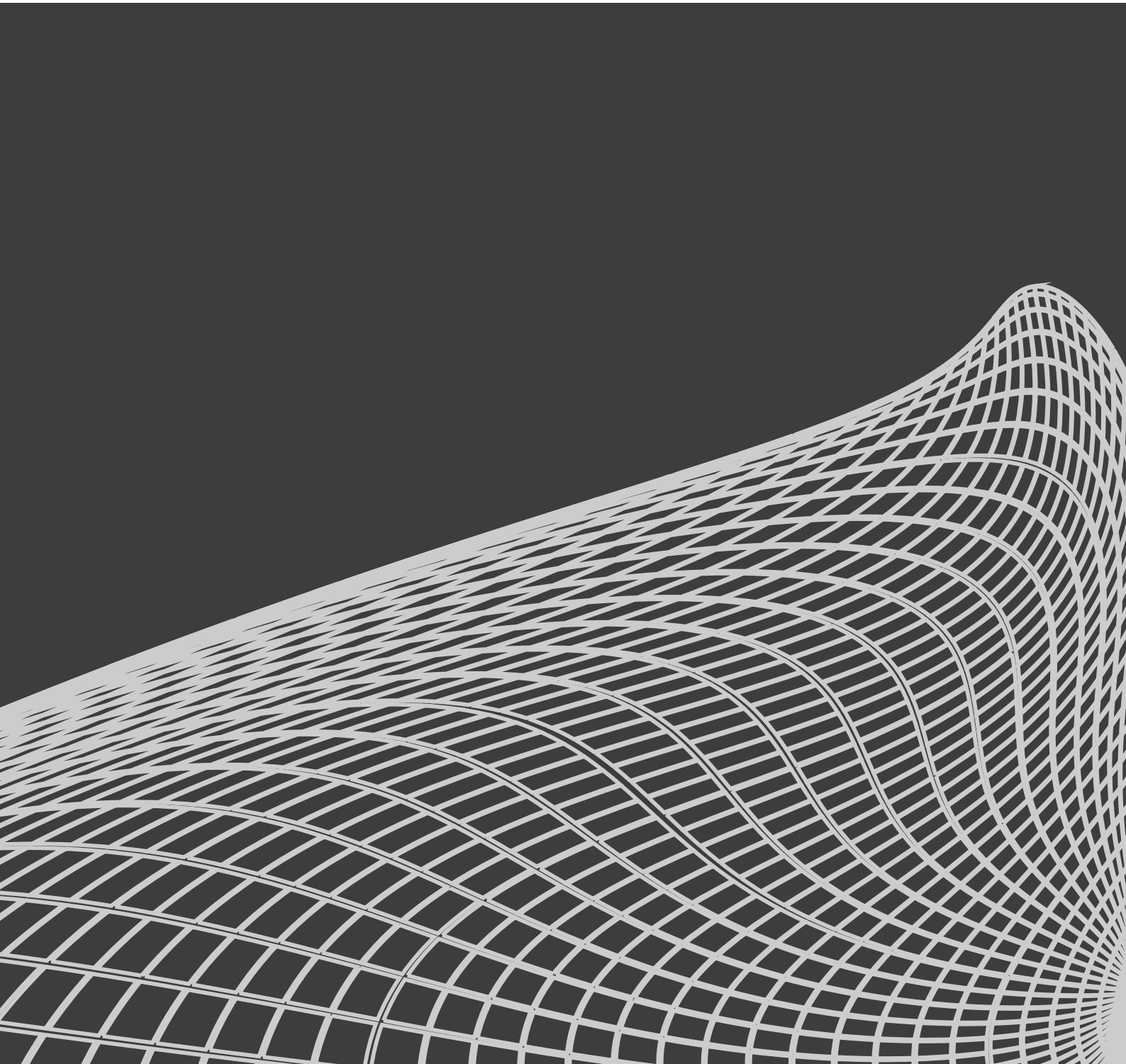
501

TI 501
d&b Soundscape System design and operation
1.12 en



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All previous versions of this document are hereby no longer valid.

Version 1.12:
Complete rework.

General information

TI 501 d&b Soundscape
System design and operation

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1 Introduction

This Technical Information paper will explain the procedure for designing and operating a d&b Soundscape system.

The aim of d&b Soundscape is to provide the most realistic spatial reproduction of sound in a given environment.

d&b Soundscape integrates into the d&b Workflow. The system is designed in the d&b ArrayCalc V10 Simulation software. DS100, loudspeakers and amplifiers are configured and controlled using the d&b R1 V3 Remote control software.

At its heart d&b Soundscape comprises the DS100 Signal Engine and the En-Scene/En-Space software modules.

1.1 DS100 Signal Engine

The DS100 Signal Engine is a 64 x 64 channel digital audio matrix. Inputs and outputs provide extensive signal processing, matrix crosspoints control level and delay. This is useful for handing the combining of multiple consoles into a single festival PA or for routing signals across a venues installation, all from with R1 software which is already in place to program and monitor d&b amplifiers.

The DS100 is available with Milan™ or Dante® network interfaces.

The DS100M is equipped with a Milan™ network interface and additional MADI inputs. Inputs can be selected in 32 channel blocks from either source. AES3 outputs to d&b amplifiers can be provided by additional DS20 Audio Network Bridges (up to 16 channels per DS20).

DS100M/DS100D are equipped with a Dante network interface. AES3 outputs to d&b amplifiers can be provided by additional DS10 Audio Network Bridges (up to 16 channels per DS10).

The DS100 hosts the optional En-Scene and/or the En-Space software modules which control all related matrix functions. Further matrix outputs can be operated manually from R1.

1.2 En-Scene: Sound object positioning

With the En-Scene software module, a d&b Soundscape system can place multiple Sound objects at individual locations on stage or in other areas of the venue.

En-Scene is a form of distance based panning between multiple loudspeakers covering the audience areas. Unlike with stereophonic reproduction it provides an authentic image of all sounds to the whole audience. Sound objects can be placed and moved to any desired position during the show.

1.3 En-Space: Virtual acoustics

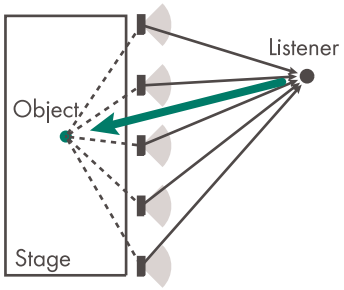
With the En-Space software module, a d&b Soundscape system can add the acoustics of different acoustic spaces to your local environment, be it outdoor or in a venue, which is an essential feature for the reproduction of acoustical instruments.

This can also be used as a theatrical effect which transports the listener to other environments. En-Space is a 3D reverb engine which comes with a set of concert venues of various characteristics and sizes. Using the technology of boundary plane emulation, each room is sampled and reproduced with the highest accuracy and spatial resolution.

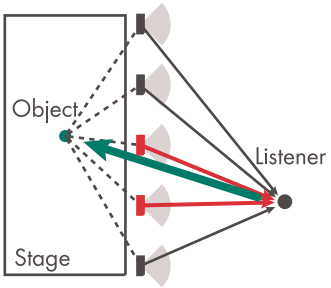
Please note that En-Space can not only be used to create an acoustic environment for the audience but also for the musicians on stage. This application is described in d&b "TI 502 Acoustic shell".

1.4 The En-Scene algorithm

The audience areas will be covered with different loudspeaker groups, so-called Function groups. A function group often is an array of identical loudspeakers equally spaced horizontally and each driven from its dedicated amplifier channel and DS100 output. A function group renders a sound object by reproducing it with different levels and delays using all loudspeakers of the group.

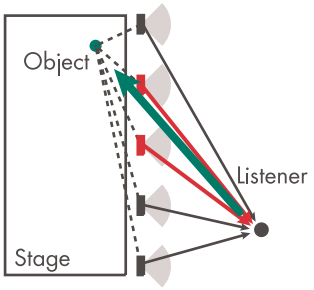


The En-Scene algorithm considers the mix of the listeners psycho acoustical perception as well as acoustic effects between the sources to calculate the transfer functions for the relevant matrix crosspoints. Maintaining the rules of the precedence effect (or "law of the first wavefront") ensures accurate spatial localization.



The loudspeakers with the earliest arrival times (red) provide the relevant direction information to the listener.

Unlike traditional deployments this method of sound reproduction requires multiple loudspeakers to cover each listener. As a consequence loudspeakers covering close audiences need to have a larger horizontal dispersion than with a conventional system design.



The perception of the direction of the object will be the best when the coverage area of the relevant (red) loudspeakers extends to the listener.

En-Scene supports any shape of loudspeaker array. Linear as well as convex or concave designs are possible.

The algorithm considers the position and orientation of the loudspeakers towards the sound object. The individual levels and delays of each source depend on the horizontal angle between Sound object and speaker axis, its distance and the Sound object properties (please also refer to ⇒ Chapter 3.5 “Acoustic properties of Sound objects” on page 12).

When objects are placed inside the coverage area of all sources a level-only panning algorithm will be applied. When objects are moved the perceived level and tonal balance will not be affected.

An En-Scene system comprises a DS100 Signal Engine, the En-Scene software and multiple loudspeakers/amplifiers to cover the audience areas.

