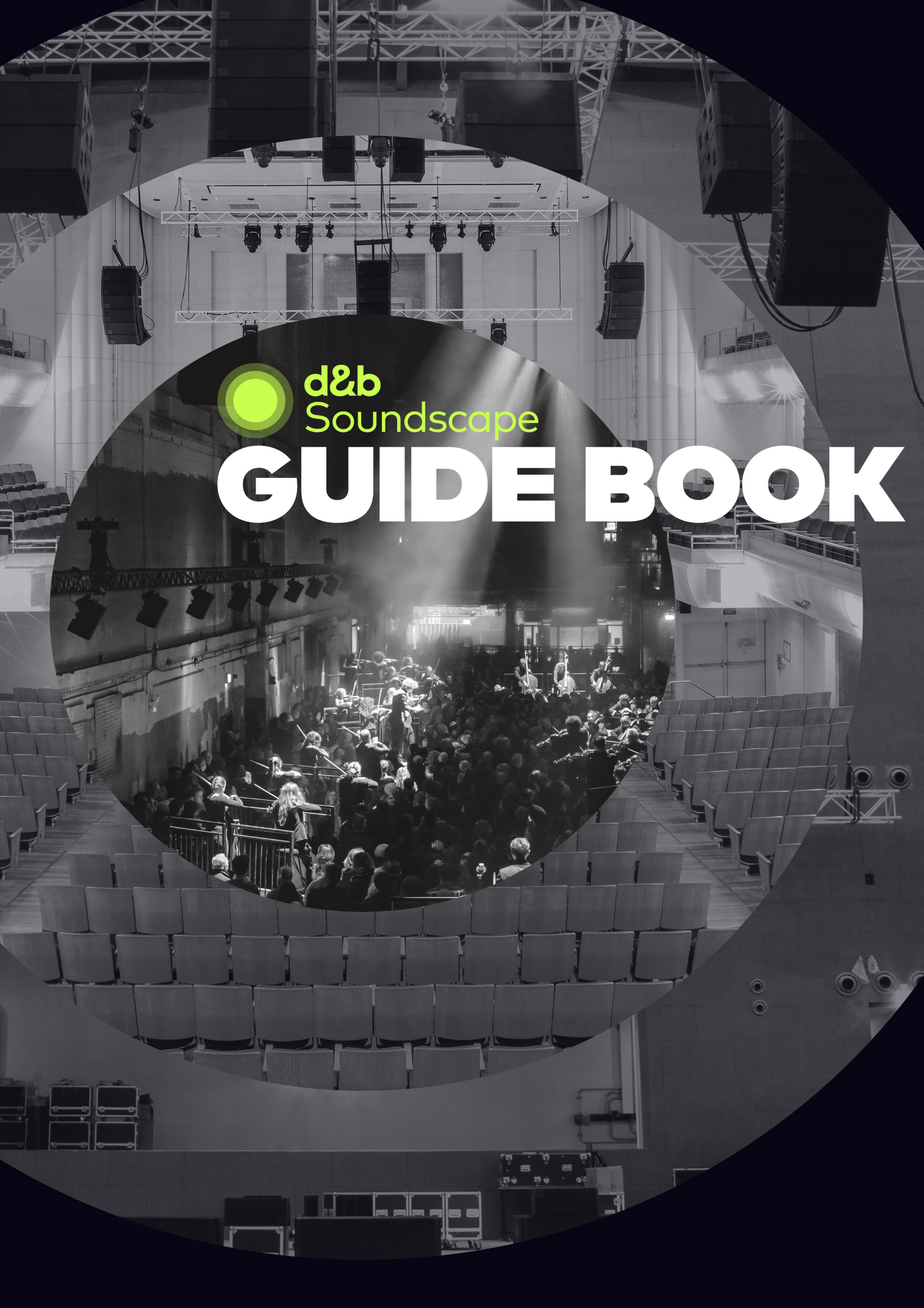
d&b Soundscape

GUIDE BOOK



Welcome!

Soundscape is a toolbox of solutions, which have different meanings and applications to different types of users. For this reason, this document is extensive and covers many topics. It is intended to represent as many perspectives and applications as possible.

If you've been with d&b on the Soundscape journey between 2018 and 2024, you may remember documents entitled *TI501: d&b Soundscape system design and operation* and *TI503: DS100 device redundancy*. The information within those documents is now covered here within the Soundscape Guide Book, rendering them obsolete.

You may also find value in these other documents, located at dbsoundscape.com (here).

- DS100/DS100M hardware manual
- DS100/DS100M datasheet
- DS100/DS100M firmware release notes
- DS100 OSC protocol
- TI 502: Acoustic shell, Stage acoustics using En-Space

Tips for the use of this document:

- Click in the the table of contents to navigate directly to that section.
- Terms in *Italics* indicate text within a software application.
- Terms in **Medium text** indicate steps to follow within an application or Soundscape specific terminology.
- Terms in **Bold text** indicate important information to pay attention to.
- Anywhere that the terms "DS100" or "Soundscape processor" are used without specifying the I/O size or network protocol indicates the process is identical for all DS100 and DS100M variants.
- Blue text is a bookmark to the section written in the link. e.g., What's new in Soundscape?
- Blue text with the word "here" is a link to an external website and will require an internet connection.
- Something missing or unclear? Email support@dbaudio.com so we can continually improve!

Products covered within this document

- Create.Control: Free Soundscape control software for mix engineers. *Soon to be released.
- ArrayCalc: Free d&b prediction software which is required to program Soundscape.
- R1: Free d&b control software which is required to program and monitor Soundscape and d&b amplifiers.
- DS100 Signal Engines: hardware for low-latency processing for d&b amplifiers and speakers.
- En-Scene: sound object positioning and auto-speaker timing add-on for the DS100.
- En-Space: real-time emulated room acoustics add-on for the DS100.
- En-Space Custom Rooms: Acoustic measurement and reproduction service.

Contents

1. What's new in Soundscape?

1.1. En-Space Custom Rooms service

Q1 2025: d&b has announced the availability of a proprietary personalized room measurement service which allows for any room acoustics to be captured by d&b for electronic replication within any En-Space enabled Soundscape system. See En-Space Custom Rooms for more information.

1.2. I/O sizes for DS100

Q2 2025: DS100 and DS100M processors now support up to 128 inputs with smaller sizes available for purchase. Owners of the DS100 or DS100M purchased before Q2 of 2025 receive a **free** firmware upgrade (version 3.xx) via R1 which expands the 64 input channels to 128. New customers now have three channel count options, which are available in both DS100 and DS100M variants:

- **Small** - 64 in x 24 out
- **Large** - 64 in x 64 out
- **XL** - 128 in x 64 out

The newly supported I/O is part of a licensing process which allows for any DS100 or DS100M to be upgraded at a later date. See Processing Hardware for more info.

1.3. Reduced DS100 Redundancy pricing

Q2 2025: To enable Processor redundancy at a reduced cost, owners of the DS100 XL or DS100M XL (including users who received the free upgrade listed above) can purchase a second processor at a greatly reduced price. This also applies to new customers who purchase a DS100 XL or DS100M XL. Contact a d&b reseller for more info. This offer is not available for DS100/DS100M Small or DS100/DS100M Large.

1.4. Create.Control software

Coming in 2025: a new Soundscape control software which allows for the spatialization of a Soundscape show to be fully configured offline or online. A project file can be transported between Soundscape systems with quick and intuitive rescaling. See the brief description of Create.Control for more information.

2. Benefits of Soundscape

Stereo deployment: 24x GSL + 14 KSL (15-19 amplifiers)

Soundscape 180° deployment: 49x KSL (13-25 amplifiers)

2.1. Added clarity with less SPL

"Writing about music is like dancing about architecture"… but we'll try our best.

With Soundscape, every audio input (musician, presenter, playback channel) has its own unique arrival time to the PA system. In other words, no two speakers carry the same mix of signals. The result is a more natural sound with a sense of depth and clarity, with less work in the mixing console.

Additionally, every speaker in the room is now in-time with every individual acoustic sound on stage. This has sonic benefits for the front seats and other audience areas that hear the acoustic level from the monitors and instruments. For applications such as traditional music, spoken word, and theater, this sense of realism will help make the speakers sound natural and transparent.

Added clarity often results in less need for SPL from the mix because additional level is no longer the primary tool to achieve separation in the mix. Engineers frequently report they are happier to run shows 6-9dB lower than with traditional PA systems.

2.2. Automatic system alignment, per source

Multiple system tunings, at the same moment
Gone are the days where a system calibration has to be based around a single one-size-fits-all approach when it comes to speaker alignment. With Soundscape, every input has its own optimum system alignment based on its unique location.

More Consistent Coverage
Delay processing allows for more consistent coverage when compared to object-based systems which utilize level-based panning only. When every speaker has a unique delay time for every sound object, the object's level can be reproduced by a larger number of speakers while giving the listeners the perception of correct localization. To say it another way: delay processing allows the listener to perceive that a sound is coming from the side when it's being amplified by speakers directly in front of them. Because more speakers can handle the signal without compromising localization, the speaker system will generally have more even coverage.

Smaller speakers or arrays
As mentioned above, more speakers work together as a team. This results in higher SPL capabilities, which can lead to smaller speaker positions and less expensive amplifier models.

Supporting more speaker deployments
Delay processing allows Soundscape to support some speaker configurations which are not technically feasible with level-based systems. This includes asymmetrical venues, compromised speaker positions that are not evenly-spaced, shows in the round that have surround speakers, and more.

Multi-use venue applications
Many multi-use spaces rely on a crew to load-in and load-out for each event configuration. Example: Tuesday has a luncheon event on the south end of the room. Wednesday hosts a small trio to perform during a dinner event at the other end of the room. Instead of moving speakers, Soundscape allows a venue to move the sound electronically. While this may cost more initially, the additional investment is recouped through the savings in labor costs over the life of the system. This approach allows for a fully capable system to be installed once and for all, and allows a non-technical operator to recall presets to quickly and easily reconfigure the sound system by moving the audio image as needed.

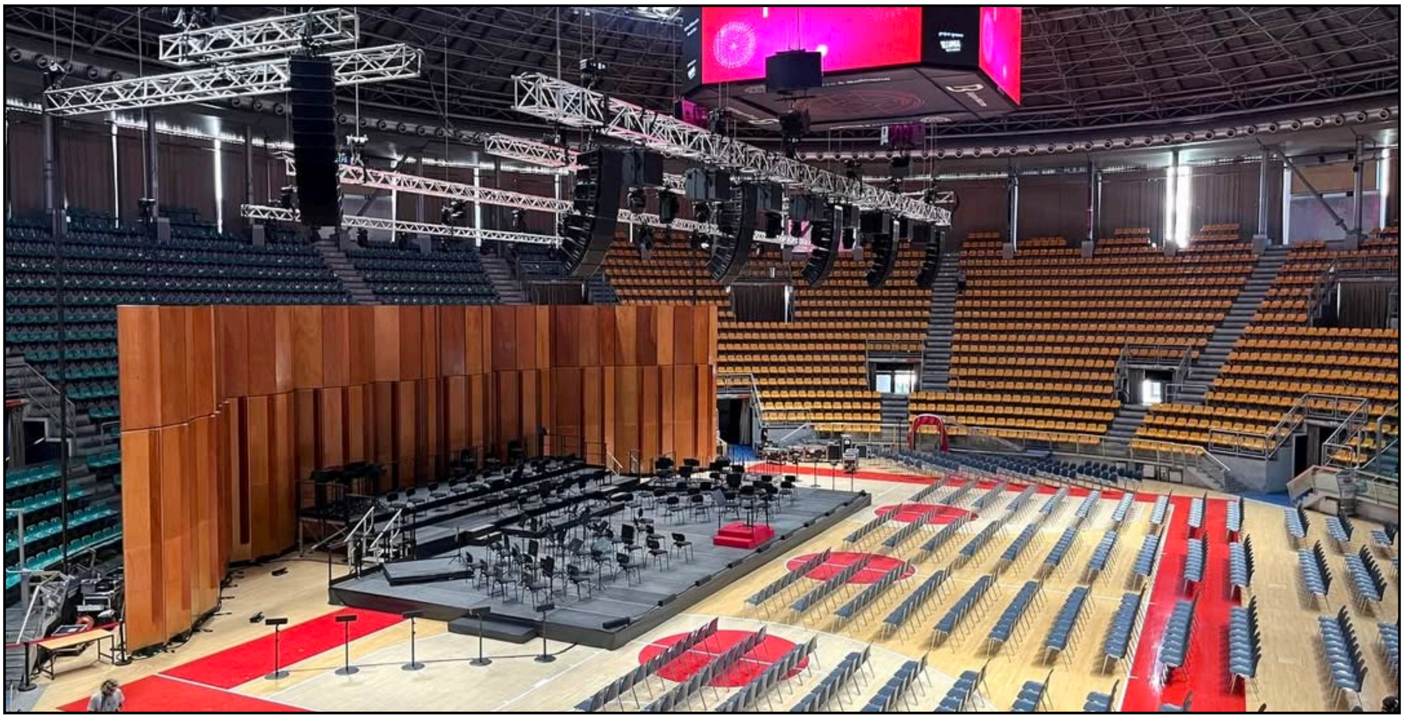
2.3. Happier performers

Most performers immediately recognize an improvement to their experience on stage, even when Soundscape is used for the audience speaker system. This is because the source-dependent algorithms don't just add clarity to the audience but also help the PA system react in a way that is more connected to what's happening onstage. This is particularly true with situations like classical music, theater, and spoken word where 'fighting' with the PA system is a hardship for performers.

Adding the acoustic characteristics of a concert hall with En-Space is a vast improvement for classical musicians, choirs, and singers who don't receive a natural acoustic response when stereo reverb is used.

2.4. Reduced sight line impact

When compared to a traditional stereo speaker deployment, Soundscape performs best with a higher quantity of smaller speaker positions. The result is reduced need for large arrays, which can be visually intrusive.



Small line arrays allow for reduced sightline impact to audience locations side-stage. Used here: 5x CL-Series arrays & mono outfill arrays.

Small line arrays allow for reduced sightline impact to audience locations side-stage. Used here: 5x T-Series arrays (each with 5x T10).

3. Soundscape is different

3.1. Applied intelligence: the self-aware PA

The Soundscape processor is aware of all speaker positions and all speaker orientations. This allows the processor to make informed choices to implement more complex routing, all while requiring less effort from the user.

The speaker information is automatically transported from the original design file in ArrayCalc (already used to design a d&b speaker deployment) to the control software, R1 (already used to program and monitor d&b amplifiers). Adding Soundscape to a speaker deployment simply adds a layer of automation and creativity based on already-available information within the project.

The user defines where the audio signals are physically located (positional audio) and where they want the audience to be virtually located (emulated room acoustics). The processor will render the musicians to the speakers and transport your listeners to another acoustic space in real time.

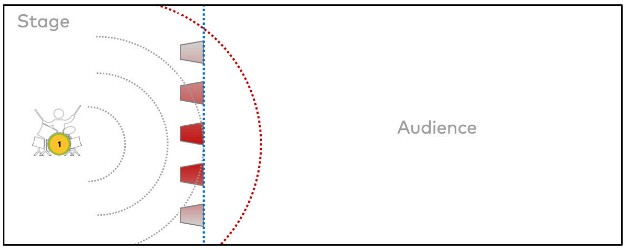
3.2. Delay-processing is critical

Soundscape is one of the only spatial systems available that sends a unique delay-time to every loudspeaker for every sound source. The result is flexibility for a wider range of applications and reduced time and effort for the operator. Unlike traditional speaker systems, delay times generally do not need to be measured or entered with a Soundscape system.

Or, turn it off and run it like a 'normal' amplitude-based spatial system. The choice is yours for every input independently. Each Sound Object can be run in your choice of 3 modes:

- **Full**: automatic delay and level processing in multiple axes at the same time (recommended generally).
- **Tight**: different delay times for each speaker, but reduced as much as possible for each speaker zone.
- **Off**: level-based panning... just like the competition.

Delay processing allows more loudspeakers to be engaged for all sound objects. Using delay and level calculations allows the speaker system to benefit from the "law of first arrival," a psychoacoustic phenomenon that allows the speaker system to maintain the listener's belief of sound localization, separate from what loudspeakers actually amplify the signal. The result is a spatial audio system with increased coverage consistency and more available SPL when compared to level-based technologies.



ArrayCalc is the only software on the market that allows for the prediction of the psychoacoustic perception of sound, which enables system designers to have higher confidence before deploying and using a Soundscape system. See the Localization simulation section for more information.

3.3. Real-world positioning

Soundscape is based on a Cartesian coordinate system (X axis and Y axis) which allows the user to place a sound object to represent a real-world performer position. This is not possible with spatial systems that utilize a polar coordinate system (pan/distance) which places objects around an egocentric view in a user-interface that is abstracted from the actual performance space.

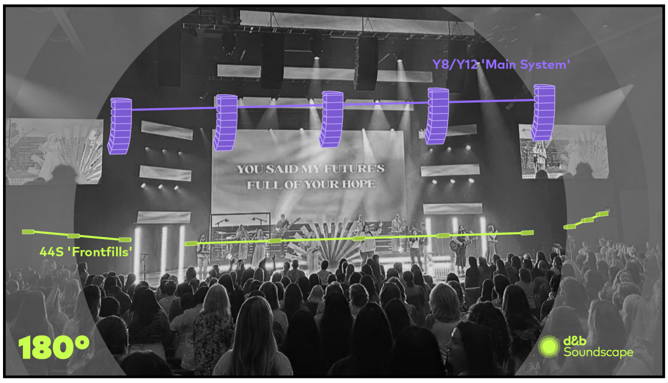
This approach also allows for image files to be placed on the Positioning View in R1 for better usability. This is not possible with polar coordinate systems.

4. Supported deployments

En-Scene supports a variety of venue configurations and system designs. However, depending on the type of event, an essential decision is whether a 180° system or a 360° system is considered. Should sound objects represent artists, instruments, or other sources on a stage, an Soundscape 180° system is sufficient. A 180° system is deployed across the stage front or proscenium. Only when sound objects should be played from other directions or located around the audience area should an Soundscape 360° surround system be considered. Alternatively, an Soundscape 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° deployment. 180°+ and 360° setups can benefit from the addition of En-Space, adding the ability to transport the listeners to differing acoustic environments.

