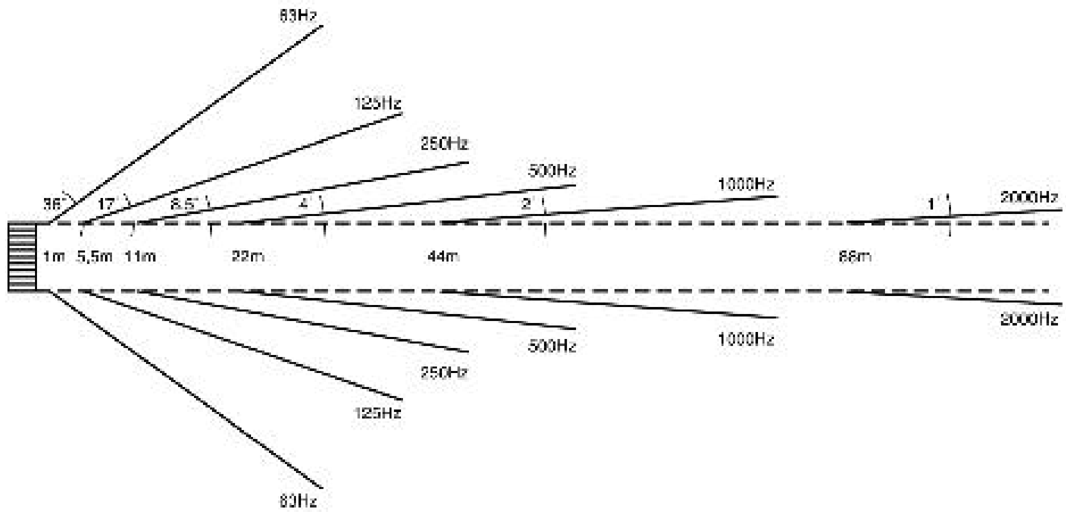
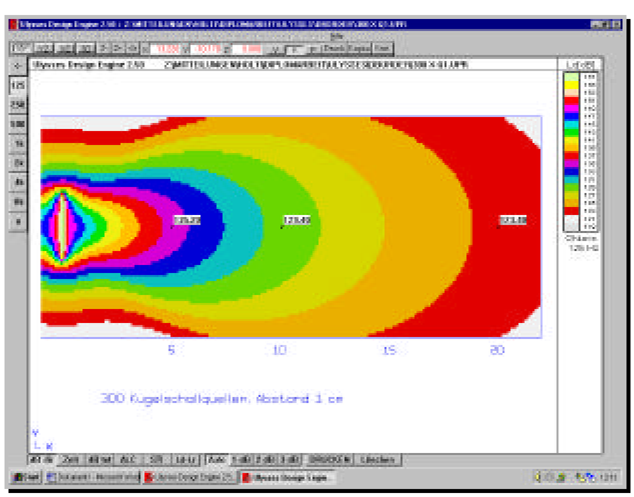


Illustration 2.8
Vertical coverage angle of a 4m line source showing the transition from nearfield to farfield

The vertical directivity of a line array is characterised by the –6 dB coverage angle of the center lobe. It is calculated by the formula

$$D_v = 2\sin^{-1}\left(\frac{0,6}{3hf}\right)$$

in which h is array height in m and f is the frequency in kHz.



Ill. 2.9
The illustration shows the directivity of 300 point sources lined up with a distance of 1cm from each other. The frequency examined is 125 Hz. The levels clicked are in the farfield which can be seen by the level drop of 6dB each time the distance doubles.

3 Acoustic Models

Calculating sound pressure levels in time and space is usually a difficult task. Several methods are applied.

Depending on the method acoustics are subdivided into wave theory, geometrical and statistic acoustics. Strictly speaking, however, there are only two methods for the analysis of sound dispersion within rooms:

1. Calculating the wave equation under consideration of the peripheral conditions at the room boundaries (wave theory acoustics)

2. Examining the multiple reflections of sound rays between room boundaries (geometrical acoustics).

Statistic methods must be integrated into both methods if the number of reflections of a sound ray is rather high.

3.1 Wave Theory Acoustics

If room dimensions are small in relation to the wavelength sound dispersion must be regarded as wave dispersion, and room resonance is an essential factor.

Ignoring the typical non-linear effects at high sound pressure levels, excess attenuation and room inconsistencies, wave dispersion in an acoustic system is completely defined by a partial differential equation, the wave equation.

Reducing the number of variables in the wave equation by a Fourier transformation results in the so-called Helmholtz equation which can be solved under consideration of the (linear) peripheral conditions. The analytical or purely numerical solution of the

Helmholtz equation is hardly practicable since the values can be determined analytically for simple room geometric only.

The *finite element method* which divides the sound field into a large number of individual fields (usually triangular pyramids or cubes) is highly practicable for complex room geometric. The Helmholtz differential equation is transformed into a linear equation which can be solved numerically.

3.2 Geometrical Acoustics

Geometrical acoustics examine the frequency range of a room in which sound is described by its particle character, i.e. wavelength is rather short relative to room dimensions. This approach ignores the wave character of sound.

Geometrical acoustics regard sound as rays which are dispersed from a source in rectilinear fashion and mirrored, i.e. reflected, at the room boundaries.

The source is regarded as the sender (of an infinite number) of sound particles with a certain amount of energy. Source directivity is determined by the directional sound particle density. The sound particles travel from the sender to a receiver directly or via single or multiple reflections (respectively scattering) at the room boundaries. Absorption at the boundaries and air damping is regarded as a frequency dependent decrease of energy, and the soundfield is calculated by the energy of the arriving sound particles.

As the number of direct rays and their reflections increases the paths of individual rays cannot be traced anymore, and only statistic data on the soundfield can be established. In such a case the number of room resonances is rather high.

3.3 Acoustic Simulation Techniques

In many technical areas simulation software has become an essential tool for the planning, design and the evaluation of complex systems. Today, simulation software is also used for electro-acoustic design to closely predict possible results.

Before the advent of acoustic simulation software model measuring techniques were commonly applied. Models were usually designed with a scale of 1:20, and signals were transformed by a factor of 20.

Today, acoustic CAD simulation techniques provide visualisation of soundfields and auralisation of acoustic signals, i.e. listening in virtual rooms.

For room simulation the geometric parameters of its boundaries must be entered as well as their frequency dependent absorption coefficients (and, if necessary, their diffusion coefficients). Sound sources are normally entered as point sources, and their directivity is shown in balloons. Direct sound dispersion is determined geometrically while both statistical and geometrical methods are used to calculate room influences.

4 The Ulysses Software

Ulysses is a computer aided acoustic design (CAAD) software for loudspeaker system and room simulation. It is a useful tool when planning acoustics and sound reinforcement systems. The software package consists of four modules which are described in the chapters below.

Illustration 4.1
The Ulysses modules

4.1 The Ulysses Design Engine

The Design Engine is the main program in which models are drawn, materials are assigned to surfaces and loudspeakers are positioned. The Design Engine also performs all calculations.

4.1.1 Reverberation Times

Ulysses calculates reverberation times (RT_{60}) according to Sabine, Eyring and Fitzroy depending on volume, surface and absorption coefficient. Results are shown with and without air absorption for each of the seven octave bands from 125 to 8000 Hz.

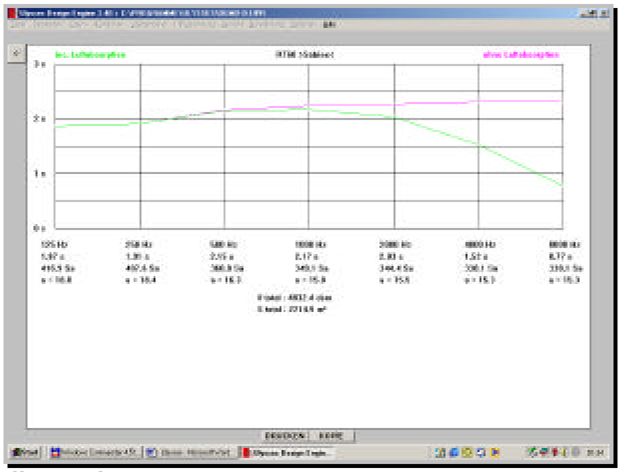


Illustration 4.2
Screenshot: reverberation time curve

4.1.2 Level And Time Calculation

Based on the calculated reverberation times and statistic data of the soundfield levels and times can be calculated. Results are shown as color coded zones on the listening area. All the values of the seven octave bands from 125 to 8000 Hz are available in a single calculation.

Clicking the frequency buttons left on the screen conveniently switches between the octave bands. Clicking the listening area shows the numerical value at the selected position. Results can be printed directly or copied to other graphic or word processing programs.

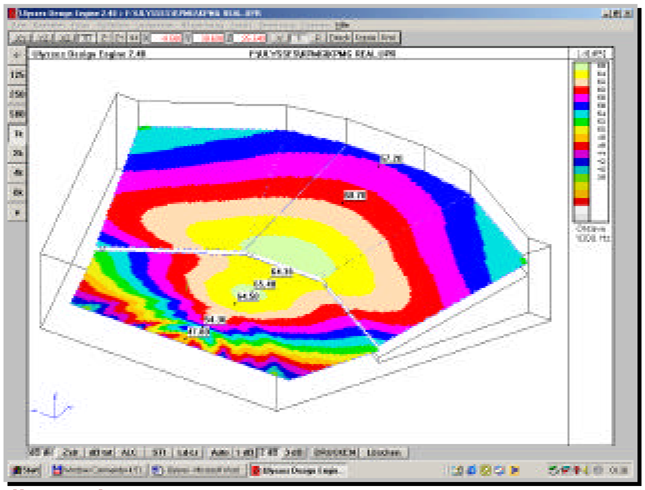


Illustration 4.3
Screenshot: level display

The following level and time calculation results are shown by clicking the appropriate buttons at the bottom of the screen:

dB dir - direct sound pressure level (including early reflections if requested)

time - arrival of the first impulse

dB tot - total of direct and reflected sound

ALC - articulation loss of consonants (%Alcons)

STI - speech transmission index

Ld–Lr - difference between direct and reflected sound

Level summation is selectable: amplitude only or coherent (complex total). For the complex total the phase relation of the octave band or the phase relations of the respective 1/3 octave bands can be selected.

4.1.3 Reflectograms

Ulysses provides raytracing up to the 40th order at any point within the room. Delay, direction and level of individual reflections are calculated up to a selected order. The resulting paths are shown in a 3D model. Impulse response, i.e. energy distribution, is shown graphically in an ETC (energy time curve).

The Ulysses software uses a combination of the geometrical methods of mirror sourcing and ray tracing. While mirror sourcing is 100% exact calculation effort increases immensely with higher orders. For that reason Ulysses uses mirror sourcing to calculate the lower orders and changes to raytracing for calculating higher orders.

Mirror Sourcing

To determine first order reflections each source is mirrored on all boundaries visible from the source. For second order reflections these mirror sources are mirrored again on all boundaries visible from their positions. The possible paths are then traced from the intersecting points on the boundaries.

This increases the number of sources exponentially. In the case of two sources in a basic room with 40 boundaries the two sources account for the direct sound, 80 sources account for the first reflection, 3.200 sources for the second order reflection and 128.000 sources for the third order reflection. It is obvious that calculation effort increases dramatically – approximately by the number of boundaries raised to the selected order of reflections.

Raytracing

A spherical 1° grid source disperses approx. 65.000 rays. Ulysses calculates if a ray strikes the listening area after it has been reflected at the boundaries or if it hits close enough to be evaluated “on target“.

The raytracing calculation effort only increases in linear fashion – approximately by the number of boundaries multiplied by the reflection order. Since the number of rays dispersed cannot be infinite the possible paths cannot all be traced. The amount of precision is linearly dependent on the number of rays that are traced. Since the rays diverge with increasing travel time the probability of a “hit“ also depends on time respectively path length. Probability is high at short travel times and diminishes with increasing travel time.

4.1.4 Auralisation

From the ETC a transformation pattern can be gained and saved as transformation file (*.UAP). This file is basically identical with the impulse response. The convolution of an exciting signal (wave file) with the calculated impulse response results in a time signal. This delineates acoustic room influence on the input signal during the transmission from sender to receiver and is equivalent to the aural reception at the corresponding position within the room. Therefore any audio file (*.WAV) can be transformed and listened to.

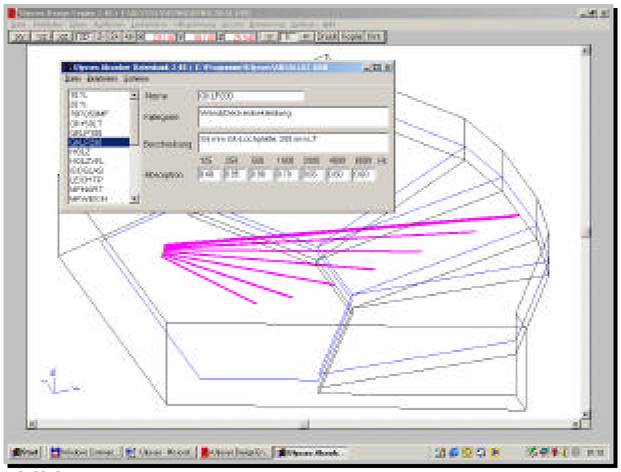
The algorithm applied has not been simplified to reduce calculation time. Every single impulse is used for calculation of the seven octave bands, and no statistical methods are used to interpolate the results.

For that reason it also provides another possibility of auralisation. Clicking a position on the listening area (with any preselected wave file) activates a real time auralisation with a simplified algorithm. Depending on room model complexity it calculates reflections up to the third order and adds statistical reverberation for higher orders. This reverberation is formed by dithered individual impulses according to the theoretical Schröder integration.

Due to the low calculation effort this type of auralisation is a compromise solution which may include severe artefacts. It is often useful, however, to provide a quick impression of the aural reception at a selected listening position.

4.2 The Absorber Base

The Absorber Base is an independent module that manages absorber materials which can be assigned to the boundaries. Their absorptive qualities are listed in octave steps from 125 to 8000Hz.

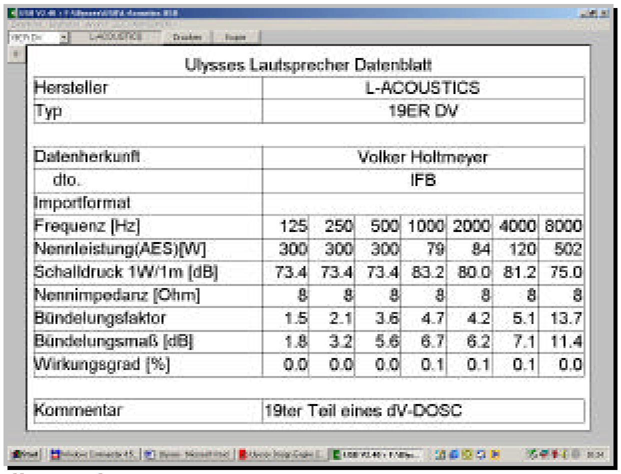


Abildung 4.4
Screenshot: Absorber Base

The data base design is open to facilitate editing of new materials by the user. The IFB homepage also provides data of numerous materials for download.