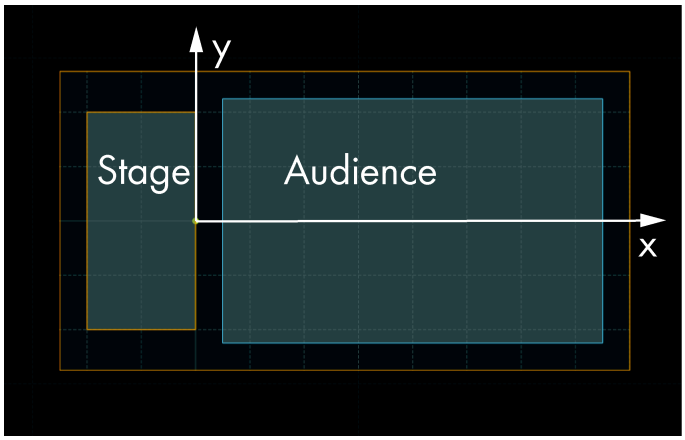
2.1 En-Scene 180 and En-Scene 360

En-Scene supports a variety of venue and sound designs. However, depending on the type of event an essential decision is if a 180 system or a 360 system is required.
Should sound objects represent artists, instruments or other sources on a stage an En-Scene 180 system is sufficient. It is arranged at and around the stage front or proscenium.
Only when sound objects should be played from other directions or located and moved in and around the audience area an additional En-Scene 360 surround system will be required.
Alternatively, an En-Scene 180+ system can be considered which extends beyond the width of the stage without fully surrounding the audience.
This provides a larger-than-life canvas while reducing cost and installation time compared to a full 360 system. 180+ and 360 setups can also be used as an En-Space system for room emulation.

2.2 Venue view

As usual the venue must be entered in the d&b ArrayCalc Simulation software. Make sure that in the ArrayCalc «Project settings» ⇒ «Advanced features» menu the «Soundscape» and «Audio networking» options are enabled.

In ArrayCalc stick to the usual orientation of stage and audience areas.



For En-Scene applications, in addition to the audience areas some additional venue elements must be created: The «Positioning» areas.
Positioning areas (e.g. the stage) need to be rectangular.

In R1, positioning areas are displayed as a reference to enable locating and moving of sound objects. The position of an object can be anywhere in the x-y plane and is not limited to the inside of the area.

Positioning areas also serve to adapt and calibrate coordinate systems of external position control devices which are connected via the OSC interface of the DS100, like tracking systems, VST plug-ins, show control systems, etc.

For further details, please refer to ⇒ Chapter 3.7 “Position control and coordinate mapping” on page 14.

2.3 Sources view and Function groups

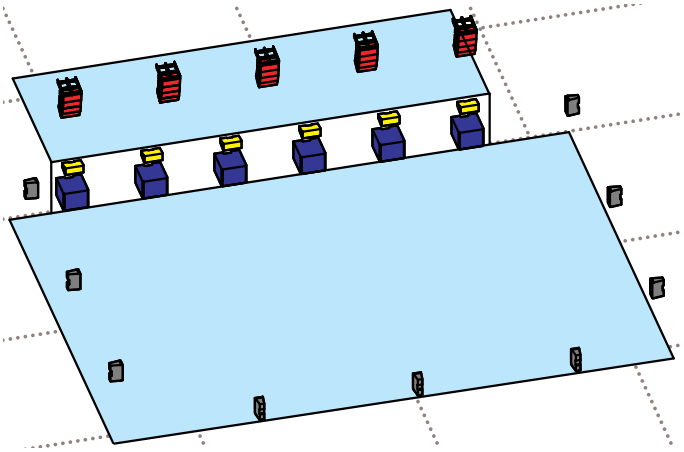
In the «Sources» view enter and place all loudspeakers and assign them to «Function groups».

En-Scene provides up to 16 function groups. There are 11 different modes for function groups available which dictate the desired behavior of En-Scene and En-Space. The corresponding mode must be selected for each group. Function groups are configured in the «Devices» view in ArrayCalc. By default one group for each available mode is provided. However, it is common to not require all the default groups or to have multiple groups with the same mode. Therefore, some editing of the default groups is generally required.

#	Name	Mode	Remarks
1	Mono SUBs	SUB array	LF, no object positioning.
2	Main	Main system	Object positioning.
3	Frontfills	Frontfill	Object positioning.
4	360	Surround	Surround object positioning.
5	SUBs	SUBs group	LF object positioning.
6	Mono outfills	Outfill	Extension to Main system.
7	Delays	Delay line	Delay line with positioning.
8	AUX	Mono out	Mono feed for fills, delays.
9	Ceiling	Ceiling	En-Space only.
10	Outfill embd.	Outfill embd.	Outfill with 360 system.
11	Delays embd.	Delay line embd.	Delay line with 360 system.

You can modify the settings or define more groups as required and select the respective mode for each of them. When multiple audience areas should be covered the best results will be achieved when the loudspeakers serving each area are assigned to their individual function group. Therefore multiple function groups with the same mode are possible.

Please refer to the table in ⇒ Chapter 10 “Function groups overview” on page 24 for a more detailed comparison of their properties.



Example setup with four Function groups:
T10 Main system (red), Frontfills (yellow), 360 speakers (grey) and Y-SUB array (blue)

In the «Sources» dialog, each line array source or group of point sources can now be assigned to **one** function group.

Depending on the size and layout of the venue and the type of program to be played, a different set of groups are required.

A Soundscape system may contain multiple Function groups with the same mode. However, in order to maintain correct imaging one audience area should only be covered by one group of each type. In above example all 360 speakers need to be in one Function group. Additional audience areas (balcony, top tier) should be given their dedicated Surround Function group.

Typical setups may consist of groups with the following modes:

2.3.1 Main (Mode: Main system)

The main system covers the central audience area. It is a horizontal array of equally spaced sources located above the stage. By placing these speakers within a Function Group, the DS100 will be informed to treat all included speaker positions as a single array which supports localization and time-alignment to the performers on stage.

In order to provide maximum efficacy for localization, the spacing of the speakers should not exceed 70% of the distance to the front part of the audience covered by the mains.

Depending on the level requirements, point source loudspeakers may be sufficient, however, line arrays provide a more controlled level distribution towards the far field. Using a higher quantity of wider dispersion cabinets (120°) in the lower section of the array enhances localization at the front while maintaining clarity at the distance.

With line arrays, the «Add array» function is used. As an example, a fivefold main system would typically consist of five identical arrays (the use of two L/R pairs and one single array is also possible but its assignment to matrix outputs is not as straightforward and needs more attention).

The use of ArrayProcessing will be helpful not only to achieve the required throw for the far field but also to create the best transition to the frontfills.

Additional Mains Function Groups can also be used for under-balcony delays (also see 'Delay Line' and 'Delay Line Embedded').

2.3.2 Frontfills (Mode: Frontfill)

Frontfills are commonly used along the stage front to cover the front area of the main listening plane. Placing all lip speakers within a Function Group with the mode Front Fill will inform the DS100 to treat them as a single array which supports localization and time-alignment to the performers on stage, separately from the mains. Unlike the mains mode, the algorithm for front fills will accommodate closer listeners with a wider level distribution from each object. This benefit can be augmented or diminished using the "Spread factor" parameter (⇒ Chapter 3.5.2 "Spread factor" on page 13).

In order to provide accurate localization of the sound objects, the spacing of the speakers should not exceed 70% of the distance to the front row of the audience.

1. In ArrayCalc, this part of the system can easily be entered by selecting «Add point sources» in the «Sources» view.
2. Define the loudspeaker type, number of cabinets, stage width, height, and cabinet aiming.
   ↳ For an easier overview when wiring the system, we recommend you to sort all speakers within each group clockwise (seen from FoH). The layout automatically created by ArrayCalc for an array of frontfills will be in reverse order. Entering a negative value for «Equally spaced along xx m» will create the required order.
3. Assign the frontfills to a function group with the mode «Frontfill».

Additional Front fill Function Groups can also be used for down-fills or other speaker groups which anticipate a close listener area.

2.3.3 360 System (Mode: Surround)

With an En-Scene 180+ or 360 design, sound objects can be moved not only on stage but also in and around the audience areas. For this purpose, additional surround speakers are required. They can be placed along the boundaries of the space and should cover a significant part of the audience area each. Using ArrayCalc, the best combination of mounting height and vertical directivity of the loudspeakers should be evaluated to achieve an even coverage.

Surround speakers automatically combine with groups with the modes Frontfill and Main system, thus they will take over an object when it is within their positioning range (i.e. behind them).

In a typical setup, surround speakers will be less powerful than the Main systems. This is sufficient for single sound objects, but it will obviously not be possible to play the whole audio program at full level from the respective direction.

In ArrayCalc, Surround speakers can be added using multiple point source groups, e.g. three groups for rear and left/right side walls. They must be assigned to a function group with the mode «Surround».

For venues which have multiple audience areas (example: floor and balcony) it may be required to have multiple Function Groups of 360 speakers. This will allow multiple surround systems to seamlessly take-over a sound object when it leaves the mains and front fills.