4.1. Soundscape 180°

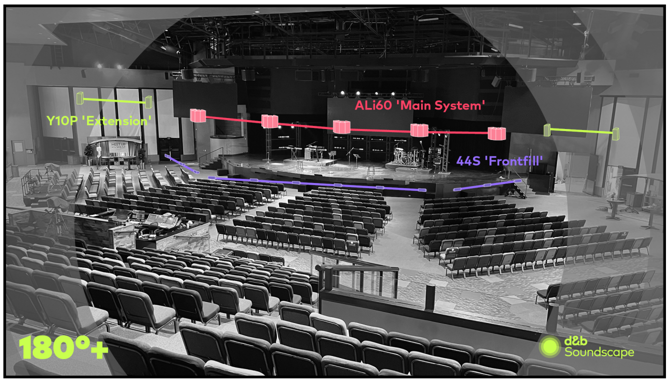
A speaker deployment which covers the width of the stage. This system is generally comprised of mains (point source or arrays), front fills, outfills, and subs. It allows for realistic positioning of performers on stage with a more realistic and compelling sonic result, when compared to stereo. This application is most commonly used with En-Scene positioning only, but some users will benefit from adding En-Space emulated room acoustics.



Shown here: small venue with a frontal system which allows for a more natural and transparent mix across the width of the stage.

4.2. Soundscape 180°+

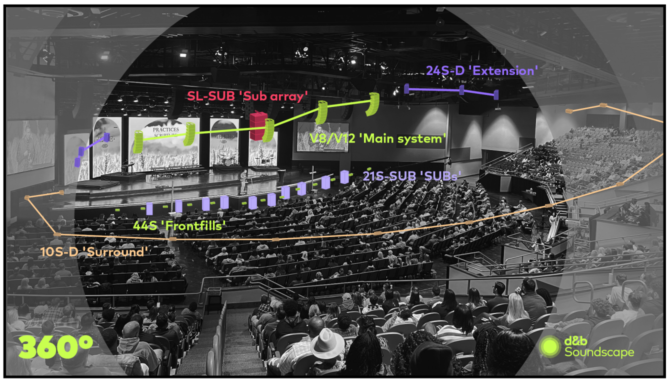
A low-investment way to 'expand' the mixing canvas beyond the width of the stage. A few point source speakers or small arrays per side can allow for a larger-than-life sonic experience which goes beyond a frontal system. En-Space can also be considered as it is very effective at utilizing the available speakers to make it feel like the reverberant behavior is immersive, although only coming from speakers in front of the audience.



Shown here: medium venue with A-Series main arrays and 2-per-side Y-Series point source to 'expand' the audio image. En-Space is also used to create a larger, more natural sound of the amplified mix.

4.3. Soundscape 360°

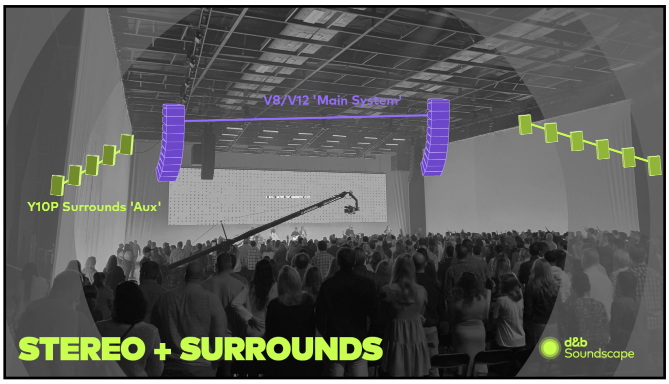
A speaker deployment which includes full surround speaker positions or even multiple surround systems. This is the most capable deployment for Soundscape as it enables full surround mixing of live sources, playback channels, and cinema content. Emulated room acoustics with En-Space allow for an even larger sound without feeling like 'surround sound.'



Shown here: large venue with Soundscape 360° for surround mixing and emulated room acoustics. This venue also utilizes microphones over the audience, which are selectively fed to En-Space to achieve a more democratic sound.

4.4. Stereo + surrounds

This technique has been used by artists who prefer to maintain their stereo mixing workflow but want to add the ability of surround mixing and/or emulated room acoustics. The processor can route a stereo mix to the left/right/sub/fill zones while utilizing all speaker zones for emulated acoustics and/or surround mixing. The hybrid approach allows for some content to be mixed in stereo while other parts of the show are simultaneously mixed using En-Scene and/or En-Space, as desired.



4.5. Traditional bus-based stereo

Having Soundscape does not force a user to use audio positioning or emulated room acoustics. Any input can be manually routed to any output, as is standard practice with legacy PA systems. In fact, parts of the mix can be routed in a traditional manner while other parts of the mix can be placed using En-Scene audio positioning. En-Space can be selectively added to all types of signals. Soundscape adds capabilities and does not take away any functions of an audio system.

4.6. Stereo deployment with En-Scene

Some users prefer to use the Soundscape processing on standard stereo PA deployments instead of sending a traditional stereo bus. This allows for some of the benefits of object-based mixing without the need to modify an existing PA deployment. En-Scene will automatically up- or down-mix all content to fit within the available speaker deployment.

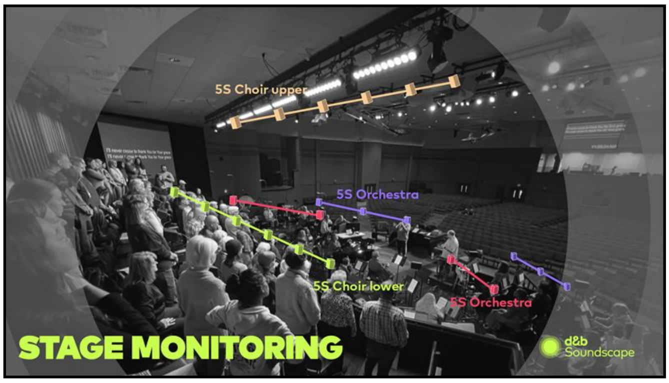
4.7. Multi-use facility

Because En-Scene Sound Object positioning includes delay-processing for every speaker, Soundscape is able to re-tune a PA system with a single sound object. Taking a mono mix and placing it virtually will instantly re-commission the speaker system for a number of applications:

- A multi-use space where the presentation area is frequently moved (lobby event space, black-box theater, or corporate presentation room).
- A sports arena or stadium which will host concert tours: a single sound object can carry the signal from the touring artist and be virtually placed to represent the position of the touring PA. This will automatically time-align the installed house speakers to play in-sync with the concert PA. This reduces hours of measurement and makes the two PA systems work better together.

4.8. Stage monitoring

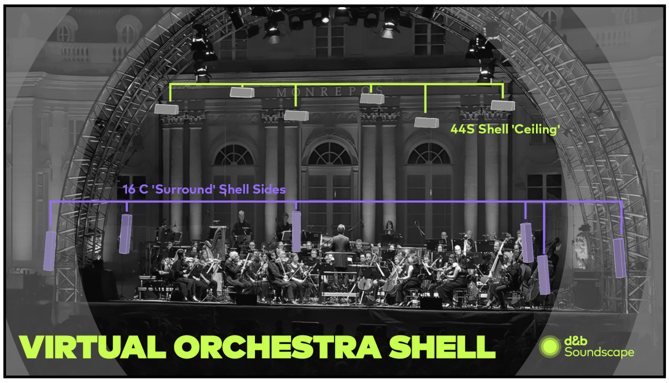
Imagine the typical "hot spot" monitor application for an orchestra or choir with lots of small speakers on top of mic stands (E4, 4S, or 5S as an example). En-Scene can provide automatic time-alignment of every musician to every speaker, which aids in delivering a natural sound that blends with acoustic sources much better for the musicians. The Sound Object Routing parameter allows for each speaker to still be mixed as if it were run on its own aux-send but with the benefits of automatic delay-processing. En-Space can also be considered to add a Virtual Acoustics Shell (VAS) element to the monitoring system, which aids the musician's ability to hear each other with a naturally-sounding acoustic environment.



Shown here: orchestra use 4S or 5S loudspeakers on mic stands while the choir has two rows of 4S/5S for monitoring. Object-based mixing allows for every monitor to benefit from automatic time-alignment to every musical source on stage. This results in a more transparent and natural sound for the musicians to hear each other with less perception of artificial amplification.

4.9. Virtual acoustics shell (VAS)

En-Space can be used (with or without En-Scene) to create a virtual acoustic space for the benefit of musicians on stage. In this application, it is common to suspend mics above the stage and feed them directly into the En-Space engine. Integration with a touch screen controller via R1 or Q-SYS, for example, allows this system to be activated and room presets changed by non-technical staff.



Shown here: some suspended mics are used to capture the sound of the acoustic performance. The mics are fed into the DS100 and En-Space is used to change the acoustics of the stage to help the musicians hear each other as if they are playing in a concert hall designed for acoustic music.

More information of VAS applications can be found within the document *TI502: Acoustic shell*, found here.

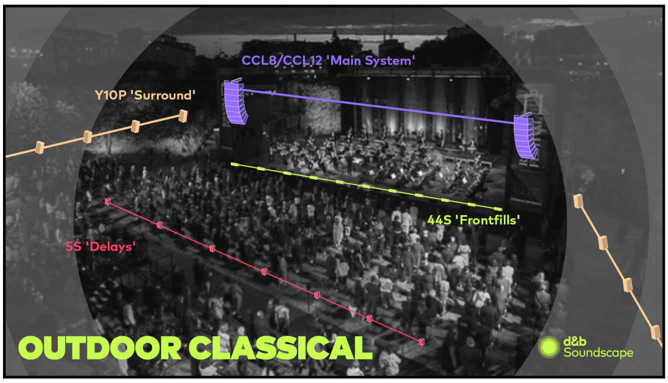
4.10. Voice lift

En-Space can be adjusted to minimally augment the acoustic response of a room while allowing more loudspeakers to contribute to the overall sound. The result is more distributed energy without the feeling of amplification. This approach can be done with close-microphones—lectern, lavalier, handheld wireless, etc.—or with area mics suspended above presentation area. The latter example is essentially the same deployment as a VAS mentioned above, except for the benefit of the audience, rather than the performers.

More information of VAS applications can be found within the document *TI502: Acoustic shell*, found here.

4.11. Outdoor events

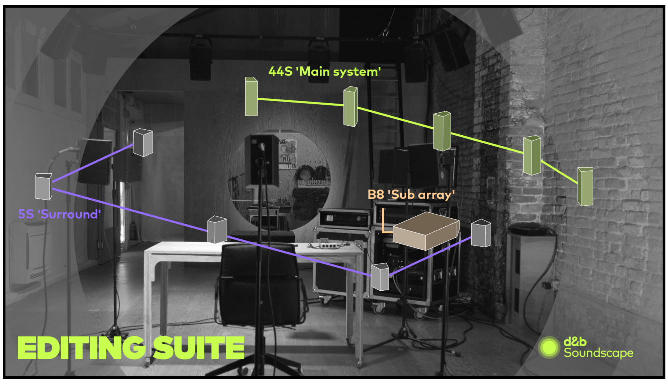
The ability to "take the concert hall with you" is a huge advantage for open-air concerts, which can otherwise sound sterile and unnatural. While the benefits for classical music are obvious, En-Space adds a feeling of immersion and intimacy for all other music genres. With the Custom Rooms service, an ensemble can bring their own home concert hall acoustics to outdoor events.



Shown here: Ravenna Festival, Italy

4.12. Mix studio or edit suite

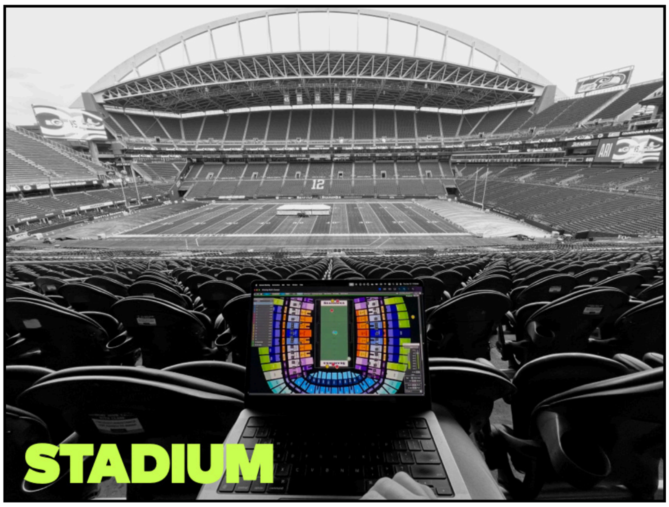
Soundscape can easily scale spatial audio from a small room (such as a mix studio) to a large room (such as a venue). This allows content to be produced outside of the event space and automatically re-scaled onto a large PA with different speaker positions, quantities, and models. En-Space can also be used within the studio to replicate the acoustic behavior of the destination venue, which greatly aids content producers in creating audio with the final venue acoustics taken into account. En-Space Custom Rooms can be considered for a total match between the acoustic environments.

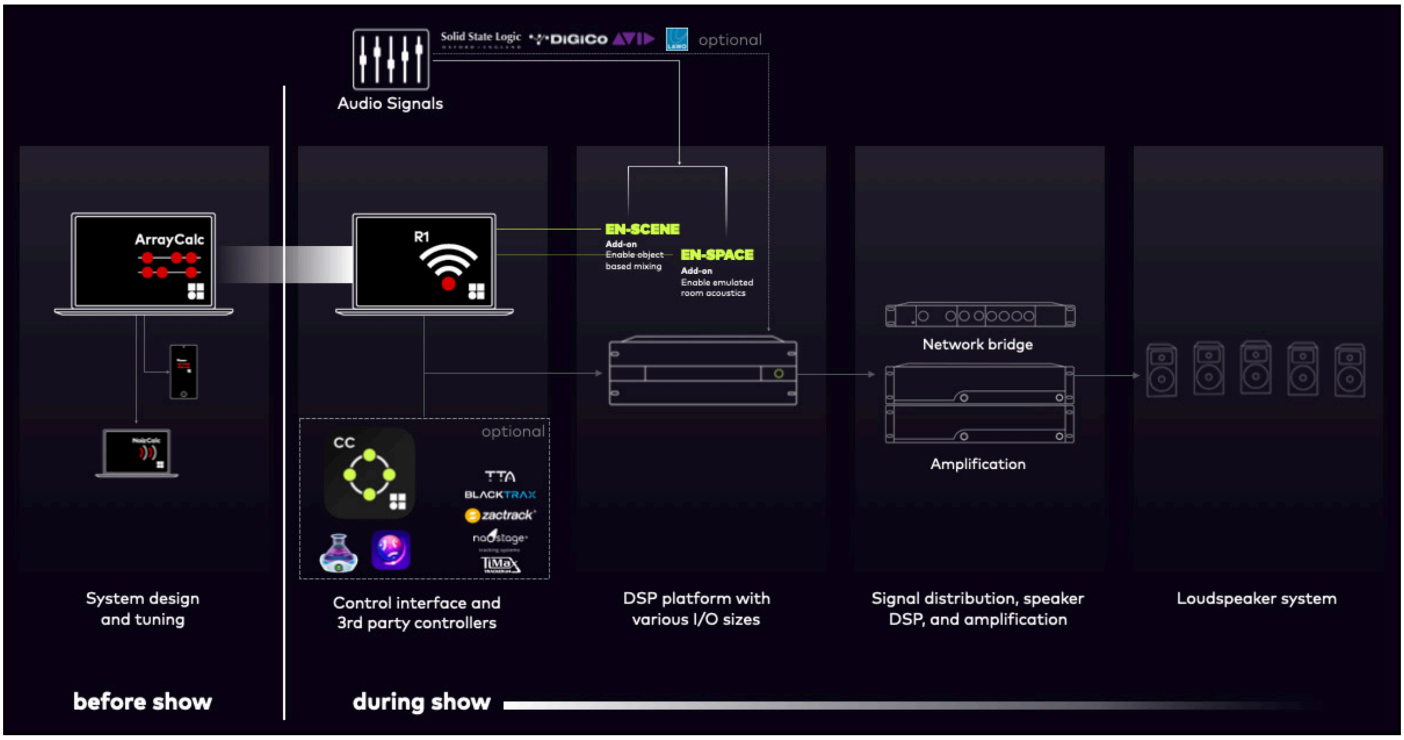


Shown here: a small mix room which utilizes Soundscape as part of the content creation. The content can then be automatically scaled-up to a large Soundscape venue with very little effort.

4.13. Non-music applications

Immersive events, corporate lectures, spectacle shows, museum installations, theme parks, art galleries, interactive displays, sporting venues, sports book in a casino, brand activations, you name it. The demand for more compelling audio experiences is everywhere. The ability to drive attention to high-value video content is a growing business need for venue operators. Increasing audience engagement requires the emotional trigger of positional audio.





5. Components of Soundscape

5.1. What is Soundscape?

Soundscape isn't a thing… it's a new way of imagining live sound systems, how they behave and how they are operated.

En-Scene and/or En-Space can be implemented in a Soundscape processor to use "Soundscape processing," which expands the capabilities of the speaker system beyond what is possible with traditional speaker systems.

However, a Soundscape processor can be used as traditional mix matrix without the Soundscape processing. This means the speaker system is considered "Soundscape-ready," as it only needs the processing software to enable Soundscape.

5.2. Hardware

5.2.1. DS100 Signal Engine

The DS100 Signal Engine is a Dante® channel-based digital audio matrix processor for low-latency live-event applications. Inputs and outputs provide extensive signal processing capabilities, including matrix crosspoints that control level and delay. This is useful for handling the combining of multiple consoles into a single festival PA or for routing signals across a venue installation, all from within R1 software, which is typically already in use to program and monitor the d&b amplifiers and speakers.

The DS100 is available in three different I/O sizes. The available inputs and outputs can be changed at any time by purchasing a different I/O license.

Model	Inputs	Outputs	Sample Rate
DS100 Small	64 Dante	24 Dante	All channels supported at 48kHz or 96kHz
DS100 Large	64 Dante	64 Dante	
DS100 XL @ 48kHz	128 Dante	64 Dante	Sample rate determines input count
DS100 XL @ 96kHz	64 Dante		

Note: the DS100 sample rate must be changed in Dante Controller. Then, the DS100 can be rebooted to adopt the new settings. This process will automatically dictate if the DS100 XL supports 128x64 or 64x64.

