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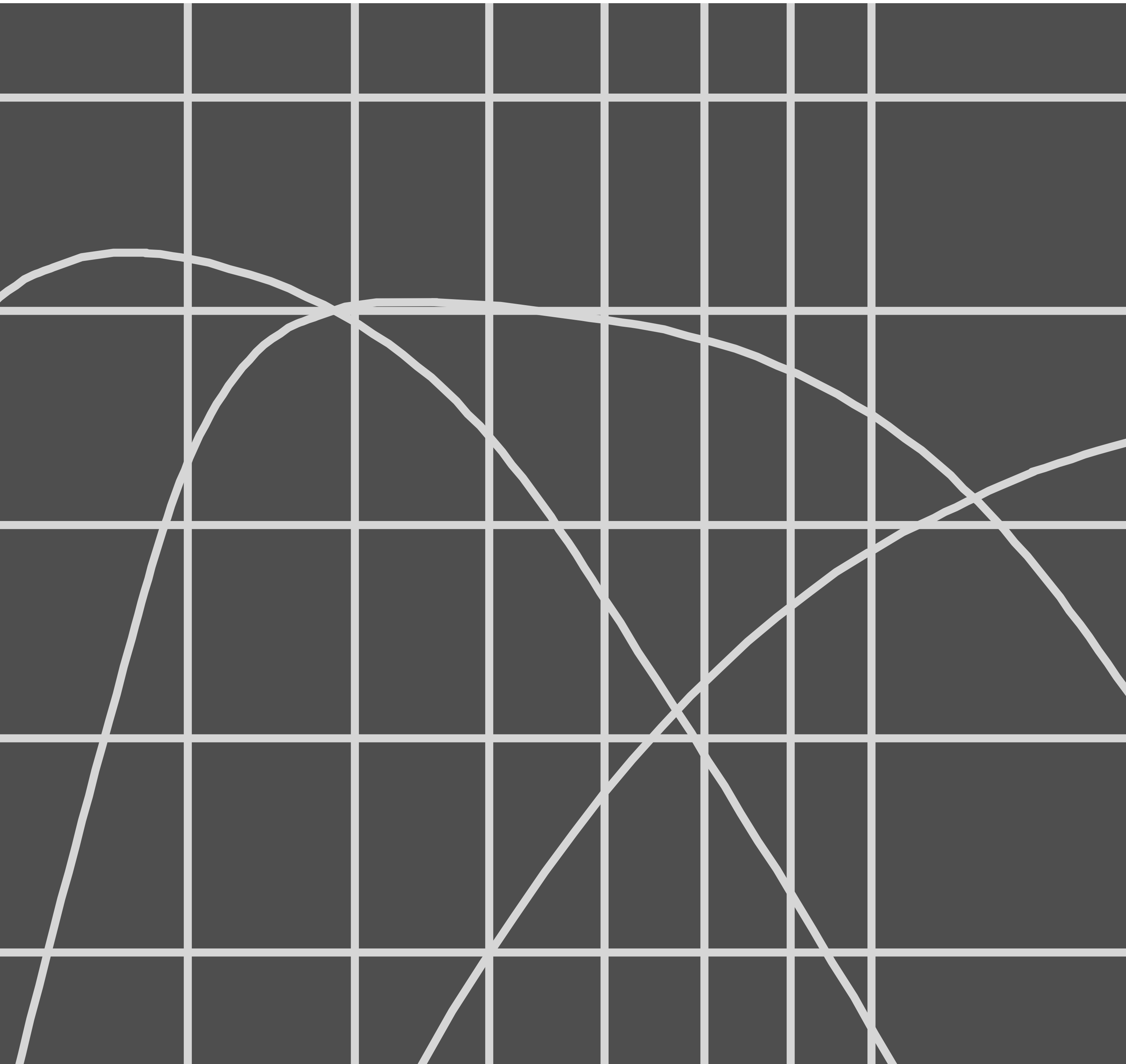
TI 385
d&b Line array design
10.9 en



Notes on document version
All previous versions of this document are hereby no longer valid.

Version 10.9:
System requirements: Mac OS aligned.

Refer to:
⇒ Chapter 1.1.1 "System requirements" on page 4.

General information

TI 385 d&b Line array design

Version: 10.9 en, 01/2023, D5385.EN .10

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Contents

1 Introduction

For both, safety and acoustic reasons, d&b line arrays must be designed using ArrayCalc.

Before setting up a system, read the respective manuals and safety instructions.

This technical information paper will explain the procedure for designing and tuning d&b line arrays, point source systems as well as column speakers from the xC-Series in a given venue using the d&b ArrayCalc simulation software.

1.1 ArrayCalc

ArrayCalc uses a sophisticated mathematical model synthesizing each line-array cabinet's wavefront using measured high-resolution dispersion data. Sound pressure level is calculated in 3D using complex data (vector summation).

ArrayCalc also provides functionality to integrate individual d&b point source loudspeakers into a simulation project. Point sources are modeled using complex measured high-resolution 3D balloon data.

ArrayCalc is available for PC and Mac and can be downloaded from the d&b website at www.dbaudio.com.

1.1.1 System requirements

PC	**Intel/AMD:** 1.5 GHz or more (Intel i5 or higher recommended) Windows 7 or higher (64-bit version recommended).
Mac	**Intel:** macOS 10.14 or later, 64-bit-processor. **Apple Silicon:** macOS 11.0 or later.
RAM	4 GB (8 GB recommended).
Graphics	OpenGL support (v2.0 or higher).
Free disk space	200 MB
Screen resolution	1920 x 1080 or higher. On viewports smaller than 1600 x 900 the viewport has to be scrolled.

1.1.2 ArrayCalc features

- Editing of three-dimensional listening planes to create audience areas in a given venue and shape.
- Help function to obtain venue dimensions using laser distance finders and inclinometers.
- Import function of venue data from an online database.
- Level distribution on all selected audience listening areas displayed as a color mapping on the planes. Calculation gives either selectable individual frequency bands from 32 Hz to 12.5 kHz or broadband linear-, A- or C- weighted summed levels using pink noise or IEC60268 spectrum as an excitation signal.
- Calculation of absolute sound pressure levels in audience areas including system headroom supervision for the selected input signals.
- Combination of up to 40 different array pairs distributed across the venue plus ground stacked subwoofers in L/R combinations or arranged as SUB array.
- Calculation of ArrayProcessing settings for line arrays.
- Flown subwoofers integrated into the line arrays or flown as separate columns.
- Additional integration of up to 30 groups of d&b point source loudspeakers.
- Integration of xC column loudspeakers.
- Supporting algorithms for vertical aiming and splay angle settings of arrays as well as SUB array settings.
- Tuning of all relevant amplifier settings like level, array coupling, crossover and cardioid modes.
- Simulation of air absorption effects depending on environmental conditions, tuning of the respective amplifier settings.
- System time alignment between different sources and subwoofers using impulse and phase response data.
- Calculation of load and space requirements for rigging points.
- Calculation and supervision of electronic and physical load conditions as well as mechanical forces within arrays.
- Design and calculation printouts, printable parts lists for inventory control and loading as well as DXF and EASE export functions.
- Project file exchange with d&b R1 Remote control software.
- Complete system design and set up toolbox for the d&b DS family of products such as DS10, DS20 or DS100 Soundscape (please also refer to TI 501, which can be downloaded from the d&b website at www.dbaudio.com).

1.1.3 Installing ArrayCalc

Do not delete any of the extracted files as long as ArrayCalc is installed on your computer.

PC

1. To install ArrayCalc, extract the zip-file to a dedicated folder.
2. Start "setup.exe" or "ArrayCalc.msi" and follow the instructions in the setup dialog.
   ↳ The default installation path is:
   `C:\Program Files\dbaudio`

Mac

To install ArrayCalc, double click the "dmg-file" and move the ArrayCalc icon to your desired application folder.

1.1.4 Starting ArrayCalc

PC

ArrayCalc can either be started via the Windows Start Menu, where it will appear in Programs ⇒ dbaudio ⇒ ArrayCalc or by double-clicking the ArrayCalc desktop icon.

Windows automatically links d&b project files (*.dbpr) to ArrayCalc. Alternatively, the program can therefore be started by double clicking on any ArrayCalc project file.

Mac

Click ArrayCalc or any ArrayCalc project file.

1.1.5 Removing ArrayCalc

PC

1. To remove ArrayCalc from your computer, go to Start ⇒ Settings ⇒ Control Panel ⇒ Add or Remove Programs in the Control Panel folder.
2. Select the ArrayCalc entry from the list and click the «Remove» button.
   ↳ The uninstall routine starts and the software is removed including all related components.

Mac

Simply move the ArrayCalc icon from your application folder into the trash bin.

1.1.6 ArrayCalc Help

Detailed information on how to use and operate ArrayCalc is provided in the help system of the ArrayCalc software.

To access the help system, press F1 or select the help button () from the ArrayCalc toolbar. This will launch the HelpViewer which provides an overview of the program as well as a search function and direct access to the related topics.

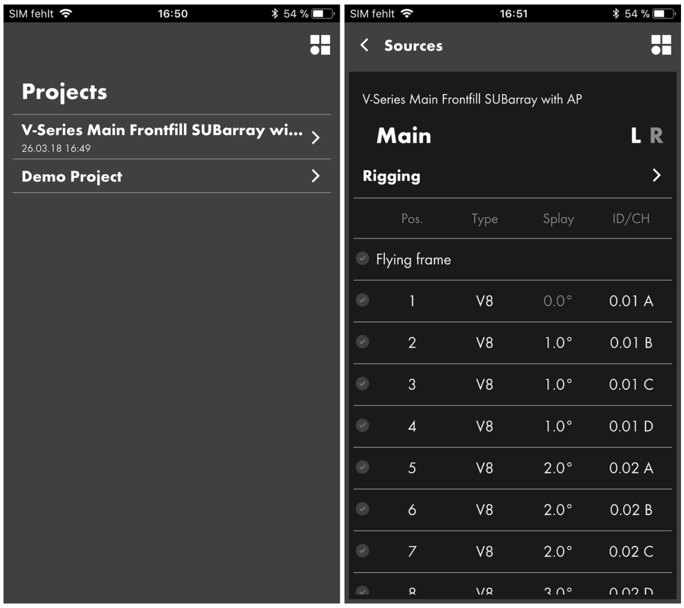
In addition, the ArrayCalc software will provide you with typical array configurations within the permitted load limits and will help you get familiar with the mechanical load conditions and limitations.

1.1.7 ArrayCalc Viewer app

ArrayCalc provides a rigging plot with all necessary coordinates, dimensions and weights of arrays and a parts list, detailing all components required. These information can be exported or printed.

However, the d&b ArrayCalc Viewer app presents this key information for positioning and flying the system on a mobile device. Once the system has been designed, calculated and optimized, ArrayCalc provides a dedicated file export (*.dbev). This file can be shared via email, AirDrop, or downloaded onto any iOS or Android device.

The app provides data such as splay angle, amplifier channel ID and cabling information, flying frame height and the height of the lowest edge, frame angle, horizontal aiming, Pickpoints, weight and load status. Any change to the system design can be distributed to each user immediately, ensuring an efficient and user friendly approach to rigging.



2 The SL-Series line array

The SL-Series is a comprehensive system containing the loudspeaker themselves with a quite unique set of features: The most noticeable is it´s well controlled dispersion over the entire audio bandwidth due to a special arrangement and processing of multiple low frequency systems. It features a multi-mode rigging system for extremely efficient and safe rigging and de-rigging procedures, complemented by frame-mounted and remote network integrated aiming laser, inclinometer, temperature and humidity sensors. Furthermore a set of matching transport and cabling solutions.

Common to all SL-Series TOPs is their full range dispersion control over the entire system bandwidth, their exceptional low frequency output capability with a high dynamic stability which results in extended system design flexibility and in most cases in a reduced number of necessary complementary subwoofers.

All SL-Series loudspeakers are operated with d&b D80 amplifiers offering d&b ArrayProcessing as standard. KSL and XSL system can alternatively be operated with d&b D40/ 40D amplifiers to their full performance including all processing functions. Two KSL loudspeakers or two XSL loudspeakers can optionally be linked in their respective Line/Arc setups.

2.1 GSL

The GSL-system consists of two fullrange loudspeakers, the GSL8 and the GSL12. The GSL8 and GSL12 are mechanically and acoustically compatible loudspeakers providing two different horizontal coverage angles of 80° and 120°. The dispersion of both systems is symmetrical and well controlled over the entire system bandwidth reaching from 45 Hz to 18 kHz.

In the vertical plane GSL8 and GSL12 produce a flat wavefront allowing splay angle settings between 7° (1° increments). An array should consist of a minimum of six cabinets – either GSL8, GSL12 or a combination of both.

The GSL8 with its 80° horizontal dispersion and very high output capability can cover any distance range up to 100 m (330 ft) depending on the vertical configuration of the array and the climatic conditions.
The GSL12 offers a wider horizontal coverage which is particularly useful for short and medium throw applications. Using a combination of GSL8 and GSL12 cabinets enables the user to create a venue specific dispersion and energy pattern.

2.2 KSL

KSL8 and KSL12 are fullrange loudspeaker systems providing horizontally symmetrical coverage angles of 80° and 120°respectively. The system bandwidth reaches from 54 Hz to 18 kHz.
In the vertical plane KSL8 and KSL12 produce a wavefront allowing splay angle settings between 0° and 10° (1° increments). An array should consist of a minimum of six cabinets – either KSL8, KSL12 or a combination of both.
The KSL8 with its 80° horizontal dispersion and very high output capability can cover any distance range of up to 100 m (330 ft) depending on the vertical configuration of the array and the climatic conditions.

The KSL12 offers a wider horizontal coverage which is particularly useful for short and medium throw applications. Using a combination of KSL8 and KSL12 cabinets enables the user to create a venue specific dispersion and energy pattern.

2.3 XSL

XSL8 and XSL12 are fullrange loudspeaker systems providing horizontally symmetrical coverage angles of 80° and 120°respectively. The system bandwidth reaches from 60 Hz to 18 kHz.
In the vertical plane XSL8 and XSL12 produce a wavefront allowing splay angle settings between 0° and 14° (1° increments). An array should consist of a minimum of four cabinets – either XSL8, XSL12 or a combination of both.
The XSL8 with its 80° horizontal dispersion and very high output capability can cover any distance range of up to 100 m (330 ft) depending on the vertical configuration of the array and the climatic conditions.

The XSL12 offers a wider horizontal coverage which is particularly useful for short and medium throw applications. Using a combination of XSL8 and XSL12 cabinets enables the user to create a venue specific dispersion and energy pattern.

2.4 SL-SUB/SL-GSUB

The SL-SUB cardioid subwoofer extends the system bandwidth down to below 30 Hz while providing exceptional dispersion control and impressive low frequency headroom paired with unmatched power efficiency. It can be deployed either flown (SL-SUB only) or ground stacked (SL-SUB or SL-GSUB) in arrays, or set up individually.

2.5 Number of cabinets required

The number of SL-Series loudspeakers to be used in an application depends on the desired level, the distances and the directivity requirements in the particular venue. Using d&b ArrayCalc will define whether the system is able to fulfill the requirements.

Depending on the program material and the desired level, additional SL-SUBs will be necessary to extend the system bandwidth and headroom.

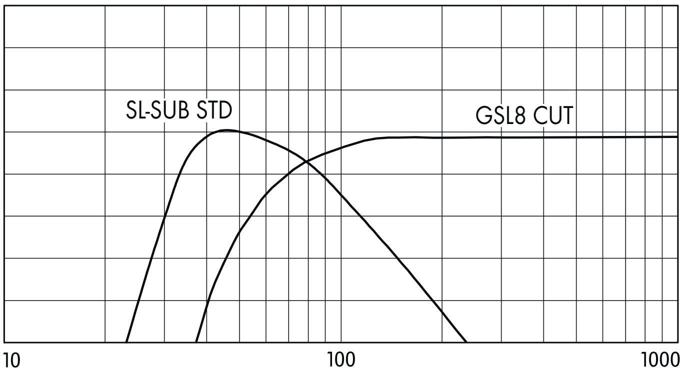
	TOP/SUB ratio	
	L/R deployment	SUB array
GSL	3:1	2:1
KSL	4:1 (SL-SUB) 3:1 (KSL-SUB)	3:1 (SL-SUB) 2:1 (KSL-SUB)
XSL	4:1 (KSL-SUB) 2:1 (XSL-SUB)	2:1 (KSL-SUB) 3:2 (XSL-SUB)

2.6 SL-SUB/SL-GSUB subwoofer setup

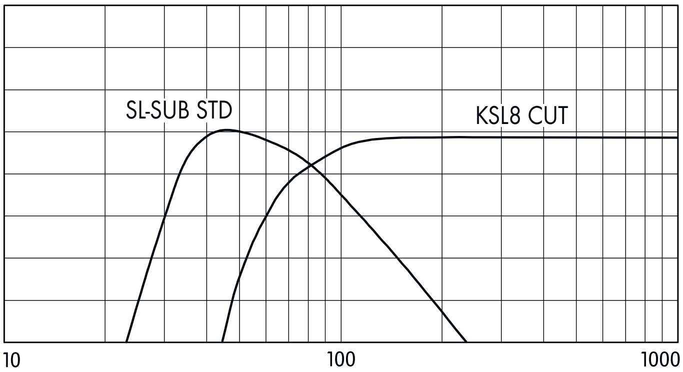
SL-SUB cabinets can be used ground stacked, as a horizontal SUB array or flown in separate columns, either beside or behind one or more columns of GSL8/GSL12 or KSL8/KSL12. SL-GSUB cabinets are acoustically identical to SL-SUB cabinets. They do not provide any rigging hardware, hence they are intended for ground stacked application only.