4.3 The Speaker Builder

The Speaker Builder is another independent module to manage loudspeaker data. The three-dimensional dispersion characteristics are recorded as sound pressure levels in a 5° grid and octaves from 125 to 8000 Hz. The software calculates efficiency, directivity and directivity factor. Dispersion characteristics are shown in polar plots and balloons. Also, loudspeaker enclosures can be drawn in the included drawing menu.



Ulysses Lautsprecher Datenblatt

Hersteller	L-ACOUSTICS						
Typ	19ER DV						
Datenherkunft	Volker Holtmeyer						
dto.	IFB						
Importformat							
Frequenz [Hz]	125	250	500	1000	2000	4000	8000
Nennleistung(AES)[W]	300	300	300	79	84	120	502
Schalldruck 1W/1m [dB]	73.4	73.4	73.4	83.2	80.0	81.2	75.0
Nennimpedanz [Ohm]	8	8	8	8	8	8	8
Bündelungsfaktor	1.5	2.1	3.6	4.7	4.2	5.1	13.7
Bündelungsmaß [dB]	1.8	3.2	5.6	6.7	6.2	7.1	11.4
Wirkungsgrad [%]	0.0	0.0	0.0	0.1	0.1	0.1	0.0
Kommentar	19ter Teil eines dV-DOSC						

Illustration 4.5
Screenshot: Speaker Builder

This data base does not have an open architecture to prevent altering of manufacturer specifications and to avoid that loudspeakers of the same type show differing specifications. Current loudspeaker data by numerous manufacturers are also available on the IFB homepage.

4.4 The Viewer

The Viewer is exclusively intended to graphically present calculations. Its structure is essentially that of the Design Engine although editing possibilities are limited. It provides the opportunity to e-mail results along with the Viewer to a customer.

5 Sound Reinforcement in Large Auditoriums

5.1 Conventional Loudspeaker Clusters

One of the main criteria for modern sound reinforcement systems is a high sound pressure level throughout the entire audible frequency range. The possibilities of a single loudspeaker are rather limited. For that reason a number of loudspeakers are grouped together as clusters. An elementary example is the simple placement of two loudspeaker enclosures next to each other.

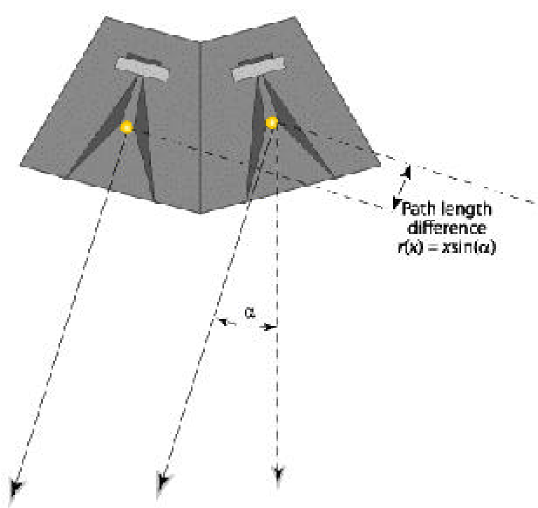


Illustration 5.1
Path length difference of two loudspeakers

It can easily be seen that the signals radiated from two sources at the same time arrive at most room positions with a delay. As described above this causes interference which forms a pattern within the room. Only at the low frequencies with wavelengths that are larger than the distance of the sources from each other the sources couple and act like a single source.

For that reason every effort is made to couple the low frequency drivers in conventional clusters. Since this is impossible in the high frequency range constant directivity horn systems with a long-throw tight pattern are used. These are aimed at individual auditorium areas while attempts are made to keep the overlapping sectors as small as possible.

Illustration 5.2
Overlapping sectors in a conventional cluster

Due to their high directivity factor these array systems have a higher sensitivity, i.e. they generate a higher sound pressure level on axis than loudspeakers with low off-axis rejection. However, the total energy dispersed remains the same.

This type of loudspeaker alignment manages to provide a wide-band increase of the sound pressure level at many listening areas. However, the overlapping sectors of the individual loudspeakers are highly critical. The resulting dips in the frequency response cannot be corrected by equalization since they are caused by destructive interference.

5.2 Modern Line Arrays

This is where the basic idea of modern line array loudspeaker systems sets in. The main objective is the generation of a coherent wavefront throughout the entire frequency range, i.e. a line array is intended to act like a single large loudspeaker.

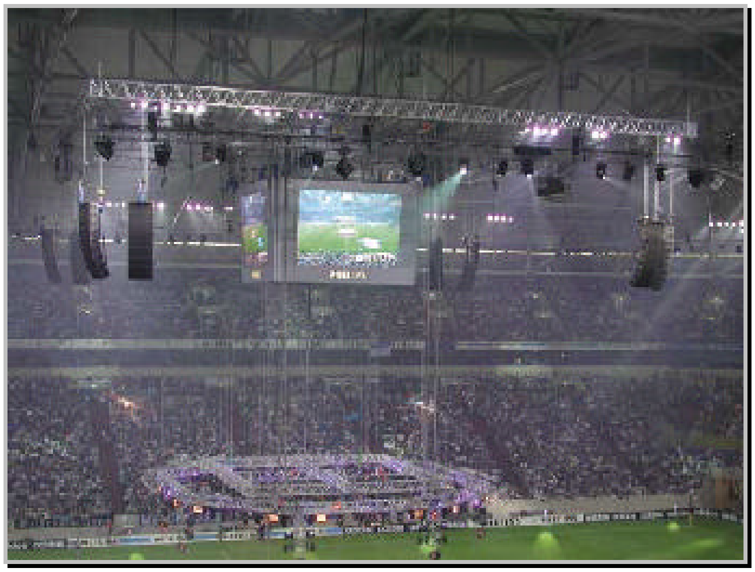


Illustration 5.3
Sound reinforcement with line arrays in a stadium

The system uses a modular construction for easy handling and a variety of configurations. It consists of multiple modules, usually of a multi-way design with conventional cone drivers and waveguides for the high frequency range.

5.2.1 Waveguides

It is essential that a line array generates a coherent wavefront. This requires coherent coupling of the individual sound dispersing modules and a maximum distance between the acoustic centers of half the wavelength. This is not a problem in the low and mid frequency range with large wavelengths, and conventional cone drivers can be used.

However, complications become manifest at the upper limits of the human hearing range. At the frequency of 16 kHz necessary for high quality music reproduction for instance wavelength is approximately 21 mm, therefore driver distance should be approximately 10 mm. This cannot be realised with conventional drivers, i.e. the problem cannot be solved by simply lining up sources.

To provide a coherent wavefront in the high frequency range throughout the height of a line array module - and the overall height of the array – loudspeaker manufacturers use a variety of approaches which are described below.

5.2.1.1 Coercive Waveguides

The most straightforward solution is probably a conventional compression driver coupled to an extended horn, the aperture of which is small relative to its length. The long sound guiding funnel achieves a reduced group delay of the frequencies radiated from the horn mouth.

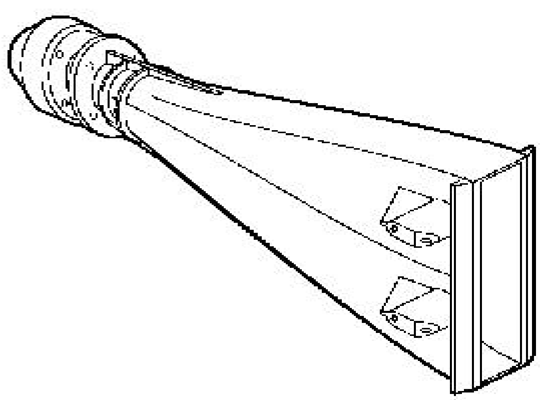


Illustration 5.4
JBL WaveFormer™

The phase difference of the wavefront which is nevertheless curved must not exceed a quarter wavelength to couple coherently with the other waveguides above or below.

5.2.1.2 Channel Sound Guiding

Very often sound waves are guided by a number of separate channels. Fed by a conventional compression driver these channels guide the sound waves to a number of vertically aligned apertures which radiate by and large spherically but couple coherently due to their short distance from each other.

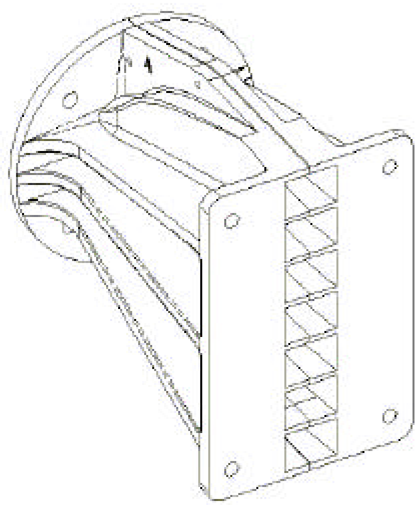


Illustration 5.5
ElectroVoice Hydra™

The individual channels of this design usually have differing lengths, and some manufacturers use additional materials such as foam to reduce sound velocity and to generate a coherent wavefront.

5.2.1.3 DOSC-Waveguide

L-Acoustics owns a waveguide patent based on the so-called Wavefront Sculpture Technology™ (WST™). A conventional compression driver is coupled to a waveguide which continuously diverts the sound waves in such a way that they are radiated from the slot aperture in phase.

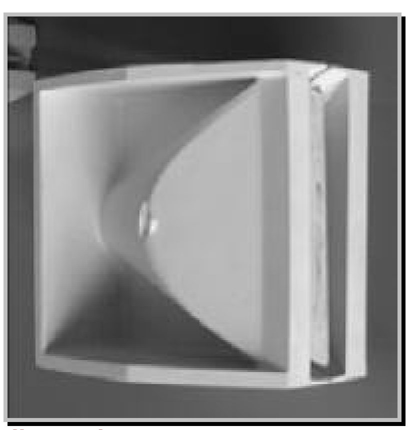


Illustration 5.6
DOSC waveguide

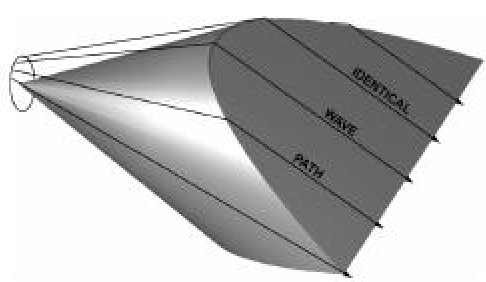


Illustration 5.7
DOSC waveguide paths

To achieve this an encased phase plug was developed. Path length around the plug remains constant from driver throat to horn mouth.

5.2.1.4 Parabolic Reflector

An innovative approach was used by NEXO for their GEO series. The sound waves generated by a conventional compression driver are reflected by a parabolic acoustic mirror. Thus they all have the same group delay when reaching the horn aperture, i.e. they are in phase.

Illustration 5.8
Parabolic reflector

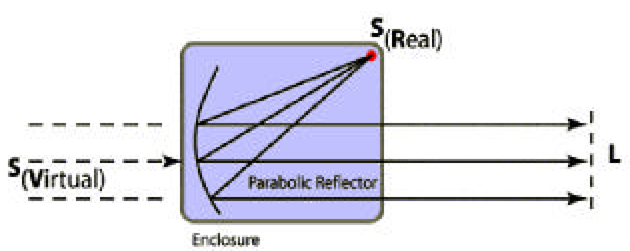


Illustration 5.9
Parabolic reflector design

Further interesting variations are elliptic and hyperbolic reflectors. The elliptic generates a concave wavefront and focuses the sound waves. The hyperbolic reflector generates a convex wavefront and tallies the focal points enabling multiple drivers to act as a point source.

5.2.1.5 Ribbon Transducers

Ribbon transducers are rarely used in large sound reinforcement loudspeakers. A classic ribbon transducer stretches a delicate, extremely low-mass aluminium ribbon between two magnets with opposite polarity. The design is basically that of a moving coil transducer with the advantage that coil and diaphragm are one and the same.

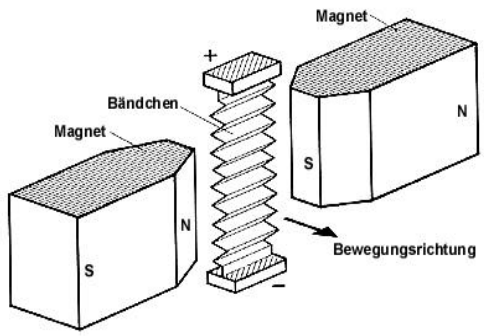


Illustration 5.10
Ribbon transducer design

Illustration 5.11
Line array module with ribbon transducers

For that reason a long continuous ribbon can be generated which is in phase throughout its length. Due to its low sensitivity this design is of importance for the high frequency range only.

5.2.2 Dispersion

By curving the vertically arrayed modules vertical dispersion can be widened to a certain extent. The small coverage angle of the vertical dispersion and the common trapezoidal enclosure design prevent interruption of the wave front. For that reason the vertical coverage angle can be graduated.

This has no influence on horizontal dispersion however which remains constant. Line array modules usually have a defined horizontal dispersion angle between 70° and 120°. The angle specifies where the level drops by 6 dB off axis. In a conventional cluster the coverage of individual loudspeakers overlaps at –6 dB resulting in a level of 0 dB at the points of constructive interference thus generating a sound field that is as even as possible.

Energy distribution within a room can also be determined by curving. Low angles for instance between the upper modules of a line array result in higher sound pressure levels at distant listening areas.

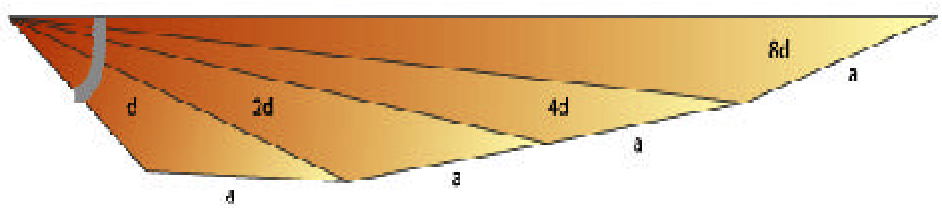


Illustration 5.12
Divergence shading

For constant sound pressure levels throughout the listening area a constant relation between the distance of a module axis from the listening area and the angle between the module concerned and the next module must be achieved. This is calculated by the formula

$$\frac{a_1}{\alpha_1} = \frac{a_2}{\alpha_2} = \ldots = \frac{a_n}{\alpha_n}$$

in which a is the distance of the module axis from the listening area, and α is the angle between the module concerned and the next module. This is also called divergence shading or constant spacing by loudspeaker manufacturers.