Model	Physical audio ports	Physical control ports
DS100 Small	Dante Primary & Dante Secondary	1x Ethernet
DS100 Large		
DS100 XL		

Dante outputs of any DS100 variant can be sent to a combination of Dante-equipped d&b amplifiers such as the 5D (info here) and/or DS10 Audio Network Bridge (info here) to feed amplifiers without native Dante support.

5.2.2. DS100M Signal Engine

The DS100M Signal Engine is a Milan™ certified channel-based digital audio matrix processor for low-latency live-event applications. Inputs and outputs provide extensive signal processing capabilities, including matrix crosspoints that control level and delay. This is useful for handling the combining of multiple consoles into a single festival PA or for routing signals across a venue installation, all from within R1 software, which typically is already in use to program and monitor the d&b amplifiers and speakers.

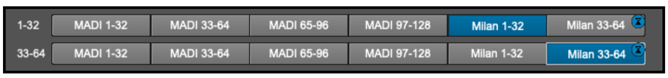
The DS100M is available in three different I/O sizes.

Model	Input DSP	Outputs	Notes
DS100M Small	64 Milan / MADI	24 Milan	input formats selectable in 32 channel blocks for max total of 128 DSP channels
DS100M Large	64 Milan / MADI	64 Milan	
DS100M XL @ 48kHz	128 Milan / MADI	64 Milan	
DS100M XL @ 96kHz	128 Milan / MADI		

Model	Available audio inputs	Physical audio ports	Physical control ports
DS100M Small	64 Milan + 64 MADI	Milan Primary + Milan Secondary + 3x BNC MADI	2x Ethernet + 2x BNC Word Clock (in+out)
DS100M Large	64 Milan + 64 MADI		
DS100M XL @ 48kHz	64 Milan + 128 MADI		
DS100M XL @ 96kHz	64 Milan + 96 MADI		

Note: MADI supports 32 channels per 3xBNC @ 96kHz and 64 channels per 2xBNC @ 48kHz.

DS100M DSP channels can be assigned in 32 channel blocks from the available 64 Milan inputs and/or the MADI inputs (96 or 128 pending sample rate). Shown below are the choices for a DS100M Small or DS100M Large which both support 64 input channels.



The Milan outputs of any DS100M variant can be sent to a combination of Milan-equipped d&b amplifiers such as the D40/40D/25D/D25/D90 (info here) and/or DS20 Audio Network Bridge (info here) to feed amplifiers without native Milan support.

5.3. Soundscape processing

5.3.1. En-Scene, audio positioning

With the En-Scene processing add-on, a d&b Soundscape system can place up to 128 sound objects (depending on DS100 variant) at separate locations on stage or in other areas of the venue. The objects can also be moved to any desired position in realtime during the show. Unlike stereophonic sound reproduction, En-Scene provides an authentic image of all sound positions for the entire audience

More details about the algorithm
En-Scene supports three forms of spatial audio algorithms, called Delay Modes, which are user-selectable per object:

- **Full Mode:** a proprietary form of distance-based amplitude panning (DBAP) which includes delay-processing (delay calculated per object and per speaker position) and different levels per speaker. This mode is recommended for most use cases, particularly with live instruments.
- **Off Mode:** vector-based amplitude panning (VBAP). This is more common in cinema applications and many of Soundscape's primary competitors in the live sound market. This mode is not generally recommended, except for use with fast-moving sound objects.
- **Tight Mode:** a proprietary hybrid approach of DBAP and VBAP. Inter-speaker delay is still applied, but the overall delay is minimized. This mode can be useful for handling multi-channel playback content like stereo, LCR, 5.1, etc.

The audience area(s) can be covered with different loudspeaker groups, or so-called Function groups, in the Soundscape approach. A Function group represents a group of loudspeaker positions, preferably of identical dispersion and equally spaced, each driven from its dedicated DS100 output. A Function group works as a spatial renderer of all sound objects by reproducing them with different levels and delays using all loudspeakers of the group.

The En-Scene algorithm considers the mix of the listeners' psychoacoustic perception as well as acoustic effects between the sources to calculate transfer functions for the relevant matrix crosspoints. Maintaining the rules of the precedence effect (aka: law of the first wavefront. More info available here) ensures accurate spatial localization. The loudspeakers with the earliest arrival times provide the relevant direction information while adjacent speakers provide additional coverage.

Unlike traditional deployments, this method of sound reproduction requires multiple loudspeakers to cover each listener. As a consequence, loudspeaker choices should prefer wider-dispersion than with traditional speaker deployments.

The algorithm considers the position and orientation of the loudspeakers in relation to the sound object. The individual levels and delays of each source depend on the horizontal angle offset between the sound object and the speaker axis, its distance, and the sound object properties. See Sound Objects for more information.

When objects are placed inside the coverage area of all sources (ie: in front of all speakers) a level-only panning algorithm will be applied automatically. When objects are moved, the perceived level and tonal balance will not be affected as this is considered by the processor's calculations.

An En-Scene system comprises a DS100 or DS100M Signal Engine, the En-Scene software running within the engine, and multiple loudspeakers and amplifiers to cover the audience areas.

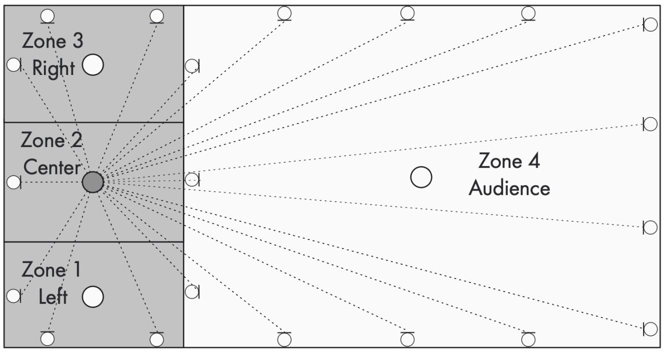
5.3.2. En-Space, virtual acoustics

With the En-Space software module added to a signal engine, a d&b Soundscape system can add the acoustic behavior of different spaces to your local environment, outdoors or within a venue. This can also be used as a theatrical effect which transports the listener to other environments.

En-Space is a spatial 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 with automatic 3D mapping onto the available loudspeakers.

Please note that En-Space can not only be used to create an acoustic environment for the audience, it can also be used for the performers on stage, or both. This is generally referred to as a Virtual Acoustic Shell (VAS).

More details about the algorithm
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 rather from hundreds of boundary plane responses distributed along the circumference of the venue at multiple heights. 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 utilizing up to 144 inline convolution filters (which act like 144 unique reverberant patterns. More info here). This reproduces the sound field of the sampled space to such a high-degree as to be nearly indistinguishable from the original space.



For each of the 64 En-Space loudspeaker positions, the venue library provides individual boundary responses for objects across zones of the stage and objects in the audience area. In other words, the real-world acoustic behavior of a concert hall depends on the location of the performers within it: and so does En-Space. The reverberant behavior applied to each speaker differs based on where the performer's location is within the venue. These performer locations for En-Space are referred to as En-Space zones, shown above as Zones #1–4.

The measurement positions within the sampled venue do not have to match the actual venue in shape, size, or speaker positions. The DS100 automatically maps the En-Space convolver outputs to the processor 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 for emulated room acoustics. The table below indicates the maximum number of positions per group which will be given individual uncorrelated boundary responses. More speaker positions are possible, thus gradually increasing correlation.

Function Group Mode	# of unique convolution filters
Main	7 (x4 for early reflections)
Frontfill	9 (x2 for early reflections)
Surround	40 (x2 for early reflections)
SUBs Group	7 (x2 for early reflections)
SUB array	1 (x2 for early reflections)
Outfill + embd	1 (x4 for early reflections)
Delay Line + embd	7 (x2 for early reflections)
Ceiling	7 (x2 for early reflections)

5.3.3. En-Space Custom Rooms

En-Space Custom Rooms is a service provided exclusively by d&b audiotechnik, where d&b acoustics specialists will travel to the venue and perform our proprietary measurement service to capture a room's acoustics for electronic reproduction with En-Space. This usually takes between 1-2 days, depending on the accessibility of the space to be captured.

Upon completion, a single file is provided to the customer for each captured room, which can then be loaded onto one a DS100 for reproduction as needed. Up to three (3) Custom Room files can be loaded into a single processor and a single Custom Room file can be loaded onto multiple processors.

The captured space will be added to the En-Space venue library. However, it can be held for exclusive use by the customer (so it's not available for other users) for an additional fee. Contact your d&b representative or email sales@dbaudio.com for more information.

Custom Rooms application A:
A symphony will be performing a summer series of outdoor concerts. En-Space Custom Rooms can be used to allow the symphony to reproduce their home concert venue electronically during the open-air concert.

Custom Rooms application B:
An ensemble will be performing in the large concert hall but rehearsing in a smaller room due to venue availability and scheduling conflicts. d&b can measure the large room for electronic reproduction within the rehearsal space. More information available in the section Virtual Acoustic Shell.

Custom Rooms application C:
A theater production or amusement park wants to feature a very specific acoustic, for example a cave, in their production to help create a sense of immersion. d&b can measure and record any space for which approval and access has been provided.

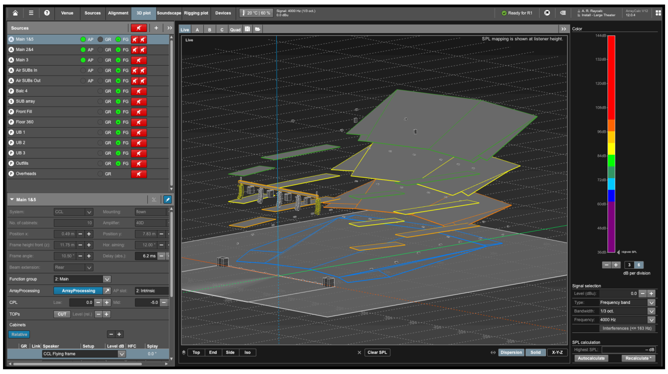
See Loading En-Space Custom Rooms for instructions on how to utilize your En-Space Custom Room.

5.4. Software

5.4.1. ArrayCalc

ArrayCalc is d&b's free prediction software available for macOS and Windows, which can be downloaded here.

An ArrayCalc file is required to program a DS100 for use with En-Scene and/or En-Space. This is because the magic behind Soundscape is its knowledge of the position and orientation of all speakers, which is entered into ArrayCalc during system design. This positional information is derived from the ArrayCalc file after it has been opened in R1 while connected to a network with a DS100 or DS100M.



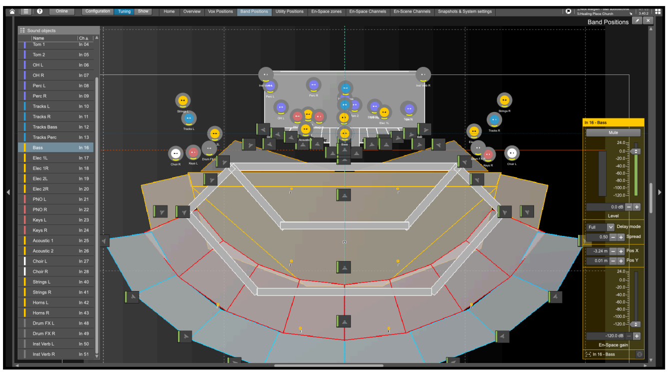
5.4.2. R1

R1 is d&b's free control software available for macOS and Windows, which can be downloaded here.

R1 allows for the control of all d&b amplifiers and all DS100 variants. R1 is also required for the initial programming of a DS100. An ArrayCalc file can be opened within R1 which will automatically:

- Create controls based on the system's design in ArrayCalc.
- Push tuning settings to all amplifier DSP.
- Inform a DS100 of all speaker positions and orientations as well as Function Group behaviors.

The R1 file can then be customized to include custom control views, recallable snapshots, and lock-out features. Multiple instances of R1 can run on a network at the same time and at the same time as Create.Control.

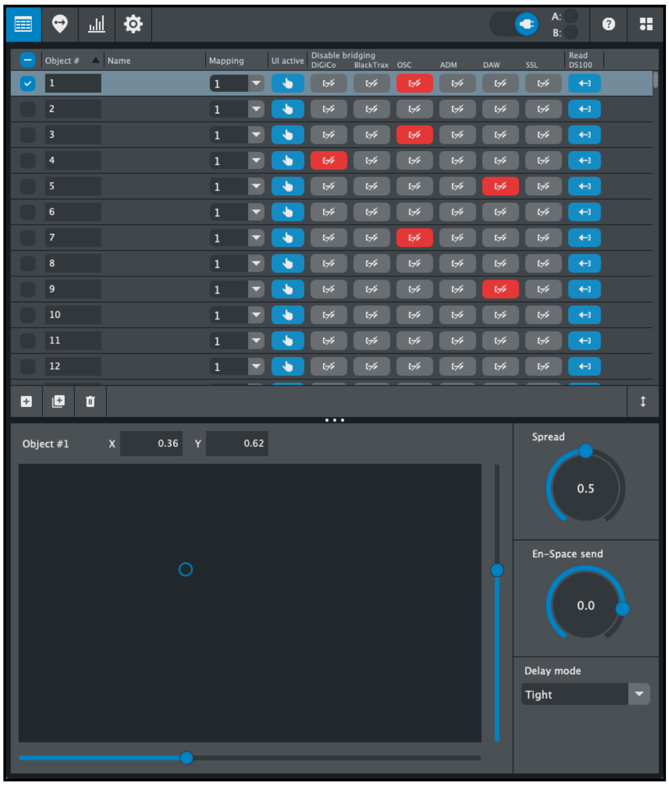


5.4.3. En-Bridge

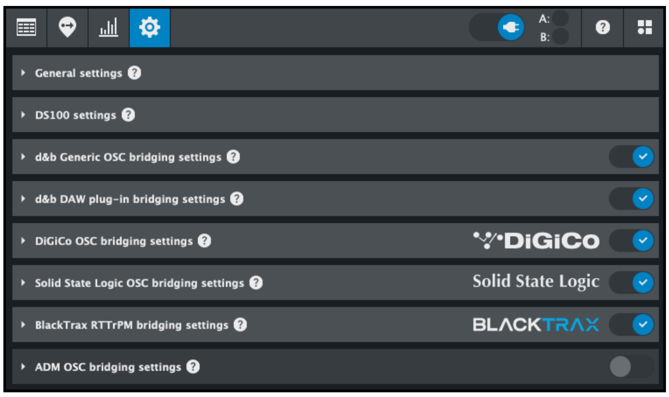
En-Bridge is d&b's free network translation software available for macOS and Windows, which can be downloaded here.

En-Bridge is a software application that allows connecting multiple hardware and software devices together for control of Soundscape. It is helpful within environments where advanced third-party control is needed. En-Bridge can accomplish a number of tasks:

- Convert some third-party network protocols to the DS100 OSC language (such as BlackTrax, DiGiCo, or SSL consoles).
- Work as a bridge between networks (example: DiGiCo control network connected to En-Bridge which also connects to the DS100 network).
- Dictate which third-party controller has the ability to change parameters of individual objects (example: DiGiCo can only control objects 1-48 and BlackTrax can only control objects 49-64)
- Forward OSC messages from a single third-party controller to multiple DS100s for redundancy or output expansion applications.



Shown here: En-Bridge being used to control which third-party devices are able to control various sound objects while monitoring all data.



5.4.4. Create.Control

Create.Control is d&b's free Soundscape control software available for macOS and Windows.

Create.Control is focused exclusively on the mixing of Soundscape, separating the sound object control from the overall system control. Users of Create.Control can fully program the spatialization of their event without any knowledge or interaction with ArrayCalc or R1. This allows Soundscape to be more widely available to users who are not involved in the design and setup of d&b speaker systems; such as artists, studio engineers, and touring mix engineers.

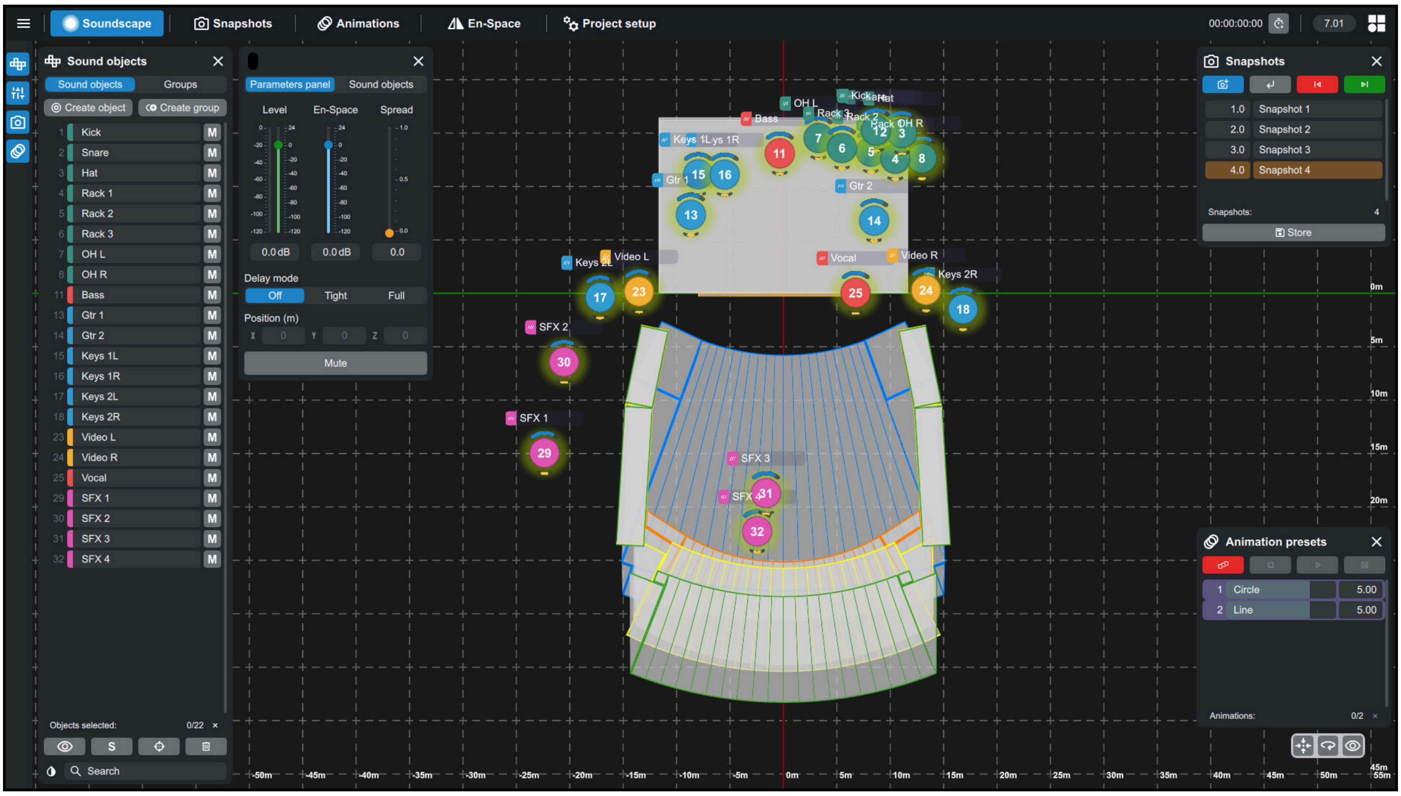
Create.Control allows for the pre-programming and live event mixing of sound object parameters such as positioning and En-Space settings, fully recallable from a powerful snapshot engine similar to mixing consoles. The snapshots can be automated using timecode or recalled manually throughout an event.