2.3.4 Subwoofers

There are two modes for Soundscape to handle subwoofers, depending on the artistic intent and the available deployment:

1. SUB array when even coverage and maximum output level is preferred, subwoofers can be configured as a mono SUB array. In this case, they are defined as a Function group with the mode SUB array. In this mode, a single DS100 output will feed all subwoofers and amp delay may be used for LF pattern control.
2. SUBs group Subwoofers can also be deployed as a function group providing imaging of low frequency sound objects. To do so, they are entered as a point source group and assigned to a function group with the mode «SUBs group». In this mode, every subwoofer position will have its own output from the DS100 for individual processing.

   Note: Signals sent to the subs can be controlled on a per-object basis within R1 using the Object Routing parameter (⇒ Chapter 3.5.5 "Sound object routing and level" on page 14). This allows subs to be selectively used (similar to running subs-on-an-aux) while retaining the benefits of Soundscape.

While it is technically possible using the DS100 matrix routing, it is not generally recommended to run the subs directly off the console as is common with traditional PAs. This is because all speakers handled by function groups will have dynamically changing delay times whereas console-driven subs do not.

2.3.5 Delay line

This mode provides localization but does not 'yield' to surrounds. In other words, all objects will be amplified by this group, all the time, regardless of their position. This behavior makes this mode beneficial for audience areas that are not covered by surrounds such as a balcony which only has a 180 deployment or extension audience-areas like an overflow lawn. This mode has also been used to generate a binaural output which can be sent to broadcast or record. For areas that are within a surround system, the mode "Mains" or "Delay line embedded" should be used.

2.3.6 Outfill

The function group mode «Outfill» will produce a mono signal but will adapt the delays of all objects to the main and far field systems in order to provide a smooth transition and time-alignment to the performers on stage. Like "Delay line" this mode is always-on for all object positions. This mode is useful for box-fills and other audience areas that are outside the coverage of the surround speakers. If an outfill is needed which covers an area within the surround system, the mode «Outfill embedded» should be used.

2.3.7 Ceiling

This mode is primarily used for En-Space and does not support object panning. However, ceiling DS100 outputs can still be routed to via DS100 matrix inputs which is useful for overhead sound FX.

2.3.8 Mono Out

This is an unprocessed output which does not support localization or delay processing. It can be useful for utility purposes like foldback, press feeds, or lobby feeds.

2.3.9 Additional Mains Groups

Another requirement may be the addition of a powerful L/R line array system for the far field where detailed imaging is less important than best intelligibility. It is not recommended to add these sources to the function group of the main positioning group since objects would be played at different levels according to their distance to the far field sources. It is best to create a second function group with the mode «Main system» which only includes the L/R sources. In order not to disturb the imaging created by the Main system, the far field arrays should really only cover the remote part of the audience area. ArrayProcessing will help to minimize the vertical overlap with the main system.

This approach has also been used in installations where two mains systems are available, depending on the needs of the event.

2.3.10 Embedded function group modes

For the «Delay line» and «Outfill» Function group modes, there are additional "embedded" (within 360) versions. Those modes should be considered when delay or outfill speakers are used to cover an area within a 360 system. Using the "embedded" mode, the level of an object will be linked to its level on the (first) Main function group. In other words, objects amplified by the first Main function group will also be amplified by these embedded function groups.

As a consequence, when an object is reproduced by the Surround system and therefore disappears from Main and Frontfill speakers, it also disappears from Outfill or Delay line speakers. This avoids disturbing the localization of surround effects by sound from those groups.

2.3.11 Level requirements

Unlike the SPL plot in ArrayCalc the reproduction of a sound object typically does not use all available loudspeakers. In order to provide a smooth level distribution and to achieve consistent headroom, the maximum output of the individual sources needs to be looked at.

In ArrayCalc, use the individual «Mute» buttons of the sources to evaluate the capabilities of the respective sources.

Please note that when different types of loudspeakers or arrays are used in a Soundscape system, the En-Scene and En-Space algorithms will not compensate for differences in system sensitivity. Level adjustments are best made at the individual amplifier channels using the SPL plot on the «Soundscape» tab in ArrayCalc.

2.3.12 Time alignment

Objects in 'Full' delay mode (see 3.5.3) are time-aligned automatically based on the speaker positions. For this reason, Full mode is generally recommended and time-alignment of your system may not be needed.

However, for objects in "tight" and "off" modes, Function groups must be correctly time aligned to each other to achieve the desired accuracy in reproducing objects for all audience areas. As the delay time for each sound object and loudspeaker is created in the DS100 matrix cross point, the time alignment between the source groups should not performed using the signal delay of the amplifiers.

Instead, it is set within ArrayCalc in the «Devices» ⇒ «Function groups» ⇒ «Configuration» table. Here each function group can be assigned a Group delay setting, which is then applied to the DS100 signal processing matrix.

Name	Mode	Group delay	Spread factor
Main	Main system	0.0 ms	1.0
Frontfills	Frontfill	0.0 ms	1.0
FX	Surround	0.0 ms	1.0

This table also provides the possibility to set the Spread factor for each Function group participating in En-Scene.

Please refer to ⇒ Chapter 3.5.1 “Spread” on page 13 and ⇒ Chapter 3.5.2 “Spread factor” on page 13 to learn more about these parameters.

> **Note:** Please note that the «Alignment» view in ArrayCalc may be used to derive the required delays for the source groups, however the settings need to be applied to the function groups as shown above. All delays set within the alignment tab pertain to amp channels and should be reset to the minimum value before taking the file to R1.

Should individual corrections of the delay time within a function group be necessary, e.g. due to speakers mounted in different heights, this must be performed using the individual delay settings of the amplifier channels on the «Sources» tab.

If other factors influence the latency of the system downstream of the DS100, amp channel delay can be used to compensate, if needed.

Example
For example, if ArrayProcessing is used on arrays, 6.2 ms (= 0.3 ms + 5.9 ms) can be entered into all other amp channels to compensate. Or, if the 5D amp is used for front fills, 1.1 ms can be set into all other amp channels to match the 5D inherent latency.

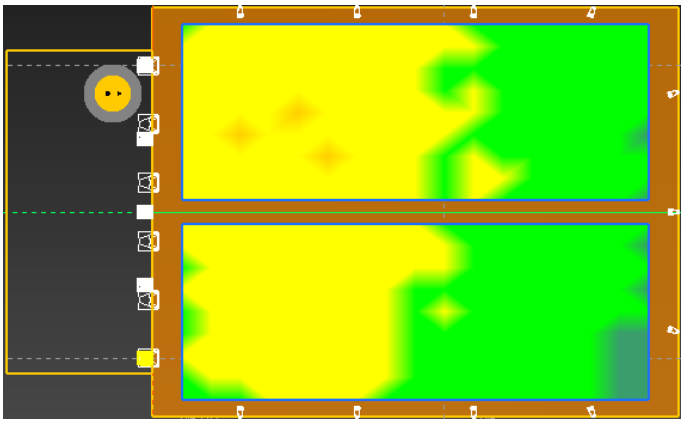
> **Note:** it is generally recommended that minimum amp latency be used for all speakers, regardless of differing latencies. This is to encourage alignment to the performers on stage instead of aligning the PA to itself.

The Function group delay settings as well as the Spread factors can be modified later when online in R1 in the «Devices» ⇒ «DS100» ⇒ «Function groups» view.

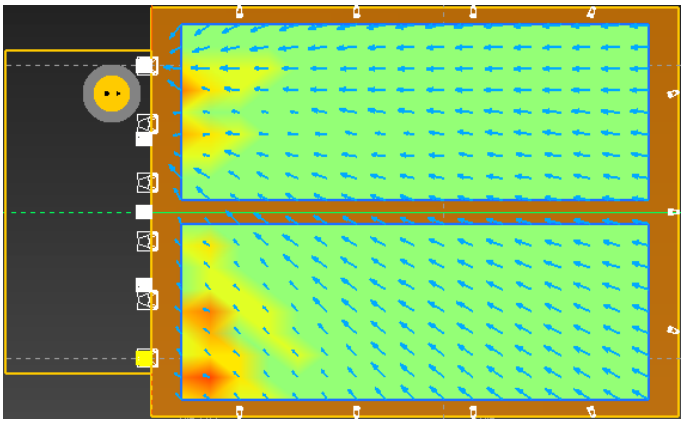
2.4 Prediction of object level distribution and localization

In ArrayCalc on the «Soundscape» tab, the level of the reproduced sound object and its acoustically perceived position can be displayed for all listening areas. Using the object position and its properties (Spread and Delay modes) ArrayCalc plots the average SPL from 1 kHz to 4 kHz using a complex summation of the signals of all sources.

The simulation takes into account all loudspeakers assigned to En-Scene function groups using their respective En-Scene processing and all relevant settings available in ArrayCalc like Function group delay, Spread factor, Mute, Level, Delay or ArrayProcessing as well as temperature and humidity.



Using the same parameter set ArrayCalc also predicts the localization accuracy of the object position for all listening areas. While the arrows point at the perceived direction of the object position, the color mapping indicates the deviation from the actual position of the object.



> **Note:** Please note that the perceived position of an En-Scene object may depend on the program material and its spectral content and transient behavior. ArrayCalc offers the choice between two different perception models.