Illustration 5.13
Curving of Monarc MLA5 modules

With line arrays it is possible to virtually realize the ideal of sound reinforcement with even frequency response and adequate sound pressure levels for a large listening area. The long-throw tight pattern characteristics of line arrays are advantageous to counteract the negative influences of acoustics.

5.2.3 Computer Aided Array Calculation

It is important to exactly determine the geometry of a modern line array before installation. Many loudspeaker manufacturers provide software to calculate all the relevant parameters. Such software is usually based on spreadsheet files which is of advantage for programming and handling.

The basis of spreadsheet files are room geometry data, i.e. the listening area position. Usually only the vertical plane is concerned although some programs support the entry of horizontal data for visualising horizontal dispersion. A sectional view based on these data is then shown into which the line array and the axes of its individual modules can be integrated.

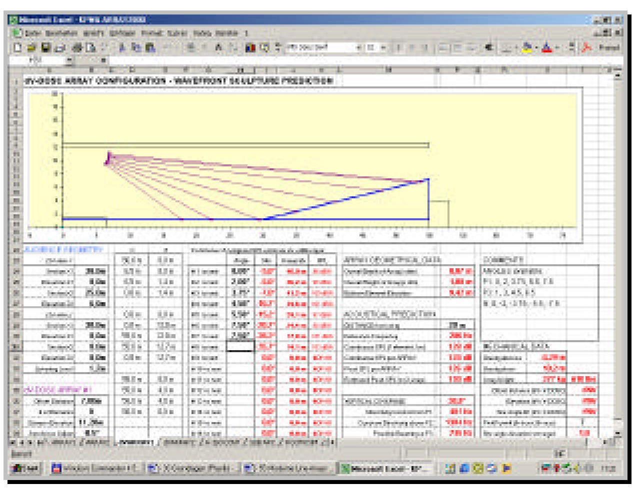


Illustration 5.14
Screenshot: L-Acoustics ARRAY2000 spreadsheet

Spreadsheet files provide important data for array configuration, e.g. angle size between individual modules and array height and alignment. Such programs are passive by nature, i.e. data must be entered by the user. They are then verified by the software.

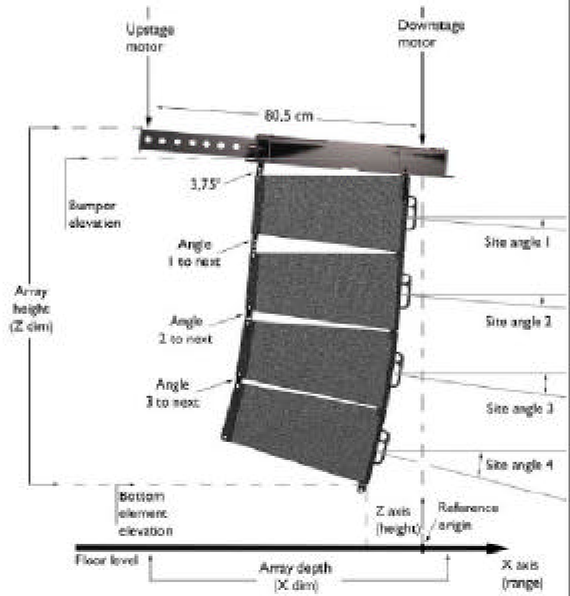


Illustration 5.15
dV-DOSC array parameters

A simple calculation of the sound pressure level to be expected is also shown. The result is only a guiding value however which cannot be compared to the results of CAAD simulation. Also, only direct sound is calculated, room acoustics and the interaction of multiple loudspeakers are not taken into account.

6 Simulation Of Actual Line Arrays

As has been shown in chapter 2 it is basically possible to simulate cylinder wave characteristics with the Ulysses software. The simulation of a line source in principle, however, is of lesser interest. In most cases the objective will be the simulation of an actual line array instead of a fictituous dicreet line source. This has a practical use.

Based on the physical law that a series of spherical waves transform into a cylindrical wave respectively that a line source can be replaced by a line of point sources one might consider to separate an actual line array into a sufficient number of individual sources. However, the question remains how to determine these sources and, most of all, how to establish the balloon data which cannot be defined by measuring.

6.1 Simulation Data

Only the dispersion characteristics of an individual line array module can be measured, for instance. The author had the opportunity to access the data of an L-Acoustics dV-DOSC module (see appendix). These consist of absolute sound pressure level values that have been measured horizontally and vertically in 5° steps around the module in a distance of 4 m under simulated anechoic conditions.

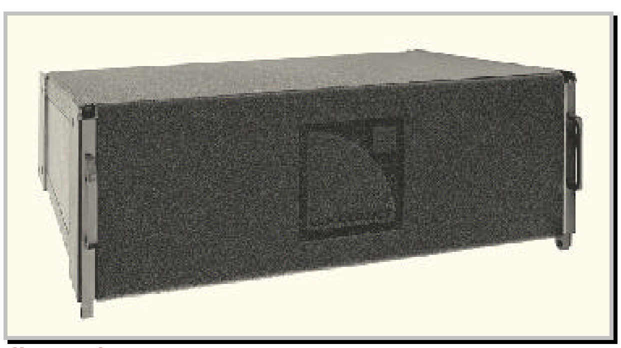


Illustration 6.1
L-Acoustics dV-DOSC line array module

It is important to measure the sound pressure levels for a data balloon in the loudspeaker farfield since the simplified analysis of a loudspeaker as a point source by balloon data is valid only in the farfield. This means that measurements must be made at an adequate distance from the source.

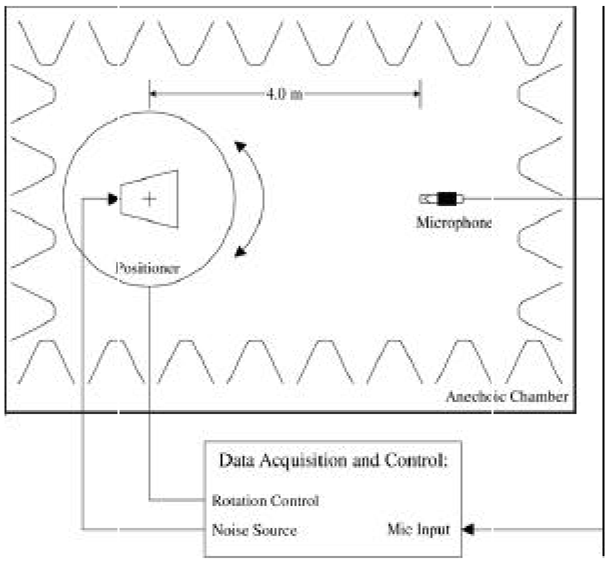


Illustration 6.2
Basic setup for measuring loudspeaker dispersion

For the actual line array module with a height of 229 mm and a maximum frequency[2] of 10.000 Hz to be considered the transition to the farfield is calculated as follows:

$$r_{Fern} = \frac{(0{,}229m)^2 \cdot 10.000Hz}{340m/s} = 1{,}54m$$

Therefore a measuring distance of 4 m is adequate.

6.1.1 Ulysses Native Format

The Ulysses Native Format (*.UNF) is a file to register data which describe loudspeaker dispersion characteristics. These files are the basis for the Ulysses calculations. They are managed in the Speaker Builder module. Currently only sound pressure levels are registered, phase relations have not been implemented yet.

The data registered in the Ulysses Native Format are shown as balloons when selecting the appropriate display.

[2] The maximum frequency used for calculation by Ulysses is 10.000 Hz since the center frequency of the highest 1/3 octave band of the 8 kHz octave is 10.000 Hz.

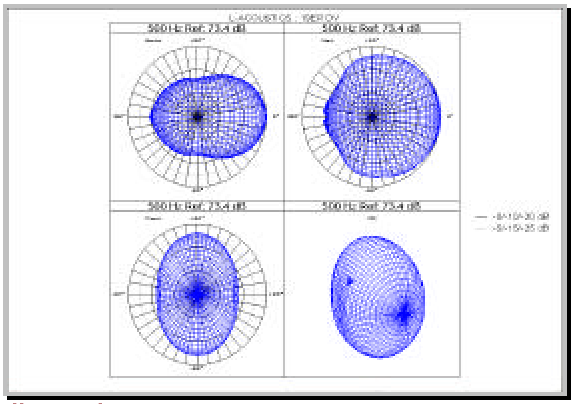


Illustration 6.3
Speaker Builder Balloon Display

The UNF data are measured in a spherical grid of 5° or 10° steps around the loudspeaker. The grid may be envisioned as the longitudinal lines of a globe. A 5° resolution results in 74 degrees of longitude with 37 measuring points each while the loudspeaker axis points to the north pole.

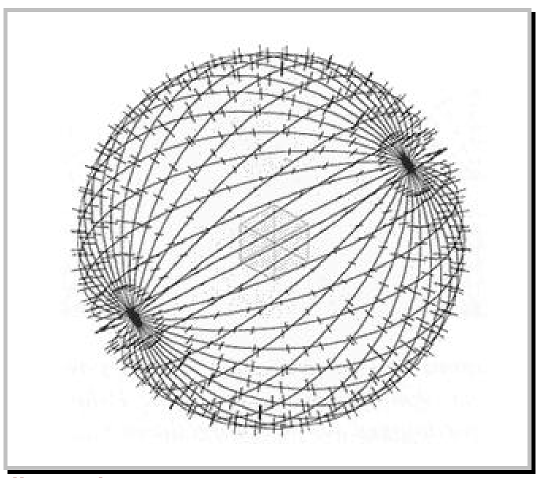


Illustration 6.4
Ulysses Native Format measuring points

A UNF file header consists of four lines with notes. Blocks for each of the seven octave bands from 125 to 8.000 Hz follow which register the absolute sound pressure levels in dB. The measuring points for each line result from a 180° horizontal turn, i.e. those of the first line result from a 180° horizontal rotation at 0° vertical rotation, those of the second line from a 180° horizontal rotation at 5° (or 10°) vertical rotation and so on.

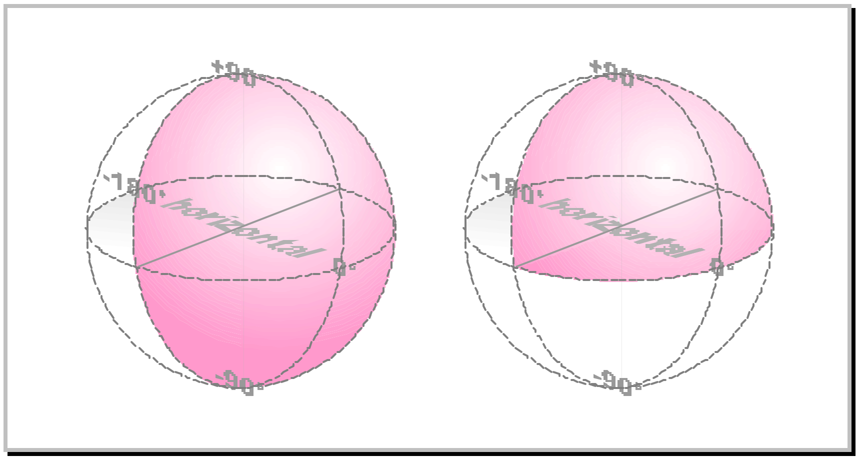


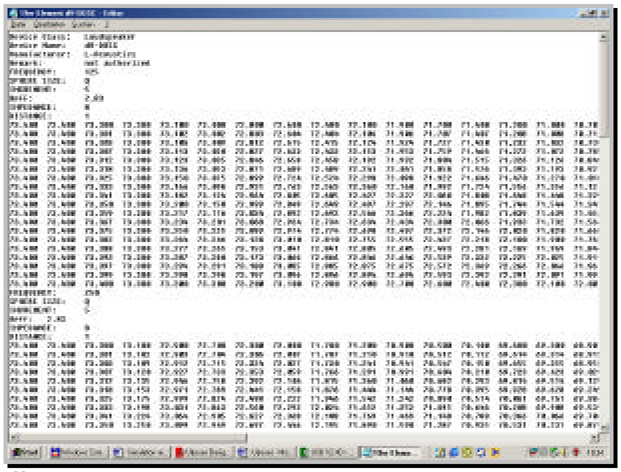
Illustration 6.5
Half and quarter formats

If the loudspeaker drivers are aligned symmetrically the number of measuring points can be reduced. Only half the sphere must be measured if the loudspeaker is symmetrical to its vertical axis (half format). If it is symmetrical to both its vertical and horizontal axes – like a coaxial loudspeaker is – only a quarter sphere must be measured (quarter format). Further reductions are possible for measuring a dynamically balanced ceiling loudspeaker.

The number of lines per block is depending on format and subformat:

5° Full: 72 lines with 37 columns
5° Half: 37 lines with 37 columns
5° Quarter: 19 lines with 37 columns

10° Full: 36 lines with 19 columns
10° Half: 19 lines with 19 columns
10° Quarter: 10 lines with 19 columns



Ill 6.6
Screenshot: Ulysses Native Format (5° quarter format)

To create a UNF file data can be copied to a standard text file. After replacing the *.TXT extension by the *.UNF extension the file can be imported into the Speaker Builder module. It can then be saved together with an enclosure drawing as a loudspeaker file for future reference.

The illustration below shows a 5° solution quarter format measuring point position. It is the 6th value (blue, 25° vertical rotation) of the 16th horizontal rotation (magenta 70°).

	0°	5°	10°	15°	20°	25°	30°	40°	
	Device Class:	Loudspeaker							
	Device Name:	dV-DOSC							
	Manufacturer:	L-Acoustics							
	Remark:	nothing							
	FREQUENCY:	125							
	SPHERE SIZE:	Q							
	INCREMENT:	5							
	Ueff:	2.83							
	IMPEDANCE:	8							
	DISTANCE:	1							
0°	73.400	73.400	73.300	73.300	73.100	73.000	72.800	72.600	...
5°	73.400	73.400	73.301	73.300	73.102	73.002	72.803	72.604	...
10°	73.400	73.400	73.303	73.300	73.106	73.009	72.812	72.615	...
15°	73.400	73.400	73.307	73.300	73.113	73.020	72.827	72.633	...
20°	73.400	73.400	73.312	73.300	73.123	73.035	72.846	72.658	...
25°	73.400	73.400	73.318	73.300	73.136	73.053	72.871	72.689	...
30°	73.400	73.400	73.325	73.300	73.150	73.075	72.899	72.724	...
35°	73.400	73.400	73.333	73.300	73.166	73.098	72.931	72.763	...
40°	73.400	73.400	73.341	73.300	73.182	73.124	72.964	72.805	...
45°	73.400	73.400	73.350	73.300	73.200	73.150	72.999	72.849	...
50°	73.400	73.400	73.359	73.300	73.217	73.176	73.034	72.892	...
55°	73.400	73.400	73.367	73.300	73.234	73.201	73.068	72.934	...
60°	73.400	73.400	73.375	73.300	73.250	73.225	73.099	72.974	...
65°	73.400	73.400	73.382	73.300	73.264	73.246	73.128	73.010	...
70°	73.400	73.400	73.393	73.300	73.287	73.280	73.173	73.066	...
75°	73.400	73.400	73.388	73.300	73.277	73.265	73.153	73.041	...
85°	73.400	73.400	73.399	73.300	73.298	73.298	73.197	73.096	...
90°	73.400	73.400	73.400	73.300	73.300	73.300	73.200	73.100	...
	FREQUENCY:	250							
	SPHERE SIZE:	Q							
	INCREMENT:	5							
	Ueff:	2.83							
	IMPEDANCE:	8							
	DISTANCE:	1							
	73.400	73.400	73.300	73.100	72.900	72.700	72.300	72.000	...
	73.400	73.400	73.301	73.102	72.903	72.704	72.306	72.007	...
	73.400	73.400	73.303	73.109	72.912	72.715	72.324	72.027	...
	73.400	73.400	73.307	73.120	...				