R1 is still available as a method to control Soundscape, either as an alternative to Create.Control, or simultaneously. For example, a systems engineer might be using R1 to control the amplifiers and monitor sound object positioning of tracked performers, while simultaneously a mix engineer is using Create.Control to trigger animations, snapshots, or En-Space settings.

Additionally, multiple instances of Create.Control can run on the same network, even at the same time as multiple instances of R1.

While the use and knowledge of ArrayCalc is not required for the use of Create.Control, venue geometry and input lists can be imported from a .dbpr file created in ArrayCalc for ease of use and to speed up pre-programming.

Create.Control will be available as a free download from dbaudio.com when it is released during the second half of 2025. A dedicated manual for Create.Control will be available upon release.



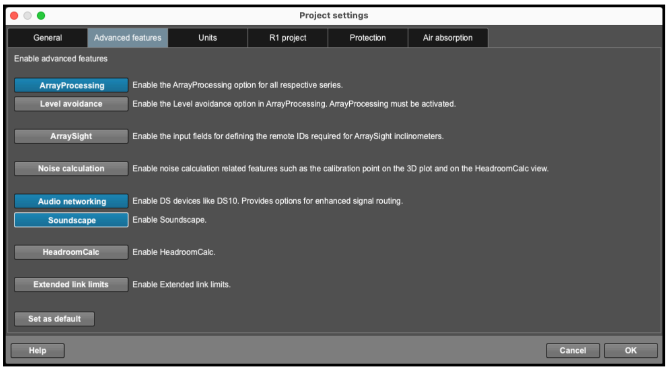
6. ArrayCalc: designing Soundscape

6.1. Introduction

A Soundscape system which utilizes En-Scene and/or En-Space must be fully configured in ArrayCalc. This is because ArrayCalc is much more than prediction software. The project file includes speaker positions and orientations, which is crucial to the self-aware nature of the Soundscape processor. ArrayCalc can be downloaded for free from the downloads section of dbaudio.com located here.

The following section assumes you already have a working knowledge of ArrayCalc as it pertains only to the additional steps required when designing a Soundscape system. For ArrayCalc training, please visit the d&b website to view tutorial videos here.

First, make sure that within your ArrayCalc project you have enabled Soundscape for your project by clicking **Settings > Advanced features** tab **>** click **Soundscape** and **Audio networking** to enable them (turn blue)**.**



6.2. The Venue tab

6.2.1. En-Scene requirements

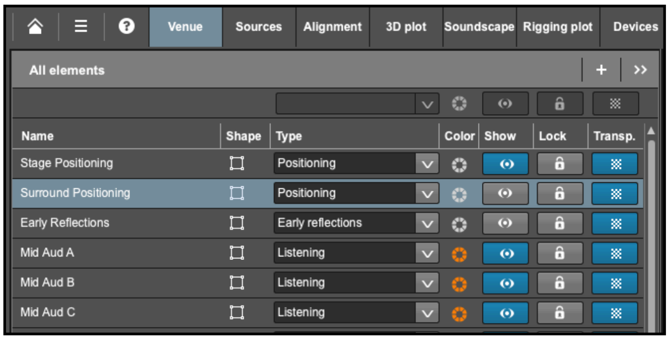
For En-Scene applications, in addition to the audience areas, some additional venue elements must be created: the Positioning planes. Positioning planes–the area in which sound objects will be positioned–need to be a quadrangular shape. At least one Positioning plane must be included in your project however up to 4 can be added for advanced third-party control requirements.

The reason for adding Positioning planes in ArrayCalc, is that when operating Soundscape within R1, they are displayed as a reference to enable locating and moving of sound objects. The position of a sound object can be anywhere in the X/Y plane and is not limited to the inside of the Positioning plane.

Positioning planes also serve to adapt coordinate systems of external position control devices which are connected to the OSC interface of the DS100. This includes tracking systems, VST plug-ins, mixing console integration, etc. The quad-panner of a third-party device will be scaled onto the Positioning Plane. For further details, please refer to the Position control and coordinate mapping section of this document.

It is recommended that you begin by adding two Positioning planes to your project, which will be sufficient for most use-cases:

1. "Surround Positioning": a Positioning Plane (quadrangle) which encompasses your entire venue and is approximately 200% the size of your venue.
2. "Stage Positioning": a Positioning Plane which encompasses the primary stage/performance area only.



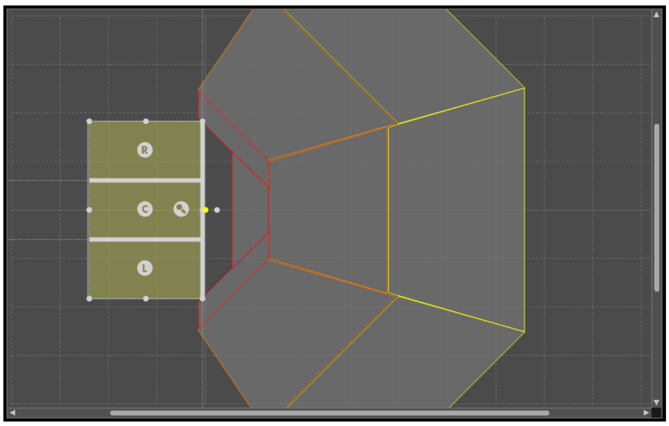
Note: The Positioning Plane option is only available when Soundscape is enabled within the project's **Settings > Advanced settings** tab.

6.2.2. En-Space requirements

For En-Space applications, an Early Reflections Plane must be defined. In most applications, the Early Reflections Plane will match the stage area.

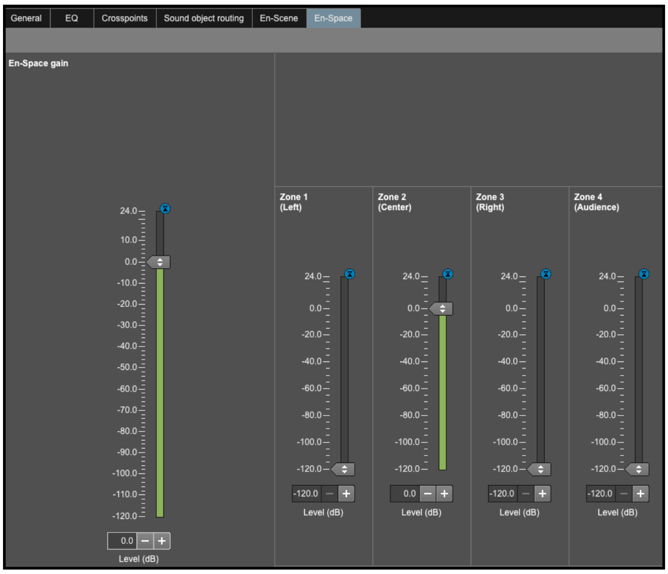
The Early Reflections Plane is used to scale the reverberant patterns of En-Space onto the available loudspeakers as well as automating how signals are sent to the appropriate En-Space engine inputs. See the DS100 block diagram for an illustration on how signals are routed into the En-Space processing.

What are early reflections?
Early reflections are the first bounces of sound waves after they leave a source and hit nearby surfaces–walls, floors, and ceilings–before they reach the listener. They arrive a few milliseconds after the direct sound and create a sense of space and dimension. Early reflections are a crucial part of how we perceive the size and character of a room.



Shown here: The Early Reflections Plane is set to the size and position of the stage area. Three zones are visualized across the plane (R, C, L) which correspond to house left, center stage, and house right.

The Early Reflections Plane splits the venue into four zones which correspond with four En-Space engine inputs. Each En-Space input represents unique early reflection patterns which mimic the real-world acoustic behavior of the modeled space. The first three zones are visualized on the plane. The fourth zone is not visualized but is the audience area. The Early Reflections Plane allows each performer to have a differing reverberant behavior which more naturally replicates their position within the venue, just like real-world acoustics.



The En-Space settings for a Sound Object are shown from R1: the master En-Space level fader is set to 0dB by the user. In this case, the object is located near the center of the stage so the "Zone 2 (Center)" fader was automatically turned up by En-Scene because the objects position is compared to the Early Reflections plane.

The Early Reflections Plane should not be thought of as Left or Right signals, as is common with traditional bus-based mixing. Any signal routed to En-Space Zone 1 (Left) will result in a reverberant pattern sent out of all loudspeakers, not just speakers on the left side. The difference between the zones is not where the reverb will be output. Rather, the difference between the zones is the type of reverb created, mimicking the nuanced differences of how an acoustic source reverberates out from the left side of the stage vs. the right, and so on.

Four En-Space inputs as shown within R1: The three inputs across the top reflect the 3 Zones (L, C, R) of the Early Reflections Plane, the fourth represents the audience area.

If En-Scene is used, the input routing to the En-Space zone 1-4 is automated based on the sound object's position. Sound objects located on the Early Reflections Plane will obtain dedicated early reflection patterns of the emulated room, depending on their spatial positions within zones 1, 2, or 3. Objects placed to the sides or behind (upstage) of the Early Reflections Plane will be sent to zones 1–3, whichever is closest, as if they are placed on top of the plane. Sound objects in front (downstage) of the Early Reflections Plane are considered to be in the audience and will be automatically assigned to En-Space Zone 4, which incurs fewer early reflections and a more even spatial level distribution, just like in a real-world acoustic environment. If En-Scene is not used, DS100 matrix inputs must be manually routed to the appropriate En-Space zone inputs.

If no Early Reflections Plane is added to the ArrayCalc file, all En-Scene objects will be sent to En-Space zone 4 as there is no reference for the source locations of zones 1–3.

Note: The Early Reflections Plane must be a quadrangle and rotation is not permitted in ArrayCalc. The Early Reflections option is only available if Soundscape is enabled within the project's **Settings > Advanced features** tab.

For more information on using En-Space for your event, refer to the En-Space - Operation within R1 section of this document.

6.3. Sources tab

6.3.1. Design concepts

6.3.1.1. Pick the right speakers

Generally speaking, wide-dispersion loudspeakers are preferred for use with a Soundscape system. This is because Soundscape works best when listeners can hear multiple speakers within a 6dB offset.

Either single point source speakers, or multiple point source speakers arrayed within a single position and linked to a single amp channel to create a single source with extra-wide coverage, can be used. For the latter, simply link the speakers on a single amp channel in ArrayCalc as normal.

Depending on the level requirements, point source loudspeakers may be sufficient, especially when you consider the available SPL of 5 or more cabinets working together. However, line arrays can also be used to provide a more controlled level distribution towards the far field, especially if ArrayProcessing is enabled. 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 mono arrays. The use of two L/R paired arrays and one single array is also possible, but this is not recommended as assignment to matrix outputs is not as straightforward as En-Scene positioning, and requires more effort.

The use of ArrayProcessing can be helpful, not only to achieve the required throw for the far field, but also to create a smooth transition from the frontfills.

6.3.1.2. Speaker placement

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

Hypothetical example for mains:
- The mains begin coverage around the fourth row. This is because front fills cover rows 1-4 very well.
- The distance from the center main to the fourth row will be approximately 8.5 meters.
- 8.5m x 70% = 5.95m (rounded to 6m).
- Therefore, 6 meters is the maximum distance between mains loudspeaker positions to achieve 100% accurate localization in the closest seats (4th row).

Hypothetical example for frontfills:
- The frontfills are placed on the stage lip in a theater and will cover beginning at the front row.
- The distance from a frontfill to the nearest listeners ears is approximately 1 meter.
- 1m x 70% = 0.7m.
- Therefore, 0.7m is the maximum distance between frontfill speaker positions to achieve 100% accurate localization.

Placing speakers further apart than the 70% recommendation will result in less-than-perfect localization for the nearest seats. When compared to a traditional system, the audience's experience is still greatly improved, even with a sub-optimal deployment for Soundscape. Furthermore, even with compromised localization resolution, the time-alignment of all speakers to all acoustic sources on stage will result in a considerable improvement, as compared to a traditional PA.

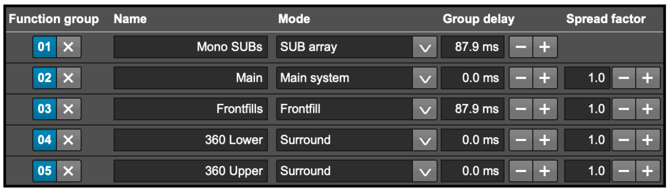
Any resulting compromises associated with your speaker placement decisions can be visualized on the Soundscape tab in ArrayCalc. This includes localization errors incurred by suboptimal speaker spacing.

Tip: Speaker placement and orientation should also consider anticipated sound object positions and how they will interact with speaker positions. It is recommended that you visit the Mixing Considerations section pertaining to Objects vs speaker positions.

Note: 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 or vertical directivity. Level compensation is best made at the individual amplifier channels using the SPL plot on the Soundscape tab in ArrayCalc.

6.3.2. Function groups

In the *Sources* view of ArrayCalc, enter and place all loudspeakers and assign them to a *Function group*.



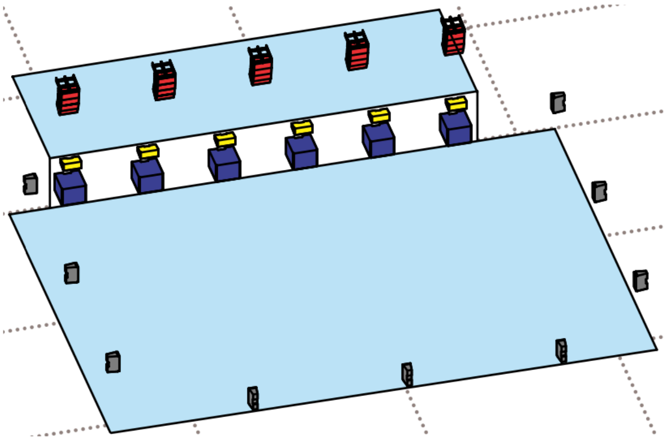
ArrayCalc provides up to 32 Function groups for use with Soundscape processing. These Function groups can be thought of as 32 individually rendered spatial audio zones, mono fills, or subwoofer deployments. Each speaker zone that is utilizing En-Scene or En-Space needs to be placed into its own Function group. Some speaker zones (such as monitors) may not need to be assigned a Function group, as they will not utilize the Soundscape algorithms.

Function groups are configured in the **Devices** tab in ArrayCalc. There are 11 available modes for each Function Group which dictate the desired behavior of En-Scene and En-Space.

Depending on the size and layout of the venue and the type of program to be played, and the desired behavior of the system, different set of groups are required. Below is a list of all the available modes for each Function group.

Mode	Remarks
Main system	Object positioning.
Frontfill	Object positioning.
Surround	Surround object positioning.
Outfill	All sound objects are summed with unique delay times per object. For use with audience areas not covered by surround speakers.
Delay line	Delay line with positioning. For use with audience areas not covered by surround speakers.
SUB array	Mono LF with unique sound object delay times.
SUBs group	LF with sound object positioning.
Ceiling	En-Space and manual routing only.
Mono out	Mono feed without any processing.
Outfill embd.	Outfill speakers embedded within a surround speaker deployment.
Delay line embd.	Delay line speakers embedded within surround speakers.

For a detailed comparison see the Function Groups overview table.



Shown here, an example setup with four Function groups: Main system (red), Frontfills (yellow), Surround speakers (grey), and SUB array (blue).

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. Example: if two surround speaker areas are required (e.g., main floor and balcony), there should be two separate Function groups, both set to the Surround Mode.

6.3.2.1. Main system

Panning & Spread?	Delay processing for each object?	Number of En-Space filters
Yes	Yes	7 x4 early reflection zones

For a detailed comparison see the Function Groups overview table.

The Main system Mode covers the central audience area. It is a horizontal array of speaker sources located across the stage, preferably equally spaced and matching speaker models or arrays. By placing these speakers within a Function group, the DS100 will be informed to treat all included speaker positions as a single spatial-array which supports localization and time-alignment to sound objects.

The Function group assignment also activates the emulated acoustics of En-Space for those speakers. The number of En-Space filters noted above represents how many unique reverberation patterns are achievable for this Function group Mode. In the case of Main system Mode, there are 7 unique reverberation patterns, for each of the 4 En-Space Early reflection zones. In other words, up to 28 unique reverberation patterns are available which are automatically mapped onto the mains loudspeakers. To say it another way, the mains Function group will have 7 unique reverberations for sound objects on the left side of the stage, 7 for sound objects located in the center, 7 for sound objects located on the right side of the stage and 7 unique filters applied to sound objects in the audience area.

Additional Main system Function groups
Another requirement may be the addition of a more powerful L/R line array system for the far field where detailed imaging is less important than increased intelligibility. It is not recommended to add these sources to the same Main system Function group because sound objects would be heard at different levels according to their position relative to each speaker position. It is best to create a second Function group in Main system Mode in order not to disturb the imaging created by the first Main system Function group. The far field arrays should only cover the remote part of the audience area.