The physical design of the SL-SUB system together with it´s exceptional efficiency enables a frequency constant sound rejection over a wide angle range around the rear halfsphere of the system. This eliminates the need for an application specific selection of cardioid or hyper cardioid modes.

When used with additional subwoofers, the GSL8/GSL12 or KSL8/KSL12 system can be operated in CUT mode, i.e. to control the low frequency dispersion entirely from a subwoofer array.

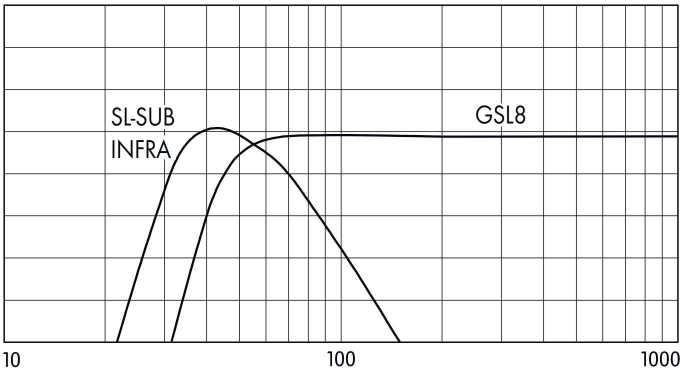


GSL8/SL-SUB crossover setup

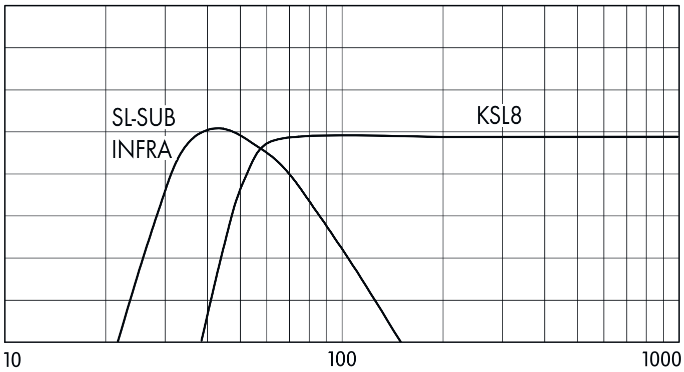


KSL8/SL-SUB crossover setup

When it is desirable to combine the low frequency output of the GSL/KSL with additional subwoofers, the GSL/KSL system can also be operated in standard mode (full range, i.e. CUT not selected). Additional SL-SUB cabinets in INFRA mode can be used to extend the system bandwidth down to 30 Hz.



GSL8/SL-SUB crossover setup, full range



KSL8/SL-SUB crossover setup, full range

2.6.1 SL-SUB ground stacks

Using SL-SUB/SL-GSUB cabinets in L/R ground stacks provides maximum system efficiency due to the ground coupling of the cabinets but will introduce power alley artefacts.

2.6.2 Flown SL-SUB columns

When complete columns of SL-SUBs are flown, the increased vertical directivity adds to the distance effect described above and thus creates a longer throw of low frequencies.

Clever positioning and horizontal aiming of flown subwoofer columns behind the main and outfill arrays of TOP loudspeakers can greatly enhance both, visual appearance and acoustic performance of the complete system through increased overall coherence between the different parts of the system.

2.6.3 SL-SUB horizontal SUB array

Arranging SL-SUBs in a horizontal array (SUB array) provides the most even horizontal coverage eliminating the cancellation zones to the left and right of the center of a typical L/R setup.

The correct alignment of both the array dispersion and the associated array-internal delay settings as well as the alignment to other parts of the system (main arrays etc.) is performed using d&b ArrayCalc.

2.6.4 SL-SUB comination with other d&b systems

SL-SUBs can be used as an efficient low frequency enhancement with any other d&b line array or point source system. It´s high output capability and it´s exceptional efficiency typically allows a reduction of the number of SL-SUBs in respect to a previous system design.

3 The J-Series line array

The J-Series consists of four different loudspeakers: the J8 and J12 loudspeakers and the J-SUB and J-INFRA subwoofers. The J8 and J12 are mechanically and acoustically compatible loudspeakers providing two different horizontal coverage angles of 80° and 120°. The dispersion of both systems is symmetrical and well controlled to frequencies down to 250 Hz, their bandwidth reaching from 48 Hz to 17 kHz.

J-Series loudspeakers can be operated with d&b D12, 30D or D80 amplifiers. With D80 and 30D amplifiers d&b ArrayProcessing is available.

In the vertical plane J8 and J12 produce a flat wavefront allowing splay angle settings between 0° and 7° (1° increments). An array should consist of a minimum of six cabinets - either J8, J12 or a combination of both.

The J8 with its 80° horizontal dispersion and high output capability can cover any distance range up to 100 m (330 ft) depending on the vertical configuration of the array and the climatic conditions.

The J12 offers a wider horizontal coverage which is particularly useful for short and medium throw applications. Using a combination of J8 and J12 cabinets enables the user to create a venue specific dispersion and energy pattern.

The J-SUB cardioid subwoofer extends the system bandwidth down to 32 Hz while providing exceptional dispersion control either flown or ground stacked in arrays, or set up individually.

The J-INFRA cardioid subwoofer is an optional extension to a J8/J12/J-SUB system. It is used in ground stacked configurations and extends the system bandwidth down to 27 Hz while adding impressive low frequency headroom.

3.1 Number of cabinets required

The number of J-Series loudspeakers to be used in an application depends on the desired level, the distances and the directivity requirements in the particular venue. Using ArrayCalc will define whether the system is able to fulfill the requirements.

Depending on the program material and the desired level, additional J-SUBs will be necessary to extend the system bandwidth and headroom. In most applications a J-SUB to J8/J12 ratio of 1:2 is sufficient. Distributed SUB arrays may require a higher number of subwoofers, such as a J-SUB to J8/J12 ratio of 2:3.

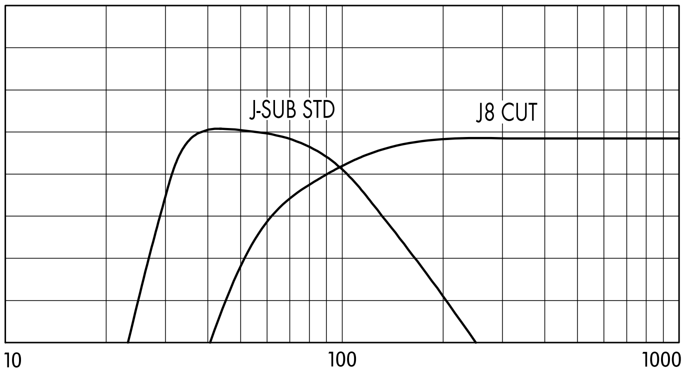
When additional J-INFRA systems are used, one cabinet provides the very low frequency extension for two J-SUB subwoofers, thus generally reducing the total number of J-SUBs required.

3.2 J-SUB subwoofer setup

J-SUB cabinets can be used ground stacked, as a horizontal SUB array or integrated into the flown array, either on top of a J8/J12 array or flown as a separate column.

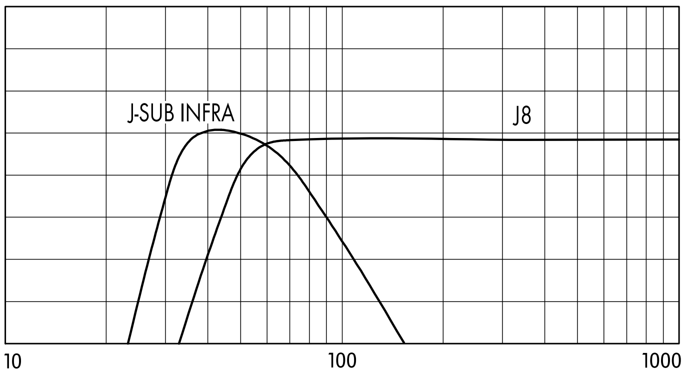
Depending on the application the dispersion pattern of the J-SUB cabinet can be modified electronically to achieve the best sound rejection where it is most effective. In cardioid mode, the standard setting of the J-SUB setup, the maximum rejection occurs behind the cabinet (180°) while hypercardioid mode (HCD selected) provides a maximum rejection at 135° and 225°. The HCD mode should also be used when J-SUB cabinets are operated in front of walls.

When used with additional subwoofers, the J8/J12 system should be operated in CUT mode to gain maximum headroom at low frequencies.



J8 / J-SUB crossover setup

When maximum low end headroom is not an issue, the J8/ J12 system can also be operated in standard mode (full range, i.e. CUT not selected) and additional J-SUB cabinets in INFRA mode can be used to extend the system bandwidth down to 32 Hz.



J8 / J-SUB crossover setup, full range

3.3 J-SUB ground stacks

Using J-SUB cabinets in L/R ground stacks provides maximum system efficiency due to the ground coupling of the cabinets.

3.4 J-SUBs flown on top of a J8/J12 array

Flown J-SUBs create a more even level distribution over distance. Compared to a ground stacked setup the area at the very front below the arrays has much less low frequency level because of the longer distance to the subwoofers. However, when a high level of low frequency energy at the front is desired, e.g. to compensate for a loud stage level, additional ground stacked subwoofers may be necessary.

3.5 Flown J-SUB columns

When complete columns of J-SUBs are flown, the increased vertical directivity adds to the distance effect described above and thus creates a longer throw of low frequencies.
Clever positioning and horizontal aiming of flown subwoofer columns behind the main and outfill arrays of TOP loudspeakers can greatly enhance both visual appearance and acoustic performance of the complete system through increased overall coherence between the different parts of the system.

3.6 J-SUB horizontal SUB array

Arranging J-SUBs in a horizontal array (SUB array) provides the most even horizontal coverage eliminating the cancellation zones to the left and right of the center of a typical L/R setup.

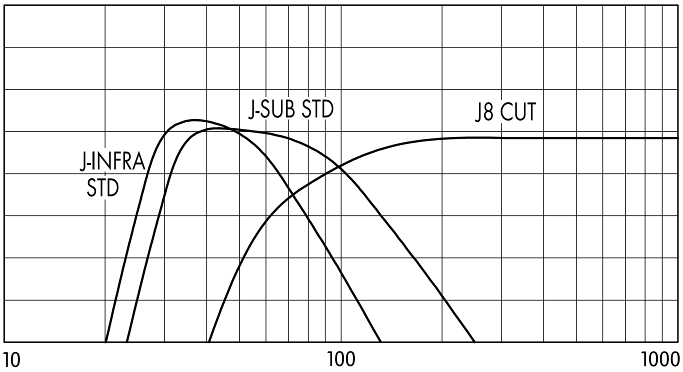
3.7 J-SUB/J-INFRA subwoofer setup

When used with J-INFRA cabinets J-SUB subwoofers are always operated in standard mode (i.e. INFRA not selected).

Depending on the application and the space requirements a combination of J-SUB and J-INFRA cabinets can be set up in several different ways.

3.8 Combined J-INFRA/J-SUB ground stacks

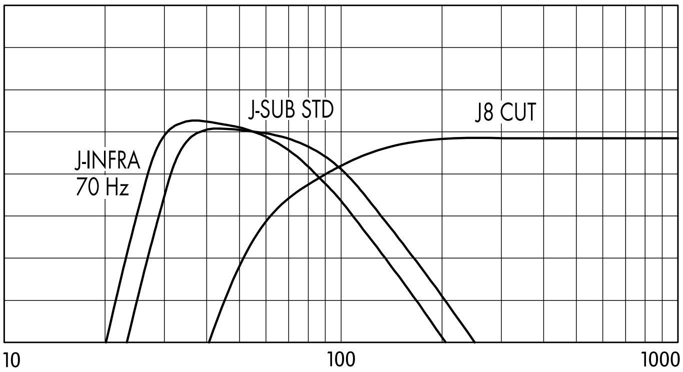
Maximum coupling and coherence of the systems are achieved when J-INFRA and J-SUB systems are stacked close to each other. However, make sure to keep a minimum distance of 60 cm (2 ft) between adjacent stacks. J-INFRA cabinets should be operated in standard mode.



J8 / J-SUB / J-INFRA crossover setup

3.9 Flown J-SUBs, J-INFRA ground stacks

Flown columns of J-SUBs provide a higher vertical directivity and thus a longer throw. Coupling with ground stacked J-INFRAs will be less coherent and therefore requires the 70 Hz setting on the J-INFRA controllers.



J8 / J-SUB / J-INFRA 70 Hz crossover setup

3.10 Flown J-SUBs, J-INFRA SUB array

As an option J-INFRA cabinets can be set up in a horizontal SUB array in front of the stage. Also in this case the 70 Hz setting on the J-INFRA controllers is advantageous. The correct alignment of the array dispersion and delay settings is performed using ArrayCalc.

4 The V-Series line array

The V-Series consists of three different loudspeakers: the V8 and V12 loudspeakers and the V-SUB subwoofer. The V8 and V12 are mechanically and acoustically compatible loudspeakers providing two different horizontal coverage angles of 80° and 120°. The dispersion of both systems is symmetrical and well controlled to frequencies down to 250 Hz, their bandwidth reaching from 65 Hz to 18 kHz. V-Series loudspeakers can be operated with d&b D12, 30D, D20, D80 or D40/40D amplifiers. With 30D, D20, D80 and D40/40D amplifiers d&b ArrayProcessing is available.

In the vertical plane the V8 and V12 loudspeakers produce a wavefront that allows splay angle settings ranging from 0° to 14° (1° increments). An array should consist of a minimum of four cabinets - either V8, V12 or a combination of both.

The V8 with its 80° horizontal dispersion and high output capability can cover any distance range up to 100 m (330 ft) depending on the vertical configuration of the array and the climatic conditions.

The V12 offers a wider horizontal coverage which is particularly useful for short and medium throw applications. Using a combination of V8 and V12 cabinets enables the user to create a venue specific dispersion and energy pattern.

The V-SUB cardioid subwoofer extends the system bandwidth down to 37 Hz while providing exceptional dispersion control either flown or ground stacked in arrays or set up individually.

The J-INFRA cardioid subwoofer is an optional extension to a V8/V12/V-SUB system. It is used in ground stacked configurations and extends the system bandwidth down to 27 Hz while adding impressive low frequency headroom.

4.1 Number of cabinets required

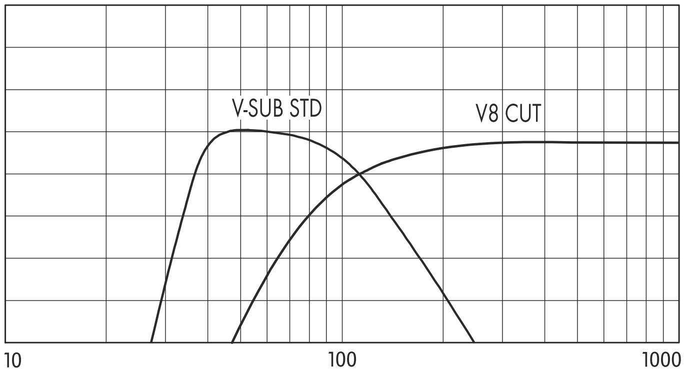
The number of V-Series loudspeakers to be used in an application depends on the desired level, the distances and the directivity requirements in the particular venue. Using ArrayCalc will define whether the system is able to fulfill the requirements.

Depending on the program material and the desired level, additional V-SUBs will be necessary to extend the system bandwidth and headroom. In most applications a V-SUB to V8/V12 ratio of 1:2 is sufficient. Distributed SUB arrays may require a higher number of subwoofers, such as a V-SUB to V8/V12 ratio of 2:3.

When additional J-INFRA systems are used, one cabinet provides the very low frequency extension for two V-SUB subwoofers, thus generally reducing the total number of V-SUBs required.