Illustration 6.7
Measuring data in a Ulysses Native Format file

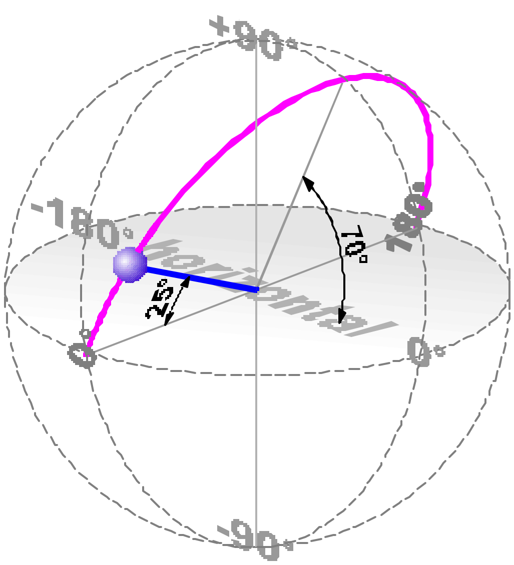


Illustration 6.8
Measuring point position

6.1.2 Data Processing

Since the data of the measured line array module consist of horizontal and vertical plots only a complete data balloon is not available yet.

Due to the horizontal and vertical symmetry of the driver alignment it is sufficient to create a UNF in the quarter format only. For that the horizontal measurements must be copied to the first lines of the respective octave blocks while the vertical measurements are copied to the last. The 35 lines in between must be interpolated. Linear completion which can easily be realised by a spreadsheet program for instance does not provide the desired results as the author found out.

An elliptical balloon shape seems closer to reality. Based on the common ellipse equation

$$\frac{x^2}{a^2} + \frac{y^2}{b^2} = 1$$

and the substitutions

$$x = r \cdot \cos \alpha \quad \text{and} \quad y = r \cdot \sin \alpha$$

the ellipse radius is calculated by the formula

$$r = \sqrt{\frac{1}{\frac{\cos^2 \alpha}{a^2} + \frac{\sin^2 \alpha}{b^2}}}$$

which calculates the proper levels in this case. The variables a and b must be replaced by the level values in the first and last lines while α is the vertical rotation angle of the horizontal rotations. The formula must therefore be written into the empty columns of the spreadsheet file.

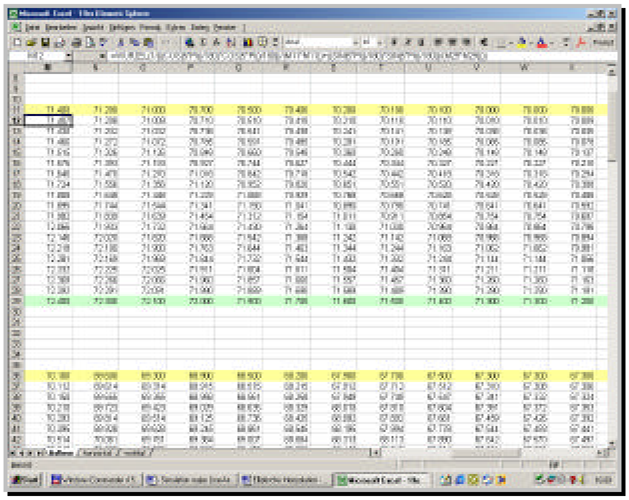


Illustration 6.9
The screenshot shows an Excel file for UNF generation. The yellow lines contain the data of a horizontal rotation while the green lines contain those of a vertical rotation. The values of the lines in between have been elliptically interpolated.

6.2 Substituting The Measured Balloon

The resulting balloon characterises the line array as a point source. Simulations based on this balloon are generally possible with the Ulysses software. However, they are valid only for the farfield (r_{Fern} = 1.54m, see above). It would be a mistake, however, to use a multiple of these sources in line array configuration for simulation. Since the acoustic coupling of the waveguides is neglected the simulation would produce false results in the mid and high frequency range.

Therefore it is important to replicate the characteristics of the waveguide which are a small line source by themselves. As described above, the approach is to separate a line source into a sufficient number of point sources. This can now be realised since a measured balloon is available for substitution by a number of balloons.

The objective is now to make a linear array of multiple sources act like the original balloon. For that the balloon data of the individual sources, their number and their position must be established. The source position is largely determined by the fact that they must be arrayed with equal distance from each other on a line the length of which is equal to the waveguide height.

6.2.1 Substitute Source Distance

First, the distance of the substitute sources from each other must be determined. To generate any level at all in array direction the distance between the individual sources must be less than half the wavelength for which the shortest wavelength is relevant.

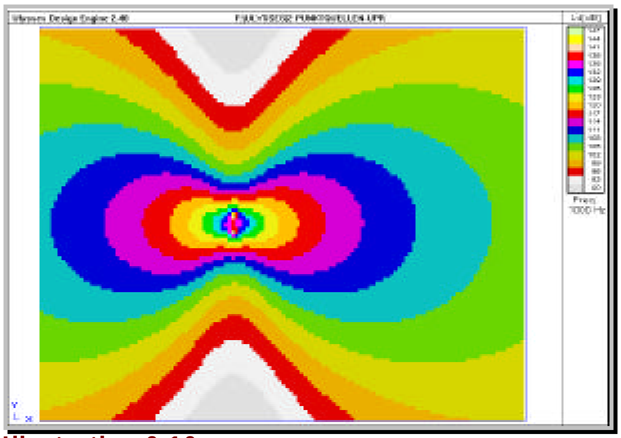


Illustration 6.10
Two sources with a distance of ½ wavelength

Illustration 6.10 shows the 1 kHz dispersion of two point sources with a distance of 34 cm from each other. A pronounced level decrease can be seen on the y-axis. It follows that the distance of the sources from each other must be shorter than half the wavelength for global coupling.

For the following studies a source distanced of 10 mm was determined. The highest frequency used for calculation by Ulysses is 10.000 Hz with a wavelength of approximately 17 mm. Therefore the chosen distance seems adequate.

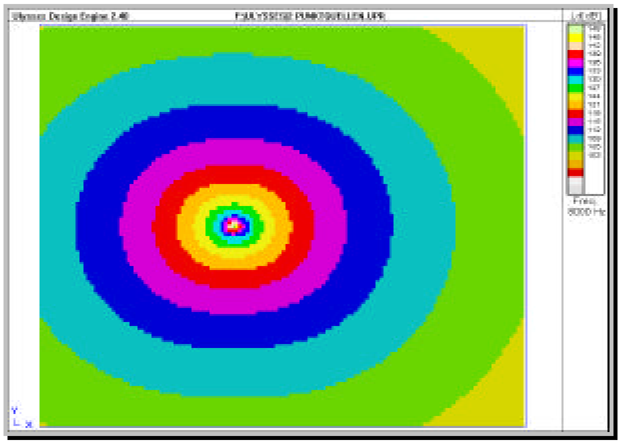


Illustration 6.11
Two sources with a distance of approx. 1/3 wavelength

Illustration 6.11 shows the 8.000 Hz calculation of a 10 mm source distance since Ulysses uses the highest frequency of 10.000 Hz for smoothing only. Coherent coupling is evident, therefore accurate results may be expected for 10.000 Hz as well.

6.2.2 Number Of Substitute Sources

The number of sources n depends on waveguide height and source distance. It is calculated by the formula

$$n = \frac{h}{d} + 1$$

in which h is waveguide height in m and d is source distance in m.

The waveguide in question has a length of 180mm while source distance is 10 mm which results in 19 sources.

6.2.3 Substitute Source Balloons

To generate a substitute source balloon seems no easy task. Therefore the dispersion characteristics of the 19 spherical sources[3] at the previously determined positions had to be calculated. Ulysses was used to design a practical model:

[3] Spherical source: point source with the directivity factor Q = 1

Small listening areas are located in 5° steps on a circle with a radius of 4m. Also on the x/y plane, the 19 spherical sources are in the center with a distance of 10mm from each other. The sound pressure levels resulting from the complex summation of the 19 spherical sources are shown on the listening areas.[4]

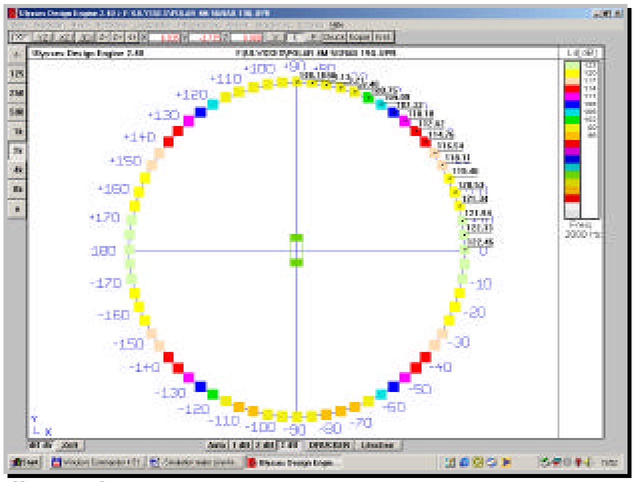


Illustration 6.12
Screenshot: Listening areas and SPL of the19 sources

The data gained by that can be listed in a matrix. The columns of each octave band line contain the sound pressure levels in circular sequence.

A second matrix listing the sound pressure levels of a single source in a distance of 4m must be generated the same way, i.e. all positions of the second matrix show the same value. This can also be calculated easily by the inverse square root law if sensitivity is known. The level which must be subtracted from the individual source sensitivity is calculated as follows:

$$20 \cdot \log 4 = 12 \text{ dB}$$

This calculation takes into account four times the distance from the source.

Subtraction of the second matrix from the first results in a vertical difference matrix that describes how the vertical dispersion of the 19 spherical sources differs from that of a single source in a distance of 4m.

[4] As proven above all "measuring values" of this geometry are in the farfield.

The difference matrix of horizontal dispersion can be generated easily since destructive interference hardly occur. Therefore the sound pressure levels for the 19 spherical sources can be calculated by simply adding the individual levels. This means a level of

$$20 \cdot \log 19 = 25{,}6 \text{ dB}$$

must be added to the level of an individual source.

These values are listed in another matrix as described above. To generate the horizontal difference matrix that of the individual source is subtracted since it contains equal values for both the vertical and horizontal planes. The horizontal difference matrix describes how the horizontal dispersion of the 19 spherical sources differs from that of a single source in a distance of 4 m.

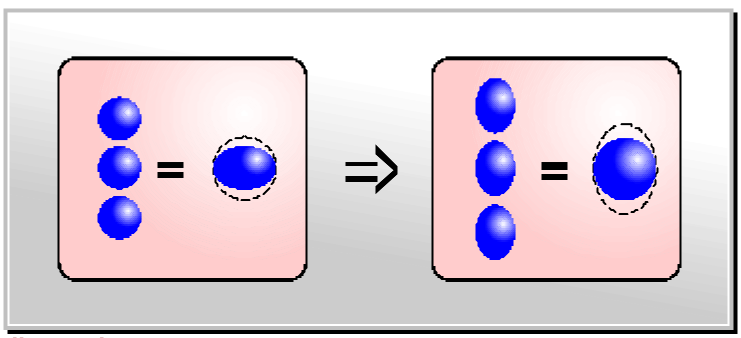


Illustration 6.13
Difference matrix

The vertical and horizontal difference matrices are the basis for generating a substitute source balloon. For that the difference matrices are subtracted from the measured balloon, i.e. the horizontal difference matrix is subtracted from the values measured at horizontal rotation, and the vertical difference matrix is subtracted from the values measured at vertical rotation. The UNF file of the substitute source can be created as described above.

6.3 Verification

The thesis following from the above is: an array of 19 balloons that are based on any given balloon and from which the difference matrices gained by the spherical source experiment have been subtracted show the same dispersion characteristics in a distance of 4 m as the original balloon.

6.3.1 Verification By A Spherical Source

First, a spherical source was used for an experiment to verify the thesis. 19 point sources were generated by subtracting the difference matrices from the spherical source. They were then lined up in the center of the model described above with a distance of 10 mm from each other.

The sound pressure levels simulated by complex level summation on the listening areas in a distance of 4m are highly convergent with the simulated levels of the original source. The differences are less than 1dB as which becomes evident with a 1 dB resolution: the listening areas are all shown in the same color.

This means that the linear array of 19 point sources from which the difference matrices have been subtracted exactly replicate the directivity of a spherical source in a distance of 4m. It follows that a spherical source can be substituted by multiple point sources that have been generated from the spherical source balloon.

6.3.2 Verification By A Line Array Module

The thesis, however, states that any given balloon can be used to generate the 19 point sources. For that reason the difference matrices gained from the spherical source experiment were subtracted from the balloon of the dV-DOSC module measurements in the next step. The sources were then lined up.

The resulting sound pressure levels are shown in the plots below. Since the simulation provides results for the octave bands only the data between the octaves have been interpolated to facilitate comparing the measuring values and the simulated results.

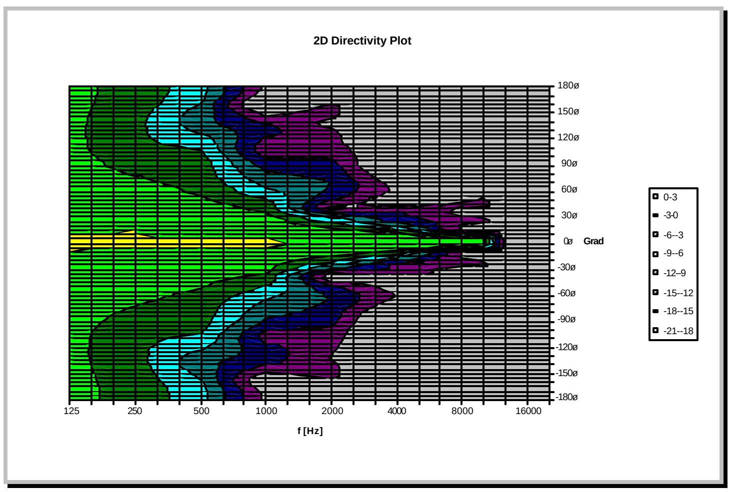


Illustration 6.14
The plot shows the measured vertical directivity of a dV-DOSC module in a distance of 4 m. The data between the octaves have been interpolated to facilitate comparison.