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

Additional Main system Function groups can be used for under-balcony delays. See Delay line and Delay Line Embedded or for additional options.

6.3.2.2. Frontfill

Panning & Spread?	Delay processing for each object?	Number of En-Space filters
Yes	Yes	9 x2 early reflection zones

For a detailed comparison see the Function Groups overview table.

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 *Frontfill* will inform the DS100 to treat them as a single array which supports localization and time-alignment to the performer's on stage separately from the mains. Unlike the mains mode, the algorithm for front fills will more effectively accommodate closer listeners with a wider level distribution from each sound object. This benefit can be augmented or diminished using the Spread factor parameter.

- In ArrayCalc, this part of the system can easily be entered by clicking **Add point sources** in the **Sources** view.
- 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 from left-to-right as seen from FoH. The automatically created layout for an array of frontfills by ArrayCalc will be in reverse order (right-to-left). Entering a negative value in the *Equally spaced along* parameter will create the recommended order.
- Assign the frontfills to a Function group with the Frontfill Mode.

In order to provide 100% 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.

Additional Frontfill Function groups
Sometimes multiple speaker zones are used to cover the front rows. Additional Function groups with the same mode can also be used for down-fills or other speaker groups which anticipate a close listener area.

6.3.2.3. Surround

Panning & Spread?	Delay processing for each object?	Number of En-Space filters
Yes	Yes	40 x2 early reflection zones

For a detailed comparison see the Function Groups overview table.

With a Soundscape 180°+ or 360° deployment, 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.

Surround speakers with the Function group Mode set to *Surround* will adapt their behavior in consideration to speakers with Function groups Modes *Frontfill* and *Main system*, by automatically taking over a sound object when it is within their positioning range.

In a typical setup, surround speakers will be less powerful than the mains system. 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 direction of smaller speakers. Using ArrayCalc, the best combination of mounting height and vertical directivity of the loudspeakers should be evaluated to achieve an even coverage across the audience area. If long distances need to be covered, and low speaker placement is unavoidable, higher vertical directivity will be required to achieve the desired effect for the whole audience. When mounting height is restricted, 16C or 24C column speakers may be useful because of the improved vertical directivity. Additionally, surround-delay speakers can be considered which are placed more towards the center of the audience to augment the outer speakers.

Additional Surround Function groups
For venues which have multiple audience areas (e.g., floor and balcony), it may be required to have multiple Function groups of surround speakers. This will allow multiple surround systems to seamlessly take-over a sound object when it leaves the mains and front fills while taking into account each surround group's unique speaker positions. This will allow both surround systems to be rendered individually for their respective audience areas.

6.3.2.4. Modes for Subwoofers

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

1. **SUB array** may be assigned when even coverage and maximum output level is preferred, subwoofers can be entered in ArrayCalc as a mono SUB array. In this case, they are defined as a Function group with the SUB array Mode selected. In this Mode, a single DS100 output will feed all subwoofers, and amp delay may be used for LF pattern control. With this Mode, the single DS100 output still has varying delay times to ensure the subs are generally in-time with the mains and their changing delay times.
2. **SUBs group** is a Mode for subwoofers which can provide imaging of low frequency sounds. To do so, these speakers must be entered into ArrayCalc 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.

Mode	Panning & Spread?	Delay processing for each object?	Number of En-Space filters
SUBs group	Yes	Yes	7 x2 early reflection zones
SUB array	No	Yes	1 x2 early reflection zones

For a detailed comparison see the Function Groups overview table.

Note: Signals sent to the subs can be controlled on a per-sound object basis within R1 using the Sound Object Routing parameter. This allows subs to be selectively used (similar to running SUBs-on-an-Aux) while retaining the benefits of Soundscape.

Notes regarding SUBs-on-an-Aux: 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 will not. This may result in unintentional alignment issues within the system. However, some users still prefer the SUBs fed from a console aux, particularly with LFE (Low-Frequency Effect sub) applications. This is possible using the processors matrix routing functionality.

6.3.2.5. Outfill

Panning & Spread?	Delay processing for each object?	Number of En-Space filters
No	Yes	1 x4 early reflection zones

For a detailed comparison see the Function Groups overview table.

The *Outfill* Function group Mode will produce a mono signal but will employ individual delay times for each sound object in order to provide time-alignment to the performers on stage and a smooth transition from neighboring speaker zones. This Mode is always-on for all sound object positions and may be useful for audience areas that are outside the coverage of surround speakers. If an outfill is needed which covers an area within the surround system, the *Outfill embedded* Mode should be used.

Because this Mode includes unique delay processing for every sound object, this allows a mono outfill to be time-aligned to the mains system for an object placed on stage and simultaneously time-aligned to the surrounds for a surround object position.

6.3.2.6. Outfill embedded

Panning & Spread?	Delay processing for each object?	Number of En-Space filters
No	Yes	1 x4 early reflection zones

For a detailed comparison see the Function Groups overview table.

This mode should be considered if fill speakers are used to cover an audience area within a surround system.

Using the Outfill embedded Mode, the level of a sound object will be linked to its level on the first Main system Function group. In other words, objects amplified by the first Main system Function group will also be amplified by this Function group mode.

As a consequence, when a sound object is reproduced by the surround system—and therefore is not amplified by the mains—it will not be amplified by the Embedded Function groups. This avoids disturbing the localization of surround effects.

6.3.2.7. Delay line

Panning & Spread?	Delay processing for each object?	Number of En-Space filters
Yes	Yes	7 x2 early reflection zones

For a detailed comparison see the Function Groups overview table.

The *Delay line* Function group Mode provides localization, but 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 balconies within a 180° deployment, or overflow seating areas that do not have surround speakers.

For example: Under balcony speakers which cover an area within a surround deployment should be assigned to the *Delay line embedded* Mode, while delay speakers for a balcony which does not have surrounds should be assigned the *Delay line* Mode. This will achieve a surround mix for the lower areas and a 180° mix for the balcony seats at the same time. In other words, the availability of this Function group Mode allows a processor to simultaneously drive a 360° deployment and a down-mixed 180° deployment at the same time, automatically.

💡 **Tip:** This Mode has also been used to generate a binaural output which can be sent to broadcast or record feeds. Simply create two loudspeakers located anywhere within the audience to create a two-channel down-mix from the perspective of the audience. In the ArrayCalc patch plan, draw 'draft patch' lines from two DS100 outputs directly to those speakers, without an amplifier in between. This will program the DS100 outputs without requiring a d&b amplifier to be added to the project. These two processor outputs can then be sent to record or broadcast and will include object spatialization and En-Space.

For areas that are within a surround system, the *Main system* or *Delay line embedded* Modes should be used.

6.3.2.8. Delay line embedded

Panning & Spread?	Delay processing for each object?	Number of En-Space filters
Yes	Yes	7 x2 early reflection zones

For a detailed comparison see the Function Groups overview table.

The *Delay line embedded* Mode should be used if delay speakers are used to cover an area within a 360° system. Using the Embedded version of the Delay line mode, the level of a sound object will be linked to its level on the first Main system Function group. In other words, objects amplified by the first Main system Function group will also be amplified by these Embedded Function groups.

As a consequence, when a sound object is reproduced by the Surround system, and therefore is not amplified by the Mains, it will not be amplified by the Embedded Function group modes.

6.3.2.9. Ceiling

Panning & Spread?	Delay processing for each object?	Number of En-Space filters
En-Scene not supported		7 x2 early reflection zones

For a detailed comparison see the Function Groups overview table.

The *Ceiling* Function group Mode is primarily used for En-Space and does not support sound object positioning via En-Scene. However, ceiling outputs can be addressed via DS100 matrix inputs which is useful for overhead sound FX.

💡 **Tip:** The ceiling En-Space impulse responses are measured along the center of the ceiling, from the stage to the rear. Therefore, all of the discrete convolution filters will be applied along the same positions. For this reason, it is generally recommended to pair speakers on a single amp channel if they are located across the ceiling, from left-to-right. This will help reduce unnecessary amplifier costs.

6.3.2.10. Mono out

Panning & Spread?	Delay processing for each object?	Number of En-Space filters
No	No	0

For a detailed comparison see the Function Groups overview table.

This Mode is an unprocessed output which does not support En-Scene positioning, delay processing, or En-Space. It can be useful for utility purposes like fold-back, press feeds, lobby feeds, etc.

6.4. Alignment tab

With Soundscape, you do not use the Alignment tab of ArrayCalc as you would with a traditional PA system. The Alignment tab corresponds with amplifier channel delay, which generally should not be implemented with a Soundscape system because:

The Soundscape processor will (generally) handle all required delay times on a per-sound object basis!

Sound objects in Full Delay mode are time aligned automatically based on the speaker positions. For this reason, Full Mode is recommended for most sounds objects and therefore 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 be performed using the signal delay of the amplifiers. Instead, it is set within ArrayCalc in the **Devices > Function groups > Configuration** table. More commonly, this alignment parameter is entered using R1 when the system is deployed.

In this table, each Function group can be assigned a Function Group delay setting, which is then applied to the DS100 signal processing matrix. It also provides the possibility to set the Spread factor for each Function group participating in En-Scene.

Note: The Alignment tab 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 for deployment.

Should individual corrections of the delay time within a Function group be necessary, (e.g., due to speakers mounted at different heights) this must be performed using the individual delay settings of the amplifier channels. If other factors influence the latency of the system downstream of the DS100, amp channel delay can be used to compensate, if needed.

For example: If ArrayProcessing is used on arrays, the inherent latency of those arrays will become 6.2ms. 6.2ms can be entered into all other amp channels to compensate, if desired. Or, if the 5D amp is used for front fills, 1.1ms can be entered into all other amp channels to match the 5D inherent latency. However, this is not strictly required and often not recommended. 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.

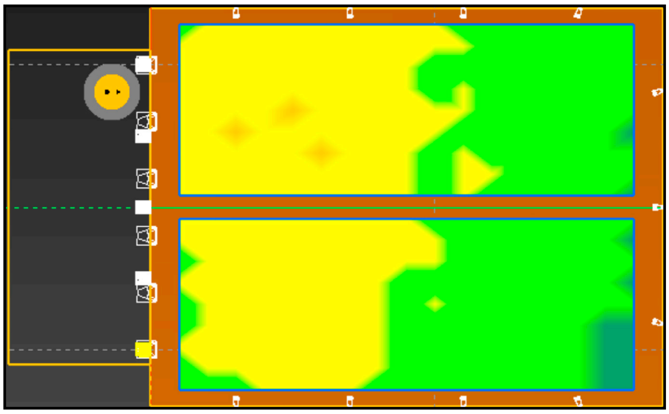
Note: The Function Group delay and Spread factor can be modified later while online in R1 via the **Devices > DS100 > Function groups** view.

6.5. Soundscape tab

6.5.1. Level simulation

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

The simulation takes into account all loudspeakers assigned to Function groups using their respective En-Scene processing and all relevant settings available in ArrayCalc like *Function Group delay, Spread factor, mute, level, amp delay, ArrayProcessing, speaker dispersion,* as well as *temperature* and *humidity*.

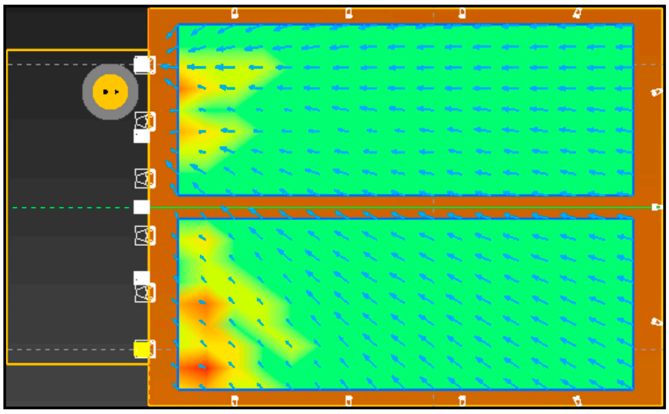


By placing the sound object in various example positions and recalculating the SPL plot, it is easy to observe if some speaker zones need to be attenuated to account for varying system sensitivities.

💡 **Tip:** Mute all but one Function group of loudspeakers and test all example sound object positions. This is an easy way to visualize if the unmuted speaker zone behaves as expected with all sound object positions. If unintentional behavior is observed, you may consider changing Function group allocation or mode.

6.5.2. Localization simulation

Using the same parameter set, ArrayCalc also predicts the localization accuracy of the sound object position for all listening areas. The arrows point towards the perceived direction of sound by each listener position. The color mapping indicates the deviation from the actual position of the object.



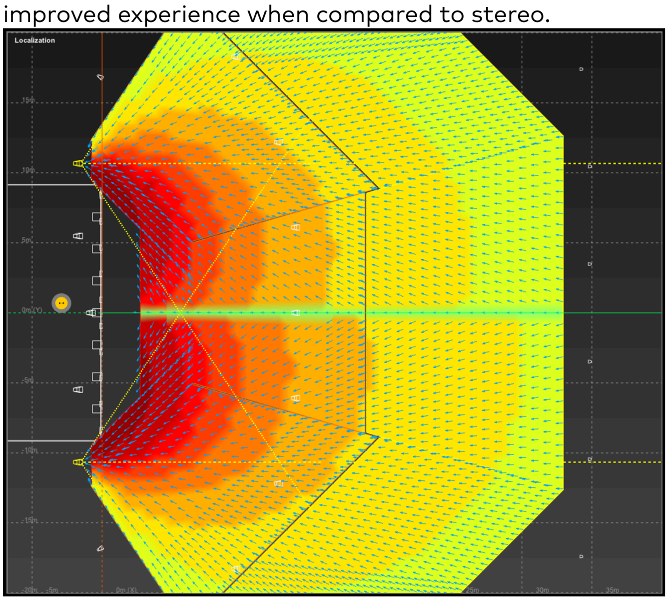
Shown here: There is insufficient speaker density for proper localization for the nearest listeners. This is visualized by the yellow/orange/red color variations near the front of the stage, which represent audience areas that perceive the sound as coming from any direction other than the object's position. In the real world, frontfill speakers usually fix this issue due to their higher speaker density. However, it should be noted that this imperfect localization is still far superior than a stereo deployment.

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

- **Precedence** (recommended): An empiric model based on the precedence effect or "law of the first wave front" (more information here). When a signal is played from multiple sources, the perceived origin is a combination of the various source positions and arrival times which are 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 4kHz.

- **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 Head-Related Transfer Functions (HRTFs, more information here). It derives the perceived direction from an analysis of the inter-aural transfer functions and resulting binaural cues in the 500Hz to 2kHz band. Areas with shorter arrows indicate a reduced convergence of the results.

The Localization plot will show areas of yellow/orange/red when a listening position is too close to an individual speaker, which causes the algorithm to fail for that listener. In the real-world, most systems have some audience areas which are slightly compromised (yellow/light orange). Keep in mind, that even in this situation, those audience members are still getting a vastly improved experience when compared to stereo.



An example of the typical localization errors common in traditional stereo systems.

Shown above is an example of how stereo can be predicted:

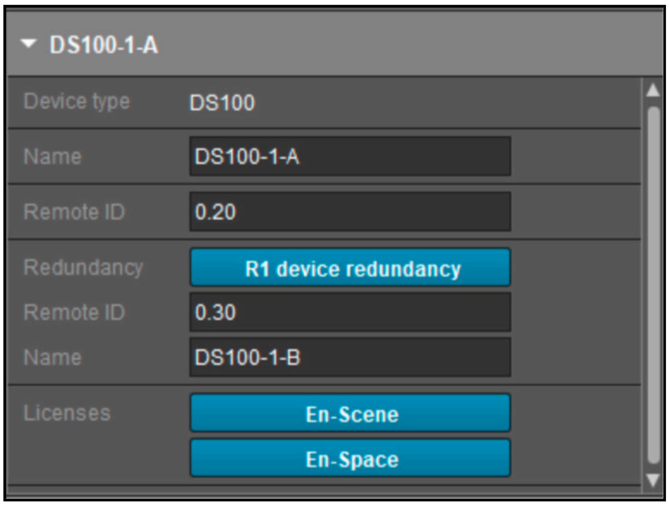
1. **Mute all speakers** except two mains
2. Change the sound object Delay Mode to **Off**
3. Press **Recalculate**

Only the small percentage of the audience located in the green area (aka “the sweet spot”) perceives the sound as coming from the sound object. Almost all of the audience is in yellow/orange/red and experiencing poor localization because they perceive the sound as coming from the speakers, not the performer.

6.6. Devices tab

6.6.1. Patching

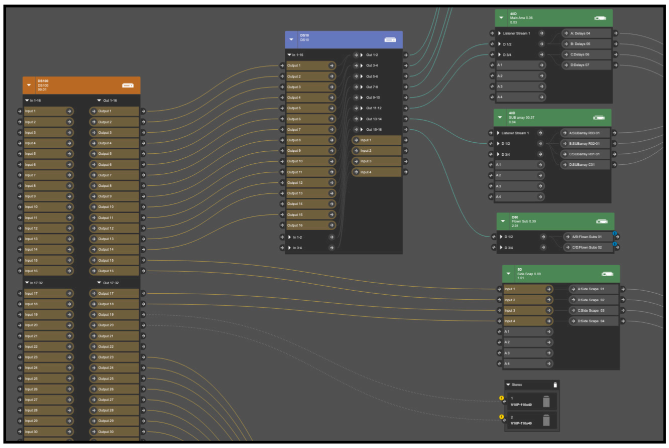
1. In ArrayCalc select the **Devices** tab and in the **Network devices** table.
2. Press the **+** button to **add a DS100 or DS100M**. You have 3 choices of I/O sizes available when adding a DS100 or DS100M. The selected I/O size must match the license installed on the processor(s) you will use.
3. If a backup processor is desired, it can be added as a redundant backup to the existing processor (not added as a separate device).
4. You can also add DS10 (Dante) or DS20 (Milan) network bridges to support amplifiers that do not have those protocols natively supported.



Two DS100 processors setup to run in redundant configuration.

By patching the entire system in this method, a lot of time is being saved. This process accomplishes the following:

- Programming processor outputs to speaker positions
- Programming processor outputs to Function group behaviors
- Programming amplifiers (input routing, output DSP, level and voicing filters)
- Generating line drawings for onsite work (and you may consider using the ArrayCalc viewer app for iOS and Android. More info here)
- For Dante network: you now have the ability to export a Dante preset file!