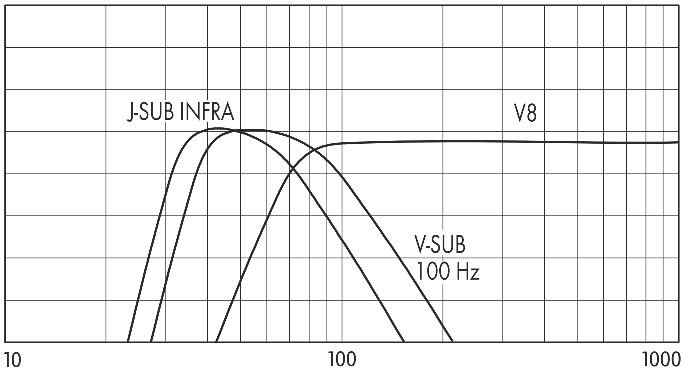
4.2 V-SUB subwoofer setup

V-SUB cabinets can be used ground stacked, as a horizontal SUB array or integrated into the flown array, either on top of a V8/V12 array or flown as a separate column. The V-SUB cabinet offers a cardioid dispersion pattern throughout its entire operating bandwidth. When used with additional subwoofers, the V8/V12 system should be operated in CUT mode to gain maximum headroom at low frequencies.



V8 / V-SUB crossover setup

When maximum low end headroom is not an issue, the V8/V12 system can also be operated in standard mode (full range, i.e. CUT not selected) and additional V-SUB cabinets in 100 Hz mode or J-SUB cabinets in INFRA mode can be used to extend the system bandwidth down to 38 Hz/32 Hz.



V8 / V-SUB / J-SUB crossover setup, full range

4.3 V-SUB ground stacks

Using V-SUB cabinets in L/R ground stacks provides maximum system efficiency due to the ground coupling of the cabinets.

4.4 V-SUBs flown on top of a V8/V12 array

Flown V-SUBs create a more even level distribution over distance. Compared to a ground stacked setup the area at the very front below the arrays has much less low frequency level because of the longer distance to the subwoofers. However, when a high level of low frequency energy at the front is desired, e.g. to compensate for a loud stage level, additional ground stacked subwoofers may be necessary.

4.5 Flown V-SUB columns

When complete columns of V-SUBs are flown, the increased vertical directivity adds to the distance effect described above and thus creates a longer throw of low frequencies.
Clever positioning and horizontal aiming of flown subwoofer columns behind the main and outfill arrays of TOP loudspeakers can greatly enhance both visual appearance and acoustic performance of the complete system through increased overall coherence between the different parts of the system.

4.6 V-SUB horizontal SUB array

Arranging V-SUBs in a horizontal array (SUB array) provides the most even horizontal coverage eliminating the cancellation zones to the left and right of the center of a typical L/R setup.

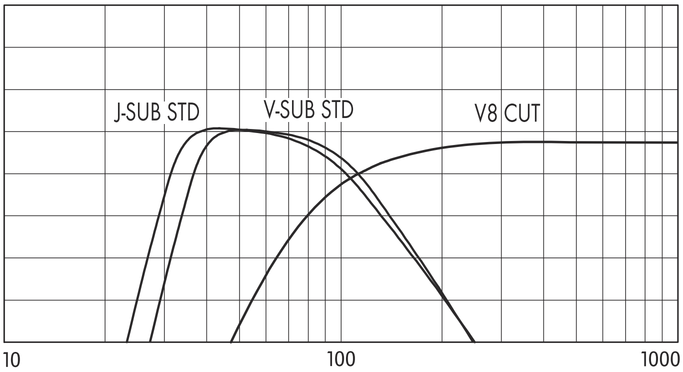
4.7 J-SUB/J-INFRA subwoofer setup

When used with J-SUB and J-INFRA cabinets, V-SUB subwoofers are always operated in standard mode (i.e. 100 Hz not selected).

Depending on the application and the space requirements a combination of V-SUB and J-SUB / J-INFRA cabinets can be set up in several different ways.

4.8 Combined J-, V-SUB ground stacks

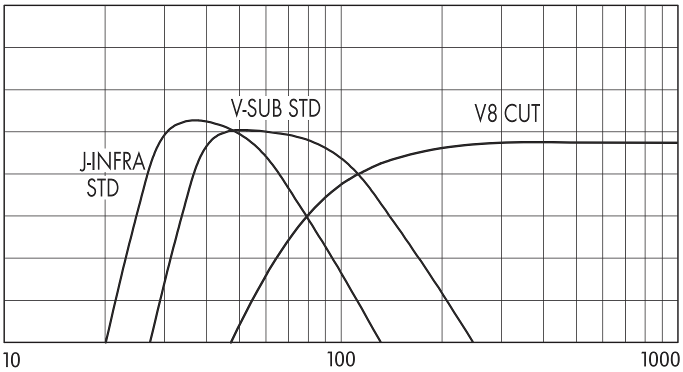
Maximum coupling and coherence of the systems are achieved when J-SUB and V-SUB systems are stacked close to each other. However, make sure to keep a minimum distance of 60 cm (2 ft) between adjacent stacks. J-SUB cabinets should be operated in standard mode.



V8 / V-SUB / J-SUB crossover setup

4.9 Flown V-, J-SUBs or J-INFRA ground stacks

Flown columns of V-SUBs provide a higher vertical directivity and thus a longer throw. Ground stacked J-SUBs or J-INFRA can be operated in either crossover mode depending on the ratio of flown to ground stacked subwoofers.



V8 / V-SUB / J-INFRA crossover setup

4.10 Flown V-SUBs, J-INFRA SUB array

As an option J-INFRA cabinets can be set up in a horizontal SUB array in front of the stage. In this case the 70 Hz setting on the J-INFRA controllers is advantageous. The correct alignment of the array dispersion and delay settings is performed using ArrayCalc.

The Y-Series line array consists of three different loudspeakers: the Y8 and Y12 loudspeakers and the Y-SUB subwoofer. The Y8 and Y12 are mechanically and acoustically compatible loudspeakers providing two different horizontal coverage angles of 80° and 120°. The dispersion of both systems is symmetrical and well controlled to frequencies down to 500 Hz, their bandwidth reaching from 54 Hz to 19 kHz.

Y-Series loudspeakers can be operated with d&b D12, 30D, D20, D80 or D40/40D amplifiers. With 30D, D20, D80 and D40/40D amplifiers d&b ArrayProcessing is available.

In the vertical plane the Y8 and Y12 loudspeakers produce a wavefront that allows splay angle settings ranging from 0° to 14° (1° increments). An array should consist of a minimum of four cabinets - either Y8, Y12 or a combination of both.

The Y8 with its 80° horizontal dispersion and high output capability can cover any distance range up to 100 m (330 ft) depending on the vertical configuration of the array and the climatic conditions.

The Y12 offers a wider horizontal coverage which is particularly useful for short and medium throw applications. Using a combination of Y8 and Y12 cabinets enables the user to create a venue specific dispersion and energy pattern.

The Y-SUB cardioid subwoofer extends the system bandwidth down to 39 Hz while providing exceptional dispersion control either flown or ground stacked in arrays or set up individually.

The J-INFRA cardioid subwoofer is an optional extension to a Y8/Y12/Y-SUB system. It is used in ground stacked configurations and extends the system bandwidth down to 27 Hz while adding impressive low frequency headroom.

5.1 Number of cabinets required

The number of Y-Series loudspeakers to be used in an application depends on the desired level, the distances and the directivity requirements in the particular venue. Using ArrayCalc will define whether the system is able to fulfill the requirements.

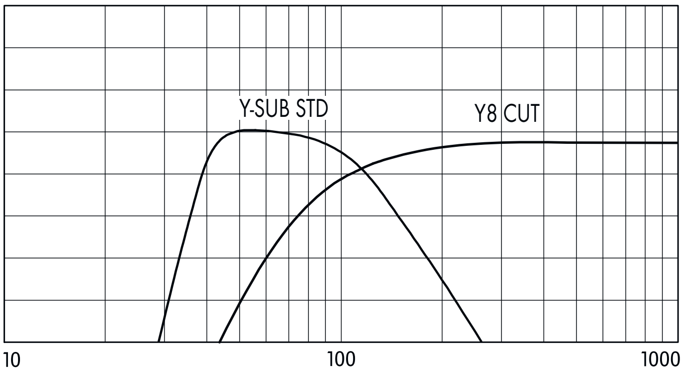
Depending on the program material and the desired level, additional Y-SUBs will be necessary to extend the system bandwidth and headroom. In most applications a Y-SUB to Y8/Y12 ratio of 1:2 is sufficient. Distributed SUB arrays may require a higher number of subwoofers, such as a Y-SUB to Y8/Y12 ratio of 2:3 or higher.

When additional J-INFRA systems are used, one cabinet provides the very low frequency extension for up to four Y-SUB subwoofers, thus generally reducing the total number of Y-SUBs required.

5.2 Y-SUB subwoofer setup

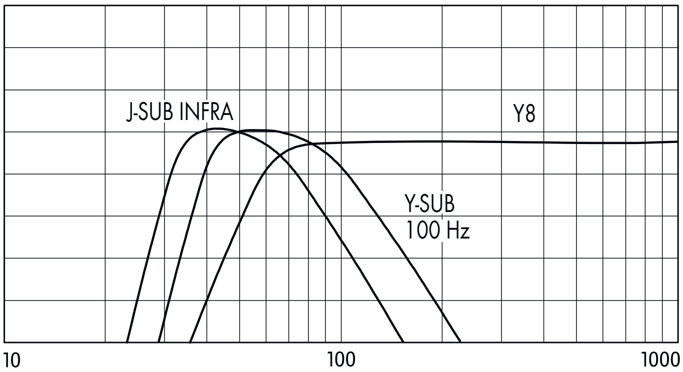
Y-SUB cabinets can be used ground stacked, as a horizontal SUB array or integrated into the flown array, either on top of a Y8/Y12 array or flown as a separate column.

The Y-SUB cabinet offers a cardioid dispersion pattern throughout its entire operating bandwidth. When used with additional subwoofers, the Y8/Y12 system should be operated in CUT mode to gain maximum headroom at low frequencies.



Y8 /Y-SUB crossover setup

When maximum low end headroom is not an issue, the Y8/Y12 system can also be operated in standard mode (full range, i.e. CUT not selected) and additional Y-SUB cabinets in 100 Hz mode or J-SUB cabinets in INFRA mode can be used to extend the system bandwidth down to 38 Hz/32 Hz.



Y8 / Y-SUB / J-SUB crossover setup, full range

5.3 Y-SUB ground stacks

Using Y-SUB cabinets in L/R ground stacks provides maximum system efficiency due to the ground coupling of the cabinets.

5.4 Y-SUBs flown on top of a Y8/Y12 array

Flown Y-SUBs create a more even level distribution over distance. Compared to a ground stacked setup the area at the very front below the arrays has much less low frequency level because of the longer distance to the subwoofers. However, when a high level of low frequency energy at the front is desired, e.g. to compensate for a loud stage level, additional ground stacked subwoofers may be necessary.

5.5 Flown Y-SUB columns

When complete columns of Y-SUBs are flown, the increased vertical directivity adds to the distance effect described above and thus creates a longer throw of low frequencies.
Clever positioning and horizontal aiming of flown subwoofer columns behind the main and outfill arrays of TOP loudspeakers can greatly enhance both visual appearance and acoustic performance of the complete system through increased overall coherence between the different parts of the system.

5.6 Y-SUB horizontal SUB array

Arranging Y-SUBs in a horizontal array (SUB array) provides the most even horizontal coverage eliminating the cancellation zones to the left and right of the center of a typical L/R setup.

5.7 V-, Y-, J-SUB/J-INFRA subwoofer setup

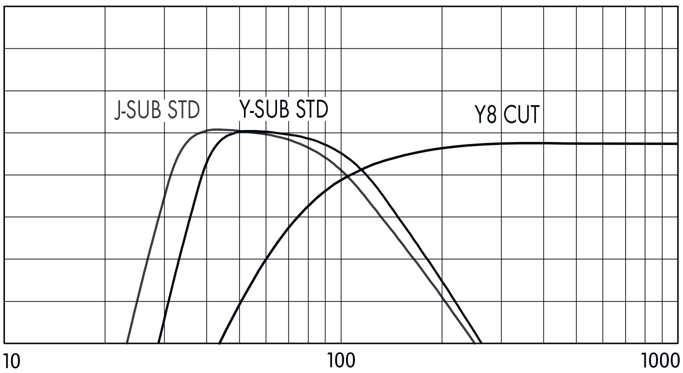
Y-SUB and V-SUB cabinets can be combined in virtually any application that does not require mechanical compatibility. Their modes should always be synchronized (i.e. both in 100 Hz mode or both in standard mode).

When used with J-SUB and J-INFRA cabinets, Y-SUB subwoofers are always operated in standard mode (i.e. 100 Hz not selected).

Depending on the application and the space requirements a combination of Y-SUB and J-SUB / J-INFRA cabinets can be set up in several different ways.

5.8 Combined J-, Y-SUB ground stacks

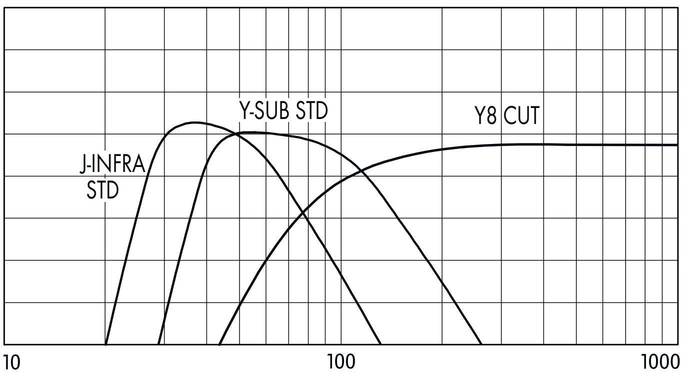
Maximum coupling and coherence of the systems are achieved when J-SUB and Y-SUB systems are stacked close to each other. However, make sure to keep a minimum distance of 60 cm (2 ft) between adjacent stacks. J-SUB cabinets should be operated in standard mode.



Y8 / Y-SUB / J-SUB crossover setup

5.9 Flown Y-, J-SUBs or J-INFRA ground stacks

Flown columns of Y-SUBs provide a higher vertical directivity and thus a longer throw. Ground stacked J-SUBs or J-INFRA can be operated in either crossover mode depending on the ratio of flown to ground stacked subwoofers.



Y8 / Y-SUB / J-INFRA crossover setup

5.10 Flown Y-SUBs, J-INFRA SUB array

As an option J-INFRA cabinets can be set up in a horizontal SUB array in front of the stage. In this case the 70 Hz setting on the J-INFRA controllers is advantageous. The correct alignment of the array dispersion and delay settings is performed using ArrayCalc.

The Q1 is a compact and lightweight line array cabinet providing a 75° constant directivity coverage in the horizontal plane down to 400 Hz. The system can be used from very small configurations of two cabinets per array up to a maximum of twenty cabinets per array for larger venues.

Q1 cabinets have a very low height of only 30 cm (1 ft) and when combined in arrays its accurate wavefront covers up to 14° vertically per cabinet, and couples coherently up to 12 kHz when configured in a straight (0° splay) long throw section. The Q1 covers the frequency range from 60 Hz to 17 kHz.

The Q7 and Q10 cabinets are mechanically and acoustically compatible loudspeakers with 75° x 40° and 110° x 40° spherical dispersion patterns which can be used as a downfill (Q7) or nearfill extension with Q1 arrays.

Smaller configurations of Q1 cabinets can also be used ground stacked, supported by Q-SUB cabinets. The most even energy distribution in the audience area will however be achieved with a flown array.

It is assumed that all Q-Series cabinets are driven by d&b D6 or D12 amplifiers. E-PAC amplifiers do not provide HFC and CSA settings.

6.1 Number of cabinets required

The number of Q1 cabinets to be used in an application depends on the desired level, the distances and the directivity requirements in the particular venue. Using ArrayCalc will prove whether the system is able to fulfill the requirements.

Depending on the program material and the desired level additional Q-SUB subwoofer systems will be necessary to extend the system bandwidth and headroom. The number of Q-SUBs needed per Q1 cabinet for serious full-range program will decrease with the size of the system. For small setups a 1:1 ratio is recommended, for example four Q-SUBs to four Q1s, while larger systems will work with a 2:3 ratio, for example eight Q-SUBs to twelve Q1s. Please note that CSA setups require a multiple of three Q-SUB cabinets.

As an option Q1 systems can also be used with J-SUB or J-INFRA subwoofers.

6.2 Subwoofer setup

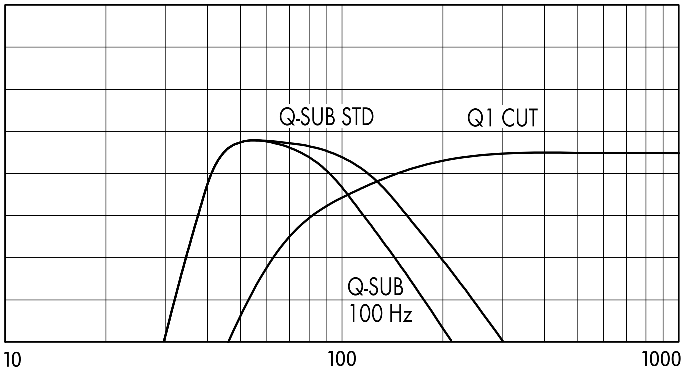
Subwoofers are operated most efficiently when stacked on the ground. For cleanest sound and coverage we recommend arranging subwoofers in a CSA configuration as described in d&b TI 330 Cardioid SUB array which is available for download from the d&b audiotechnik website at www.dbaudio.com. When used with subwoofers, the Q1 systems should be operated in CUT mode to gain maximum headroom at low frequencies.