1. **Precedence:** An empiric model based on the precedence effect or “law of the first wave front”. It follows the observation that when a signal is played from multiple sources, the perceived origin is a combination of the various source positions dominated by the one with the wavefront arriving first at the listener's ear. The model focuses on transient sounds and the frequency range above 4 kHz.
2. **Binaural:** A model based on the research of *M. Dietz* and *H. Wierstorf* using the impulse responses of all sources and a generic set of binaural HRTFs. It derives the perceived direction from an analysis of the interaural transfer functions and resulting binaural cues in the 500 Hz to 2 kHz band. Areas with shorter arrows indicate a reduced convergence of the results.

The two types of predictions (level distribution and localization) allow you to test for design decisions such as:

1. Speaker density: The localization plot will show areas of yellow/orange/red as a listening position is too close to an individual speaker position which causes the algorithm to fail.
2. Function group mode: Does sound come out of an unexpected speaker for a given object position?
   ↳ You may have selected the wrong mode. Switch the function group mode and recalculate for better results.

2.5 Amplifier and channel assignment

After all loudspeakers have been placed they need to be assigned to amplifier channels, DS10 and DS100 matrix outputs.

For ease of use we recommend to maintain to the default order of the function groups and to sort the arrays and point source groups in the «Sources» view accordingly. The order of arrays or point source groups forming one «Function group» should be clockwise around the FoH. To avoid any wiring mismatches during the setup, the order should also be kept within a point source group. Source groups and cabinets can be easily resorted using drag an drop within the «Sources» view.

Amplifier settings

In the «Devices» ⇒ «Sources» table start editing the default settings of amplifier type, amplifier ID, output channel and input source for each «Source» on the «Cabinets» tab to effectively use all amplifier channels.

In installed applications, it might be useful to assign all channels in a straight forward way to avoid unused amplifier and DS10/DS20 channels. With mobile setups, however, the existing amplifier rack configurations as well as cable runs and possible rack locations need to be considered. Main arrays with an uneven number of amplifier channels may leave some channels unused.

The «Configure amps» function in the respective «Devices» ⇒ «Sources» dialog automatically sets inputs to digital when sources are assigned to a DS100 function group. It puts the input source settings of point source groups in an ascending order (1 for output A, 2 for B and so on). Line arrays only have one single input setting which has to be set manually.

2.5.1 Audio network devices

1. In ArrayCalc select the «Devices» view and in the «Audio network devices» table enter a DS100 and the required number of DS10s.
2. Assign the respective DS10 output for each source.
   ↳ Consider that AES3 outputs come in pairs which should feed a 1/2 or 3/4 input pair of one amplifier.
3. Finally assign DS100 outputs to the sources.
   ↳ As this is a Dante network connection to the DS10s it can be done in a straight forward way starting from output 1. The «Configure patch» function in the respective «Devices» ⇒ «Sources» dialog will do this automatically for each source group.

   ArrayCalc will check the validity of IDs, input settings and the channel assignments of all devices before saving the project.

   For an immediate check go to «Devices» ⇒ «Sources», open the «>>» menu and select «Validate patch».

2.5.2 Additional loudspeakers

Additional loudspeakers that are not participating in the d&b Soundscape processing, such as e.g. stage monitors, can also be part of the project, however, they are not assigned to a function group. If required, they can also be linked to DS10 channels and available matrix outputs of the DS100. They can be controlled from the manual matrix controls of R1 (place them at the top of the «Sources» table and use consecutive DS100 outputs.
This is useful for the layout of the respective matrix controls in an R1 «Remote view»).

Cabinets | Devices | Function groups

Cab.	Status	Speaker	Setup	Level/dB	Delay/ms	Filter			Link	Channel name	Input mode	Input src.	Output mode	Ch.	ID	DS device	Output	DS device	Input/Output	ID
Front Fill																				
1		T10 PS-15x105	---	0.0	8.0	CUT	--	CPL0		Front Fill 01	--	D2	Dual Channel / Dual Chai	B	1.02	DS10-1	Digital out 10	DS100	Dante 6	7.01
2		T10 PS-15x105	---	0.0	8.0	CUT	--	CPL0	1	Front Fill 02	--	D3	Dual Channel / Dual Chai	C	1.02	DS10-1	Digital out 11	DS100	Dante 7	7.01
3		T10 PS-15x105	---	0.0	8.0	CUT	--	CPL0	2	Front Fill 03	--	D4	Dual Channel / Dual Chai	D	1.02	DS10-1	Digital out 12	DS100	Dante 8	7.01
4		T10 PS-15x105	---	0.0	8.0	CUT	--	CPL0	3	Front Fill 04	--	D2	Dual Channel / Dual Chai	B	2.02	DS10-1	Digital out 14	DS100	Dante 9	7.01
5		T10 PS-15x105	---	0.0	8.0	CUT	--	CPL0	4	Front Fill 05	--	D3	Dual Channel / Dual Chai	C	2.02	DS10-1	Digital out 15	DS100	Dante 10	7.01
6		T10 PS-15x105	---	0.0	8.0	CUT	--	CPL0	5	Front Fill 06	--	D4	Dual Channel / Dual Chai	D	2.02	DS10-1	Digital out 16	DS100	Dante 11	7.01

Example channel assignment for a 6 x T10 frontfill setup

2.5.3 Dante preset

ArrayCalc can create a Dante preset file which provides the routing of DS100 outputs to DS10 outputs for the project.

1. Go to «Devices» ⇒ «Audio network devices».
2. At the top right of the «Devices» view open the «>>» menu and select «Export as Dante preset file».
3. Load the Preset file and apply it to the Dante network using the Dante Controller software.
4. Then patch the desired inputs/objects from the transmitters in the network (e.g. mixing console or DAW) to the respective DS100 input channels.

   Note: It is recommended that all Dante devices be set to the smallest stable latency in Dante Controller. This is because all alignment to the performers is being handled within the DS100 so additional system latencies should be mitigated.

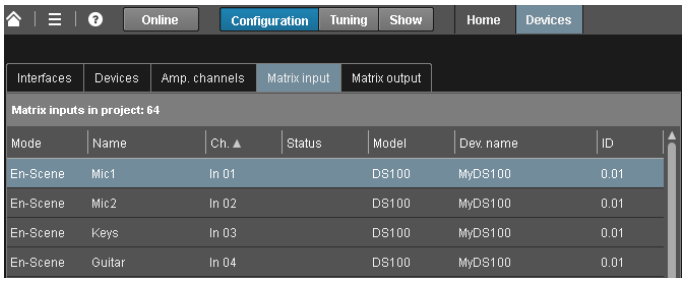
R1 downloads the project configuration to the devices, controls all user parameters, and manages the properties and positions of the En-Scene Sound objects.

> **Note:** Please note that in contrast to the older **"*.dbac2"** and **"*.r1p"** file formats, R1 and ArrayCalc now share the same **"*.dbpr"** project file format.
> Changes and additions made in R1 can be saved at any time while the file can still be further modified in ArrayCalc, e.g. to adjust loudspeaker positions or change channel assignments and create a new Dante Preset file.

1. Open the project file in R1
2. Select «Tuning» mode and go «Online».
3. In the «Overview» tab, reload the default snapshot created in ArrayCalc.

3.1 DS100 Matrix input settings

1. In «Configuration» mode open the «Devices» view and the «Matrix input» tab.



2. On the «Properties» tab on the right-hand side, enter a «Name» and select «En-Scene» as «Input configuration» for all required input channels.
   ↳ When changing to «Tuning» mode, the «General» and «EQ» tabs on the right-hand side provide the input processing options for each channel: «Gain», «Delay», «Mute», «Polarity», and an 8-band parametric «EQ».

   Matrix inputs can be manually routed to any output (be it assigned to a function group or not).

   En-Scene inputs are automatically routed to function group outputs based on their position. However, they can also be manually routed to outputs which are not assigned to a function group. This can be handy for building a press-feed, holdback mix, or binaural mix of your sound objects.

3.2 DS100 Matrix output settings

The «Matrix output» tab is also to be found on the «Devices» view.

In «Tuning» mode, the «General» and «EQ» tabs on the right-hand side provide the processing options for each output channel: «Gain», «Delay», «Mute», «Polarity», and a 16-band parametric «EQ».

3.3 Positioning view

In the R1 «Home» view in «Configuration» mode, create a new «Positioning view» and assign a «Positioning area» to this view.

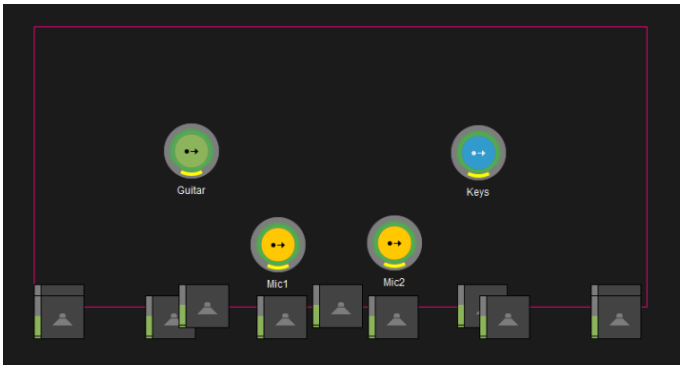
The positioning area serves as a reference to place and move the respective objects within the venue. However, the position of an object can be anywhere in the x-y plane and is not limited to the inside of the area.

There can be more than one positioning area in the design (e.g. a setup with multiple stages). You can create as many positioning views as you like.