Shown here: DS100 outputs have been patched through a DS10 (purple) for amplifiers that do not support Dante, whereas other outputs have been patched directly to some 5D amplifiers which have native Dante support. Additionally, outputs 19-20 have been assigned directly to loudspeakers using a “draft patch” visualized as grey dotted lines.

💡 **Tip:** It is recommended that the order of speaker sources forming one Function group should be clockwise (left-to-right) 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 reordered using drag within the Sources view. This approach will help to more easily test the final routing using the DS100’s noise generator, allowing you to simply and quickly verify that all DS100 outputs produce sound, starting from house left and continuing clockwise around the system.

Additional loudspeakers
Additional loudspeakers that are not participating in the d&b Soundscape processing, such as stage monitors, can also be part of the project; however, they are not assigned to a Function group. If desired, they can be assigned to available matrix outputs of the DS100. This way they can be controlled from the manual matrix controls of R1.

6.6.2. Channel and device names

You may choose to rename channels and devices to best fit your desired nomenclature. These device names will be pushed to all available device and Dante Controller listings when operating the system.

6.6.3. Processor input names and config

DS100 and DS100M input channels can be named and configured to operate in Matrix mode or En-Scene mode. This pre-configures the .dbpr file for when it is opened in R1, saving a lot of time programming the online system.

Patch plan | Channels | Devices | Function groups

Inputs | Outputs

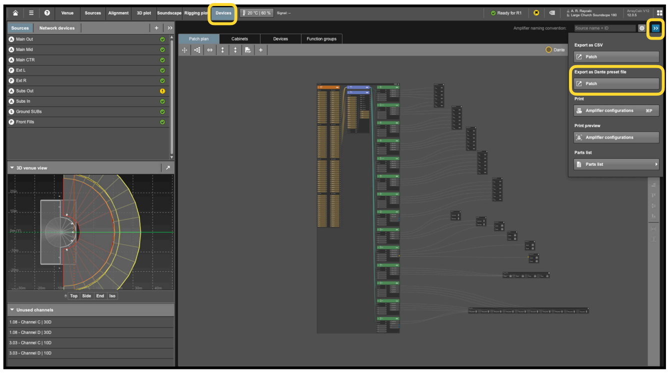
Status	Inputs of	DS100	Input Mode
✓	1	Kick	En-Scene
✓	2	Snare	En-Scene
✓	3	OH	En-Scene
✓	4	Bass	En-Scene
✓	5	Piano Lo	En-Scene
✓	6	Piano Hi	En-Scene
✓	7	SFX to FX speaker 1	Matrix
✓	8	Monitor feed to stage	Matrix
✓	9		Matrix

Shown here: Some inputs will be run in En-Scene mode, where also the output from speakers will be dictated by sound object settings such as position. Other inputs are left in Matrix mode to allow those signals to be manually routed to any parts of the system in a fixed method.

6.6.4. Export Dante preset

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

1. In ArrayCalc, click on **Devices > Network devices**.
2. At the top right of the **Network devices** view, click the **>>** menu and click the ↗ **Patch** button under *Export as Dante preset file*.
3. Choose a location to save your preset file and click Save.
4. **Open Dante Controller**
5. In Dante Controller, **load and apply the preset file** to the Dante network. Doing so will name all Dante devices within the file, name all the channels for those devices, and applies crosspoint matching between devices.
6. Lastly, 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 lowest stable latency in Dante Controller. This is because all alignment to the stage performers is handled within the DS100 without considering downstream system latencies. With most Dante network setups, latency settings for DS100 and DS10 devices below 1ms are stable.

6.6.5. Export Milan Manager file

ArrayCalc can create a Milan Manager preset file which provides the routing channel names of DS100M outputs to any DS20s or d&b amplifiers which have native Milan support. It works very similarly to the process described above for Dante systems.

7. R1: Controlling Soundscape

7.1. Introduction

R1 is free software for macOS and Windows that allows for the control of all d&b amplifiers and all DS100 variants and can be downloaded here.

An ArrayCalc .dbpr file can be opened within R1 which will:

- Automatically create controls based on the system's design in ArrayCalc.
- Automatically push tuning settings to the amplifier DSPs from ArrayCalc.
- Automatically inform a DS100 or DS100M of all speaker positions and orientations.

The .dbpr file can then be customized to include lock-out features, custom control interfaces, and recallable snapshots within R1. R1 is the only spatial control software that allows for control over routing, spatialization, emulated room acoustics, and amplifier DSP at the same time.

The following section assumes you already have a working knowledge of ArrayCalc & R1 as it pertains only to the additional steps required when designing a Soundscape system.

For education on the operation of ArrayCalc, please view our ArrayCalc tutorial videos here. Educational information on the operation of R1 can be found here.

7.1.1. R1 push/pull behavior

Opening an R1 file does not automatically reset all settings and recall a previous state. If you want to recall settings from a previous state, you will need to recall an R1 Snapshot or so-called System Settings. See Storing and recalling settings for more information.

If you are setting up a system for the first time, ensure that all devices have been reset before continuing. Otherwise, settings from previous events may be left running within the amps and processors.

What Soundscape data is automatically 'pushed' from R1 when going online with amps and Soundscape processors?

- En-Space and/or En-Scene enable (licenses still required to be installed within the processor in order to operate)
- Coordinate mapping/positioning planes
- En-Space room dimensions
- Function Group allocations/modes/names
- Input and output channel names
- Input mode (Matrix vs En-Scene)
- Speaker positions and orientations

Tip: These are all settings which are not variable within R1. In order to change these settings, R1 must be in Configuration Mode or the file must be re-opened in ArrayCalc.

What Soundscape data is 'pulled' from a DS100 to R1 when going online:

- Matrix input & output settings (mute, gain, delay, EQ, polarity)
- Matrix node settings (gain, delay, enable)
- Object settings (Spread, Delay Mode, Position, En-Space send level)
- En-Space Settings (Room Selection, Predelay Factor, Rear Level, Zone Send Gain, Mute)
- Function Group Settings (Spread Factor, Delay)

To recall these settings when going online, a Snapshot, System Setting, or Device scene must be recalled.

Note: Device Scene parameters (Scene index, Scene name, Scene comment) are compared between the online device and the R1 file. The user is then given a choice to pull from the online device or to push from R1.

7.2. Initial processor configuration

Before connecting a DS100 processor to your project file, the unit's firmware must be updated, network settings configured, and remote ID assigned to match your desired settings.

Control network & firmware update

1. The device's control network port has a default address of 192.168.1.100. Set your computer to an IP address within that range (192.168.1.xxx) and connect an ethernet cable to the control port of the device.
2. Within a new project in R1, click **ONLINE** and navigate to the **Devices** page **>** **Service** tab. Select the DS100 within the left column (the device should show a green connection indicator if connected properly).
3. Within the right window, navigate to the firmware tab and select the newest DS100 firmware from the list on the right. You must have an internet connection for firmware versions to appear in the list.
4. After firmware update: Within the right window, select the **Parameters** tab and filter the parameters by **Network**. Change the remote ID to match the ID which is preset within your ArrayCalc file. The DS100 will go offline and reappear with the new ID as the parameter is changed.
5. Change the IP address to be within your desired range or to an automatic mode (DHCP or Link-Local). The DS100 will go offline as soon as you enter the new IP address. You may now need to change the address of your computer to connect to it again.

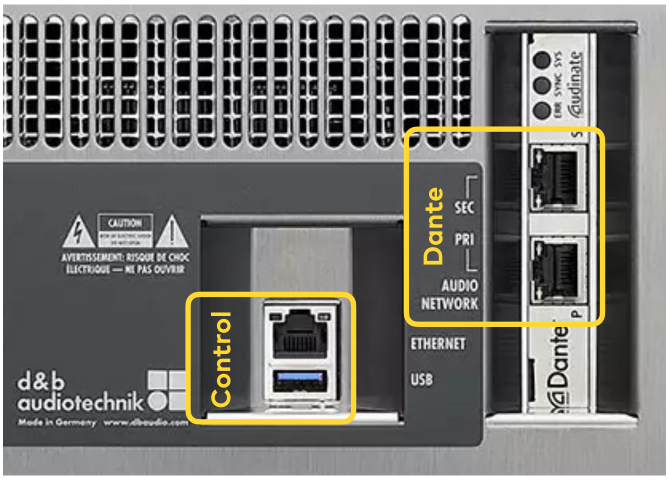
Dante network configuration

Dante Controller software should be connected from a computer to the primary port of the Dante card in the DS100 and configured in the same method as any other Dante device.

The Dante configuration is not adjustable via R1. Likewise, DS100 control parameters cannot be accessed through a network connection on the Dante card.

More information regarding the configuration of Dante devices can be found on Audinate's website here.

Note: Dante and control of the DS100 are always separated onto two different ethernet ports on the device. However, both control and Dante Primary can be connected to the same network via a network switch.



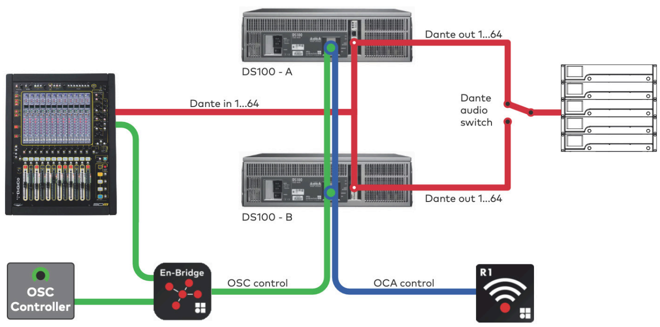
Milan network configuration
A Milan network is configured in a similar way to Dante except instead of using Dante Controller, you will require Milan Manager which can be downloaded here.

Generally, Patching Milan is very similar to Dante except one key difference: each group of 8 DS100 output channels are fixed within a "stream."

7.2.1. Processor redundancy

Introduction
The d&b DS100 signal engine is a powerful multi-channel signal processing device which very often fulfills a central function in a sound system. Therefore, a system topology using redundant DS100 devices may be desired. The R1 Remote control software supports this functionality. Device redundancy covers En-Scene, En-Space and matrix operation of the DS100, and any combinations thereof.

Even OSC control of sound objects can be applied to redundant DS100 devices using optional En-Bridge software.



Components
A redundant setup of DS100 requires at least one pair of DS100 signal engines, both equipped with matching Add-Ons (En-Scene and/or En-Space) and I/O Size licenses.

In redundant operation, both devices simultaneously process identical signals. They are always configured identically and controlled simultaneously to enable swapping the signals without changing the system response and sound design. Parallel operation, configuration, and synchronization of the devices is performed using the respective functions in R1, which is outlined on the following page.

As usual, ArrayCalc can be used to prepare the configuration and create the Dante Preset File for the audio routing.

DS100 audio routing
Identical input signals must be sent to both the main and redundant DS100s to function. In order to switch between DS100 A and DS100 B, the output signal routing to the amplifiers needs to be modified.

Depending on the desired failover performance and hardware effort, different means can be used for this purpose. Three possible switch-over solutions are described below:

- **Dual Dante preset files**
  Load the Dante Preset File created by ArrayCalc into Dante Controller. This will patch DS100 A to the entire system, ready for operation as the primary processor. Then, change the network patch so DS100 B is connected to all speakers and ready for operation. Save this as the backup Dante preset file. Now the user has two Dante Preset Files which can be loaded, depending on which processor is needed.

  In this scenario, swapping devices will interrupt the audio signal for until the Dante subscriptions in the network are reestablished.

- **Third-party audio switch**
  Using a hardware audio switch (for example, a Dante enabled router or matrix), the outputs of both DS100 devices can be connected simultaneously. A setup like this allows swapping DS100 devices without the latency of reconfiguring the Dante network. Some users have opted to use the XDante-1 by Autograph Sound (here).

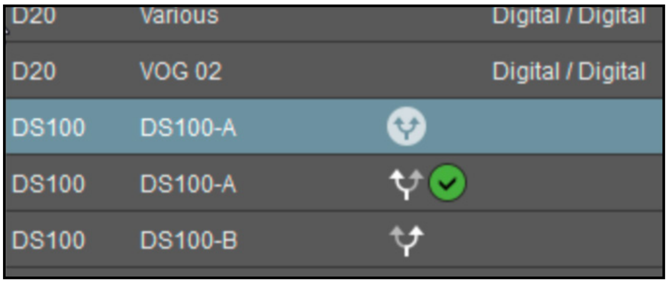
  This method introduces a new single point-of-failure—the hardware audio switch—but does not require much downtime when transitioning between primary and redundant processors.

- **Amplifier input switching**
  When sending the output from DS100 A to the speakers through a DS10 and AES3 amp inputs, the DS100 B signal can be sent to the analog inputs of each amp (using a third-party Dante to analog converter). The analog/digital amp inputs can then be swapped from R1 or triggered automatically by the amplifiers using Input Fallback. With most d&b amps (10D, 30D, D20, 25D, D25, 40D, D40, D80, D90), the Fallback function provides an automatic failover from digital to analog inputs by monitoring the DS-Data in the AES3 stream.

  Not all amplifiers support these functions equally. For example, The 5D amplifier is not compatible with AES3 signals. Second example: D20 and D80 amplifiers do not support all four AES inputs at the same time as analog inputs.

  This method not only avoids any latency or down-time but also does not add a single point-of-failure.

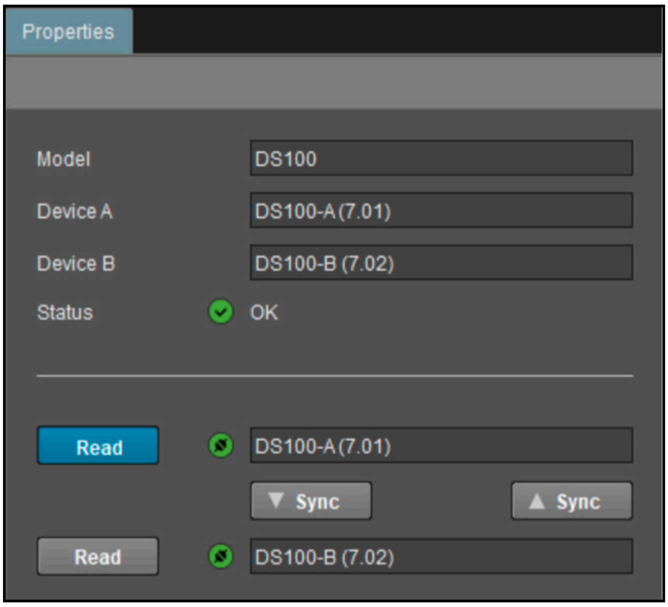
Configuration in R1
If not already configured in ArrayCalc, redundant DS100 devices can also be assigned in R1 with the following steps:
1. Edit the **Devices** table and **Add** a second DS100 unit.
2. Select the DS100 A device and choose the respective DS100 B device from the drop-down list provided.



Shown here: In the **Devices** table, three devices will be listed—the combined redundant device and both physical devices. The device that is selected for readout is marked with a green checkmark.

Operation in R1
In R1, redundant DS100 devices are operated using the same functions and controls as for a single unit.

The *Device redundancy* system view shows redundant pairs of DS100s and allows you to select which one of the paired devices will be used to *Read* the parameters for R1 controls in the workspace and in the *Devices* view.



As all R1 controls and commands act on both devices, their status will normally be identical. However, after initially going online with R1, you should use the **Sync** function to push all parameters and scenes from one device to the other. Sync can be performed in either direction.

A typical use case is to Read the device which is initially active in the signal chain (e.g., DS100 A). Should it be necessary to switch to the redundant DS100 B (timeout messages in R1, audio misbehavior, or interruption), the readout should also be switched to DS100 B. When DS100 A is up and running again, syncing DS100 B to DS100 A will replicate all current settings.

If in doubt or after a failure of one device or an interruption of the communication with one of the devices, please repeat the sync of the devices within R1.

OSC control
All the features written above are functions of R1. If external OSC control is used (DiGiCo, DAW plugin, etc.), then extra effort must be made to ensure that the OSC data is received by both DS100s.

En-Bridge is a great tool to achieve this: simply have external devices target En-Bridge. En-Bridge will then relay the messages to a primary DS100 and a secondary DS100 at the same time to ensure that both units are in the same state.

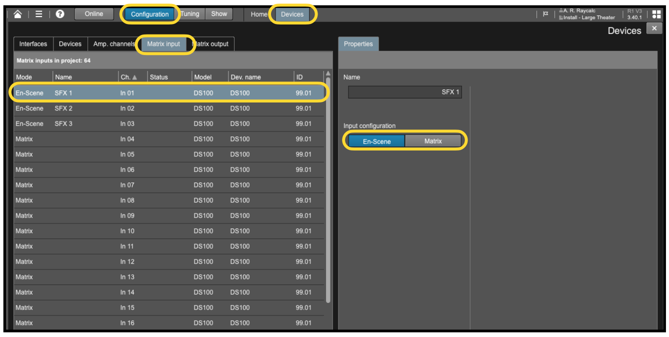
See Intro to OSC control for more detailed information.

7.3. DS100 input & output settings

7.3.1. DS100 input settings

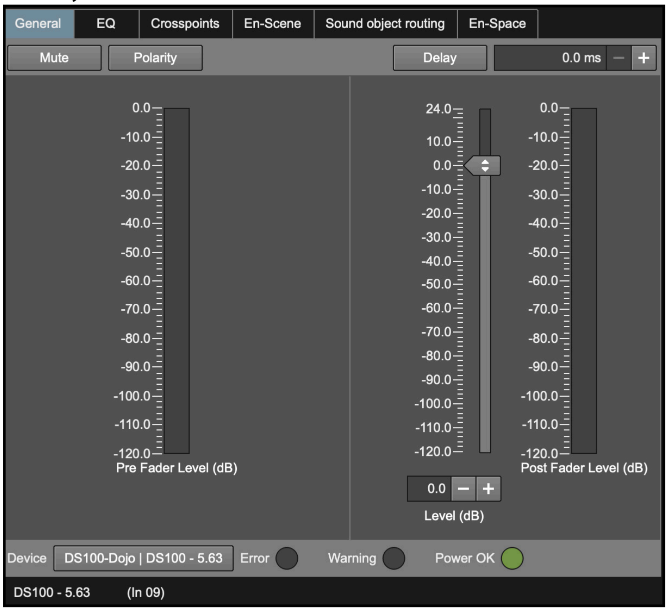
Any DS100 input can be operated in one of two modes:
1. **Matrix**: Allows for manual routing of this input to any DS100 output, as is typical in legacy signal processors.
2. **En-Scene**: Output of the signal will be determined by the sound object position. However, it can also be manually routed to outputs which are not assigned to a Function group. This can be handy for building a press-feed, fold-back mix, or binaural mix of your sound objects.