Q-SUB (40 – 100/130 Hz)

Q-SUB cabinets can be used ground stacked or integrated into the flown array, either on top of a Q1 array or flown as a separate column.

Flown Q-SUBs create a different level distribution in the audience area than ground stacked ones. In particular the area at the very front below the arrays has much less low frequency energy when subwoofers are included in the array. This can be very useful in applications that do not require high levels of low frequency energy at the front, however for an event with high stage level additional ground stacked subwoofers may be necessary.

For Q1 arrays consisting of three or more cabinets we recommend the use of the 100 Hz setting for the Q-SUB systems. Smaller Q1 arrays providing less coupling at low frequencies may benefit from the higher crossover frequency of the standard mode of the Q-SUBs (130 Hz).



Q1/Q-SUB crossover setup

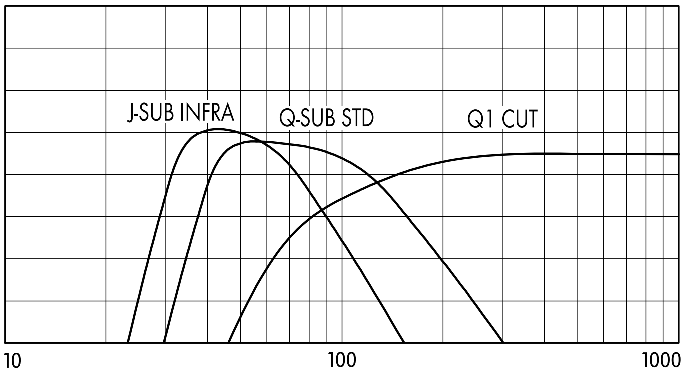
Compared to a standard Q-SUB configuration a CSA setup produces slightly less level above 70 Hz, so it may be advantageous to use the standard (130 Hz) amplifier setting.

J-SUB (32 – 70/100 Hz)

J-SUB cabinets can be used to supplement a Q1 system in different ways.

If the system is equipped with a sufficient number of Q-SUB cabinets, J-SUBs can be used to extend its bandwidth to below 32 Hz. Driven by D12 amplifiers set to INFRA mode one J-SUB will supplement up to four Q-SUB cabinets.

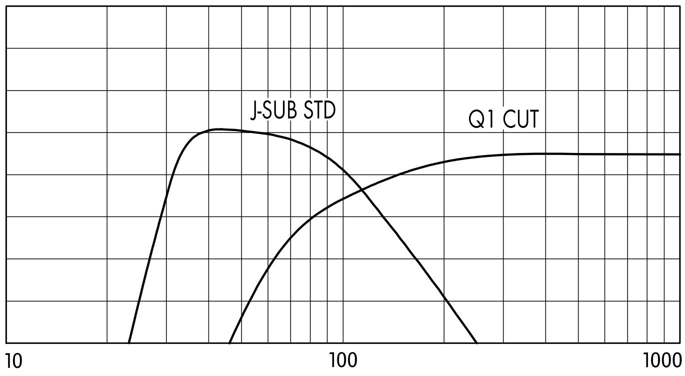
This combination will achieve its maximum headroom when the Q-SUBs are operated in the 130 Hz mode. If for audio reasons the lower crossover frequency to the Q1s is desired you may also reduce the gain of the Q-SUB amplifiers. Decreasing the gain by 2.5 dB will create the same downward shift to the upper slope as switching to the 100 Hz setting, but with less low frequency boost.



Q1/Q-SUB/J-SUB crossover setup

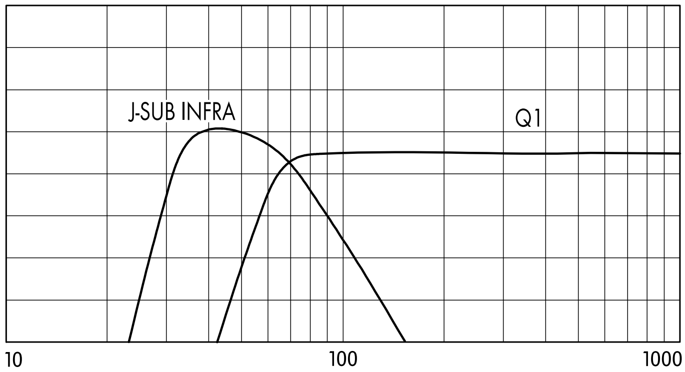
Please note that a combined ground stack consisting of Q-SUB and J-SUB cabinets will only provide a consistent directivity when Q-SUBs are used in CSA setups. Also make sure to keep the required distance of 60 cm (2 ft) between the stacks in order to not adversely affect the cardioid directivity of the systems.

J-SUB subwoofers can also be used as an alternative to ground stacked Q-SUBs. In this case J-SUB cabinets are operated in standard mode with a crossover frequency of 100 Hz. One J-SUB will replace three Q-SUB cabinets in a CSA setup and extends the system bandwidth down to 32 Hz.



Q1/J-SUB crossover setup

J-SUB cabinets in INFRA mode can be used to extend the bandwidth of a Q1 line array operated in full-range mode, without Q-SUBs. As this application does not expand the headroom of the Q1 array it is only useful when medium levels but very low frequencies are required, for example for special effects.

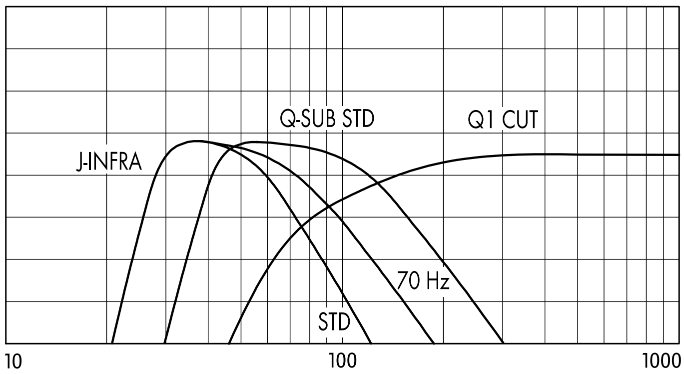


Q1/J-SUB crossover setup, full range

J-INFRA (27 – 60/70 Hz)

To achieve the ultimate low frequency extension for a Q system consisting of Q1 and Q-SUB cabinets, additional J-INFRA subwoofers can be used. They provide a standard (60 Hz) and a 70 Hz mode. The selection of the mode depends on the coupling between J-INFRA and Q-SUB cabinets in the actual setup. When combined in a ground stack the standard (60 Hz) mode provides more headroom at very low frequencies.

Please note that a combined ground stack consisting of Q-SUB and J-INFRA cabinets will only provide a consistent directivity when Q-SUBs are used in CSA setups. Also make sure to keep the required distance of 60 cm (2 ft) between the stacks in order not to adversely affect the cardioid directivity of the systems.



Q1/Q-SUB/J-INFRA crossover setup

The T10 is a very compact loudspeaker system which can be used both as a line array and as a high directivity point source speaker. For these applications the T10 cabinet provides two different dispersion characteristics which can be swapped over without any tools.

In line array mode the T10 provides a 105° constant directivity coverage in the horizontal plane allowing for vertical splay angles of up to 15° per cabinet. The system can be used from very small configurations of three cabinets per array up to a maximum of 20 cabinets per array for larger venues. The T10 covers the frequency range from 68 Hz to 18 kHz. The T-SUB subwoofer extends the system bandwidth down to 47 Hz either flown or ground stacked.

Smaller configurations of T10 cabinets can also be used ground stacked supported by T-SUB cabinets or mounted on a high stand. The most even energy distribution in the audience area will however be achieved with a flown array.

T-Series loudspeakers can be operated with d&b D12, 30D, D20, D80 or D40/40D amplifiers. With 30D, D20, D80 and D40/40D amplifiers d&b ArrayProcessing is available.

7.1 Number of cabinets required

The number of T10 cabinets to be used in an application depends on the desired level, the distances and the directivity requirements in the particular venue. Using ArrayCalc will prove whether the system is able to fulfill the requirements.

Depending on the program material and the desired level additional T-SUB subwoofer systems will be necessary to extend the system bandwidth and headroom. The number of T-SUBs needed per T10 cabinet for serious full-range program will decrease with the size of the system. For small setups a 1:3 ratio is recommended, for example one T-SUB to three T10s.

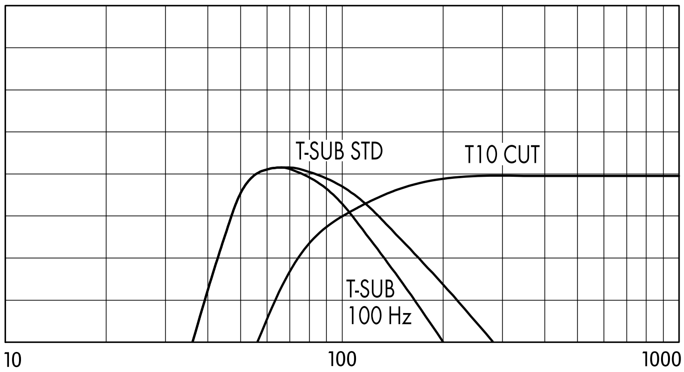
7.2 Subwoofer setup

When used with subwoofers, the T10 systems should be operated in CUT mode to gain maximum headroom at low frequencies.

T-SUB (47 – 100/140 Hz)
T-SUB cabinets can be used to supplement the LF headroom of the T10 loudspeakers in various combinations. They can be used ground stacked or integrated into the flown array, either on top of a T10 array or flown as a separate column.

Flown T-SUBs create a different level distribution in the audience area than ground stacked ones. In particular the area at the very front below the arrays has much less low frequency energy when subwoofers are included in the array. This can be very useful in applications that do not require high levels of low frequency energy at the front, however for an event requiring a loud stage level additional ground stacked subwoofers may be necessary.

For T10 arrays consisting of three or more cabinets we recommend the use of the 100 Hz setting for the T-SUB systems. Smaller T10 arrays providing less coupling at low frequencies may benefit from the higher crossover frequency of the standard mode of the T-SUB (140 Hz).

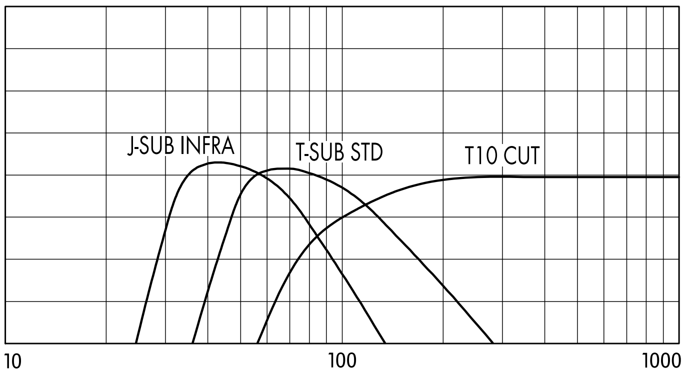


T10 / T-SUB crossover setup

B4-SUB (40 – 100/150 Hz)
E15X-SUB (37 – 100/140 Hz)
As an option T10 systems can also be used with B4-SUB or E15X-SUB subwoofers. These cabinets cannot be integrated into a flown T-Series rig. However, they allow the deployment of T10 cabinets on their M20 flanges using either the T-Series Base plate or the T-Series Cluster bracket.

The T-Series Base plate connects directly to the M20 flange and supports an array of up to 6 x T10 cabinets while the T-Series Cluster bracket is pole mounted on the M20 flange and supports up to three T10 cabinets. To achieve the best acoustic results in critical venues, we recommend to use the B4-SUB. It is a compact and effective solution providing a cardioid dispersion from a single amplifier channel. Like the T-SUB these systems provide a 100 Hz circuit on their controller which can be set accordingly.

J-SUB (32 – 70/100 Hz)
J-SUB cabinets in INFRA mode can be used to extend the frequency range of a T-Series system. To gain maximum headroom T-SUBs should be operated in standard mode (i.e. 100 Hz not selected).



T10 / T-SUB / J-SUB crossover setup

The 10AL and 10AL-D line array modules of the xA-Series have been specifically designed for fixed installations with visually unobtrusive integrated rigging systems.

For these applications, the cabinets are available with two different constant directivity dispersion characteristics in the horizontal plane:

The 10AL provides a 75° coverage while the 10AL-D version provides 105° of coverage. In the coupling plane, both allow for vertical splay angles of up to 15° per cabinet. Both versions may be combined in one array, for example with 10AL cabinets at the top for longer distances and one or two 10AL-D to cover the areas near the stage.

Both systems can be used from small configurations of three cabinets per array up to a maximum of 9 cabinets per array.

The 10AL (-D) covers the frequency range from 60 Hz to 18 kHz. 18A-SUB or 27A-SUB subwoofers extend the system bandwidth down to 37 Hz or 40 Hz, respectively. They can be flown in a separate column, integrated at the top or within an array or used as ground stacked applications. When they are flown together with line array modules, the maximum number of total cabinets is reduced due to the additional weight.

Configurations of up to six 10AL / 10AL-D cabinets can also be used ground stacked, supported by 18S-SUB or 27S-SUB cabinets. The most even energy distribution in the audience area will however be achieved with a flown array.

xA-Series loudspeakers can be operated with d&b D6, D12, 10D, 30D, D20 or D80 amplifiers.

8.1 Number of cabinets required

The number of 10AL or 10AL-D cabinets to be used in one application depends on the desired level, the distances to be covered and the directivity requirements of the particular venue. Using ArrayCalc will prove whether the system is able to fulfill the requirements.

Depending on the program material and the desired level additional 18A-SUB or 27A-SUB subwoofer systems may be necessary to extend the system bandwidth and headroom. The number of subwoofers required per 10AL (-D) cabinet to provide a serious full-range program decreases with the size of the system. For small to medium size setups, a 1:3 ratio is recommended, for example one 27A-SUB to three 10ALs.

8.2 Subwoofer setup

When used with subwoofers, the 10AL(-D) systems should be operated in CUT mode to gain maximum headroom at low frequencies.

27A-SUB/27S-SUB (40 – 100/140 Hz)

Subwoofers can be used to supplement the LF headroom of the 10AL loudspeakers in various combinations.

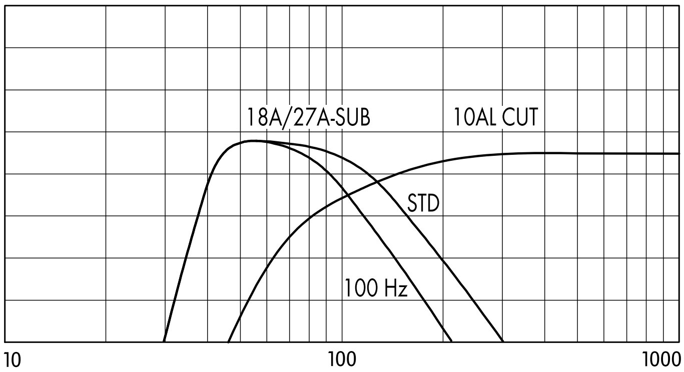
To achieve the best acoustic result in critical venues, we recommend the use of 27A-SUB or 27S-SUB subwoofers. They offer a compact and effective solution by providing cardioid dispersion from a single amplifier channel.

They can be used ground stacked (27S-SUB and 27A-SUB) or integrated into the flown array (27A-SUB), either at the top or within a 10AL array, or flown as a separate column.

Flown subwoofers create a different level distribution in the audience area than ground stacked ones. Particularly the area directly at the front below the arrays provides less low frequency energy when subwoofers are included in the array.

This can be very useful in applications that do not require high levels of low frequency energy at the front, however for an event requiring a loud stage level, additional ground stacked subwoofers may be necessary.

For 10AL arrays consisting of three or more cabinets, we recommend the use of the 100 Hz setting for the subwoofers. Smaller 10AL arrays providing less coupling at low frequencies may benefit from the higher crossover frequency of the standard mode (140 Hz).



10AL / 18A/27A-SUB crossover setup

18A-SUB/18S-SUB (37 – 100/140 Hz)

18A-SUB or 18S-SUB cabinets can be used in the same way as 27A-SUB or 27S-SUB cabinets but without the benefit of cardioid dispersion.

For these systems, just like for the 27S/A-SUBs, a 100 Hz circuit is available on the controller, which can be set accordingly.

The A-Series consists of two different loudspeaker models with variants for mobile and install applications. They can be used either as a single point source or as horizontal or vertical arrays of up to four loudspeakers with the captive rigging components and the normal mounting accessories. The AL60/ALi60 and AL90/ALi90 are mechanically and acoustically compatible loudspeakers providing two different coverage angles of 60° x 30° and 90° x 30°. Splay angles between adjacent cabinets in the coupling plane can be set from 20° to 40° in 5° increments. This results in a total coverage of 50° up to 70° for an array of two loudspeakers, with a maximum total coverage of 150° per array of four loudspeakers.