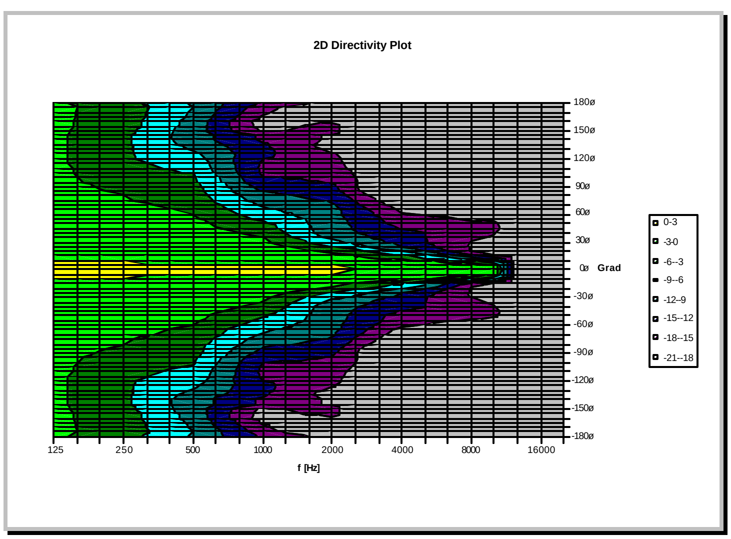


Illustration 6.15
The plot shows the simulated vertical directivity of a dV-DOSC module in a distance of 4 m. The simulation data result from the 19 substitute sources.

This comparison also shows that the simulated data and the predetermined data tally to a large extent. It proves that a simulated line array module which has been substituted by multiple sources very closely replicates the directivity of the actual line array module in a distance of 4 m.

6.3.3 Verification By Line Array Configurations

Since a line array is no longer defined as a single point source but rather as a source that is expanded in space it should be possible to achieve coherent coupling of the modules of a typical line array configuration.

For that purpose two line array configurations of four dV-DOSC modules each with 0° and 7.5° angles between modules were simulated.

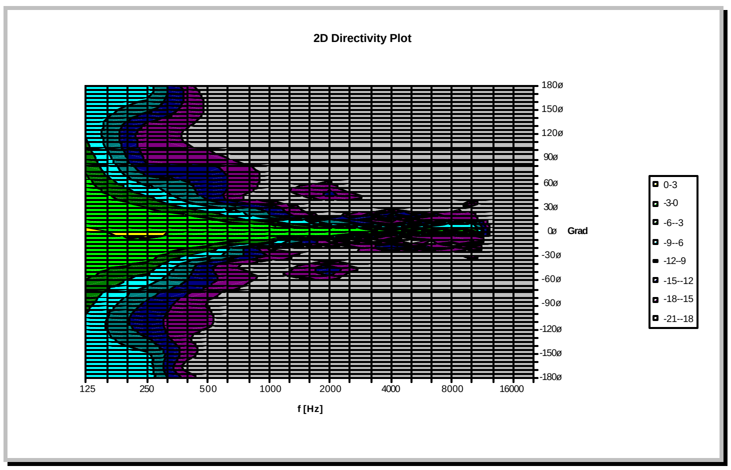


Illustration 6.16
The plot shows the measured vertical directivity of a dV-DOSC module in a distance of 4 m. The angle between the array modules is 0°. The data between the octaves have been interpolated to facilitate comparison.

Line Array Directivity Plots

These plots cannot show line array directivity correctly. It must be noted that line array directivity is highly dependent on distance. Therefore such plots are valid only up to a certain distance from the array.
Also, the results of line array measuring in the nearfield cannot be compared to those of point sources since the coverage angle of line sources in the nearfield is 0°.

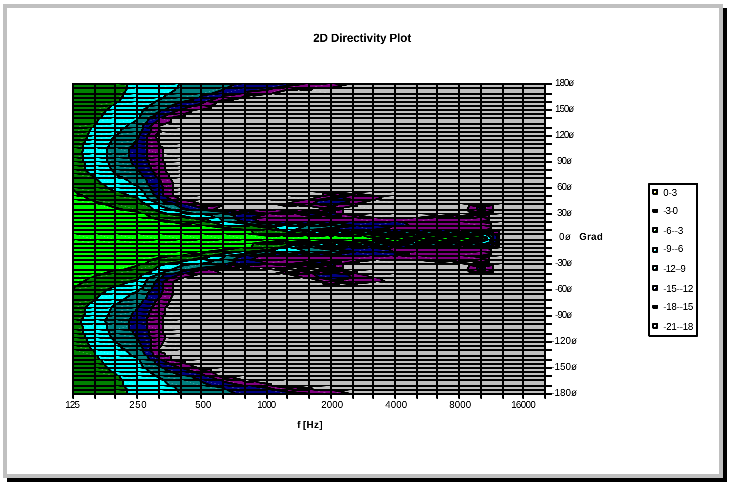


Illustration 6.17
The plot shows the simulated vertical directivity of a dV-DOSC module in a distance of 4 m. The angle between the array modules is 0°. The simulation data result from a total of 76 substitute sources.

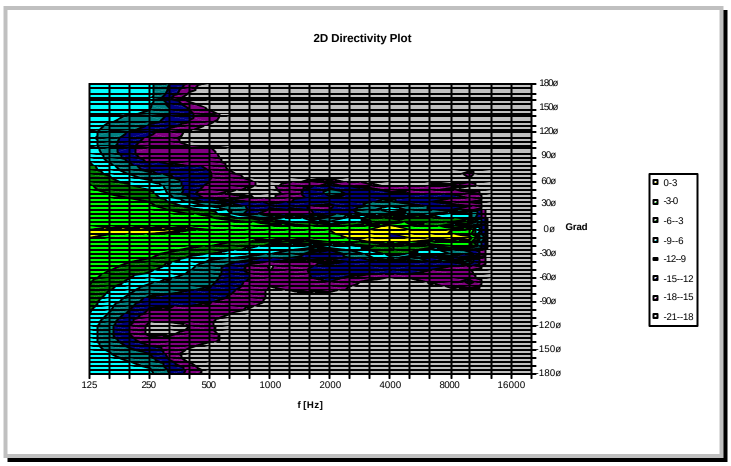


Illustration 6.18
The plot shows the measured vertical directivity of a dV-DOSC module in a distance of 4 m. The angle between the array modules is 7.5°. The data between the octaves have been interpolated to facilitate comparison.

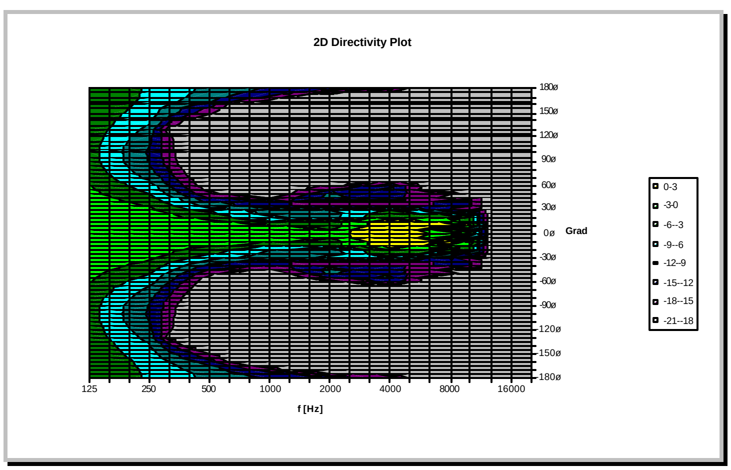


Illustration 6.19
The plot shows the simulated vertical directivity of a dV-DOSC module in a distance of 4 m. The angle between the array modules is 7.5°. The simulation data result from a total of 76 substitute sources.

These comparisons also show that the simulated data and the measured data are highly convergent. The coherent coupling of the individual models is clearly evident since the simulated results exactly replicate the coherent wavefront of the actual array.

However, the simulation results show noticeably increasing levels in the low and mid frequency range at angles above +90° and –90° which are not evident in the measuring results of the actual array. This cannot be explained without further efforts. In reality a possible reason might be loudspeaker enclosure influence at the lower frequencies which would indicate a simulation deficiency.

By researching arrays with differing heights appropriate high-pass filters depending on array height might be implemented, however, this exceeds the scope of this study.

Another possible explanation for the differing simulation and measuring results might be the fact that the measuring points for the low and mid frequencies at the given array height of approximately 90cm are located in the nearfield already. This would also require further research.

Nevertheless accurate results can be expected when simulating direct sound dispersion since the loudspeaker axis is aiming at the listening area in common simulation situations. The energy radiated from the back of a loudspeaker is of importance only for raytracing and where reverberated sound pressure levels are relevant.

7 Epilogue

7.1 Summary

This study has examined to which extent modern line array loudspeaker systems can be simulated with the Ulysses CAAD software. The chosen approach was to separate the line array into a sufficient number of individual point sources.

Experimental research has shown that it is basically possible to simulate line sources with Ulysses using this approach. Theory and simulation have always been convergent.

Furthermore a method has been developed to simulate actual line arrays. From the measured balloon data of an individual line array module sources were generated which had the same dispersion characteristics as the original line array module when aligned in linear fashion.

The linear substitution of an individual line array module made it possible to simulate complete line array configurations by extending the line length. Simulation data and measured data were largely one and the same. With this method it should be possible to simulate all kinds of line array configurations with any height and angle between modules.

7.2 Annotations

For further verification several line array configurations were measured in full space with a level meter, and a temporary array installation was measured in a hall with the MLSSA system. Unfortunately these measurements did not yield relevant results for comparison with a Ulysses simulation of those arrays.

I wish to extend my cordial thanks to Dr. Anselm Goertz for the measuring results of the examined loudspeakers and Joachim Birner and Jakob Kraft for their support during the verification of the results.

I am especially grateful to IFB for their continuous support throughout this study.

8 Literature

[1] Crawford, S. Jr.: Schwingungen und Wellen. Vieweg-Verlag

[2] Meyer, E. & Neumann, E.-G.: Physikalische und technische Acoustics. Vieweg-Verlag

[3] Goertz, A.: L-Acoustics dV-DOSC. Production Partner 2/2001, S. 126-136

[4] Ureda, M.: Line Arrays: Theory and Application. 110th AES Convention Amsterdam 2001

[5] Ureda, M.: „J“ and „Spiral“ Line Arrays. 111th AES Convention New York 2001

[6] Engebretson, M.: Directional Radiation Characteristics of Articulating Line Array Loudspeaker Systems. 111th AES Convention New York 2001

[7] Urban, M.; Heil, C.; Bauman, P.: Wavefront Sculpture Technologie. 111th AES Convention New York 2001

[8] Goertz, A.: Theoretische Grundlagen und die praktische Anwendung von Line arrays in der Beschallungstechnik und ihre Berücksichtigung in Simulationsprogrammen. CAVIS 2002

Illustration Sources

Electro Voice	5.5
IFB	4.1
JBL	5.4
L-Acoustics	2.2, 2.4, 2.8, 5.6, 5.7, 5.15, 6.1
LLW Insider	2.1, 2.3
McCauley	Title, 5.13
Meyer Sound	6.2
NEXO	5.1, 5.2, 5.8, 5.9, 5.12
Pat Brown	6.4
SLS Loudspeakers	5.11
Trius	5.3
Visaton	5.10

Appendix A

Geo S805 Line Array Simulation

In addition to the L-Acoustics dV-DOSC module measurements there was a sudden opportunity to measure a NEXO Geo S805 line array for similar verifications. The results are shown below.

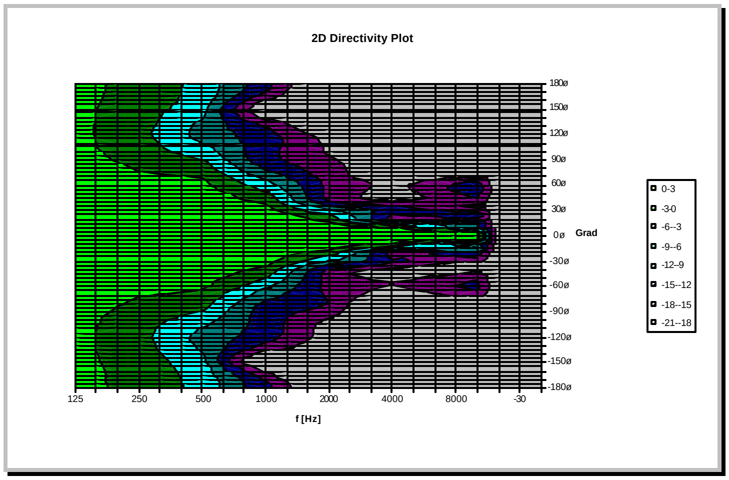


Illustration A1
The plot shows the measured vertical directivity of a Geo S805 module in a distance of 4 m. The data between the octaves have been interpolated to facilitate comparison

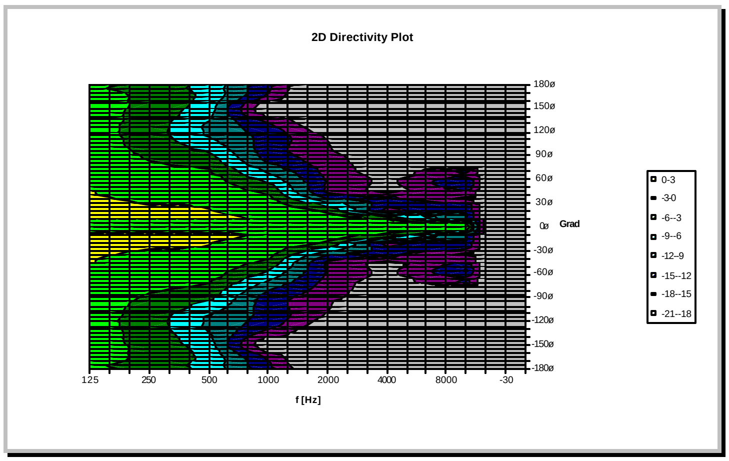


Illustration A2
The plot shows the simulated vertical directivity of a Geo S805 module in a distance of 4m. The simulation data result from 19 substitute sources.