3.4 Sound objects

Each En-Scene input channel may be represented by a «Sound object» in a «Positioning view».

1. In «Configuration» mode, drag sound objects either from the «Controls» menu on the right or from the «Matrix inputs» table on the left into the view.
2. In the «Properties» menu «Name» and «Color» of each object can be defined.



Positioning area with sound objects, indicators for level (green) and spread (yellow)

Sound objects can be moved individually or in groups.

1. To group sound objects, go to the «Groups» view.
2. Add a new group to the tree, and assign the respective «Matrix inputs» to this group.
3. Back in the «Positioning view» add another «Sound object» and assign it to the group.
   ↳ With «Relative» selected, all objects of the group can be placed individually to each other but the group can be moved as a whole. The triangle at the upper right corner of the icon indicates its exact position.

   It may be helpful to create multiple positioning views for one positioning area, e.g. using one for moving the groups and another for arranging objects within a group.

> **Note:** While sound objects can be freely added to a positioning view, the associated inputs must be in En-Scene mode for those controls to operate (see ⇒ Chapter 3.1 “DS100 Matrix input settings” on page 12). Otherwise, if the object is targeted to an input which is in Matrix mode, the object position will not affect the sound.

3.5 Acoustic properties of Sound objects

In «Tuning» mode, the acoustic properties of each sound object (i.e. Matrix input) can be configured. This includes level, delay, EQ, delay mode and spread.

3.5.1 Spread

The «Spread» of an object defines whether it is reproduced rather focused or wide. Wide objects deliver a less sharp image and provide a more even level coverage.

Wide objects make use of the level of more speaker positions than focused ones and therefore are less demanding regarding the SPL capability of the individual loudspeakers.

The «Spread» of an object is defined in the «Devices» view on the «Matrix input» ⇒ En-Scene tab for each input. It can also be controlled by a «Digital» control placed in a «Remote view» and assigned to the respective «Matrix input» or group of inputs. The «Spread» value ranges from 0 (focused) to 1 (wide).

The amount of spread incurred by an object is dependent on each function group. For example, front fills will become 'wider' at a faster rate than mains when an objects spread is widened. This affect can be controlled via the spread factor control for each function group (see ⇒ Chapter 3.5.2 "Spread factor" on page 13).

3.5.2 Spread factor

Using the Spread factor, the relative spread of all objects can be adjusted for each function group. In R1 in the «Devices» ⇒ «DS100» ⇒ «Function groups» table, a value between 0.5 and 2 can be set for each group. The default factor is 1.
The Spread of all objects is multiplied with the set value and the result applies to the respective function group. The resulting Spread of an object is limited to 1.

Spread factor can be used to help overcome sub-optimal speaker spacing as it 'morphs' a function group back towards a traditional speaker system.

> **Note:** A front fill function group already has a wider spread factor optimized for closer listeners. It can be widened further or its spread can be reduced to act more like a mains function group.

Spread and Spread factor are not effective for function groups without object positioning (Outfill, Outfill embedded, SUB array, Ceiling, Mono out).

3.5.3 Delay mode

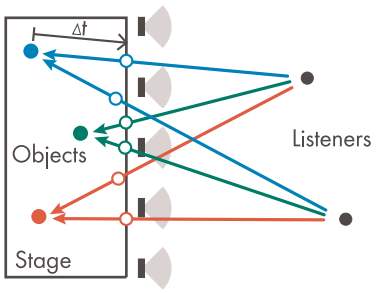
Three delay modes are available for sound objects. «Tight», «Full» and «Off». The delay mode can be configured individually for each object. The «Delay mode» option is also found in the «Devices» view on the «Matrix input» ⇒ «En-Scene» tab.

When the Delay mode is set to «Off», En-Scene only uses level shading to image an object by a function group. All relevant sources of the function group will reproduce the objects simultaneously, only the delay setting of the whole function group will be applied for alignment with other function groups. Delay mode «Off» may provide less precise localization of an object, however, it avoids signal artifacts with fast moving objects.

When the delay mode is set to «Full», level and delay are used. Objects will be reproduced by all function groups with the latency equaling the actual acoustical path length providing consistent time alignment in the entire venue. All objects in «Full» mode ignore the "Function Group Delay" parameter as time-alignment is handled in multiple axis simultaneously. For this reason, «Full» mode is generally recommended.

For acoustic or locally amplified instruments the «Full» mode should be selected in order to preserve the image and timing of direct to reproduced sound of those sources.

When the delay mode is set to «Tight», the total latency of its reproduction through all function groups is minimized. The signal delay of each object is reduced by Δt equaling the distance to the closest loudspeaker of the Function group. Relative delay values between the sources of the group are kept therefore the localization of the object is not affected.



«Tight» mode beneficial for a mix of electronic instruments and/or pre-recorded material, in order to reduce relative delays between the channels depending on the placement of the objects in the «Positioning view» (i.e. stage), thus keeping the mix "tight".

«Tight» mode also is of advantage for moving Sound objects as it reduces the variation speed of the signal delays.

> **Note:** Please note that the delay modes «Off» and «Tight» are dependent on correct delay settings of all function groups in the R1 «Devices» ⇒ «DS100» view. The delay mode «Full», however, does not apply these settings, instead it uses the geometry of the system layout to determine the delay values.

3.5.4 Object placement

With delay modes «Tight» or «Full» selected, the sound of an object is reproduced by multiple loudspeakers with different delay times each depending on the distance between the object and the respective loudspeaker. Consequently, objects in different positions **are** played through the same loudspeakers with different delay times. This behavior corresponds to the natural propagation of sound. However, it may create audible effects regarding the relative timing of instruments. Therefore, like with acoustic music, make sure that the physical distribution of the band or orchestra in relevant listening directions stays within acceptable limits. As a rule of thumb, a distance of 10 m (30 ms) between rhythmic instruments should not be exceeded.

3.5.5 Sound object routing and level

As a default, all objects will be routed to all function groups or - in a 360 configuration - to the combined function groups of Main and Surround or Front fills and Surround.

Using the Sound object routing matrix for each En-Scene input, the send gain to each function group can be set manually. This can be useful to modify the mix for a particular function group or to completely remove an object from it. At each crosspoint, either "Mute" or the "Gain" can be set using absolute or relative faders. The latter being particularly useful when multiple crosspoints are selected.

As an example, to factor in the direct sound from stage, the level of all voices on the front fills could be reduced. Also removing the vocals from subwoofers may be beneficial.

The matrix can be accessed under «Devices» ⇒ «DS100» ⇒ «Sound object routing» or with the respective control in a Remote view.

3.6 OSC control

Most parameters within the DS100 Signal Engine cannot only be controlled by R1 but also via OSC messages, for example, level and delay of matrix inputs, outputs, and crosspoints as well as Scene recall and object positions. A detailed description of the DS100 OSC protocol is provided at dbSoundscape.com under Interoperability.

d&b provides plug-ins for DAWs and several console manufacturers supporting the OSC protocol (see https://github.com/dbaudio-soundscape).

The d&b En-Bridge software is also available to route control messages from a number of third-party controllers for advanced setups. It can also be downloaded at dbSoundscape.com under Interoperability.

3.7 Position control and coordinate mapping

En-Scene object positions can be controlled by external devices like mixing consoles, show control systems, DAWs or tracking devices.

In most cases, it will be necessary to map the coordinate systems of the controlling device to the En-Scene system. En-Scene uses the coordinate system given by ArrayCalc internally.

The mapping is done in «Configuration» mode in the R1 «Devices» view on the «DS100» ⇒ «Coordinate mapping» tab. Make sure the coordinate system of the positioning device is parallel to one side of the positioning area and input the values the device provides for two diagonal points of the rectangle. The OSC message to address sound objects using this mapping is displayed on the tab.

3.8 Temperature

In R1 on the «Devices» ⇒ «DS100» ⇒ «Ambient conditions» tab, the current ambient temperature can be set. The value is used to align signal delays with the actual speed of sound. Delay values set manually (function group and matrix crosspoint delays) are not modified by this parameter.

An En-Space system comprises a DS100 Signal Engine, the En-Space software and multiple loudspeakers/amplifiers to cover the audience with the reverberation signature of the sampled space.

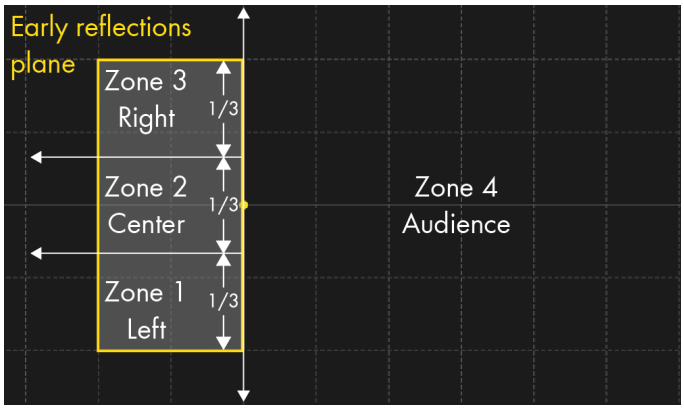
The En-Space convolver engine provides 64 independent output channels for up to 64 loudspeaker sources to create the sound field.