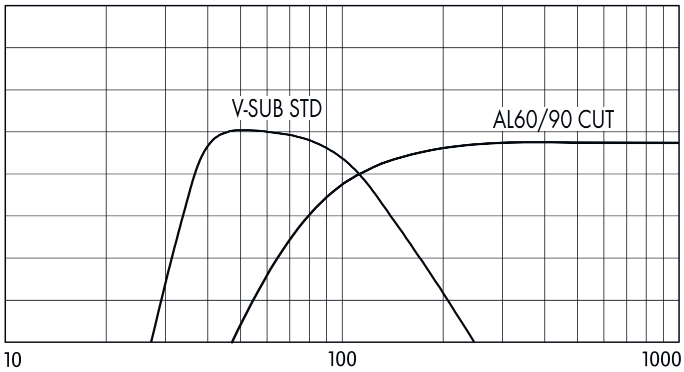
The nominal coverage angle of both systems in the non-coupling plane is symmetrical and well-controlled down to frequencies of 550 Hz (AL60/ALi60) and 370 Hz (AL90/ALi90) respectively. The bandwidth of the A-Series extends from 60 Hz to 18 kHz. A-Series loudspeakers can be driven by all d&b four-channel system amplifiers except the 10D.

With regard to performance, the A-Series sits between individual point sources and line arrays and is designed for coverage distance ranges of up to 30 m (100 ft), depending on the configuration of the array, climatic conditions and SPL requirements.

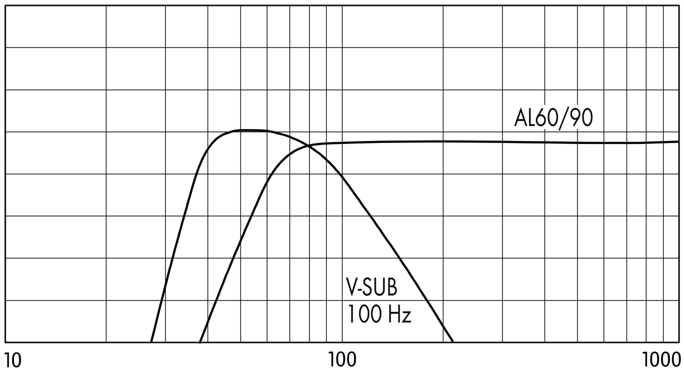
9.1 Number of cabinets required

The number of A-Series loudspeakers to be used in an application mainly depends on the coverage requirements. Using ArrayCalc will define whether the system is able to fulfill these requirements. Depending on the program material and the desired SPL, additional subwoofers will be necessary to extend the system bandwidth and headroom. V-SUB or variants thereof is the recommended model of subwoofer, but subwoofers from other d&b series are also applicable in appropriate amounts.

When used with subwoofers, A-Series loudspeakers are typically operated in CUT mode for extended headroom. For applications with low headroom demands, and if this is artistically preferable, A-Series loudspeakers can also be driven in standard mode. In this case, the 100 Hz or INFRA mode for the subwoofers is recommended. The necessary number of subwoofers depends on SPL and coverage requirements. In most L/R applications, a TOP/SUB ratio of 2:1 is sufficient.
Distributed SUB arrays typically require a larger number of subwoofers in order to enable an appropriately tight spacing while physically extending across the necessary width in front of the stage. When additional INFRA systems are used, one cabinet provides very low frequency extension for two subwoofers, thus generally reducing the total number of V-SUBs required. For more details please see the relevant chapters in this document.



AL60/AL90 crossover setup, standard/CUT



AL60/AL90 crossover setup, 100 Hz

9.2 A-Series horizontal array

Horizontal arrays are particularly useful when a wide horizontal coverage paired with significant low-mid headroom beyond the capabilities of a single point-source loudspeaker is required, while at the same time the available mounting height is limited. Typical examples of such applications are low-ceiling installs or low mobile stages. This means that the vertical directivity needs to be controlled in order to limit the increase in SPL close to the system. Orienting the cabinets upright makes use of the properties of the bipolar low-frequency section in the vertical plane and provides the necessary directivity control down to low-mid frequencies.

A horizontal A-Series array can consist of up to four loudspeakers. Two loudspeakers of the same type (AL60/ALi60 or AL90/ALi90) can be linked and driven by one amplifier channel.
For horizontal arrays of more than two A-Series loudspeakers, the use of MDC (**M**idrange **D**irectivity **C**ontrol) is recommended. MDC counters the “beaming” of arrays of 3 or more loudspeakers, where the physical size of the array produces an exaggerated narrowing in coverage for frequencies between 200 Hz and 2 kHz. With MDC, an even distribution of the lower-mid frequency range is achieved, thus matching the coverage of the high frequency sections which is set by the splay angles.
MDC requires two setups to be used correctly within the same array: 'ALx0 Out' for the 'outer' loudspeakers and 'ALx0 In' for the 'inner' loudspeakers of an array. For arrays of three or four loudspeakers, the 'Out' setup is used for the outer two loudspeakers of the array, while the 'In' setup is used for the inner one or two loudspeakers. For arrays of only two loudspeakers, the 'Out' setup is used for both. In all MDC applications, loudspeakers with the same setup can be linked in pairs.

9.3 A-Series vertical array

Vertical arrays of A-Series are very useful in situations where a very large coverage angle in the vertical is required and all listeners are roughly equidistant from the system, such as when covering a section of raked seating from far above. This frequently results in a necessary coverage angle of 70° or more. In this application, a vertical A-Series array typically outperforms clusters of regular point sources or line arrays in terms of evenness of coverage, weight and required amplifier channel count.

For vertical A-Series arrays, both MDC (please refer to ⇒ Chapter 9.2 "A-Series horizontal array" on page 18) or ArrayProcessing can be employed.
When using ArrayProcessing, each loudspeaker in the array must be driven from an individual amplifier channel.

A-Series vertical arrays and ArrayCalc
A vertical A-Series array can consist of up to four loudspeakers for mobile and installed applications where the captive rigging components and the normal mounting accessories are to be used. For installed applications where more than four loudspeakers are required in a vertical array, ArrayCalc allows the prediction of arrays with up to six A-Series loudspeakers.

> **Note:** Please note that arrays of more than 4 loudspeakers can only be mounted using custom cabinets and custom-designed mounting accessories and require the array to be driven with ArrayProcessing. For these special applications, please consult with your d&b sales partner.

9.4 Mixed A-Series array

A mixed deployment of different A-Series loudspeakers within the same array is only recommended for vertical arrays and when ArrayProcessing is used. This is due to the differing crossover functions between AL60 and AL90 that would otherwise lead to unexpected acoustic results.

9.5 A-Series point source applications

A-Series loudspeakers can also be used individually in a point source application. As an example, this is highly useful in front fill applications with high output demand, where for example an AL90 in landscape orientation provides a coverage of 90° in the horizontal and 30° in the vertical plane. A respective option is available in ArrayCalc, and as a matching 'ALx0 PS' setup in the system amplifiers. Please note that the HF horn of A-Series loudspeakers is not rotatable.

10.1 Point sources

All current top cabinets of the E-Series, V(i)P-Series, Y(i)P-Series, Q(i)7, Q(i)10, T(i)10PS , xC-Series and xS-Series can be selected for simulation within an ArrayCalc project. Please note that a T(i)10L loudspeaker that is deployed horizontally may also be used as a single nearfill with the T10PS setup although its polar dispersion does not reflect a "point source".

For cabinets that are equipped with rotatable HF horns, both horn orientations can be selected separately. Each selectable orientation for a specific loudspeaker type uses its own measured polar data set. This is defined by the chosen nominal horizontal and vertical dispersion angles and follows the convention [SystemName] [horizontal dispersion] x [vertical dispersion] while the cabinet itself remains in its typical mechanical orientation, i.e. in an upright position (e.g. 10S 75° x 50°; E6 55° x 100°; Q7 40° x 75° etc).

If a system is used lying on its side, the standard dataset must be used and the cabinet rotation must be set to either 90°(on its left side, seen from a listener's position) or 270°(on its right side, seen from a listener's position). The cabinet can be rotated in steps of 90° degrees. Each individual cabinet can be freely positioned within the room with horizontal or vertical aiming.

Selecting a loudspeaker optionally displays a balloon polar plot or its vertical aiming into the room. More specific loudspeaker data can be found in the relevant documentation of the respective d&b products.

10.1.1 Number of cabinets required

The number of point source cabinets is primarily defined by their specific application, for example as nearfill or delay systems or as the main system. Of course, the number of cabinets also depends on the desired level, the distances to be covered and the directivity requirements in the particular venue or project. Using ArrayCalc will prove whether the system is able to fulfill the specific requirements.

Depending on the program material and the desired level, additional d&b subwoofer systems may be necessary to extend the bandwidth and headroom.

When used with subwoofers, the point sources should be operated in CUT mode to gain maximum headroom at low frequencies.

10.2 Column loudspeakers

The xC-Series column loudspeakers are passive 2-way designs with a passive bandpass system providing a cardioid dispersion control with an 18 dB average broadband attenuation to the rear of the loudspeakers.

The 16C behaves as a standard point source cabinet with a 90° x 40° (h x v) dispersion and is treated accordingly in ArrayCalc. Its HF horn orientation is fixed, as a result there is one single set of data available. You can, of course, change the orientation of the cabinet itself like with all point sources.

The 24C provides a special 90° x 20° pattern with a variable vertical aiming to produce an even level distribution over a typical audience area. This is achieved by adjusting the vertical angle of the complete HF array between 0° and –14°combined with a 5° down tilt to the dispersion of low and mid frequencies.

When the 24C-E Cardioid column extender is attached, vertical dispersion control is extended towards low frequencies by another full octave.

11.1 Time alignment

Within a line array column it is absolutely essential to maintain perfect time alignment, otherwise the whole principle of creating a coherent wavefront will fail. Therefore, all amplifiers used to drive one column must be fed from the same input signal. Should a delay for the complete line array be necessary, the delay function in the amplifier channels can be used. The setting must be identical for all amplifier channels used in the column.

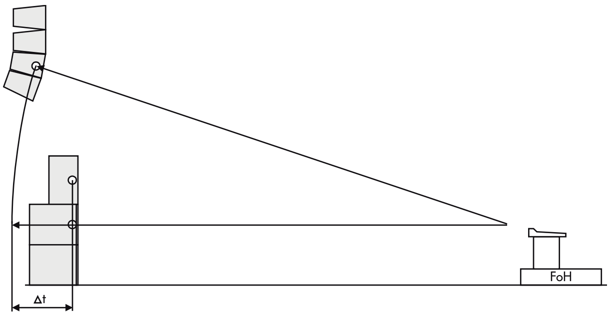
11.1.1 Subwoofers

A correct time alignment of subwoofers to the main line array is very important. If the required delay settings cannot be calculated using ArrayCalc (unknown geometry of the setup, unknown latency of devices in the signal chain), an acoustic measurement system should be used.

The signal arrival at the FoH position can be used as a reference for time alignment.

In the following example the subwoofer amplifiers have to be set to a delay time of Δt, equivalent to the physical offset Δx divided by the speed of sound (343 m/s or 1126 ft/s).

$$\Delta t = \frac{\Delta x}{v_{sound}}$$



Time alignment of ground stacked cabinets

> **Note:** An automatic time alignment using impulse responses - sometimes called "delay finder" – cannot provide correct results when systems cover different frequency bands, as this is the case with line array cabinets and subwoofers.
>
> Therefore use the response of the flown array and a full-range speaker placed on top of the subwoofer (e.g. nearfill) to determine the delay setting.

11.1.2 Nearfills

If nearfill loudspeakers are placed on top of subwoofer cabinets, the respective amplifier channels have to be set to the same delay value as the subwoofer.

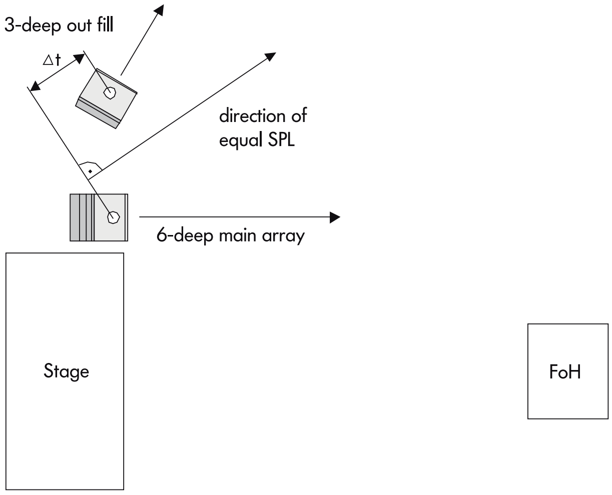
When applying SUB arrays with extensive delay settings to form the wavefront, equal delay settings of nearfill speakers placed on top of the subwoofers may not allow for correct source imaging in some positions.

In this case, the correct timing of the nearfill speaker is of greater importance and a possible local phase mismatch to the subwoofer is acceptable.

11.1.3 Horizontal array

If multiple J/V/Y/Q/T-Series columns or any combination of them are used per side, the arrays should be time-aligned at a meaningful position around the direction where both produce equal level. Use the ArrayCalc Alignment view to perform the alignment.

Should the procedure result in additional delay for the array selected for the subwoofer alignment, make sure this is compensated for in the delay settings for the SUB array. To measure the delay setting between subwoofers and main column on site turn off the outfill column.



Time alignment of a horizontal array

11.2 Equalization

If additional equalization of the system is required, use the d&b amplifier's multi-band fully parametric equalizer. It is important when applying the equalization that all channels are set identically within one column. Using the d&b Remote network and the R1 Remote control software the amplifier channels and their equalizers can be operated in user-defined functional groups, such as arrays, subwoofers, or outfills.

Using the amplifier's parametric equalizer for the system EQ provides the sound engineer with a flat FoH EQ for his personal sound design.

ArrayProcessing (AP) is a feature to calculate and design the holistic behavior of a line array. It is an additional feature to enhance the performance of d&b SL-Series, J-Series, V-Series, Y-Series and T-Series line array systems when powered by the d&b D80, D40/40D, D20, 10D or 30D amplifiers, while the 10D amplifier supports the Y-Series and T-Series line array systems only.

Physically, ArrayProcessing employs a conventional line array setup that is properly designed and positioned. The array must provide the required vertical dispersion and sufficient acoustic output to cover the audience areas effectively. Within one array it is possible to combine loudspeakers with different horizontal dispersions, for example GSL12 loudspeakers below GSL8.

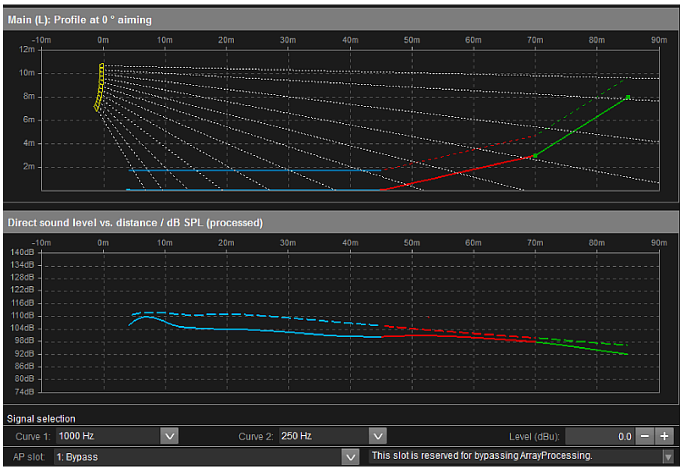
ArrayProcessing creates individual sets of FIR and IIR filters for every single cabinet of the array, each of which requires a dedicated amplifier/processing channel. These filters shape the sound generated by the array to precisely match a user defined level distribution and obtain a uniform frequency response over a given audience geometry.

In addition to individual amplification for each loudspeaker of an array, ArrayProcessing requires OCA/AES70 Ethernet remote control for these amplifiers. The use of ArrayProcessing is optional, meaning the function can be applied for specific applications or not, as and when required. ArrayProcessing adds 5.9 msec of latency, this is in addition to the 0.3 msec of the d&b amplifiers, arriving at a total of only 6.2 msec.

12.1 Motivation and benefits

Spectral differences in audience areas

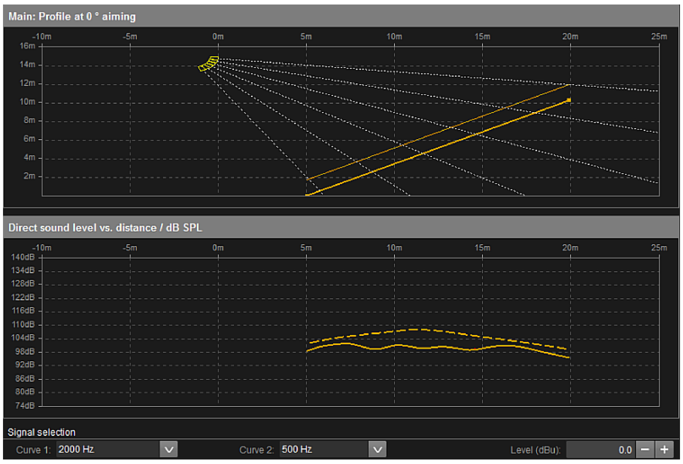
Typically, a line array setup for a given situation is planned in a way that optimizes the level distribution over distance in the high-mid frequency range (i.e. 2 kHz to 4 kHz bands). This requires a specific vertical aiming for the individual cabinets that is defined by the splay angles between them. However, the array dispersion at lower frequencies (100 Hz to 1000 Hz, depending on the array length) is a direct result of the total array curvature created by the splay settings (and not the individual aiming of a cabinet). This often creates a different level over distance distribution to that in the high-mid range.