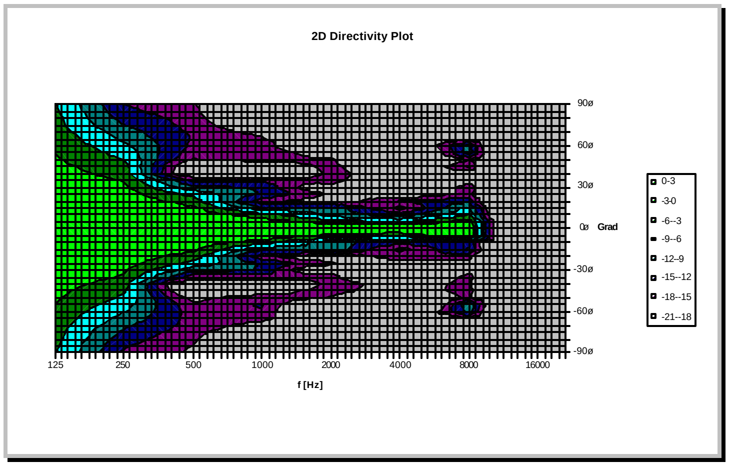


Illustration A3
The plot shows the measured vertical response of a Geo S805 array in a distance of 4 m. The angle between modules is 0°. The data between the octaves have been interpolated to facilitate comparison.

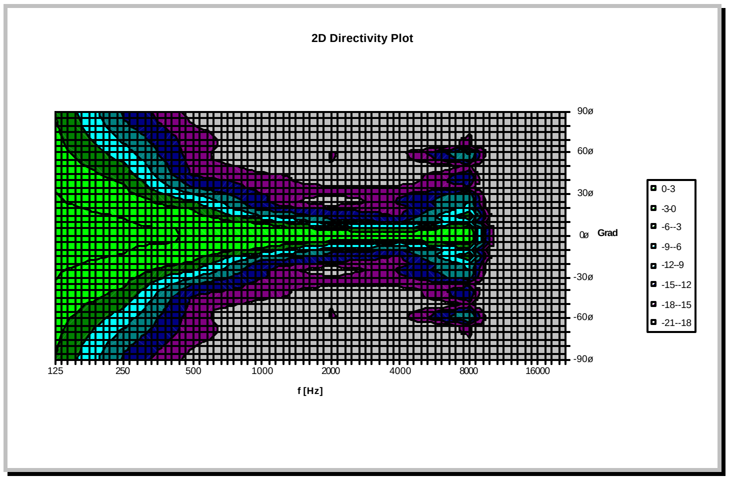


Illustration A4
The plot shows the simulated vertical directivity of a Geo S805 array in a distance of 4 m. The angle between modules is 0°. The simulation data result from a total of 76 substitute sources.

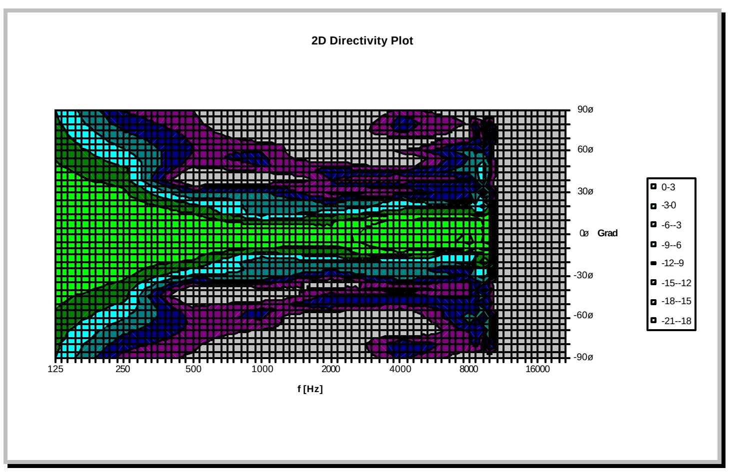


Illustration A5
The plot shows the measured vertical directivity of a Geo S805 array in a distance of 4 m. The angle between modules is 5°. The data between the octaves have been interpolated to facilitate comparison.

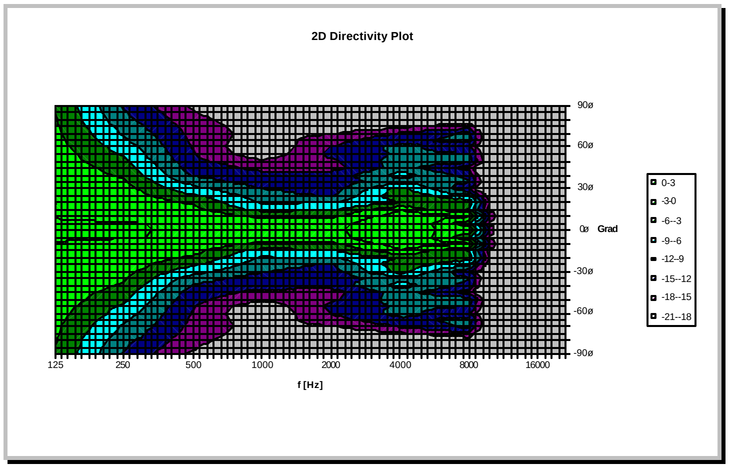


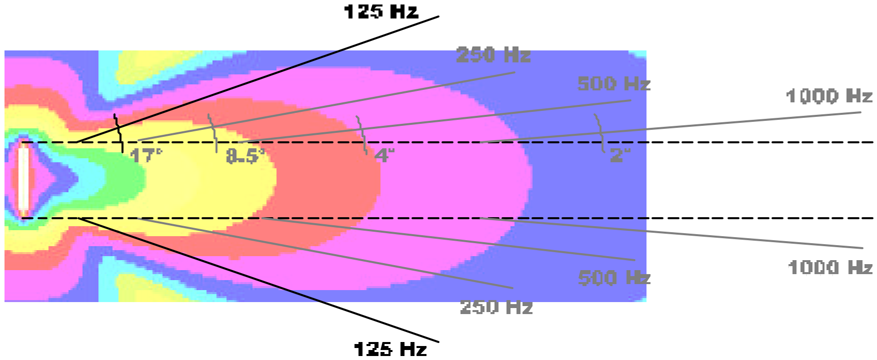
Illustration A6
The plot shows the simulated vertical directivity of a Geo S805 array in a distance of 4 m. The angle between elements is 5°. The simulation data result from a total of 76 substitute sources.

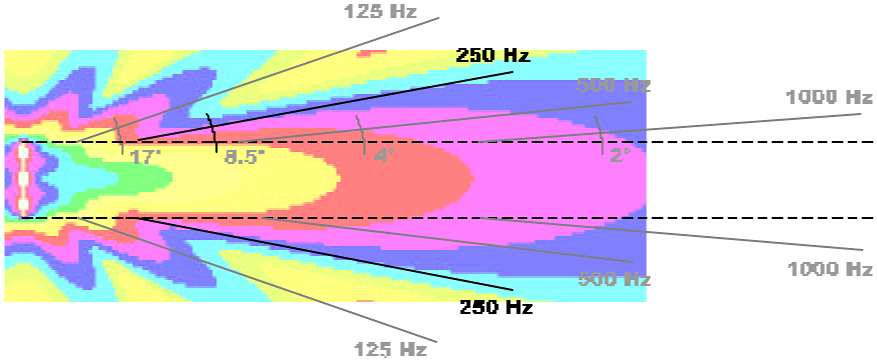
The results shown in the plots above are basically similar to those of the dV-DOSC array. In the individual simulated module plot a small discrepancy is noticeable slightly off axis in the low frequency range. A slight level increase of up to 0,35 dB at 0° results in the next higher color value.

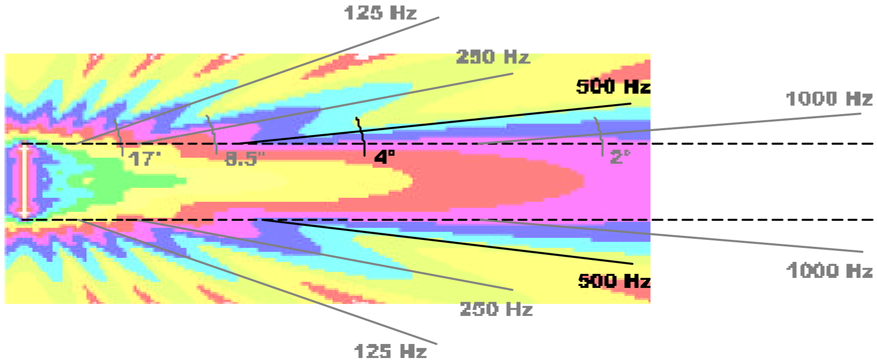
Appendix B

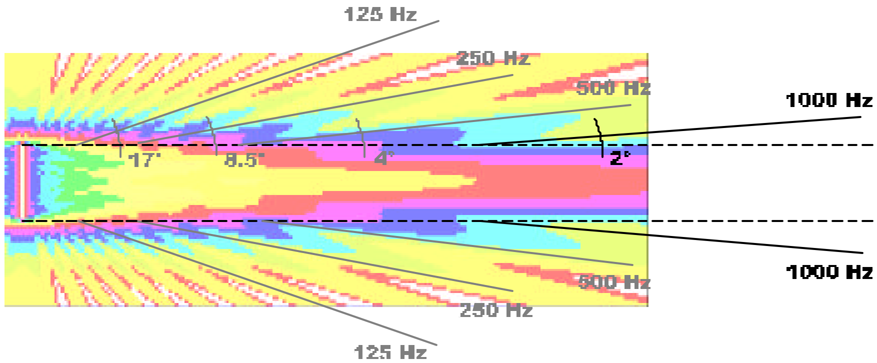
Coverage Angle

The illustrations below show the seven octaves of a 4 m line source simulation and their corresponding coverage angles.



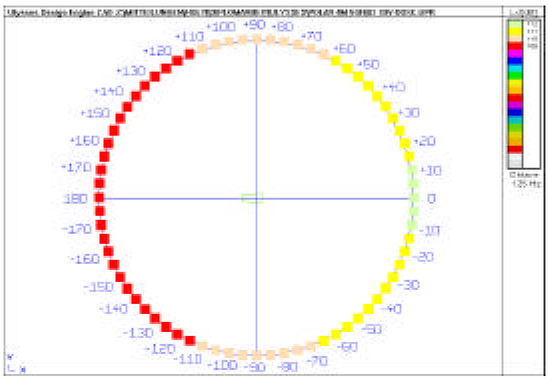
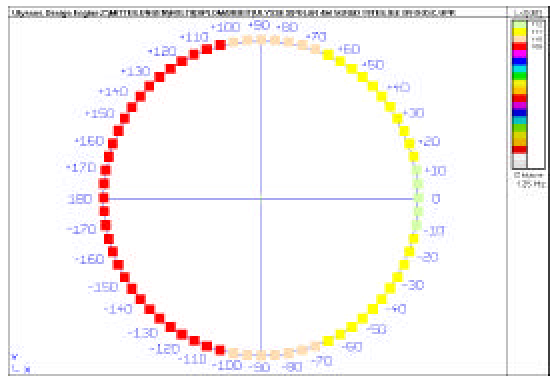
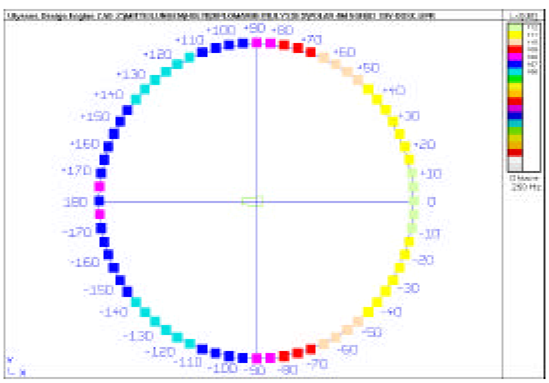
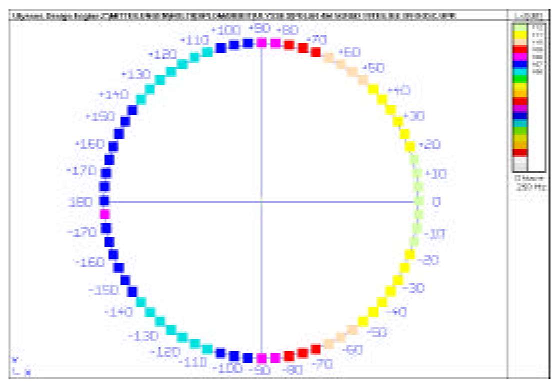
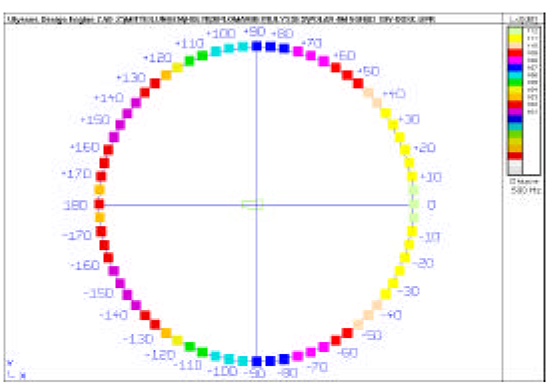
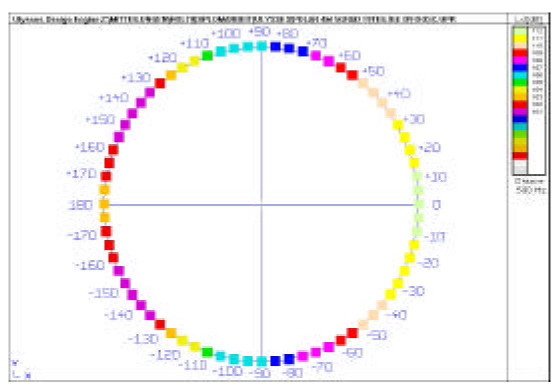