The En-Space and En-Scene software modules can be operated in parallel on the same DS100 Signal Engine. They can share matrix inputs and outputs as well as loudspeaker sources.

4.1 Venue view

As usual, the venue must be entered into the d&b ArrayCalc Simulation software. Make sure that in the ArrayCalc «Project settings» ⇒ «Advanced features» tab the «Soundscape» and «Audio networking» features are enabled.

For En-Space applications, in addition to the audience areas, the «Early reflections» plane must be defined. The Early Reflections plane is used to scale the reverberant patterns onto the available loudspeakers as well as automating signals sent to the appropriate En-Space engine input. In most applications this will be the stage. The «Early reflections» plane splits the venue into four zones with individual early reflection patterns.



If En-Scene is used, the routing to the En-Space zone inputs 1-4 is automated based on the objects position. Sound objects located on this plane will obtain dedicated early reflection patterns of the measured stage, depending on their spatial positions (Zones 1, 2, 3). Sound objects outside the Early Reflections plane are assumed to be in the audience and will be automatically assigned to En-Space Zone 4 which incurs less early reflections and a more even spatial level distribution, just like in the original acoustic environment.

> **Note:** Please note that the «Early reflections» plane must be rectangular and a rotation of it is not permitted.

4.2 Sources view

Speakers should surround the whole audience, if possible. When not possible, a 180+ speaker deployment can be used to create the sensation of surround speakers. Surround speakers will typically be placed along boundary walls or - in open-air environments along the boundaries of the audience area including the stage front.

In order to achieve the best emulation result, every single loudspeaker should cover as much of the audience area as possible. Therefore speakers with wide horizontal dispersion are advantageous. Additionally, point source speakers can be linked to create an array with extra-wide coverage. Simply link them on a single amp channel as you normally would in ArrayCalc.

The ideal vertical coverage pattern depends on the loudspeaker mounting height and the size of the audience area. The larger the distances to be covered and the lower the speaker is placed the higher will be the required vertical directivity to achieve the desired effect for the whole audience. When mounting height is restricted like for example in outdoor situations, 24C column speakers may be useful.

Often a system will combine both En-Space and En-Scene performance. All En-Scene function groups will be used for the En-Space reproduction simultaneously and no additional configuration in ArrayCalc is required.

If no En-Scene setup is present the En-Space sources also have to be assigned to the respective Function groups that match their physical position in the venue. This will ensure the correct match of convolver response to loudspeaker position.

The configuration of amplifier channels, DS10 and DS100 outputs as well as patching of the Dante network is done in the same manner as described above for En-Scene.

En-Space comes with a set of sampled concert venues, modern architecture and classical ones in different sizes which range in reverberation time from 1.3 seconds to 5.6 seconds. This extremely wide range allows En-Space to be useful for a wide range of musical, theatrical, and architectural purposes.

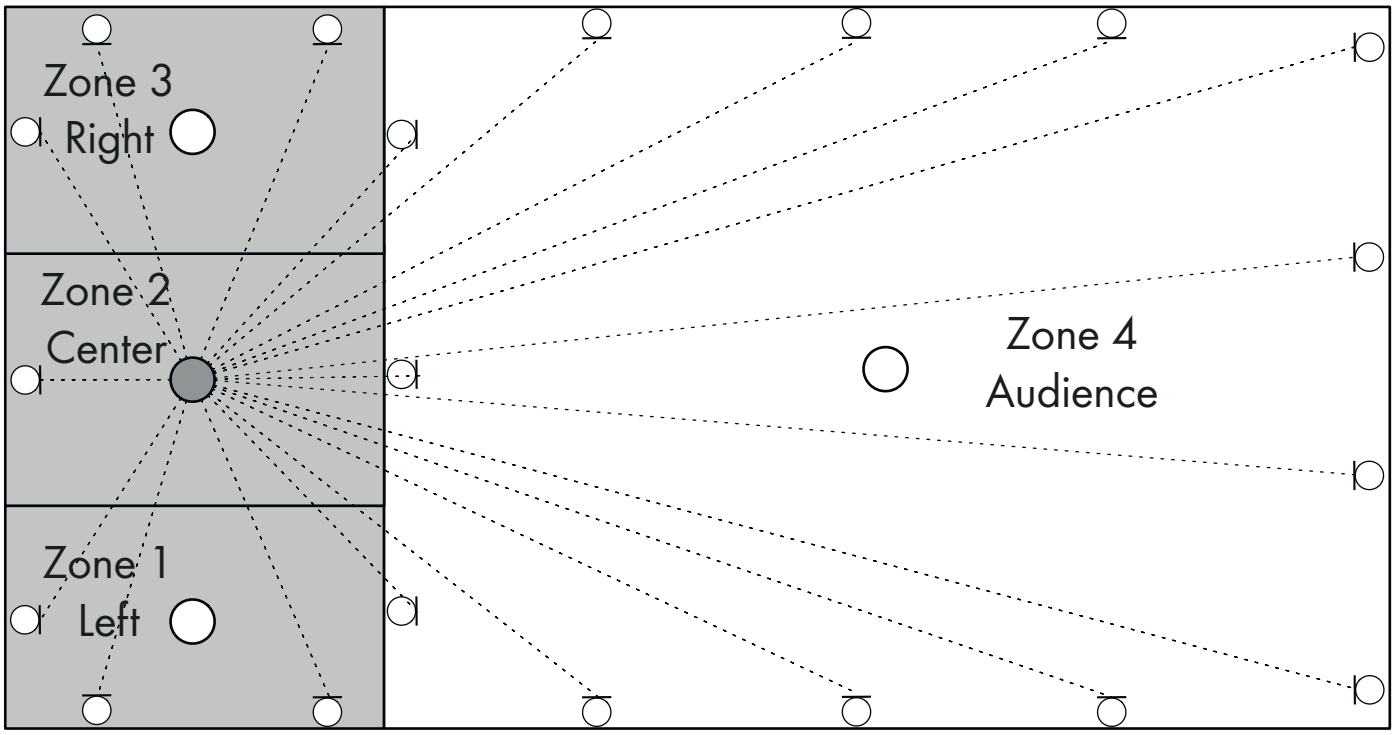
When combining a sampled space with your local environment be aware that the acoustic responses of both rooms will add up. It is not possible to shorten the reverb of the actual venue, it will always be extended. Therefore, the venue should have a considerably shorter reverberation time than the sampled space otherwise the audible effect is limited.

En-Space applies the unique technology of boundary plane emulation. The room response is not created from free field measurements taken from within the space but from a set of 144 boundary plane responses for 64 positions distributed along the circumference and stage lip of the venue. The sampled responses are taken from boundary measurements at the walls, which is exactly the location from where the En-Space loudspeaker sources will later reproduce them. This generates the sound field of the sampled space with highest accuracy.

For each of the 64 En-Space loudspeaker positions, the library provides individual boundary responses for objects on stage and objects in front of the stage. For the Main system there are individual responses for all 4 zones in order to most accurately reproduce early reflections.

The 64 positions were chosen to easily match the sources of an En-Space setup, however, sampled space and actual venue do not have to have the same size or shape. The DS100 automatically maps the En-Space convolver outputs to the matrix outputs in such a way that the respective boundary responses of the sampled space match each actual loudspeaker position and function.

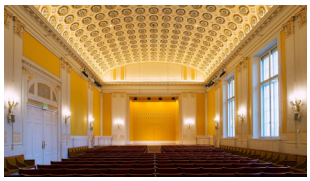
En-Space uses all available Function groups of the types SUB array, Main system (7), Frontfill (9), Surround (40), SUBs group (7), Outfill, Mono out, Delay line (7) and Ceiling (7). The number in brackets indicates the maximum amount of positions per group which will be given individual uncorrelated boundary responses. More positions are possible, thus gradually increasing correlation.



5.1 Space #1: Modern - small

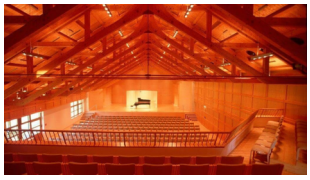
Blaibach Concert Hall
Capacity: 200 seats
Reverberation time: 2.0 s
(T40: 200 Hz - 2 kHz)

5.2 Space #2: Classic - small



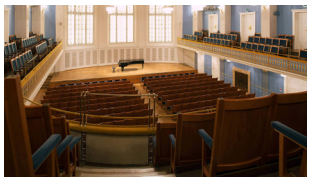
Schubert Saal, Vienna Concert Hall
Capacity: 350 seats
Reverberation time: 1.9 s
(T40: 200 Hz - 2 kHz)

5.3 Space #3: Modern - medium



Angelika-Kauffmann -Saal, Schwarzenberg
Capacity: 600 seats
Reverberation time: 1.7 s
(T40: 200 Hz - 2 kHz)

5.4 Space #4: Classic - medium



Mozart Saal, Vienna Concert Hall
Capacity: 700 seats
Reverberation time: 2.1 s
(T40: 200 Hz - 2 kHz)

5.5 Space #5: Modern - large

KKL Luzern
Capacity: 1900 seats
Reverberation time: 2.6 s
(T40: 200 Hz - 2 kHz)

5.6 Space #6: Classic - large

Großer Saal, Vienna Concert Hall
Capacity: 1850 seats
Reverberation time: 2.4 s
(T40: 200 Hz - 2 kHz)