Typical level/distance high-mid vs. low-mid:
Changing tonal balance over distance with progressive curvature.

The effect is well known and has been a cause for criticism from the very beginning of line array usage in modern sound reinforcement. The result is an uneven spatial balance and spectral response from the front of a venue to the back - a rich and (too) warm sound close to the array, which then becomes thin and almost aggressive in remote areas.

Another well-known example is the difference in spectral response when covering steep seating areas with a strongly curved array, as it is often used in outfill and 270° applications for tiers or balconies. In the highest seats it sounds very thin, in the seats around the middle there is a strong and annoying midrange beam, which disappears again when approaching the stage. In these situations it can often be perceived that the lower midrange dispersion does not follow the array shape.



Typical level/distance high-mid vs. low-mid:
Changing tonal balance over distance with constant curvature.

ArrayProcessing can eliminate these issues by providing a consistent frequency response throughout all listening positions. The resulting effect is that what you hear at FoH is what you will hear everywhere else. The mix is valid for everyone.

Compensating air absorption effects

ArrayProcessing includes air absorption effects in its calculations and provides a precise and seamless correction for all relevant cabinets. This not only provides a more consistent sound balance over distance, in many applications where the system has sufficient headroom, its throw can be extended and the need for delay systems is greatly reduced.

With the **T**emperature-**H**umidity-**C**ontrol (THC) functionality, compensation of air absorption effects can easily be adapted to typical changes of atmospheric conditions during operation for multiple arrays at once.

Flexibility

The level distribution in the audience area can be modified and tailored to reduce the level towards the front of the audience area and modify the level drop over distance over the audience area. Different ArrayProcessing settings for the array can be compared at a mouse click.

Intelligibility

In many applications, achieving a more accurate directivity control causes less stimulus to the reverberant field and leads to improved intelligibility.

Health & safety
Using ArrayProcessing, the level increase towards the front of the venue can be adjusted. Reducing it may help avoiding harmful sound pressure at the front while keeping the desired level for the rest of the audience.

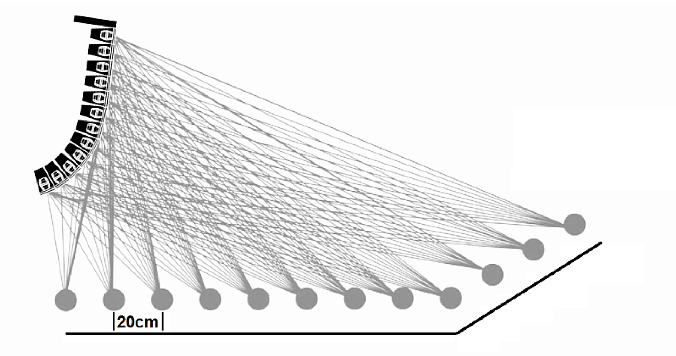
12.2 How does it work?

With the introduction of ArrayProcessing, for speaker simulation a completely new unified, more accurate and adaptive speaker model was developed and implemented. This speaker model provides exactly the necessary degree of detail for the type, size and the frequency range of each source - the highest resolution to provide a precise description of the behavior of a line array's sharp HF dispersion, a medium resolution to cover the dispersion characteristics of point sources and directional subwoofers or a rather coarse resolution for omnidirectional subwoofers.

The ArrayProcessing algorithm also considers and corrects diffraction effects produced by neighboring cabinets.

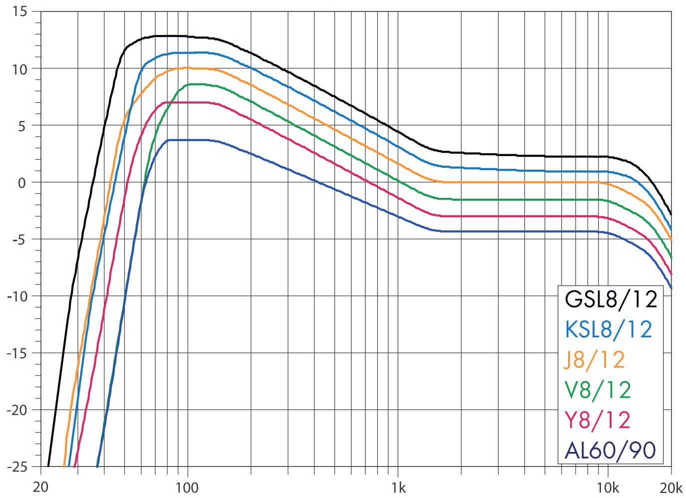
Target points are distributed along the listening area profile with a 20 cm spacing (along the intersection of the array profile with all matching listening planes).

When ArrayProcessing is enabled, it first calculates the contribution of each individual source to each listening position using a high spectral resolution of 24 frequencies per octave, making a total of 240 individual frequencies per target point over the entire 10 octave audio band.



The resulting data are stored in a matrix and serve as a basis for all further calculations.

The ArrayProcessing optimization routine will then create a unified/standardized frequency response at all these points. This target frequency response is exactly the reference response that is initially defined when tuning and voicing the controller setups for the d&b line arrays in conventional (unprocessed) setups. This response is identical for all systems above approximately 140 Hz, below that frequency each system has its own individual LF extension, depending on the specific cabinet design.



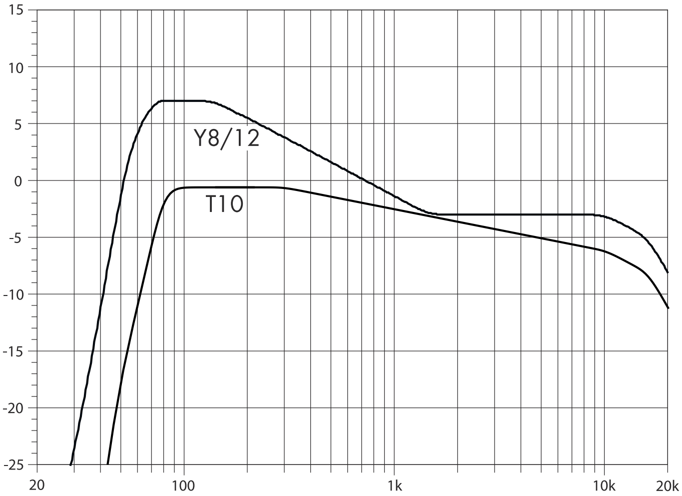
Target frequency responses for SL-, J-, V- Y- and A-Series

Note: Please note that the response created by the ArrayProcessing algorithm is independent of the array length, curvature and system type. Any ArrayProcessing line array design will provide the same sonic characteristics. Any combination using multiple columns of ArrayProcessing line arrays (rear fills, outfills, delays, etc.) does not require individual tuning and maintains this uniform sonic footprint.

Any further adjustment to the system response, either by using the CPL (Coupling) function or by applying master equalization is then carried out identically on the entire system for all listening areas.

d&b T-Series arrays are often used in theaters with large balcony sections which demand a spectral and spatial balance, and corporate events such as conventions or presentations, where intelligibility is a necessity and flexible level drop characteristics may be required. With ArrayProcessing, the T10 and Ti10L line array systems will be more consistent across the entire audience listening area in the vertical plane.

Due to the compact cabinet size and typical length of a T10 or Ti10L line array, the ArrayProcessing optimization will not be as effective in the low-mid region as the larger J, V and Y-Series arrays. Additionally, the typical frequency response (particularly in the vocal range) of a T-Series system does not fully correspond with the current d&b target frequency response within ArrayProcessing. This means it was necessary to develop a specific target frequency response for the T-Series for optimization with ArrayProcessing. The graph below shows T-Series target frequency response in comparison to the standard Y-Series response. This means the familiar T-Series voicing is still achieved in combination with the additional advantages of ArrayProcessing.



Schematic AP target response T10 vs Y8/Y12

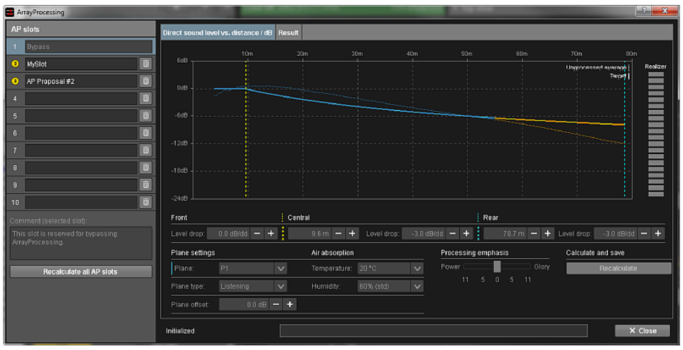
To optimize the T-Series with ArrayProcessing, each loudspeaker must be driven individually using either the D80, D40/40D, D20, 10D or 30D amplifiers and a **minimum cabinet count** of 6 x T10/Ti10L loudspeakers is required.

Due to the differing target frequency responses, a T-Series array with ArrayProcessing does not necessarily have the same sonic characteristics to a neighboring Y- or V-Series Array.

Any further adjustment to the system response, either by using the CPL (Coupling) function or by applying master equalization is then carried out identically on the entire system for all listening areas.

User parameters

The user can specify a desired level distribution along the listening profile. This is done in a simple way by specifying the level drop (dB per doubling of distance) for up to three different sections of the listening area profile (Front/Central/Rear). Additionally, a level offset can be applied to specific listening planes.



Furthermore there is another powerful parameter.

The «Power/Glory» fader, which defines the processing emphasis. Special focus on either maximum SPL and system headroom («Power») or on a best match of the target level distribution and frequency responses («Glory») can be selected. The center position usually provides a good balance between both of these parameters.

Up to nine different combinations of user parameter settings can be prepared and stored in the «AP»-slots of the amplifiers. These can be selected using the R1 Remote control software.

Switching between different slots is performed in near to real time, but as it will interrupt the audio program for some tenths of a second, it is not recommended.

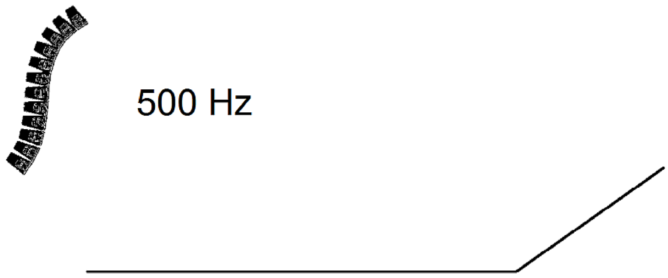
Keep it "organic"

Individual FIR filtering for each line array element can easily destroy the sonic integrity of a system. The secret lies in useful constraints to the algorithm and all resulting transfer functions.

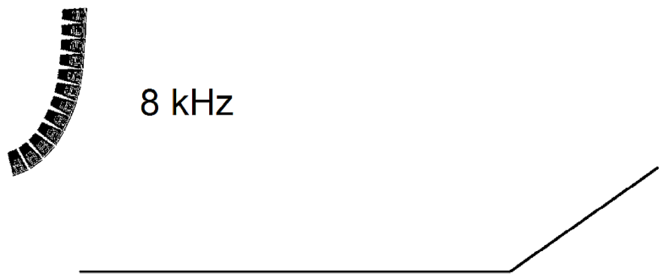
Algorithm results for each frequency need to relate to the neighboring frequencies to ensure a continuous filter response. System efficiency, headroom and time correlation must be preserved.

Different strategies for different frequencies

For the lower frequency range, where all sources contribute to most listening positions, processing basically only modifies the time alignment, but keeps equal level for all sources. You can picture the result as a varying virtual curving of the array over frequency.



For higher frequencies, where the individual sources cover only a small part of the listening area, the algorithm gradually shifts towards individual magnitude equalization of the transfer functions.



The transition between these ranges is continuous, always considers coherence relations between all elements of an array, ensuring the acknowledged d&b sonic footprint.

Processing is precisely matched to compensate for the air absorption under the actual atmospheric conditions and geometric relations. This replaces the manual process of selecting specific HFC (**H**igh **F**requency **C**ompensation) settings for each loudspeaker.

Subwoofers

ArrayProcessing is also available for flown SL-Series, J-Series, V-Series and Y-Series subwoofer arrays in mixed arrays with subwoofers at the top of the column (when applicable). However, to preserve a latency of 5.9 msec, ArrayProcessing will not significantly modify the directivity of subwoofer columns, but rather ensures their time alignment and frequency response correctly match to the line arrays.

Speed

For mobile applications, the speed of the calculation is an essential aspect. The user should always be able to immediately react on changing requirements in atmospheric conditions, audience attendance, level adjustments at the front or back. From

ArrayProcessing initialization to the filter set being active in the amplifiers, the typical calculation time for a 20-deep array covering an audience profile of 100 m is in the range of one minute – on a standard laptop computer.

12.3 ArrayProcessing workflow

Finally, as part of ArrayCalc, ArrayProcessing integrates seamlessly into the d&b workflow without compromising the renowned d&b sonic character or ease of use.

The planning process starts in a well known way; the array is positioned and splayed mechanically to achieve a useful level distribution for the 2 kHz and 4 kHz bands.

Enabling the loudspeaker specific ArrayProcessing option in ArrayCalc/R1 provides access to the additional processing functionality.

Settings for the array shape (Arc/Line) as well as for the compensation of air absorption (HFC) are obsolete as they are now embedded in the ArrayProcessing algorithm.

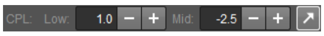
ArrayProcessing sets the target frequency response of the applied system to its original reference response. The optional CUT mode functions as usual: the low frequency level is reduced. The source is now configured for use with the system's dedicated subwoofers.

The CPL functionality is still available with ArrayProcessing being active. However, its traditional functionality of compensating for array length and curvature has been taken over by ArrayProcessing as it provides a uniform target frequency response for every array. With ArrayProcessing, CPL now provides an additional user parameter to adjust the system's tonal balance, for example to cater for the venue acoustics or individual taste. Its characteristics are identical for all ArrayProcessing line arrays. All arrayprocessed line arrays used in a system should be set to the same CPL value.

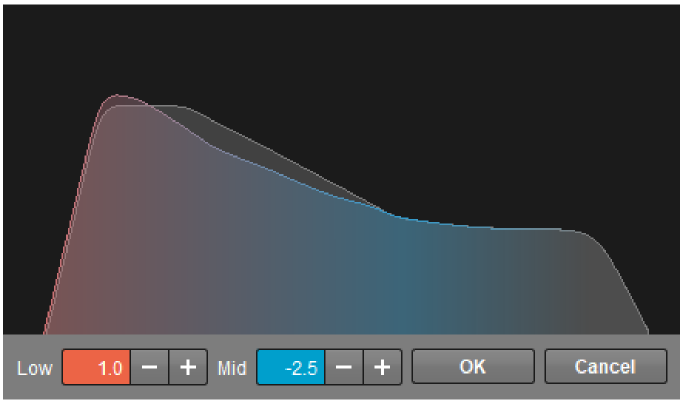
The classic CPL-function for J-, V-, Y- and T-Series arrays reduces the rising in low- and low-mid response towards lower frequencies with a low shelving-like filter response.

CPL-control: 

The Coupling function for SL-Series arrays provides a two-stage filter (Low/Mid) that allows for individual shaping of the low and low-mid frequency range. The increased headroom capabilities of the SL-Series systems are a necessary prerequisite to provide this in a meaningful way. In essence it provides an even more flexible way to adapt a systems response to individual mixing techniques.

SL-Series Coupling-control: 

By clicking on the rising arrow button on the right of the control, the control can be enlarged, providing a schematic graphical indication of the modified array response:



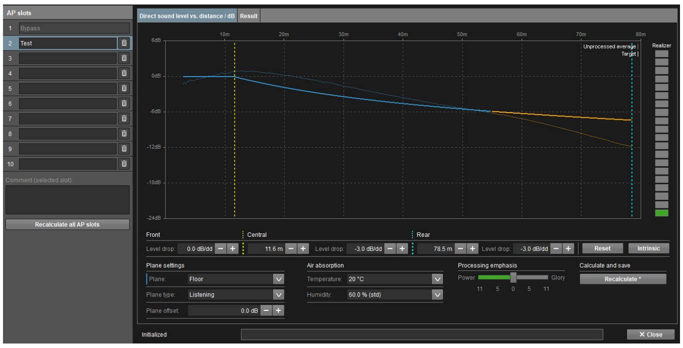
An identical control is available within the d&b R1 Remote control software.