To change the mode of a DS100 input:
1. When in **Configuration** mode, open the **Devices** view, then the **Matrix input** tab.
2. On the **Properties** tab on the right-hand side, enter a name and select **En-Scene** or **Matrix** as the Input configuration for all required input channels.

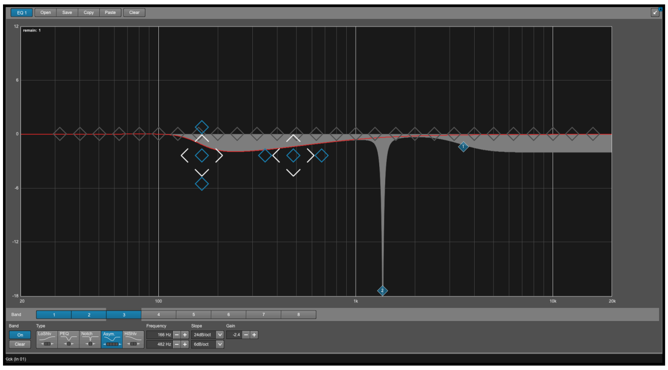


Note: When changing to **Tuning** mode, the *General* and *EQ* tabs on the right-hand side provide the input processing options for each channel: Gain, Delay, Mute, Polarity, and an 8-band parametric EQ. Changing input mode is not permitted when in Tuning or Show modes.

All DS100 or DS100M inputs have the following processing available at all times: Mute, level, polarity, 500ms of delay, and 8-band parametric EQ. Processing is available regardless of input mode (matrix or En-Scene).



Shown here: all the tabs for a matrix input.



Shown here: an expanded channel EQ with an asymmetrical filter, a notch filter, and a shelf with adjustable slope.

7.3.2. DS100 matrix output settings

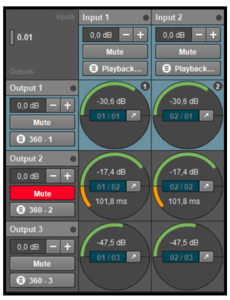
The *Matrix output* tab is also to be found in the *Devices* view. When in Tuning mode, the *General* and *EQ* tabs on the right-hand side provide the processing options for each output channel: Gain, Delay, Mute, Polarity, and a 16-band parametric EQ.

7.3.3. Manual matrix routing

For all inputs in Matrix mode, the DS100 signal matrix can be operated manually by the controls on the **Devices** page > **Devices** tab > **DS100** tabs - or more conveniently by adding **Matrix crosspoint** controls to a Remote view.

When Matrix crosspoint controls are added to a Remote view, there are level and delay controls for a user-defined range of matrix crosspoints which can be operated individually, or by multi-selection.

Shown to the right is an example of a matrix control which was added to a remote view. In this example, it is set to only display inputs 1-2, outputs 1-3 and the associated crosspoint controls. Of course, this type of control can be set to display any number of inputs and outputs, as desired by the user. The menu button on the input and output



fields of the matrix control opens the respective input and output processing options.

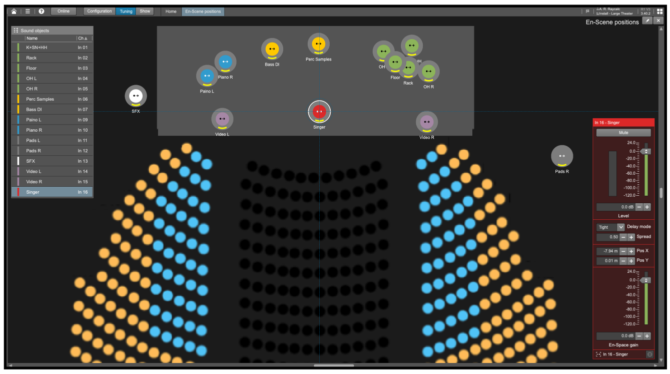
There are two situations where adjusting matrix crosspoints is permitted:

a) Any DS100 input in Matrix mode - all nodes are adjustable.
b) Any DS100 input in En-Scene mode can be manually routed to any DS100 output that is <u>not</u> associated with a Function group.

Note: A matrix crosspoint cannot be adjusted if the DS100 input is set to En-Scene mode and the output is assigned to a Function group. This is because all the level and delay values are being handled by the En-Scene object position.

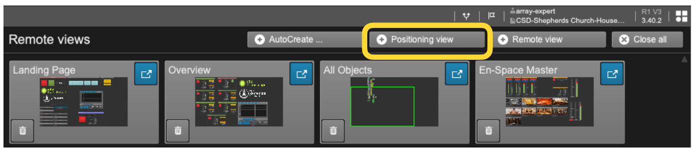
7.4. En-Scene operation within R1

7.4.1. Positioning view



Shown here: a positioning view with background image loaded to show the stage and seating area.

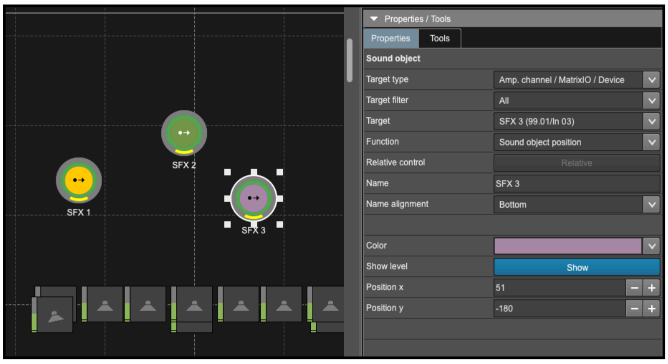
With R1 in Configuration mode, from the **Home** view, select **+ Positioning view** and assign a Positioning plane to this view.



7.4.2. Adding sound objects

Each En-Scene input channel may be represented by a sound object on a positioning view.

1. In **Configuration** mode, drag sound objects from the **Matrix inputs** table on the left into the view. This will use the default color and the channel name, if one exists.
2. In the **Properties** menu on the right, the name and color of each sound object can be defined as needed.
3. Switch to **Tuning** mode to freely move the sound object and control your system in real-time.



Shown here: when R1 is in configuration mode the selected sound object can have its control parameters edited on the right.

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. Otherwise, if the object is targeted to an input which is in Matrix Mode, the object position will not affect the sound. See DS100 input settings for more information.

The positioning plane selected for a given Positioning view serves as a reference to place and move the respective objects within the venue and to anchor the center point of the view. 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 up to four positioning planes in the design, and you can create as many positioning views as you like.

An image file (.png, .jpg, .jpeg) can be added to a positioning view (such as a stage plot, seating chart, or architectural drawing) to improve usability by providing a visual reference for the end user.

💡 **Tip:** Double-clicking on the background will re-center the positioning view on the selected Positioning Plane.

Grouping Sound Objects

1. In R1, click the *Groups view* from the home screen.
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 while maintaining the relative offset of their positions.

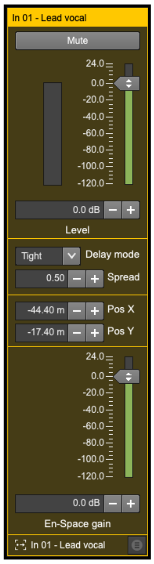
7.4.3. Sound Objects parameters

When R1 is in Tuning mode, the acoustic properties of each sound object (i.e., DS100 input) can be configured. This includes level, delay, EQ, Delay Mode, and Spread. If enabled, En-Space send level can also be controlled.

7.4.4. Spread

The Spread of a sound object defines whether it is reproduced in a focused (less speakers) or wide (more speakers) way. The Spread value ranges from 0 (focused) to 1 (wide) with the default value set to 50% (0.5).

Sound Objects with a Spread value of less than 50% (<0.5) have a more precise (focused) localization, but utilize less speakers for coverage. Because focused objects make use of the level of fewer speakers, they are more demanding of the SPL capabilities of the individual loudspeaker(s) they play through.



Conversely, Sound Objects with a Spread of more than 50% (>0.5) deliver a less focused image yet provide a more even level coverage. Wide objects make use of the level of more speaker positions and therefore, are less demanding regarding the SPL capability of the individual loudspeakers. However, wide objects may not have consistent localization.

The default value of 50% (0.5) is recommended for general use and does not necessarily require adjustment.

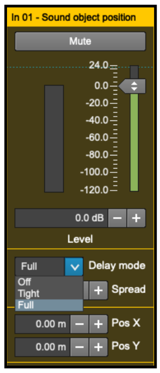
When the Spread value for a sound object is changed, the overall level of that object does not change. This is because En-Scene will automatically compensate the level to each speaker so the overall SPL output stays consistent. For this reason, you may not hear a difference when listening from a typical mix position. Spread is best experienced in the seats which are closest to the loudspeakers.

The Spread of an object is controlled on a positioning view by selecting an object by clicking the object itself or within the object list on the left, then expanding object controls in the lower-right corner of the positioning view.

The amount of Spread incurred by an object is also dependent on each Function group. For example, when the Spread of an object is widened, the front fills will become wider at a faster rate than mains. This is because of the inherently different behavior of those two Function group modes. This effect can be controlled via the Spread factor control for each Function group.

7.4.5. Delay mode

Three Delay modes are available for sound objects: *Full*, *Off*, and *Tight*. The Delay mode can be configured individually for each sound object in R1 on the **Devices view > Matrix input > En-Scene** tab, or within the sound object controls on a **Positioning view**, shown right.



Full mode: 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 which provides consistent time alignment across the entire venue in both X and Y axes. 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, such as guitar amps or wedges, Full mode should be selected in order to preserve the image and timing between the direct and reproduced sound of those sources.

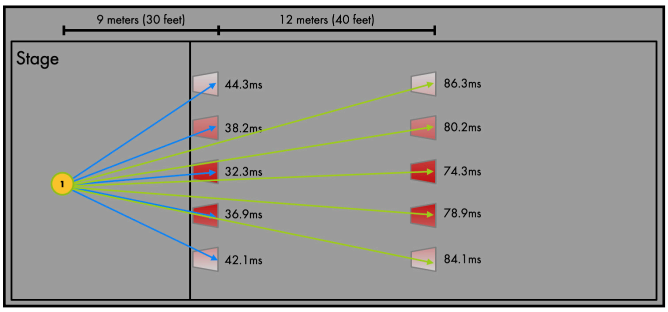
Off mode: When the Delay mode is set to Off, En-Scene only uses level adjustments to localize a sound object, without any delay times. All relevant sources of the Function group will reproduce the objects simultaneously; only the delay setting of the whole Function group will be applied for alignment with other Function groups. Off mode will provide less consistent coverage (as it inherently will be amplified by fewer speakers), less overall SPL capability, and may cause less precise localization of a sound object. However, it avoids audible artifacts of fast-moving objects as a result of the rapidly changing delay times associated with Full and Tight modes. Off mode is equivalent to VBAP (vector-based amplitude panning) which is common among other object-based mixing platforms.

Tight mode: When the Delay mode is set to Tight, level and delay are used to reproduce localization, but the total latency of its reproduction through all Function groups is minimized. The signal delay of each sound object is reduced by Δt (delta of time) equaling the distance to the closest loudspeaker of each Function group. Relative delay values between the sources of the group are kept, therefore the localization of the object is not affected.

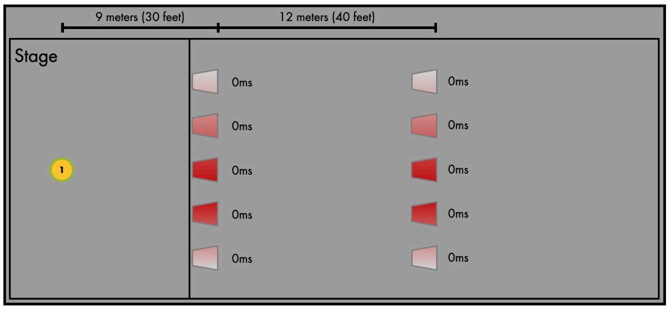
Tight mode is beneficial for a mix of electronic instruments and/or pre-recorded material in order to reduce relative delays between the channels depending on the placement of the objects, thus keeping the mix "tight." Tight mode is also advantageous for moving sound objects, as it reduces the variation speed of the signal delays.

Note: Full and Tight modes are only compatible with sound objects placed behind loudspeakers. Objects placed in front of all loudspeakers will automatically behave in Off mode, as delay is no longer possible. The object's Delay mode parameter will not visually change to Off, as it is still possible that the object may be delayed through other speakers in the venue. For example, under-balcony fills which point away from the object's position although the object is in front of the mains.

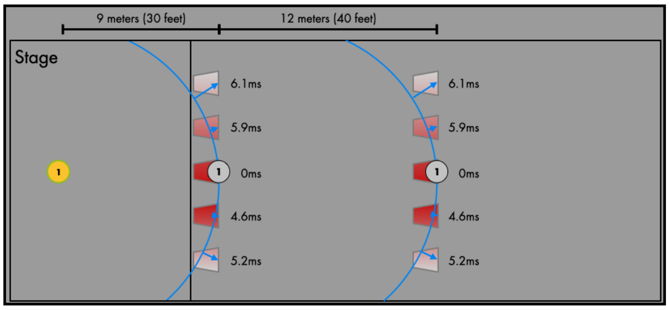
Note: Tight and Off modes will not properly time-align to the speaker automatically. For both these modes, a meaningful Function Group delay parameter input is required when commissioning the system with R1. Function Group delay is typically set once, during initial setup.



Full mode: The sound object has full delay processing to every speaker position, in both X and Y axes.



Off mode: The sound object has no delay processing and operates in level-only (VBAP or Vector-Based Amplitude Panning).



Tight mode: The sound object has differing delay times per speaker position for more consistent coverage and localization, but the overall delay is minimized separately for each Function group. Shown above is the actual object position (yellow) and the algorithm's presentation of the object (grey).

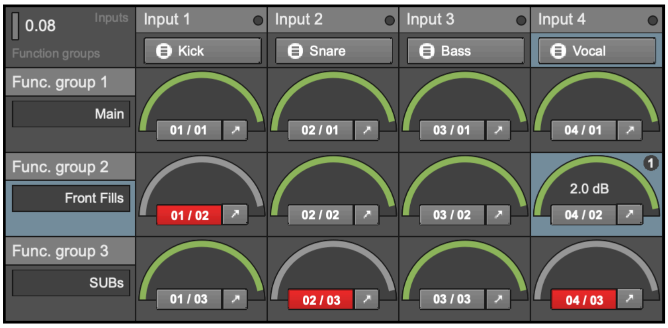
Mode	Recommendations for use
Full	Recommended most applications. Especially for objects that represent a source which makes acoustic sound (vocal, drums, guitar amp, etc.).
Tight	Recommended for sources where the overall delay needs to be reduced. Can work well for multichannel bus-based content which is routed to multiple objects (stereo, LCR, 5.1, 7.2, etc.).
Off	Recommended for fast-moving sound objects if artifacts can be heard. This mode is automatically used for objects which are in front of the speaker system. Therefore, this mode can be preemptively selected which avoids the sound of it changing to Off mode automatically during the transition from behind to in-front of speakers.

7.4.6. Sound Object Routing

As a default, all sound objects will be routed to all Function groups. This includes subwoofers, fills, delay speakers, etc.

Using the Sound Object Routing Matrix for each En-Scene input, the send gain from each sound object to each Function group can be set manually. This can be useful to modify the mix for a particular speaker zone (i.e., Function group) or to completely remove an object from it, such as subs. At each crosspoint, the *Gain* or *Mute* parameters can be set using absolute or relative faders. The latter is particularly useful when multiple crosspoints are selected.

As an example, to factor in the direct sound from stage wedges and amplifiers, the level of voices on the front fills could be boosted. Also, removing the vocals from subwoofers may be desired. In the example below, Input 1 *Kick* has also been muted in the front fills and Input 2 *Snare* has been muted in the subs, denoted by the respective red rectangles.



Shown here: A *Sound object routing* control on remote view in R1.

The Object Routing matrix can be accessed under **Devices > DS100 > Sound object routing** or with the respective control added to a Remote view. Object routing can also be found by expanding the sound object details from a positioning view as shown below.

7.4.7. Mixing considerations

Now that we've described Spread and Delay modes, let's apply some of these concepts in real-world applications to best understand the audible effects and how to properly utilize them.

7.4.7.1. The positioning approach

The Cartesian coordinate system (X/Y) used for sound object positioning allows the user to place a sound object to represent a real-world performer position. This is not possible with object-based systems that utilize a polar coordinate system (pan/distance) which controls objects around an egocentric sweet-spot in a method that is abstracted from the actual venue.

7.4.7.2. Low frequency impact

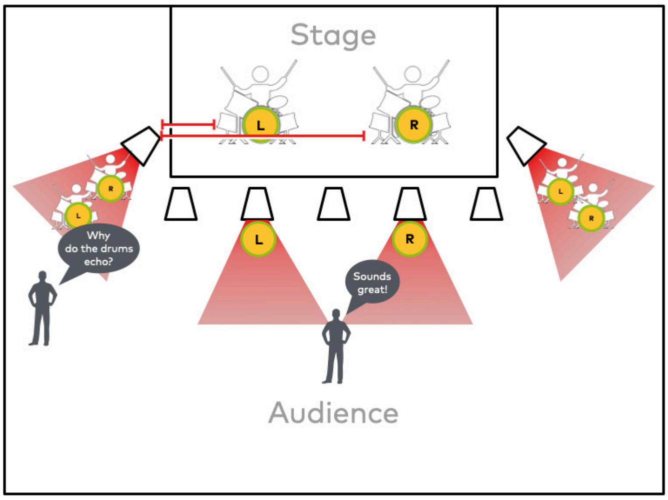
When listening at the mix position, which is typically in the center of the venue, it may sound like Off mode provides more impact or presence. This is the typical result of any speaker system without varying delay times applied to speakers: more energy is focused towards the center of the room, and this effect is commonly referred to as the "power-alley." In other words, while more impact is perceived in the middle of the venue, less impact will be perceived in all other areas of the venue, which creates inconsistency.