5.7 Space #7: Modern - medium 2

Bing Concert Hall, Stanford
Capacity: 850 seats
Reverberation time: 2.2 s
(T40: 200 Hz - 2 kHz)

5.8 Space #8: Theater - small

Teatro Alighieri, Ravenna
Capacity: 830 seats
Reverberation time: 1.3 s
(T40: 200 Hz - 2 kHz)

5.9 Space #9: Cathedral

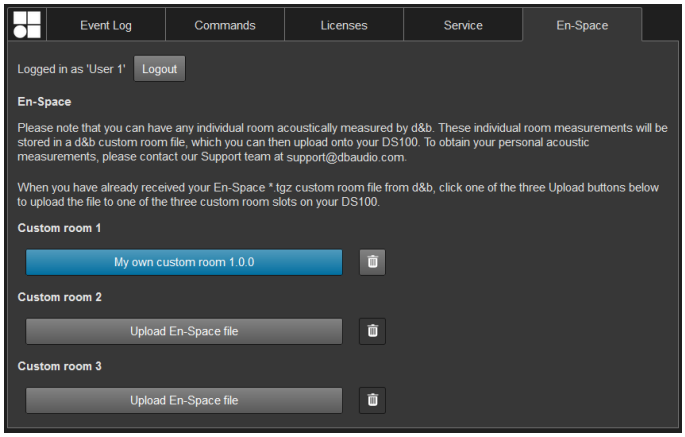
Basilika San Vitale, Ravenna
Capacity: - -
Reverberation time: 5.6 s
(T40: 200 Hz - 2 kHz)

5.10 En-Space Custom rooms

En-Space provides three storage slots for individually recorded room responses. En-Space room measurements are a paid d&b service.

A Custom room comes in form of a database (**Roomname**.tgz) which needs to be loaded into to your DS100. To do so...

1. Connect your web browser to the web interface of your DS100 using its IP-address and navigate to the «En-Space» tab.
2. Log in using the password **dbaudio**.
3. Select one of three «Custom room» slots to upload the file.



1. Open the project file in R1, select «Tuning» mode and go «Online».
2. In the «Overview» view reload the default snapshot created by ArrayCalc.
3. In the «Devices» view under «Devices» ⇒ «DS100» on the «En-Space Room» tab, select the sampled space.
   ↳ The «En-Space Zones» tab contains level faders and an 8 band EQ for each of the four source zones of the space. It may be useful to create a new Remote view with the relevant controls, such as a List element for the Room selector and Faders for the «En-Space Send» of the input channels.

Any DS100 input in either En-Scene or Matrix configuration can be used for En-Space reproduction. The mix to the four zones can be performed in different ways as described below.

6.1 Zone mixing at matrix inputs

On the «Devices» ⇒ «Matrix inputs» ⇒ «En-Space» tab, each matrix input provides a control for the En-Space reverb level of the channel. It is supplemented by four more controls for the sends to the Zones Left, Center, Right, and Audience.

Each matrix input will need the master En-Space fader adjusted as well as the Zone send. The Zone should be chosen based on the general location of the source within the venue.

Example:
- All mics on stage-right should be sent to En-Space Zone 1 (“left”).
- All mics center-stage should be sent to En-Space Zone 2 (“center”).

This approach is common with systems that do not use En-Scene such as Virtual Orchestra Shell applications (VOS).

Note: Identical signals (including left and right of a stereo) should NOT be sent to two zones simultaneously. This will result in the system replicating the same concert hall twice and may sound muddy. Instead, the two nearly identical signals should both be sent to the same Zone (probably Zone 2 “center”).

Note: DS100 inputs in En-Scene mode will automatically overwrite and automate En-Space Zone sends when they are moved (see ⇒ 6.2 "Zone mixing at the mixing console").

6.2 Zone mixing at the mixing console

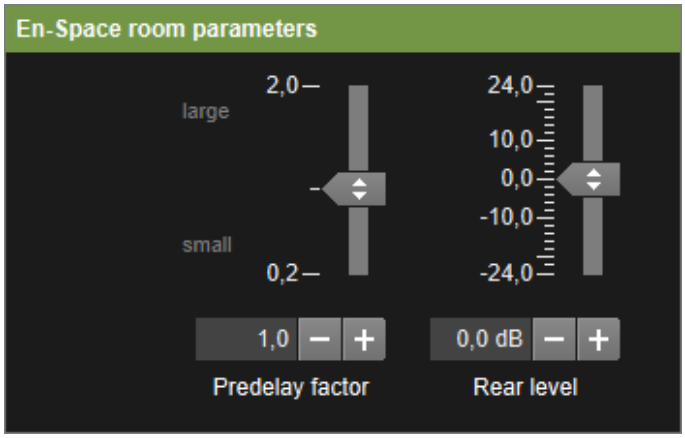
Alternatively, the mix to zones can be performed at the mixing console using for example four AUX sends for each zone routed to four DS100 inputs in Matrix configuration, each is feeding one zone.

6.3 Zone mixing by En-Scene

For all matrix inputs configured for En-Scene operation, the zone mixing is performed automatically according to the position of the object.

On the «Devices» ⇒ «Matrix inputs» ⇒ «En-Space» tab, only set the overall reverb level for the channel, the four zone levels will be controlled by En-Scene.

Independent of the En-Space venue selected, its response can be modified to match the characteristics of your Soundscape design. The room parameters can be set on the «Devices» ⇒ «Room» tab or using the respective template in your workspace.



7.1 Predelay factor

The «Predelay factor» scales the predelays of all boundary responses of the selected venue. The range extends from 0.2 to 2. The default value of 1 maintains the response of the original measured venue. Larger values delay the onset of the room response, while smaller values shorten this time.

The «Predelay factor» can be used to modify the perceived size of the room. Please note that Predelay factors smaller than 1 should only be applied when the actual venue is smaller than the En-Space room selected. Otherwise the En-Space reproduction of an object may occur earlier than the direct sound.

7.2 Rear level

The «Rear level» fader adjusts the En-Space level gradient from the front to the back of the room. The range extends from –24 dB to +24 dB, where positive values gradually increase the reverberation level towards the back.

«Rear level» can be used to adjust the direct to reverberant field ratio along the depth of the room. A main system with very high directivity (line arrays) will cause a smaller level drop over distance than point source speakers and therefore may need a higher En-Space level at the rear.

The «Rear level» fader can also compensate for the higher level drop towards the rear of a large measured venue when reproduced in a considerably smaller room.

7.3 En-Space output faders

If necessary, the En-Space level of individual loudspeaker positions can be modified using the Output faders («Devices» ⇒ «En-Space Outputs») of the respective En-Space output.

8.1 Manual matrix

The DS100 signal matrix can also be operated manually by the controls on the «Devices» ⇒ «Devices» ⇒ «DS100» tabs - or more conveniently by controls in a Remote view. The «Matrix crosspoint» control provides an array of level and delay controls for a user defined range of matrix crosspoints which can be operated individually or by means of multi-selection.

The menu button on the input and output fields of the matrix control opens the respective input and output processing options. Also with En-Scene and/or En-Space at work, the available DS100 matrix output channels can be used for manual matrix operation.

8.2 Grouping of channels

Matrix input and output channels, just like amplifier channels, can be grouped in R1.

Note: Please note that this does not "link" the channels.

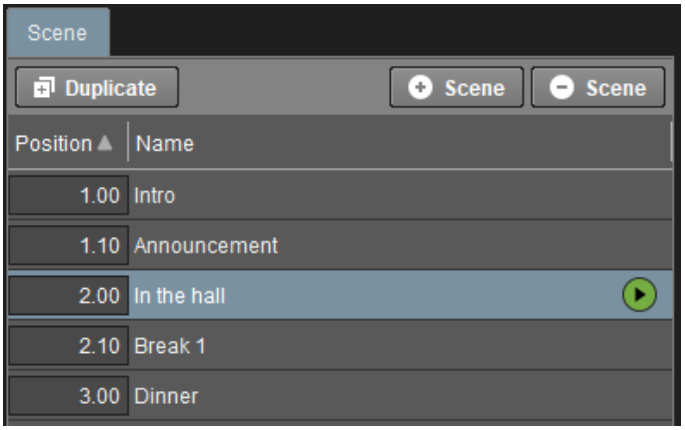
The known controls such as EQ and Faders - also relative - can be placed in a remote view and applied to the group.

8.3 Snapshots

The Snapshot options of R1 also apply to all DS100 control elements. Make sure all parameters to be captured in a snapshot are represented by respective controls in a Remote view - for example input or output processing, En-Scene sound object positions, En-Space sends or the room selector.

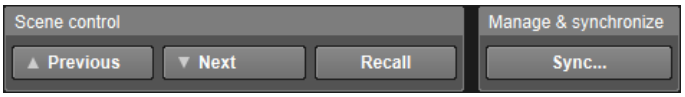
8.4 DS100 Scenes

The DS100 provides a local memory for Scenes. Scenes are organized numerically in a range from 0.01 to 999.99 and contain a user selectable set of parameters which may include En-Scene, En-Space and/or Matrix settings.