All arrayprocessed line arrays used in a system should be set to the same CPL value. When the classic CPL and two-stage Coupling are mixed, the classic CPL setting should be matched to the Coupling-Mid of the two-stage Coupling.

> **Note:** Enabling ArrayProcessing for an array locks all mechanical settings of that particular array.
>
> To change the mechanical settings again, first disable ArrayProcessing.
>
> Keep in mind that disabling ArrayProcessing deletes all calculated ArrayProcessing data for that array. However, ArrayCalc provides the option to keep the ArrayProcessing user settings and when ArrayProcessing is enabled again, you can make use of the «Recalculate all slots» option.

12.4 ArrayProcessing dialog

To access the above mentioned user parameters of ArrayProcessing, open the «ArrayProcessing» dialog by clicking the «Process» button in the ArrayProcessing section of the respective array. The ArrayProcessing dialog is subdivided into two sections.



On the left hand side, the ArrayProcessing («AP slots») can be selected for editing and saving user parameters and resulting ArrayProcessing data. This includes a name and comment which will later be visible in R1. For each slot, a «Clear» button () is provided which allows you to clear all data stored to that particular slot.

> **Note:** The «Clear» button is only available when the respective slot contains data.

The other section on the right hand side includes two tabs, «Direct sound level vs. distance/dB» and «Result».

On the «Direct sound level vs. distance/dB» tab, you can define the target level distribution for the ArrayProcessing calculation. The following user parameters are available:

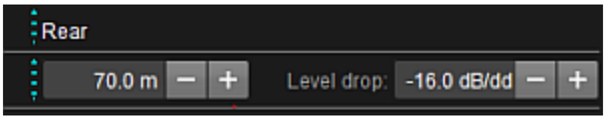
Level drop

Specifies the desired level drop in dB per doubled distance for the respective venue area (Front/Central/Rear).

Automatic level drop positions (intrinsic curve)

To find out which are the best level drop positions and levels (slopes), it is useful to start with a default setting which fits the intrinsic curve. ArrayCalc can calculate these settings automatically. Select the «Instrinsic» button to start the calculation.

Area borders



Two distance values setting the borders of the venue areas (Front/ Central/Rear) for which individual level drops can be defined.

Plane settings

In this section, the properties of the individual planes intersecting with an array's main axis are defined.

Plane type

For each plane intersecting with an array's main axis different plane types can be chosen: "Listening", "Reflecting" or "Level avoidance" (the latter only when this option is enabled on the ArrayCalc ⇒ Preferences ⇒ ArrayProcessing tab).

Listening plane type

The Listening plane type is the standard setting which is used for all planes occupied by an audience. When this type is chosen, an additional level offset (positive or negative) can be defined. The resulting target level vs. distance is shown in the diagram.

The ArrayProcessing algorithm now aims to reach exactly the offset level specified for this plane.

Reflecting plane type

The «Reflecting plane type» can be chosen for a highly reflecting surface such as, for example, glass walls or balcony fronts.

ArrayProcessing will then reduce the SPL only for the individual sources aiming at the reflecting plane. In practice, this will mostly affect the range above approx. 800 Hz and corresponds well with the range in which discrete reflections are most audible and annoying.

The benefit of the «Reflecting plane» option is that the performance of the entire array below 800 Hz is not affected; i.e. the sources aiming at the reflecting plane still fully contribute to the broadband directivity improvement in the lower frequency range.

It is important to notice that an appropriate resolution is necessary to achieve the effect:
At least one dedicated source has to hit the respective plane.

A plane that has been defined as “Reflecting” is illustrated by a dashed line in the «Target level vs. distance diagram».

Level avoidance plane type

The Level avoidance plane type option does not appear in the drop-down list unless it has been enabled on the ArrayCalc ⇒ Preferences ⇒ ArrayProcessing tab.

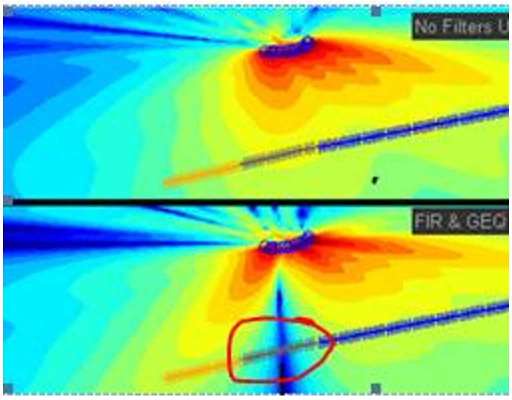
There is a very good reason for this. It is recommended that users only apply this very special function when they have gained adequate experience with the associated impacts.

The Level avoidance setting can drastically change the behavior of an array in a non-intuitive way. The Level avoidance plane type is used to prevent the transmission of energy to a particular area. This vertical plane is then acoustically “excluded” in a broadband way down to a frequency that depends on the FIR filter latency and array length. This split beam mode can only be achieved by creating a dipole bias.

The «Level avoidance plane type» can be used within an AP-slot to temporarily exclude a certain stand or balcony from the broadband electroacoustic program.

In case the selected slot contains an avoidance plane, the «Realizer» is disabled. This is because the realizer's red LED would glow in (almost) every case that includes an avoidance plane due to the large differences of target and unprocessed curve and therefore the slot calculation would be blocked.

Please keep in mind that the behavior of an array can be drastically changed in a non-intuitive way, when using avoidance planes and will no longer be acoustically compatible with any other array of TOPs or SUBs.



Schematic beam splitting in Level avoidance mode.

A plane that has been defined as Level avoidance plane is illustrated by a dashed and dotted line at the bottom of the «Direct sound level vs. distance» diagram.

Direct sound over distance

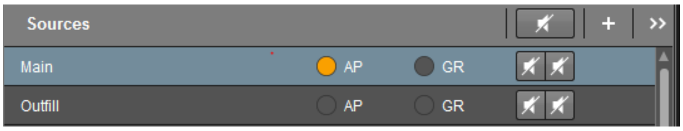
The graph shows the current target curve resulting from these settings as a continuous line. The dotted line displays the unprocessed level distribution (average level across all frequency bands).

Modification of venue data after AP calculation

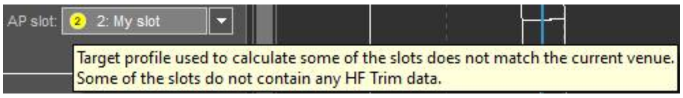
It is possible to modify the venue data after having calculated some AP-slots. This is a very common procedure in modern multi-purpose venues, for example, where entire grandstands can be removed.

A typical procedure could be to prepare one slot for a situation “grandstand in use” and another one for the situation “grandstand removed”.

When you modify the venue after having calculated some ArrayProcessing slots, the AP indicator in the headline of the relevant source group lights up yellow. The yellow LED also indicates when an AP-slot of an older project does not contain any HF Trim data (for further details, see below).



Detailed information about the cause for an AP indicator lighting up orange can be obtained by opening the source group and placing the mouse pointer over the yellow icon relating to the relevant slot:



All slots based on target profiles that no longer match the current venue setup are highlighted and the following tool tip is displayed: «Target profile used to calculate some of the slots does not match the curre
.

For all slots that do not include any HF Trim data the following tool tip is displayed:
«Some of the slots do not contain any HF Trim data.»

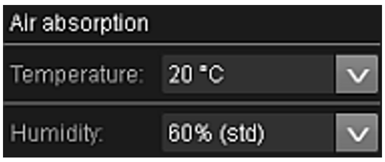
Realizer

The Realizer meter indicates the match between target and unprocessed curve. A good match means the processing effort is low and the system headroom and coherence will not be significantly affected by it (green area). A poor match has the opposite effect and will be indicated by yellow/orange/red LEDs.

Red means the array is not capable of providing the desired level distribution and the calculation will be blocked. In this case, either the target levels or the actual physical array design has to be changed.

Yellow or orange means the system is reaching its limits and you should not demand too much 'Glory' from it without sacrificing headroom and coherence.

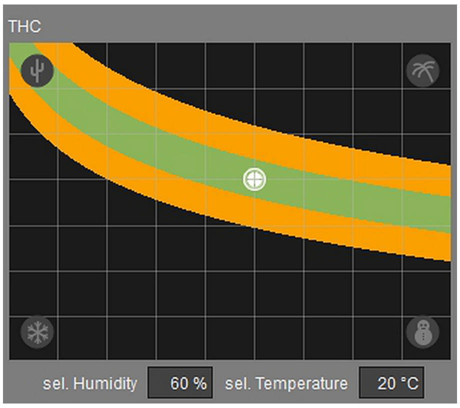
Air absorption



Note: The «Bypass» slot always has the global air absorption settings of ArrayCalc. If different air absorption settings are used on different arrays, the 3D plot will display a warning that your settings do not match.

Air absorption/THC
Each ArrayProcessing slot has its own settings for temperature and humidity which will be taken into account for the optimization and the creation of the processing filter per channel. These initial settings, plus information on the average distance an individual cabinet within an array has to cover, is included in the AP slot data transferred to the amplifier processing channel via the d&b R1 remote control software.

Workflow within R1
Within R1, with recall of an AP slot, these initial settings for the environmental conditions are displayed and loaded into the amplifiers. In a 2-dimensional control for temperature (vertical) and humidity (horizontal) on-the-fly changes to the AP processing in the amplifier channels can be made:



The initial parameters for temperature and humidity the AP slot has been calculated for are indicated with a white cross-hair cursor in the diagram. Moving the mouse pointer over the diagram changes its appearance to another cross-hair cursor. The exact position for temperature and humidity is shown by tool-tip indicators. By clicking within the colored area, new compensation parameters are selected and sent to all amplifier channels of the respective array group. These new active parameters are then displayed underneath the diagram. Within the amplifier, the new compensation response to be applied to each individual channel is calculated and merged into the AP-processing FIR filters.

The colored area indicates the variation range of temperature and humidity combinations for which on-the-fly changes are possible. Within the green area, the accuracy of the modified processing is comparable to an initial calculation of an AP slot, while within the orange area a modification of processing is still reasonable, however some reduced accuracy in absorption-compensation might occur.

In practice the vast majority of applications will be adjustable within the green range, as typically a drop (rise) in temperature is almost always accompanied by a rise (drop) of relative humidity, keeping the physically most relevant parameter for air absorption, the molar concentration of water molecules, nearly constant.

Grouping THC controls
When the initial parameters for air absorption compensation (temperature and rel. humidity) have been set identically for specific AP slots over multiple arrays, the THC controls of the respective arrays can be grouped and controlled from a higher ranked THC group control. This provides easy on-the-fly modification of air absorption compensation within AP processing for all associated arrays with a single click! As the necessary calculations happen within the amplifiers, the change is always very quick and does not change with total system size. A recalculation of AP slots is only necessary when drastic changes in atmospheric conditions occur.

HF Trim
In some applications, unpredictable effects like deflection of sound caused by temperature gradients in the air may disturb the precisely matched frequency response. Most often the result is that the processing which has been calculated to cover a specific listening range is deflected to an area slightly more distant. While this has a linear effect on the low- and mid-band frequency range, the excessive absorption of the high-frequency range can disturb the overall response.

To control and compensate for these effects in a meaningful way, a special function is provided: The ArrayProcessing algorithm automatically calculates additional air absorption compensation for both 10% and 20% of the additional target distance for each source and stores the additional data in the respective AP slot. However, the additional compensation is limited to an absolute additional distance of 30 m (100 ft).

This additional filtering information is transferred to the relevant amplifier channels using R1. It can be accessed in the amplifier channels and the R1 user interface as «HF Trim» option with the settings «0» (off, no additional target distance) «1» (10% additional target distance) or «2» (20% additional target distance). This option can be set,, grouped and handled in R1 in the same way as the conventional HFC controls. Live and in real time. The HF Trim filters for Slot 1 (Bypass) are identical to the conventional HFC filters of the respective "Arc" setup of the system.

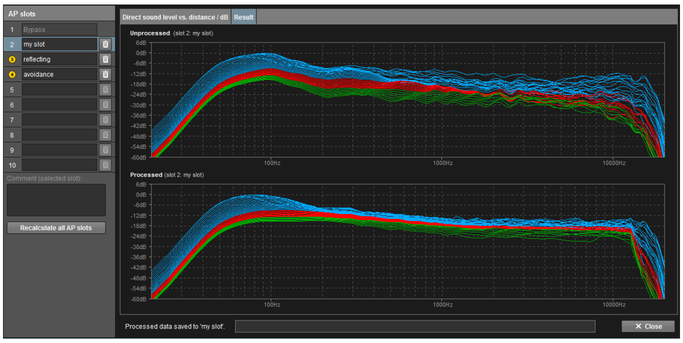
Due to its unpredictable nature, there is no meaningful reason for these data to be used in simulation. As a result, it is only accessible in R1 when the actual system is connected and in operation.

Processing emphasis

Sets the above mentioned Power /Glory option.

Calculate and save
Click the «Recalculate» button to start the optimization process. This saves all settings made and the calculation results to the selected ArrayProcessing slot.

Results



On the «Result» tab, ArrayCalc displays the frequency responses over distance along the array axis before (unprocessed array, top graph) and after the optimization process (processed array, bottom graph).

Note: Please note that the «Result» tab is automatically displayed as soon as the optimization process for a slot is completed.

HeadroomCalc offers a more specific and precise SPL calculation as well as system headroom evaluation during the system design process than the conventional method on the Sources and 3D Plot Views.

The conventional SPL calculation uses generic spectra or sinusoidal excitation signals, all of which are static and do not contain any temporal structure. Also, it is strictly linear and cannot consider non-linearity, which makes it somewhat inaccurate at high levels. Furthermore, the implementation cannot consider the individual capabilities of different d&b system amplifiers.

In contrast, HeadroomCalc works in the time domain and is therefore able to completely simulate the behavior of the digital signal processing in d&b system amplifiers, including non-linearity that is introduced by the limiter circuits.

For this to work, the user must provide the assumed input signal of the system in form of a WAV file to which the simulation is then applied. The result is specific to this input material and up to five user-definable virtual measuring points.

13.1 Motivation and Benefits

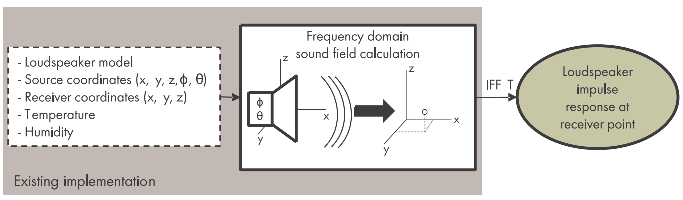
In order of relevance, system designs focus on achieving coverage of the intended listening areas, and, within that coverage, on uniform tonality. With these criteria satisfied, the remaining parameter is the achievable SPL.
While some applications are already well-served with a more generic calculation, others require a very precise prediction, possibly of specific metrics. One of the possible specific application fields here is PAVA-relevant systems (PAVA = Public Address and Voice Alarm). In this case, regulations specify the precise level a system must be capable of producing in order to make emergency announcements that overcome background noise.

Here, the conventional method is not accurate enough and is subject to individual interpretation. Only HeadroomCalc is able to provide precise simulation results, especially when there is specific pre-recorded audio content to be considered.

Similarly, the necessary size of the system amplifier required to achieve a certain minimum SPL is a crucial factor when dimensioning budgets for installation projects in particular, since the difference between two models can mean a substantial variation in total system costs. While the conventional SPL simulation method involves a level of uncertainty and estimation, HeadroomCalc enables precise comparisons, as long as the assumed excitation signal is available.

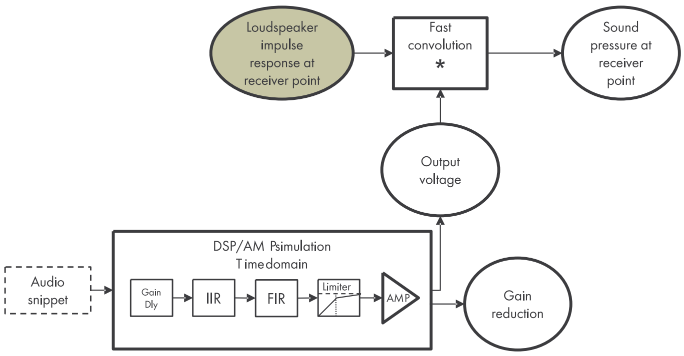
13.2 How does it work?

HeadroomCalc drastically expands on the existing linear model already implemented in ArrayCalc, which yields sound pressure transfer functions in the frequency domain. Those are transformed to the time domain to create the impulse responses of all contributing sources for every receiver point.