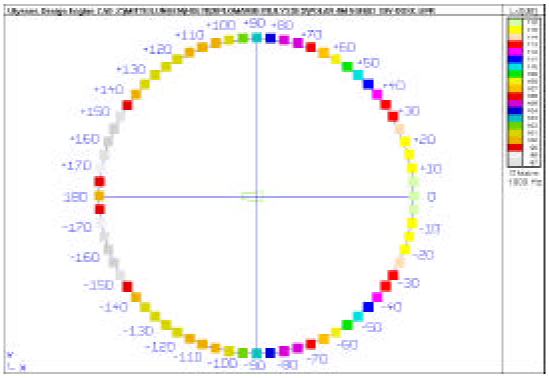
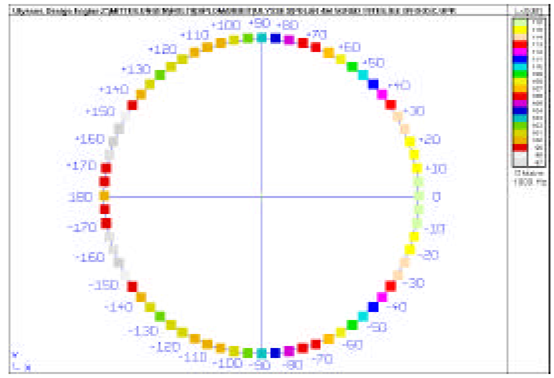
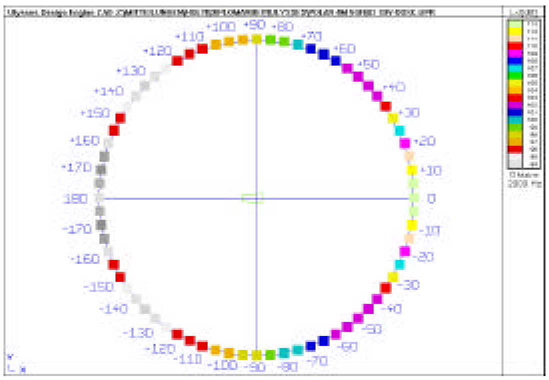
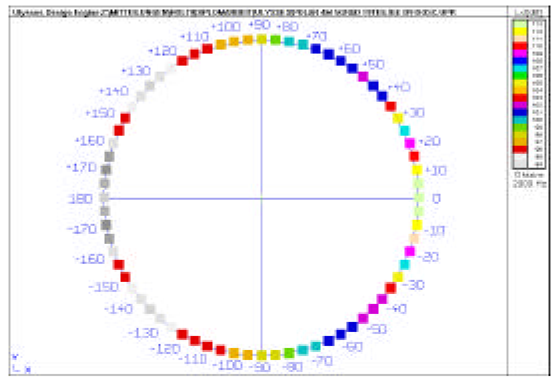
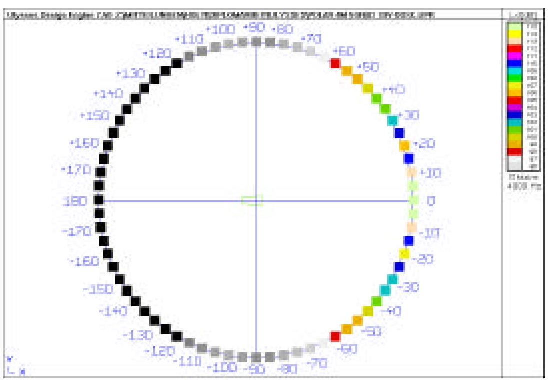
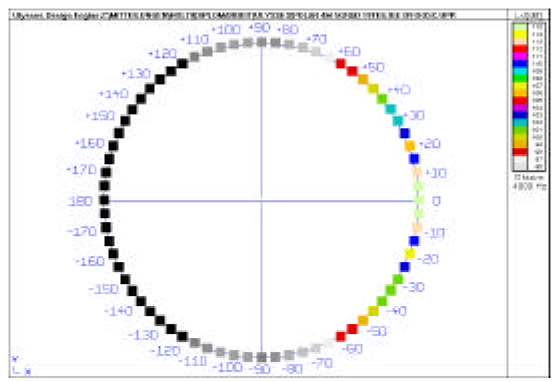
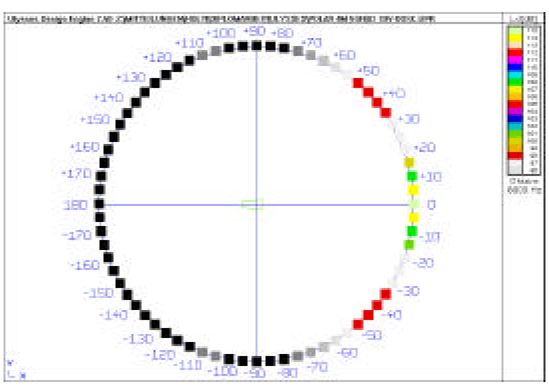
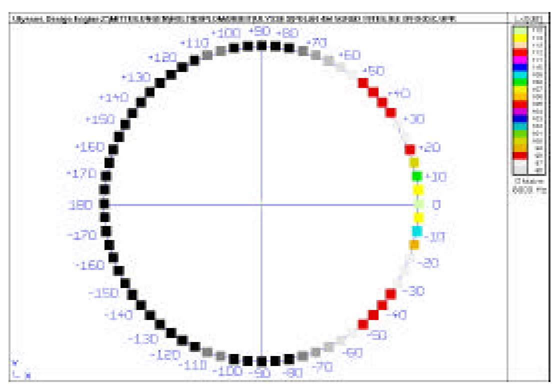




Comparing Meassurements And Simulation

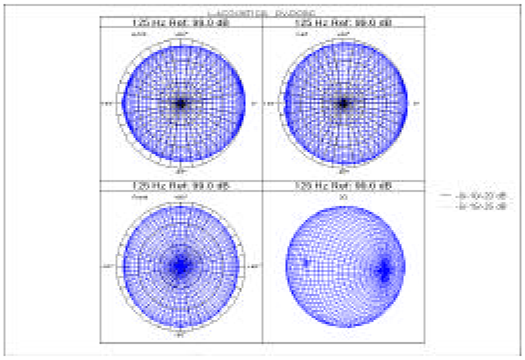
The following simulations show the measured sound pressure levels of a dV-DOSC module and the substitute sources. The sound pressure levels in a distance of 4m are shown color coded in 5° steps with a resolution of 1dB. The left column shows the measured data of the seven octaves, the right column shows the simulated values of the 19 substitute sources.

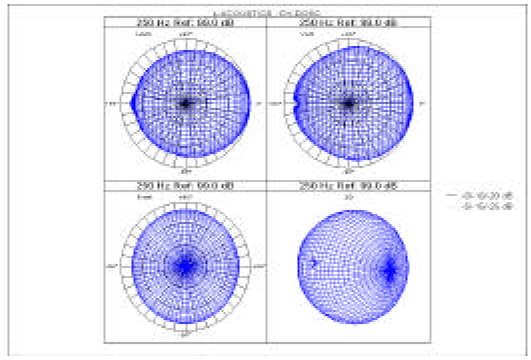







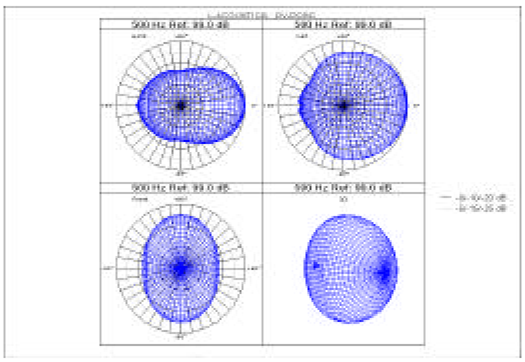









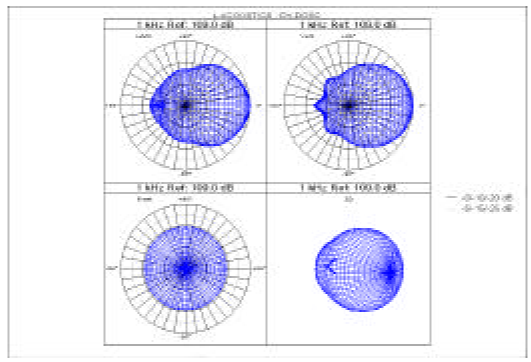
dV-DOSC Balloon Data

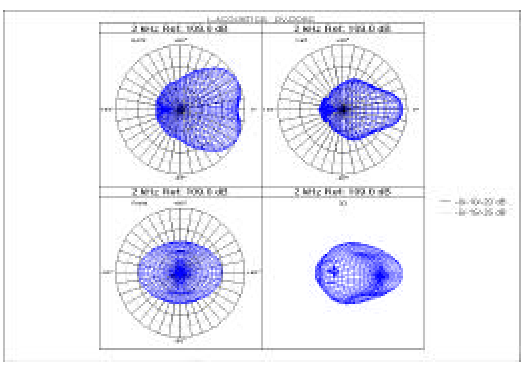
The plots below show the dV-DOSC module balloons of the seven octaves which have been generated directly from the measured data.

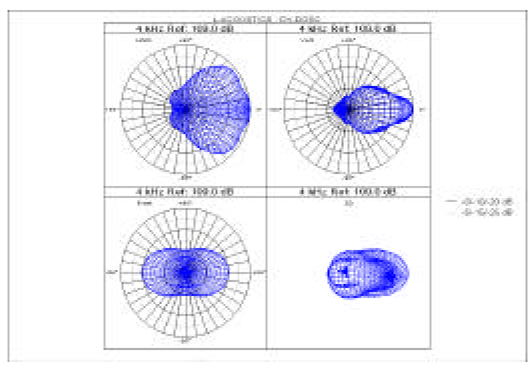


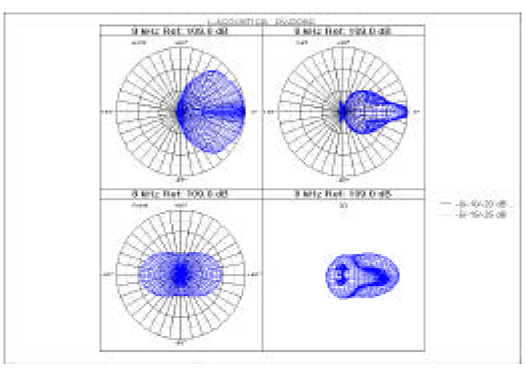




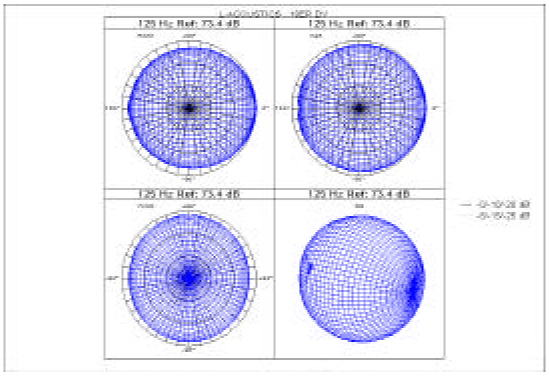


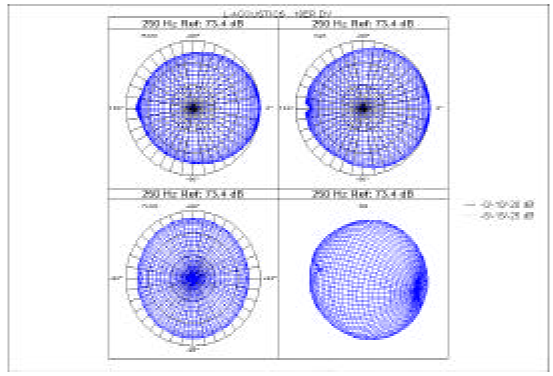


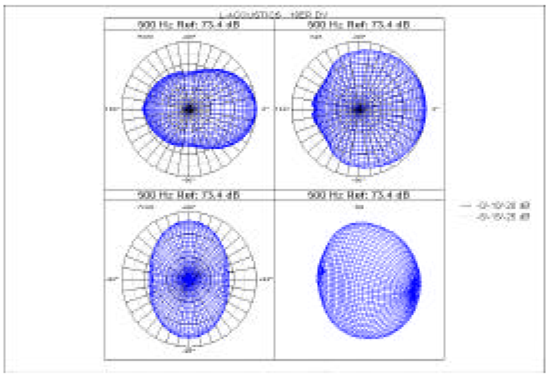


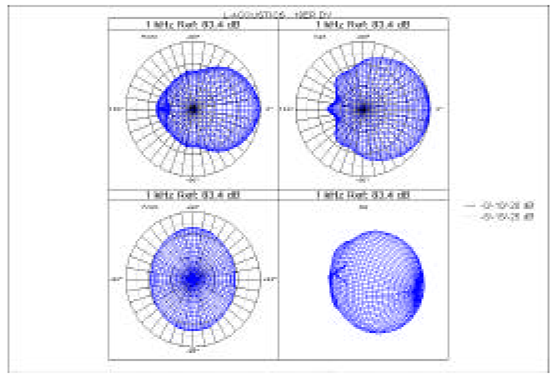


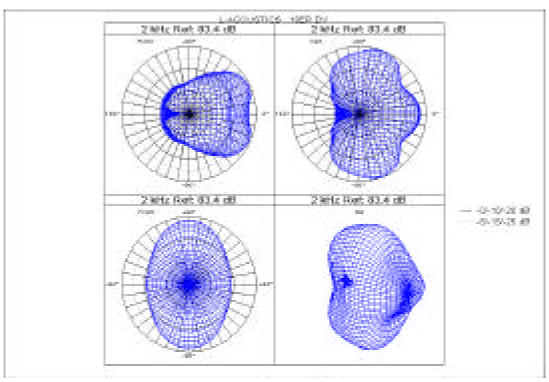
The plots below show the substitute source balloons of the seven octaves.

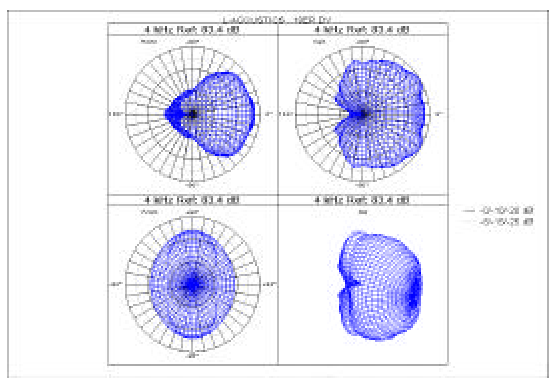


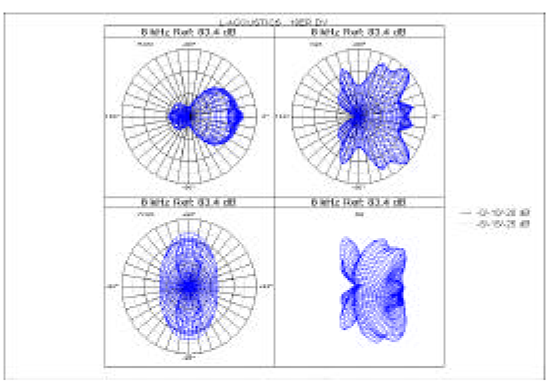












Appendix C

L-ACOUSTICS

dV-DOSC™

THE INCREDIBLE INNOVATION OF
WAVEFRONT SCULPTURE TECHNOLOGY®

APPLICATIONS

L-ACOUSTICS® dV-DOSC™ provides the same performance benefits as V-DOSC® and ARCS® in an extremely compact format suitable for both fixed installation and touring applications. The 120° horizontal coverage of dV-DOSC combined with the power of Wavefront Sculpture Technology® in adapting vertical system directivity to match the audience area results in well-defined, predictable coverage combined with exceptionally even frequency response and SPL.