Scenes are created in R1 («Device Scenes»). This is done by choosing the Positioning and Remote views that contain the desired DS100 control elements, and assigning a name and a scene number (n.mm). If a set of Scenes with an identical selection of controls but different settings needs to be created the «Duplicate» function can be used followed by an «Update» of the duplicated Scenes without having to select the relevant views for each Scene.

Object positions on a «Positioning view» can be stored with absolute coordinates or relative to a coordinate mapping of a «Positioning area» (e.g. the stage, see ⇒ Chapter 3.7 “Position control and coordinate mapping” on page 14). Using a coordinate mapping allows the Scene to be used in different venues with different stage sizes simply by creating the mapping with the respective number in the new project.

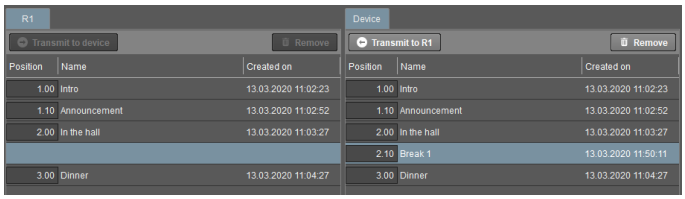


Scenes can be recalled using the «Recall» button or by stepping through the DS100 scene memory using «Previous» or «Next». All three functions can also be assigned to switches in a Remote view or triggered by OSC commands. A Scene recall by OSC is done on the basis of the scene number.

While having the DS100 connected, new scenes or updates to existing scenes are directly applied to both the DS100 and the R1 scene memory.

Scenes can also be created offline in R1 without the DS100 being present. When recalling a scene offline R1 simulates its behavior. The «Previous» or «Next» commands are not available.

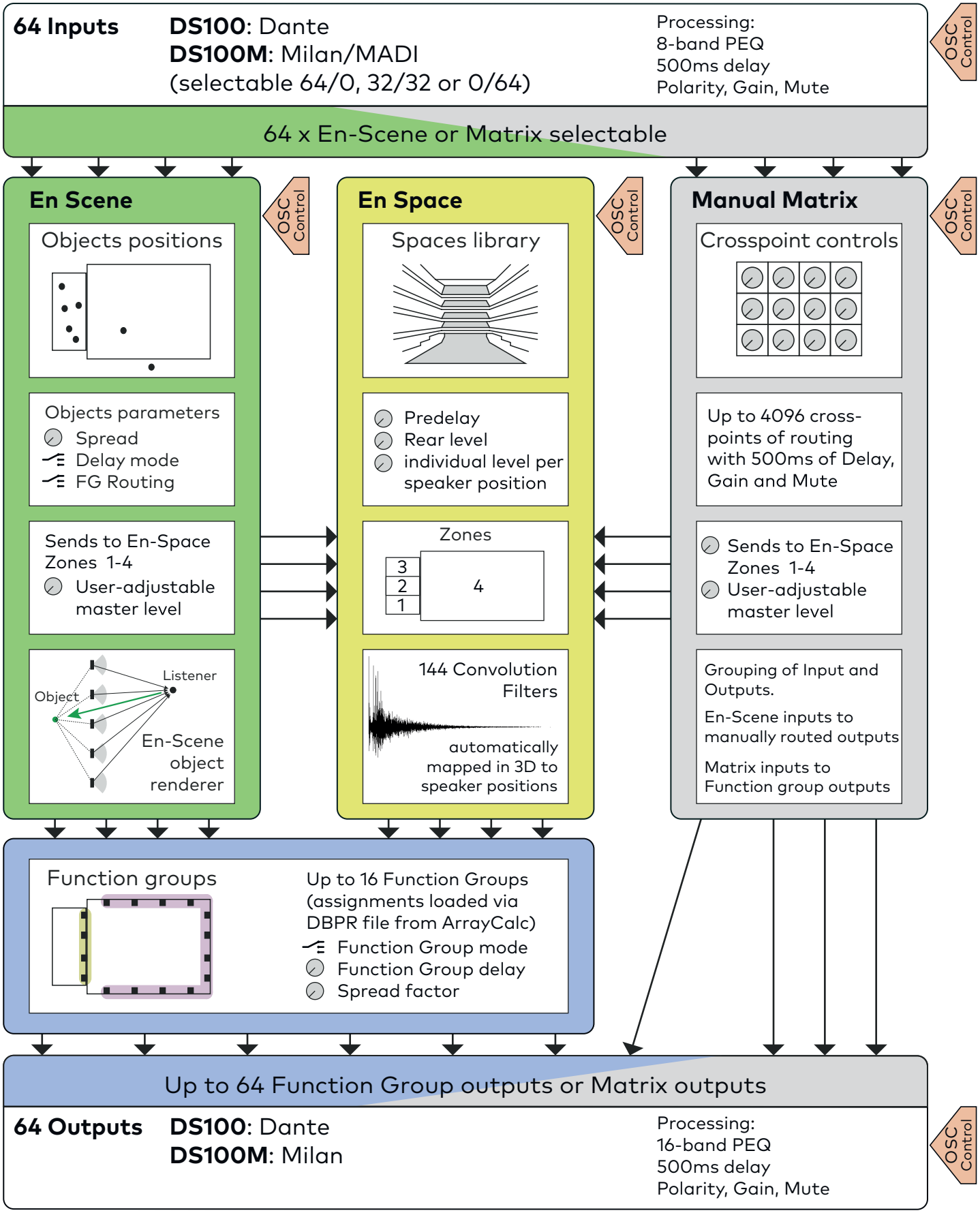
As soon as the DS100 is connected, the «Manage & synchronize» dialog allows the synchronization of Scenes between R1 and the DS100 in both directions.



Within the Scene list, a yellow 'different values' icon (≠) is shown for all scenes where R1 data does not match the device data. Scenes available within the device only, can be identified by a light gray font being used for showing the scene name.

Having different scene contents and scene lists within the device and R1 might lead to unexpected behavior.

> **Note:** Please note that a Scene does not include the DS100 channel input modes (En-Scene/Matrix). It only contains parameters of the selected DS100 and no other settings of R1 or any connected devices.


64 Inputs
DS100: Dante
DS100M: Milan/MADI
(selectable 64/0, 32/32 or 0/64)
Processing:
8-band PEQ
500ms delay
Polarity, Gain, Mute
OSC Control
64 x En-Scene or Matrix selectable
En Scene
Objects positions
Objects parameters
Spread
Delay mode
FG Routing
Sends to En-Space Zones 1-4
User-adjustable master level
Listener
Object
En-Scene object renderer
En Space
Spaces library
Predelay
Rear level
individual level per speaker position
Zones
3
2
1
4
144 Convolution Filters
automatically mapped in 3D to speaker positions
Manual Matrix
Crosspoint controls
Up to 4096 cross-points of routing with 500ms of Delay, Gain and Mute
Sends to En-Space Zones 1-4
User-adjustable master level
Grouping of Input and Outputs.
En-Scene inputs to manually routed outputs
Matrix inputs to Function group outputs
Function groups
Up to 16 Function Groups (assignments loaded via DBPR file from ArrayCalc)
Function Group mode
Function Group delay
Spread factor
Up to 64 Function Group outputs or Matrix outputs
64 Outputs
DS100: Dante
DS100M: Milan
Processing:
16-band PEQ
500ms delay
Polarity, Gain, Mute

FG Mode		Main	Surround	Frontfill	Delay Line	Delay Line Embedded	SUBs group	SUB array	Outfill	Outfill Embedded	Mono Out	Ceiling
Default name in ArrayCalc (user adjustable)		Main	360	Frontfills	Delays	Delays embd.	SUBs	Mono SUBs	Mono outfills	Outfill embd.	AUX	Ceiling
En-Scene	Supports localization and Spread Factor?	✓	✓	✓	✓	✓	✓	No, mono down mix	No, mono down mix	No, mono down mix	No, mono down mix	Not applicable - does not support En-Scene. Only used for En-Space and matrix routing.
	Supports dynamic time alignment per Sound Object?	✓	✓	✓	✓	✓	✓	✓	✓	✓	No	
	Supports 'Sound Object Routing' parameter for customized mixes?	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	
	Requires FG delay for objects in Full mode?	No. Ignores 'Function Group Delay'									✓	
	Requires FG delay for objects in Tight mode?	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	
	Requires FG delay for objects in Off mode?	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	
	'Always on' for all object positions?	No	No	No	✓	Mimics the level behavior of the 1st Main function group	✓	✓	✓	Mimics the level behavior of the 1st Main function group	✓	
	Yields object amplification to other function groups?	✓	✓	✓	No		No	No	No		No	
En-Space	Number of impulse response files available to replicate unique reverberation.	7 x 4 zones	40 x 4 zones	9 x 4 zones	7 x 4 zones	7 x 4 zones	7 x 4 zones	1 x 4 zones	1 x 4 zones	1 x 4 zones	1 x 4 zones	7 x 4 zones
	Supports En-Space for 'Matrix' mode DS100 input?	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Matrix Routing	Can En-Scene mode DS100 inputs be manually routed using matrix crosspoints?	No. Object position and spread dictates level distribution. All matrix crosspoint controls for En-Scene inputs routed to function group outputs are disabled. 'Object Routing' feature is available for this purpose and can be adjusted per object and per function group.										
	Can Matrix mode DS100 inputs be manually routed and delayed using matrix crosspoints?	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓

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