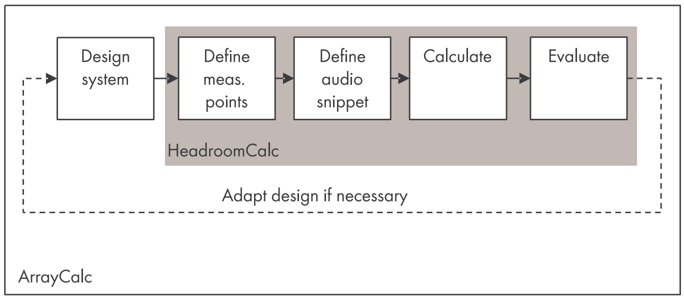
The impulse responses of all contributing sources at a receiver point are convolved with their respective discrete-time amplifier output voltage signals. Those are derived from feeding the user-supplied audio snippet into a simulation of the system amplifier's DSP program in the time domain. The simulation considers all user-adjustable settings that can be made in ArrayCalc, all non-linearities that are incurred by activity of the limiter circuits and the chosen model of system amplifier.

For every receiver point, this results in as many discrete-time sound pressure signals as there are contributing sources, which are summed up to produce the total discrete-time sound pressure signal at this receiver point.
Finally, the SPL metrics are calculated by applying the required time and frequency weightings to the sound pressure signal. As an additional calculation result, the DSP and amplifier simulation also yields the discrete-time gain reduction signals of all sources for display on the GUI.



13.3 Workflow and GUI dialogs

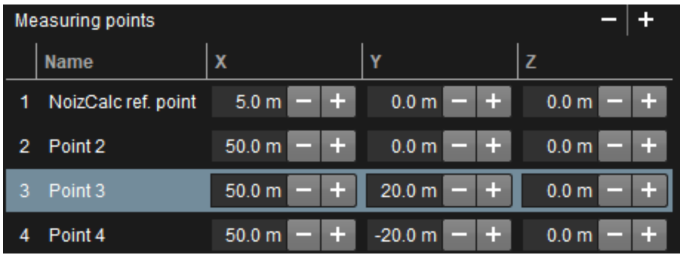
In terms of the workflow, HeadroomCalc is applied as an additional analysis after all regular system design steps that define the overall coverage, tonal uniformity, and alignment have been performed.



13.3.1 Defining measuring points

As HeadroomCalc requires specific receiver or measuring points for which the simulation is then performed, you need to define these first.

1. On the top right of the HeadroomCalc view, click «+» to add a measuring point.
2. Then enter coordinates in the respective fields to position it or click into the 3D view displayed on the «Set measuring points» tab to place it.
3. To name a measuring point, double click its name.
4. Repeat this procedure for up to five measuring points.

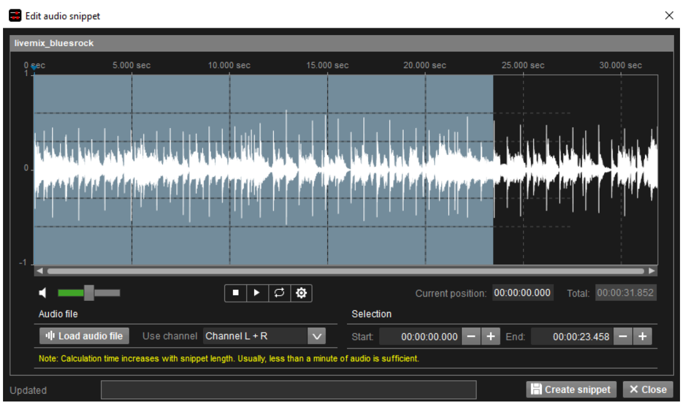


13.3.2 Define audio snippet

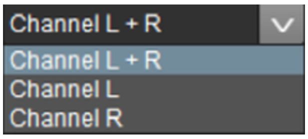
Next you need to define an audio snippet on which the simulation will be performed.

1. Go to the «Signal Selection» section directly below the measuring points.

2. Click «Audio data» to open the selection dialog.
3. Here, load the desired audio file, which needs to be in WAV format.
   ↳ File lengths of up to 90 minutes are possible, but keep in mind that calculation time increases with file length. Usually a representative snippet of shorter length is more than enough.
4. Grab and drag with the mouse to mark a specific section of the audio file or all of it for further use.
   ↳ This will be highlighted. You may also manually enter values in the field under «Selection» to precisely mark a specific piece of the recording.



5. Use the volume and playback controls to listen to the audio file and set your preferred audio interface.
   ↳ The «Use channel» dropdown menu determines whether both or only one channel of a stereo recording will be considered. The playback also reflects those settings.



Note: When «Channel L+R» is selected, HeadroomCalc will use the sum of both channels as the input signal for all sources. Depending on the stereo content of the mix, this may result in parts of the signal being canceled out (for example, when the same guitar track is panned hard left and right and polarity-inverted on one side for an extreme “wideness” effect) or, in extreme cases, in some sources being fed with content that is significantly different from what it would have been in real life (for example, when a loud signal is hard-panned to one side only).

6. Once you are satisfied with your selection and the channel settings, click «Create snippet» to import it into HeadroomCalc.
7. Before starting the calculation, you might want to set the level of the imported audio with the «Gain trim» controls.
   ↳ As you will usually be interested in the maximum achievable SPL, the gain trim should be set so that the system experiences only mild gain reduction. Since the amount of gain reduction incurred can only be seen after performing the calculation, this might be an iterative process of calculating, evaluating the gain reduction, readjusting the gain trim and recalculating.
8. To speed up the process, start by clicking «Auto trim».
   ↳ This will estimate and set the gain trim necessary for the system to just touch gain reduction. Revisit the gain trim later to adjust further as necessary.

13.3.3 Perform calculation

When all parameters are set, click «Calculate».
HeadroomCalc will then calculate the gain reduction for all unmuted sources and the SPL metrics for all defined measuring points. The calculation time depends on the following parameters:

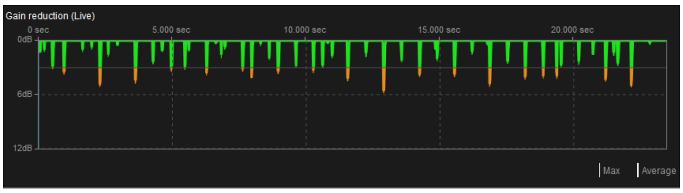
- Number of sources
- Use or non-use of ArrayProcessing
- Length of the audio snippet
- Number of measuring points
- Performance/memory of your computer

During the calculation, the screen changes to the «Results» tab and all graphs and the results table are gradually populated as the calculation proceeds.
At the end, it will show the gain reduction graph, the SPL metrics graph, and the waveform of the audio snippet below one another. All graphs are always aligned with the waveform so that you can easily relate particular events to a time index in the audio.

13.3.4 Evaluate results

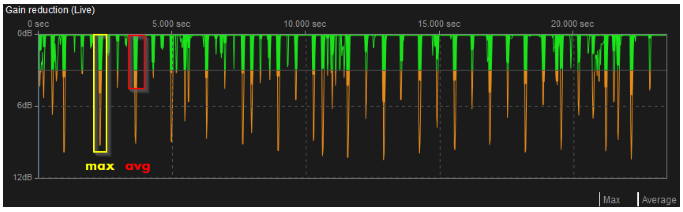
13.3.4.1 Gain reduction

As a first step, it makes sense to evaluate the gain reduction graph at the top.



It is color-coded in the same way as the LEDs on d&b system amplifiers, meaning that green indicates zero to 3 dB of gain reduction, yellow greater than 3 and up to 12 dB of gain reduction, and red anything above 12 dB.

In addition to the amount of gain reduction, the graph also differentiates between the maximum and average gain reduction across all sources. They are calculated per data block of 2048 samples of audio each. The maximum gain reduction represents the highest occurring value of all sources within each data block and is shown as a thin line. The average gain reduction is calculated by taking the mean value of the maximum gain reduction per block and source and is shown as a thick line.
In systems with a very non-uniform headroom distribution which are driven at high levels, this will make the average and max curves diverge significantly. If you encounter such behavior (see screenshot), you should investigate which source(s) contribute most to the divergence and consider whether it may make sense to change the source's settings or change the loudspeaker type to make the headroom distribution more uniform across the overall system.



One way to identify the respective sources more easily is to play the audio snippet with the controls at the bottom of the screen. While doing so, observe the activity of the GR LEDs in the «Sources» widget on the left. They will light up in sync with the audio playback whenever the gain reduction for that source exceeds 3 dB, as they would in a real-world system. Also, a blue playback cursor will pass through all time graphs while the audio is played back to exactly identify the current value of each graph.
Alternatively, you may also perform a recalculation for the same parameters but with some sources muted to better identify which one experiences excessive gain reduction.

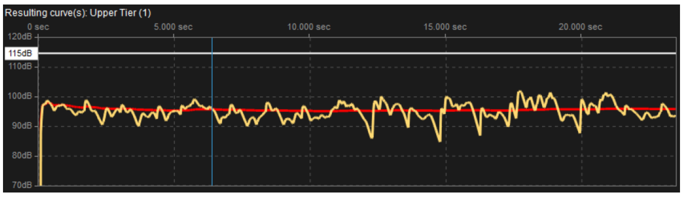
There is one thing that the Gain reduction graph cannot show, which is how much gain reduction will still sound acceptable. This becomes relevant when you attempt to push the system to its absolute maximum by continuously increasing the gain trim. For this, you need to use your own judgement in context with the system and application.
Generally, a graph that has occasional peaks but is completely in the green and would therefore not light up any yellow LEDs can be considered unnoticeable in terms of audible gain reduction. This perception is likely to change with increased and regular instantaneous gain reduction, especially when it falls onto beats of drums, for example, or if the overall graph never returns to zero.
At this point, it also makes sense to briefly listen to the audio snippet using the playback controls to check whether the mono summing mentioned in the section about the audio snippet selection has produced any unwanted artifacts that could influence the simulation in an unexpected way.

13.3.4.2 SPL results

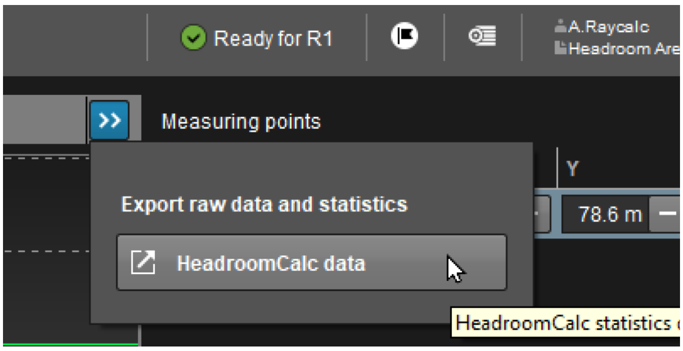
After checking the gain reduction performance of the system, you can choose from a variety of industry-standard SPL metrics and some statistical data in either tabular or graphic form to evaluate them for the selected measuring point.

Result	Live cursor	Live max.	Mem. cursor	Mem. max.
LAeq	95.8 dB	98.1 dB		
LCeq	103.9 dB	105.8 dB		
LAF	96.0 dB	102.1 dB		
LAS	96.3 dB	98.8 dB		
LCF	104.7 dB	108.1 dB		
LCS	104.4 dB	104.7 dB		
LCPk	114.4 dB	118.9 dB		
LCeq-LAeq	8.1 dB	8.7 dB		
LZPk		118.3 dB		
Gain reduction avg.	0.0 dB	5.6 dB		
Gain reduction max.	0.0 dB	5.8 dB		

Use the buttons on the left to show the curves of individual metrics in the graph.



All metrics shown in the GUI, their data, and many more statistics and metrics can also be exported in 1 second resolution in CSV format for individual further analysis/visualization. The data format conforms to what industry-standard professional sound level meters typically offer. Open the pop-out menu next to the measuring points to access the export function.



The «Save to memory» option is a handy tool for comparisons, for example, how the system performs with different system amplifiers. The recommended procedure would be as follows:

1. Perform a calculation with one model of system amplifier.
2. Save the results to memory and observe how much gain reduction the system incurred.
3. Change the model of system amplifier and recalculate.
   ↳ You may need to adjust the gain trim and recalculate a couple of times so that the gain reduction curve is comparable to the previous simulation.
4. Compare the «Live» results to the «Mem.» results to make your evaluation.

13.4 SPL metrics

All SPL metrics are calculated as defined in:
IEC 61672-1:2013-09: Electroacoustics - Sound level meters Part 1: Specifications.

They are derived by applying specific time and frequency weightings or a combination thereof to the raw sound pressure data. Which one is the most appropriate in any given situation depends on what the goal of an analysis is or what has been defined by tender specifications or applicable regulations.

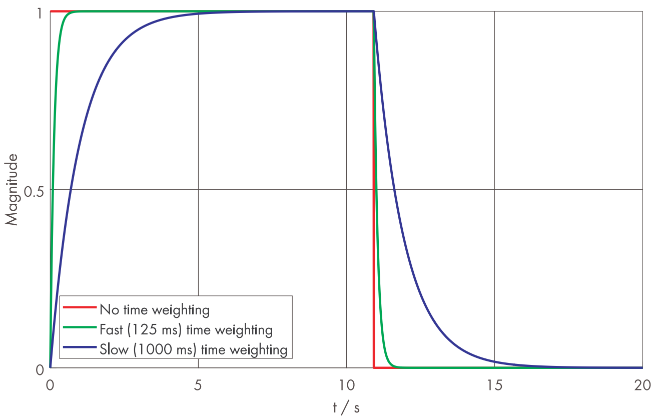
As such, we can only give general guidance in the following paragraphs.

13.4.1 Time weightings

Time weightings with rather short time constants (no time weighting, fast, and slow) result in readings that are dominated by individual, transient sounds even though the overall level is much lower in comparison. This is most obvious when in addition only the maximum occurring value is considered. Lmaxpeak values, for example, only tell you the level of the loudest sample during your observation period. Still, they are sometimes required to verify compliance with health and safety standards, as even a single, extreme transient may already lead to hearing damage, but they are not as relevant in judging the overall loudness or performance of a system.

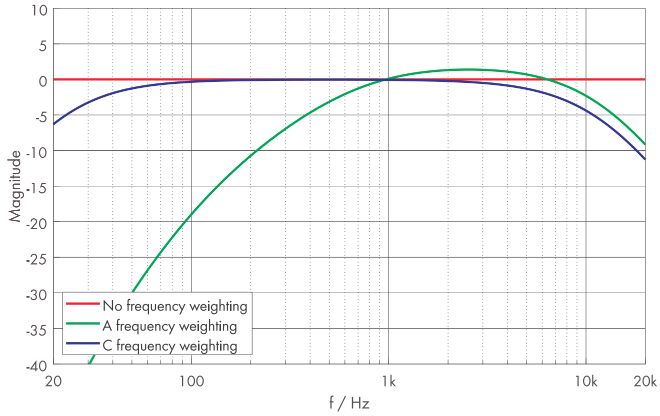
Time weightings that average the sound pressure level over several seconds or minutes to produce an equivalent continuous SPL (Leq) provide a much more accurate representation of the overall loudness. If a given Leq value can be reached with only occasional, mild dips on the gain reduction graph, the system possesses enough overall and dynamic headroom to cleanly reproduce the desired program signal. These metrics are often used by health and safety regulations to evaluate the overall energy dosage received.

The graph shows the effects of commonly used standard time constants with regard to their exponential averaging time when applied to a constant level signal of 11 seconds.



13.4.2 Frequency weightings

Frequency weightings have been defined to accommodate how the sensitivity over frequency of the human hearing changes in relation to SPL. Many standards and regulations do not use the frequency weighting that would be the technically most appropriate for the expected overall SPL.



A-weighting emphasizes the frequency range where the human hearing is most sensitive, and that is most relevant for spoken word communication. It is often applied in health and safety regulations even though technically, A-weighting applies only to level ranges that are significantly lower than typical sound reinforcement levels. Given the spectrum of a typical program signal, an A-weighted value will usually be perceived as much louder than the same numerical value with C-weighting. For the reasons mentioned above, it is usually more meaningful to look at A-weighted values when evaluating the achievable SPL for emergency announcements.

C-weighting is technically much more appropriate for typical sound reinforcement levels because it considers the lower frequencies to a much greater extent and refers more closely to the sensitivity of the human hearing at these levels. Many health and safety standards include a limit for C-weighted peak levels to properly represent transient and very loud bass frequencies in contemporary music.

Z ("Zero"-weighting) is what the name says – no frequency weighting whatsoever. Technical datasheets usually also provide such unweighted data for better comparison.

13.5 HeadroomCalc vs. measurements?

Extensive laboratory and real-world measurements have been performed to evaluate the accuracy of HeadroomCalc and thus the validity of its calculation results.
Both the amplifier output voltage and resulting SPL at the measuring point where analyzed. Unsurprisingly, the amplifier output voltage matched the prediction quite exactly. The SPL measurement showed more deviation that was rooted in the measurement conditions: In laboratory free field conditions, the most significant and expectable influence found was about 2 dB of power compression resulting from running the loudspeakers under test in continuous gain reduction.

Test in real venues with larger and also multiple systems were performed as well. Here, depending heavily on the ratio of reverberant to direct sound, the measurements surpassed the simulation because of the energetic contribution of the reverberant field. In general, it can be stated that the simulation is always conservative in comparison to the measured SPL when room-acoustics are pronounced.

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