As a full-range, two-way system, dV-DOSC can be used for speech reinforcement in corporate applications or as a voice-only system for television and theatrical productions. The compact profile of dV-DOSC is ideal for installations where visually unobtrusive sound design is an important issue.

dV-DOSC is also well-suited to large-scale fixed installations such as single or multiple distributed arrays for stadium and arena sound reinforcement. For these applications, the generous 120° horizontal pattern combined with the seamless transition between short- to long-throw zones obtained using Wavefront Sculpture principles allows the sound designer to achieve excellent intelligibility and cost-effective coverage even under difficult, highly reverberant acoustic conditions.

When combined with subwoofers for extended bandwidth applications, dV-DOSC can be used as a front-of-house system for small, medium and large venues. For these applications, the 120° horizontal coverage pattern provides excellent stereo imaging in the standard left-right format while the flexibility provided by Wavefront Sculpture allows the sound designer to cover virtually any room geometry.

For touring applications, dV-DOSC can be used as a down-fill enclosure for flying underneath V-DOSC arrays or as a long-throw extension of the system when stacked on top of the flying bumper. Other applications include flown dV-DOSC arrays for center cluster or offstage fill, and stacked configurations for stereo front-fill. When operated in conjunction with subwoofers, dV-DOSC can also be used for stage monitoring applications.

Given all these possibilities, that's why the "dV" in dV-DOSC stands for "Definitely Versatile".

L-ACOUSTICS PROFESSIONAL SOUND SYSTEM

- **WST-based active two-way enclosure**
- **Perfect coupling, predictable coverage**
- **Ergonomic, fast, secure rigging system**
- **Medium- and long-throw applications**
- **Excellent speech intelligibility**
- **Extremely compact**
- **Trapezoidal design for curved vertical arrays**
- **Highly versatile for both installation and touring**
- **Perfect for corporate, theatre, club and concert reinforcement**

SPECIFICATIONS

L-ACOUSTICS specifications are based on measurement procedures which produce unbiased results and allow for realistic performance prediction and simulations. Some of these specifications will appear very conservative when compared with other manufacturer's specifications. All measurements are conducted under free field conditions and scaled to a 1 m reference distance unless otherwise indicated.

Frequency Response

Frequency response	160 - 18k Hz (±3dB)	single unit
	100 - 18k Hz (±3dB)	coupled array
Full system bandwidth[1]	25 - 18k Hz (±3dB)	

Sensitivity[2]

LF (2.83 Vrms @ 1m)	99 dB SPL	100 - 800 Hz
HF (2.83 Vrms @ 1m)	109 dB SPL	800 - 18k Hz

Power Rating[3] (Long Term)			**Amplification** (Recommended)	**Impedance** (Nominal)
LF 49 Vrms	300 Wrms	1200 Wpeak	600 W	8 ohms
HF 25 Vrms	75 Wrms	300 Wpeak	300 W	8 ohms

Nominal Directivity (-6dB)[4]

Horizontal	symmetrical	120°
Vertical	defined by the array	

Array[5]	**Continuous SPL (flat array)**	**Continuous SPL (maximum curvature array)**	
One enclosure	128 dB	128 dB	
Two enclosures	134 dB	132 dB	15° vertical coverage
Four enclosures	140 dB	136 dB	30° vertical coverage

Components

LF 2 x 8" weather-resistant loudspeaker (2" voice coil)

HF 1 x 1.4" neodymium compression driver mounted on custom DOSC waveguide and acoustic lens

[1] Full system bandwidth with SB218 subwoofer

[2] Sensitivity is the average SPL measured over the component's rated bandwidth

[3] Power rating displays the long term RMS voltage handling capacity using pink noise with a 6 dB crest factor over the component's rated bandwidth

[4] Directivity is averaged over the 1-10 kHz range

[5] Array data gives the continuous unweighted SPL output of the system referenced to 1 m, including preset equalization and band-leveling adjustment using pink noise with a 6 dB crest factor over the system's rated bandwidth

Enclosure

• Width	695 mm	27.4 in
• Front height	257 mm	10.1 in
• Rear height	171 mm	6.7 in
• Depth	476 mm	18.7 in
• Trap angle	2 x 3.75°	
• Shipping dims	800 x 360 x 560 mm	31.5 x 14.2 x 22 in
• Weight (net)	31.8 kg	70.1 lbs
• Shipping weight	35 kg	77.2 lbs

- Connectors : 2x 4-pin Neutrik speakon
- Material : Baltic birch plywood, aluminum top and bottom plates
- Finish : Maroon-gray™
- Grill : Black epoxy-coated perforated steel with acoustically-transparent foam
- Rigging : Integrated flying hardware and handles

Additional Equipment

- L-ACOUSTICS approved digital crossover with custom presets
- L-ACOUSTICS SB218 subwoofer
- L-ACOUSTICS LA 24 or LA 48 power amplifier

L-ACOUSTICS®, ARCS®, V-DOSC® and Wavefront Sculpture Technology® are registered trademarks

ARCHITECT SPECIFICATIONS

The loudspeaker shall be a full-range active two-way enclosure covering the frequency range of 160 Hz - 18 kHz (± 3 dB). For coupled arrays of more than six elements, low-frequency response shall extend to 75 Hz / 100 Hz depending on preset. The loudspeaker shall be used with an approved digital crossover and dedicated software presets. The loudspeaker shall function as either a standalonesystem for speech reinforcement applications or be used in conjunction with subwoofers for extended bandwidth operation.

The loudspeaker enclosure shall consist of two high-efficiency, high power- handling 8-inch speakers mounted in V-shaped configuration combined with a 1.4-inch neodymium compression driver coupled to a waveguide. The waveguide employed in the loudspeaker system shall generate a flat isophasic wavefront. Components shall be configured in a coplanar symmetrical arrangement and provide stable 120-degree horizontal coverage independent of the number of vertically arrayed elements.

When vertically arrayed, multiple enclosures shall function according to the principles of Wavefront Sculpture Technology, whereby the distance of separation between acoustic centers of individual sound sources shall be less than the size of half the wavelength at the highest frequency of its operating bandwidth, or the sum of the individual areas of the isophasic radiating elements shall be greater than 80 percent of the target radiating area.

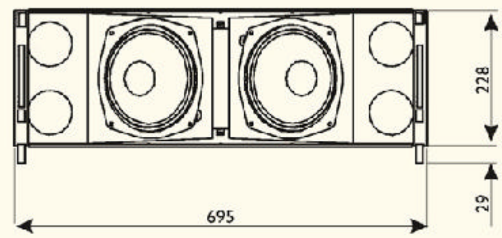
When multiple loudspeakers are arrayed vertically, they shall radiate a wavefront that varies from flat to a maximum of 7.5 degrees of curvature per element with optimized high-frequency coupling performance that is free of destructive interference effects. When installed according to Wavefront Sculpture Technology principles, the combination of cylindrical wavefront generation produced by the waveguide and proper focus of the arrayelements shall allow the system to produce 1/R attenuation properties(-3 dB per doubling of distance).

The loudspeaker shall have low profile, trapezoidal cabinet construction and an integral rigging system that allows arrays to be assembled with variable angles between enclosures up to a maximum of 7.5 degrees. Cabinet finish shall be maroon-gray, high-resilience paint and all external hardware shallbe stainless steel or black powder-coated to protect against rust.The front of the enclosure shall be covered with open-cell, acoustically-transparent foam.

The loudspeaker system shall be the L-ACOUSTICS dV-DOSC.

ACCESSORIES

dV-PIN25	Locking quick release pin (25 mm grip length) for dV-DOSC enclosure interconnection and connection to dV-BUMP
dV-PIN81	Locking quick release pin (81 mm grip length) for attaching dV-DOSC to dV-DOWN
dV-ANGLEP	Rear angle bar for varying vertical angle between dV-DOSC enclosures to form convex arrays (positive curvature) dV-ANGLEP1 values: 0, 2, 3.75, 5.5, 7.5 degrees dV-ANGLEP2 values: 1, 3, 4.5, 6.5 degrees
dV-ANGLEN	Rear angle bar for varying vertical angle between dV-DOSC enclosures to form concave arrays (negative curvature)
dV-BUMP	Flying bumper for standalone rigging or flying on top of V-DOSC
dV-DOWN	Flying bars (two) for rigging dV-DOSC under V-DOSC for down-fill applications
dV-FLIGHT	Flight case for three dV-DOSC



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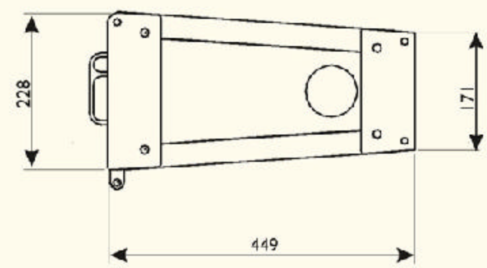
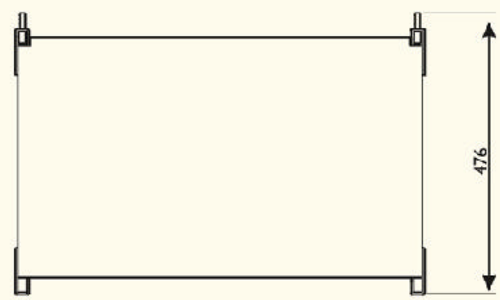
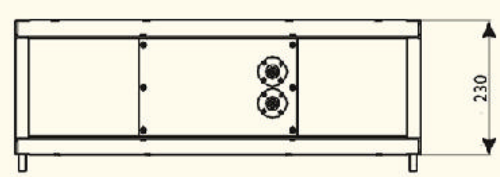


SIDE



TOP



REAR

Features & Specifications for GEO S8-Series and CD12 Sub

geo

The new shape of sound

SYSTEM SPECIFICATIONS	GEO S805 / S830 with NX241 TDcontroller	CD12 Sub with NX241 TDcontroller
Frequency Response [a]	67 Hz – 19 kHz ± 3 dB	42 Hz – 200 Hz
Usable Range @-6dB [a]	60 Hz – 20 kHz	39 Hz – 250 Hz
Sensitivity 1W @ 1m [b]	99 dB SPL Nominal – 97 dB SPL Wideband	102 dB SPL Nominal
Peak SPL @ 1m [b]	125 to 128 dB Peak for a single cabinet. Configuration Dependent when arrayed [d]	131 to 134 dB Peak (500 to 1200 W RMS Amp)
Dispersion [c]	GEO S805: Coupling Plane: Not useable as a single cabinet Configuration Dependent [d] Non Coupling Plane: 120° (Configurable to 80°) GEO S830: Coupling Plane: 30° for a single cabinet. Configuration Dependent when arrayed [d] Non Coupling Plane: 120° (Configurable to 80°)	SuperCardioid pattern 120°x120° over the entire useable bandwidth. Directivity Control is achieved through DSP algorithms in the NX241 (two channels of NX241 are dedicated to the process).
Directivity Index [c]	GEO S805: Not useable as a single cabinet. Configuration Dependent [d] GEO S830: DI = 12 Nominal (f > 1.5 kHz) for a single cabinet. Configuration dependent when arrayed. [d]	Q=3.73 DI = 5.7dB over the entire useable bandwidth
Crossover Frequency	1.8kHz Passive	150 Hz Active through NX241 TDcontroller
Nominal Impedance	16 Ohms	6 Ohms
Recommended Amplifiers	1500 to 3000 W into 4 Ohms / 4 cabinets per channel. Up to 6 cabinets per channel may be connected to large amplifiers capable of operating into low loads.	2 amplifier channels required for SuperCardioid operation. Same amplifier model as for the GEO S8 main system. Up to two complete CD12 can be connected on a 2-channel amplifier.
PRODUCT FEATURES	**GEO S805 / S830**	**CD12 Sub**
Components	LF: 1 x 8" (20cm) Neodymium Hi-Flux 16 Ohms Driver HF: 1 x 1" Throat ND Driver + HRW™	2 x 12" (30 cm) Long Excursion 6 Ohms Neodymium Drivers
Height x Width x Depth	406 x 250 x 219 mm (16" x 9 7/8" x 5 5/8")	400 x 600 x 754 mm (15 3/4" x 23 5/8" x 29 11/16")
Shape	GEO S805: 5° Trapezoid GEO S830: 30° Trapezoid	Rectangular
Weight : Net	GEO S8-Series: 10.5 kg (23 lb) GEO S8-Series with flying hardware: 14.5 kg (32 lb)	36 kg (79 lb)
Connectors	2 x NL4MP SPEAKON 4 poles (In & Through)	2 x NL4MP SPEAKON 4 poles (In & Through)
Construction	Baltic Birch Ply finish with structured black coating. Dark gray carpet finish also available	Baltic Birch Ply finish with structured black coating. Dark gray carpet finish also available
Front Finish	Perforated Steel Grille	–
Flying points	Integral flying system. Intercabinet Angle Adjustments= 0 to 5° (step 0.5°), 17.5° & 30°	Integral flying system.
SYSTEM OPERATION		
Electronic Controller	The NX241 TDcontroller presets are precisely matched to the GEO S8-Series cabinets, and include sophisticated protection systems. Using GEO S8-Series without a properly connected NX241 TDcontroller will result in poor sound quality and can damage components.	
HF Dispersion Configuration	After quick release of the front grille from its fittings, the HF Waveguide can be configured for 80° or 120° dispersion in the non-coupling plane.	
Array design	Arrays of less than 4 GEO S805 will provide poor dispersion control and are not recommended. S805 and S830 Wavefront being tangent, they can be mixed within the same array. Please refer to the user manual. Correct understanding, design and implementation of GEO arrays is paramount to ensure even coverage over the audience area.	
Sub-bass	The GEO S805 & S830 can be used without optional CD12-Sub. In this case the NX241 can be used in stereo. With the CD12-SuperCardioid Sub each Sub channel will require two NX241 outputs and the NX241 will operate in mono.	
Speaker Cables	GEO S805 & S830 are wired 1- & 1+ on Speakon connectors. 2 + & 2- are not connected. The front loudspeaker of the CD12-Sub is wired 2+ & 2- while the back loudspeaker is wired on 1- & 1+ GEO S8-Series cabinets and CD12 must use separate cables.	
Rigging system	Please refer to the GEO User Manual before any operation	

As part of a policy of continual improvement, NEXO reserves the right to change specifications without notice.

[a] Response Curves and Data : Anechoic Far Field above 200 Hz, Half-space Anechoic below 200 Hz. Usable Range Data : Frequency Response Capability with TD crossover slopes removed.

[b] Sensitivity & Peak SPL : will depend on spectral distribution. Measured with band limited Pink Noise. Refers to the specified +/- 3 dB range. Data are for Speaker + Processor + recommended amplifier combinations.

[c] Directivity Curves and Data : 1/3 octave smoothed frequency response, normalized to On-Axis response. Data obtained by computer processing on off-axis response curves.

[d] Please refer to GEO User Manual.

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