Furthermore, speaker zones such as outfills and front fills may now sound too loud and lack transparency because of their less precise delay times.

If you prefer the sound of more impact, use the channel processing on the mixer instead of biasing the object behavior.

7.4.7.3. Stereo content

Two-channel stereo content can be applied to two sound objects. Be careful when applying delay processing to stereo objects because the two channels share much of the same data. This means that the shared content could be summed into mono outfills with differing delay times. This easily goes unnoticed if the mix position is not within the coverage of the outfills.



In the illustration above, you can see two sound objects labelled L and R, which represent the position of a stereo drum bus. As is typical, the drum bus has kick, snare, and hi-hat all panned in the center. This means those signals will be sent to object "L" and object "R" at the same level. If the Delay mode of the two objects are set to Full, they will both combine into the mono outfills with differing delay times because they each have different distances from each mono outfill. The result is that the kick, snare, and hi-hat will come out of those speakers twice with different delay times. Channels which are hard-panned (such as overheads) will not exhibit this behavior because they are handled by only one object.

💡 **Tip 1:** Don't use stereo mixes if you can avoid it. Once signals are mixed to stereo, they can no longer benefit from the advantages of Soundscape. When stereo content can't be avoided, those objects should be used in Off mode, or you may consider switching those DS100 inputs from En-Scene mode to Matrix mode and manually routing them in a traditional stereo method.

💡 **Tip 2:** If you want stereo objects to be run in Full mode, you can consider utilizing the Sound object routing function to ensure that only one of the objects is sent to the Function group which handles the mono outfills.

Tip 3: You may also consider utilizing a third-party plugin to convert the 2-channel stereo content into stems or a surround upmix.

7.4.7.4. Musical timing & object distances

With *Full* mode selected, the sound of a sound object is reproduced by multiple loudspeakers with different delay times, each depending on the distance between the object and the respective loudspeaker. Furthermore, objects in all Delay modes are still subject to the natural propagation delay of sound (sound traveling through the air after leaving the loudspeakers. More info here).

Consequently, objects in differing positions are played through the loudspeakers with differing delay times, to differing listener positions. This unavoidable effect may create audible artifacts regarding the relative timing of instruments and negatively impact musical timing. Therefore, ensure that the physical distribution of the sound objects stays within acceptable musical limits. As a rule of thumb, a distance of roughly 15m/50ft (or 45ms) between rhythmic instruments should not be exceeded.

For non-rhythmic sounds, you have more freedom to place objects much further without disturbing the musical tempo/rhythm (for example: pads, choir, organ, and sound FX). Far distances may also work well with signals that have intentionally distorted timing such as stereo guitars with delay and reverb returns.

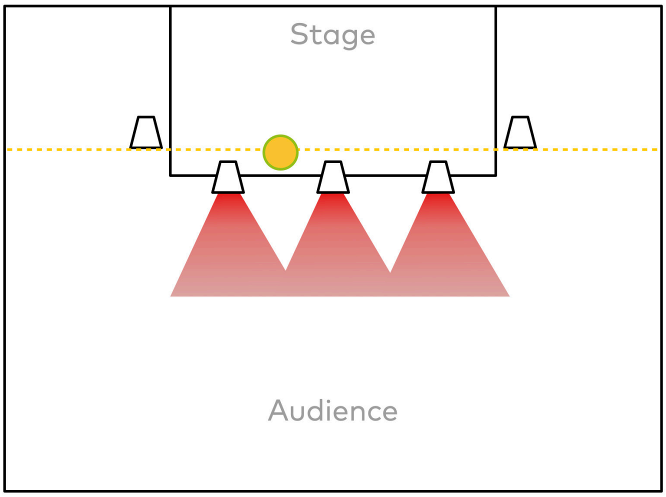
In general, objects should be placed to accurately represent the performer's actual position on stage. Off and Tight Delay modes may also help to reduce timing issues in some situations.

7.4.7.5. Objects vs. speaker positions

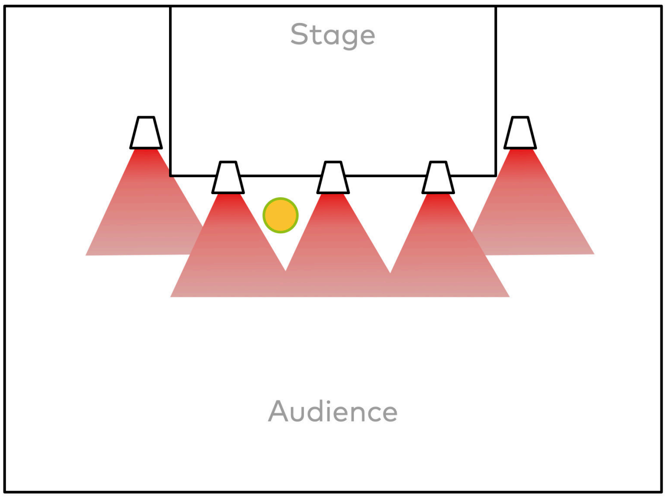
The En-Scene algorithm considers every object position in combination with speaker position and orientation. Furthermore, delay-processing is only possible when an object is behind speakers. As an object gets closer to a given speaker, the delay time will be gradually minimized. When the object crosses in front of the speaker, the delay time will become zero. The sound object now behaves in level-based panning mode (VBAP).

Additionally, there are a couple of rules built into the algorithm which are constantly being managed by En-Scene and may influence considerations for object placement, speaker placement, and speaker orientation:

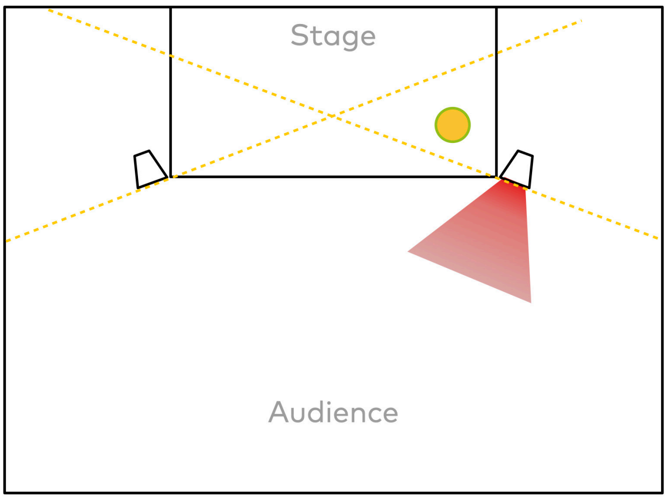
- The overall acoustic level of a sound object should never change, regardless of its position or how many speakers are amplifying it.
- En-Scene will always prioritize speakers which face away from a sound object's position. This is a requirement of the delay-processing.



Shown above: an object which is in front of the outer speakers (represented by the yellow dashed line). Therefore, the signal comes out of the remaining speakers within the Function Group. This illustrates one reason why it is generally recommended to deploy speakers within a linear line. Alternatively, if it is desired to have the outer speakers amplify this object position, they can be turned outward within ArrayCalc.



Shown above: an object which is in front of all speakers in the Function group. Therefore, the signal will come out of all speakers, without delay processing (level-based panning only).



Shown above: as a result of the horizontal angle of the speakers, this object is considered in front of the left speaker and behind the right speaker. If it is desired to have both speakers working to amplify this object position, rotating the speakers towards the audience within ArrayCalc (not in the venue) may be a suitable solution.

7.5. Acoustic properties of Function groups

Some *Function group* modes provide additional parameters to customize the behavior: *Group delay* and *Spread factor*. Both of these parameters are typically set once during the initial system commissioning. This is done in R1 in the **Devices** page **>** **Devices** tab **>** select **DS100 > Function groups** tab.

Function group	Name	Mode	Group delay	Spread factor
01	Mono SUBs	SUB array	0.0 ms	
02	Main	Main system	0.0 ms	1.0
03	Frontfills	Frontfill	0.0 ms	1.0
04	360	Surround	0.0 ms	1.0
05	SUBs	SUBs group	0.0 ms	1.0
06	Mono outfills	Outfill	0.0 ms	
07	Delays	Delay line	0.0 ms	1.0
08	AUX	Mono out	0.0 ms	
09	Ceiling	Ceiling	0.0 ms	
10	Outfill embd.	Outfill embedded	0.0 ms	
11	Delays embd.	Delay line embedded	0.0 ms	1.0

7.5.1. Spread factor

Using the *Spread factor*, the relative *Spread* of all sound objects can be adjusted for each *Function group* independently. A value range between 0.5 and 2 can be set for each group, with the default being 1. The *Spread* of all objects is multiplied with the set *Spread factor* value and the result applies to the corresponding *Function group*. With the default value of 1, the resulting *Spread* of all objects is multiplied by 1 (no change). However, the *Spread* of all objects within a *Function group* can be diminished as much as half (0.5) or increased by 200% (2.0). The parameter is also automatically added to the *Soundscape Outputs* remote view during R1 AutoCreate.

Example: audience areas to the extreme ends of the front rows do not hear enough level from object on the opposing side of the stage. In this case, the *Spread factor* of the Frontfills *Function group* can be increased to a value 1.4 in order to prioritize more even level distribution instead of accurate localization.

Spread factor can also be used to help overcome sub-optimal speaker spacing as it 'morphs' a *Function Group* back towards a traditional speaker system when the value is increased above 1.0. In this case, if the speaker system does not provide adequate resolution (or seats are too close to some speakers), you may prefer to expand the *Spread factor* to prioritize consistent coverage instead of consistent localization.

The default *Spread factor* of 1.0 will maintain the En-Scene algorithm as intended and will work for most applications.

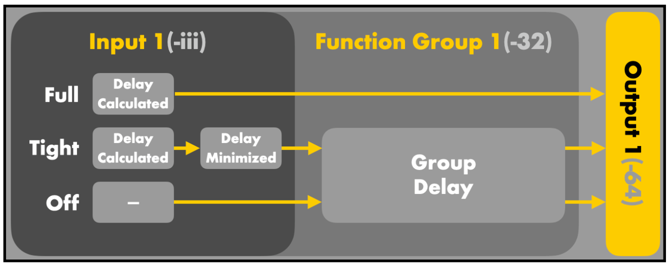
💡 **Tip:** A *Frontfill* Function group mode already has a wider *Spread factor* optimized for closer listeners. Using its *Spread factor* control, it can be widened further or it can be reduced to act more like other *Function group* modes.

Spread and *Spread factor* are not available for *Function group* modes which do not support object positioning (Outfill, Outfill embedded, SUB array, Ceiling, Mono out).

7.5.2. Function Group delay

Sound objects running in *Tight* or *Off* modes do not automatically process all delay times for all speakers like Sound objects in *Full* mode. Tight and Off mode objects will require some manual delay times to be entered. However, unlike a traditional system, delay times should not be entered into the amplifier channels as this will destroy the behavior of objects in Full mode.

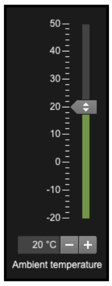
The entered delay time will be added to the entire *Function group* for all objects in *Tight* and *Off* modes. *Function Group delay* is completely ignored by objects in *Full* mode as their delay times are calculated more precisely in two axes.



7.5.3. Temperature

In R1 on the **Devices** page **>** **Devices** tab **>** select **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 which varies slightly by air temperature. Delay values set manually (*Function group delay* and matrix input/output/crosspoint delays) are not modified by this parameter.

Because temperature only has a minor effect on the propagation delay of sound, changing the parameter regularly is only recommended for systems that span large distances (≥100m/330ft) and have substantial temperature swings throughout the system's use (≥10°C/15°F). Most users leave the temperature control at the default value of 20°C/68°F without issue.



7.6. En-Space operation within R1

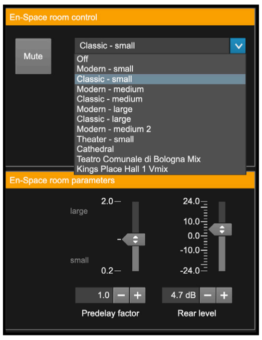
7.6.1. Venue Library

En-Space comes with a set of sampled concert venues (En-Space rooms) which range in reverberation time from 1.3 seconds to 5.6 seconds. They are also categorized by either modern or classical architecture, which generally affects the high frequency reverb time (≥5kHz). This extremely wide range of available En-Space rooms allows En-Space to be useful for a wide range of musical, theatrical, and architectural applications.

When combining an En-Space with your local environment, be aware that the acoustic responses of both rooms will sum together. It is not possible to shorten the reverb of the actual venue; it will only be extended. Therefore, the venue should have a considerably shorter reverberation time than the selected En-Space room, otherwise the audible effect is limited. There is no recommended baseline reverb time for venues that will use En-Space. Generally speaking, shorter is better. Most venues which are effectively designed for amplified music will work well with En-Space.

7.6.2. Selecting an En-Space room

In R1, click the dropdown list to choose which room you'd like applied to your Soundscape system. Currently, there are 9 rooms available which have a range of reverb times from 1.3 seconds (theater) to 5.6 seconds (cathedral). The available rooms have been measured by d&b and represent some of the world's most highly regarded acoustic spaces.



Number	Name	Reverb time
0	En-Space OFF	Reverb is OFF
1	Modern - small	2.0 seconds
2	Classic - small	1.9 seconds
3	Modern - medium	1.7 seconds
4	Classic - medium	2.1 seconds
5	Modern - large	2.6 seconds
6	Classic - large	2..4 seconds
7	Modern - medium 2	2.2 seconds
8	Theater - small	1.3 seconds
9	Cathedral	5.6 seconds
101	Custom Room A	empty by default
102	Custom Room B	empty by default
103	Custom Room C	empty by default

The list of currently available selections and spaces can be emulated in real-time.
The numbers correlate with their numbering via OSC recall.

See En-Space venue library for more information.

7.6.3. Predelay factor

The *Predelay factor* in R1 scales the predelay of all boundary responses of the selected venue. The range extends from 0.2 to 2, with the default value of 1 maintaining the original response of the measured venue. Values larger than 1.0 will 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. A Predelay factor of 0.2 will shrink the emulated room to 20% of its original size, while a Predelay factor of 2.0 will double the predelay, and thus the perceived size of the emulated space.

Example:

- *Bing Concert Hall at Stanford University* is listed in the En-Space venue library as "Modern medium - 2"
- Predelay differs by source and listener location, but let's generalize a predelay time of 87ms for Modern medium - 2.
- This roughly represents the amount of time it takes for a performer's acoustic sound to propagate to a side wall and reflect back to a listener in the center of the main seating area. In other words, the sound leaves the performer, travels towards the side wall (approximately 43.5ms) and then returns to the listener's ear (approximately 43.5ms).
  - 43.5ms x 2 = 87ms.

	Minimum Predelay Factor (0.2)	Default Predelay Factor (1.0)	Maximum Predelay Factor (2.0)
Original predelay time for "Modern medium - 2"	87ms	87ms	87ms
Multiplier applied to the En-Space engine	x0.2 (20% of the original value)	x1.0 (100% of the original value)	x2.0 (200% of the original value)
Resulting predelay applied by Soundscape	17.4ms	87ms	174ms

Shown here: A table illustrating the range of Predelay factor settings and how they impact the onset of reverb when the "Modern medium - 2" En-Space room is selected.

Therefore, the available range of predelay available for "Modern medium - 2" is between 17.4ms and 174.0ms. In reality, the math is highly variable and three-dimensional and cannot truly be reduced to single values. This is why the user is presented with a single fader control which equally affects all 144 convolution filters in three-dimensions.

It is recommended to use your ears and pick a Predelay factor that is sonically appealing with your content. In smaller venues, a reduced Predelay factor generally helps the emulated acoustics seem more believable. This is because the onset of reverb will more closely match the actual size of the venue, which the listener can plainly see with their eyes. This helps drive up the En-Space level without it sounding artificial.

With the range of reverb times within the venue library and the range provided by the Predelay factor, an extremely wide range of acoustic environments is possible with En-Space.

Note: Predelay factors smaller than 1 should only be applied when the actual speaker deployment is smaller than the selected En-Space room. Otherwise, the En-Space reproduction of a sound object may occur earlier than the direct sound.

7.6.4. Rear level

The *Rear level* fader in R1 adjusts the En-Space output levels in a ratio from the front to the back of the entire speaker deployment. The range extends from –24 dB to +24 dB, decreasing or increasing the reverberation level towards the rear of the speaker deployment.

Rear level can be used to adjust the direct-to-reverberant-field ratio along the depth of the speaker deployment. A main system with very high directivity (such as 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.

Alternatively, use the Rear level control to adjust the artistic goal of the En-Space response without affecting En-Scene or matrix-routed content.

Note: Rear level adjustments will not be reflected by the En-Space outputs faders mentioned below. Those faders run independently of rear level.

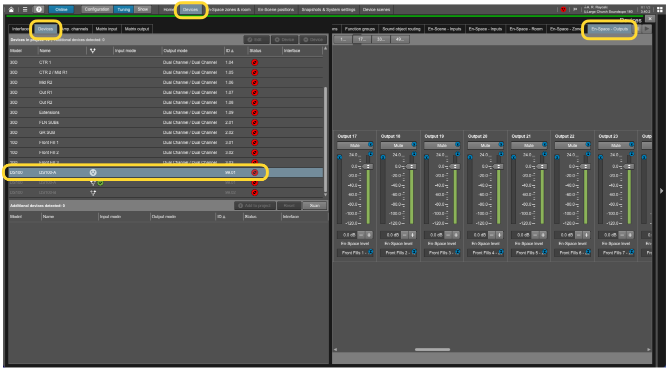
7.6.5. En-Space outputs

By default, En-Space will feed all Function group outputs with the level faders at 0dB. This will result in accurate emulated acoustics and a great starting level without any adjustment required.

Adjusting the En-Space output levels on a per-speaker basis is uncommon and generally left on the default settings. This is because the level of En-Space is already well controlled by the En-Space engine, which takes into account speaker distances and positions.

However, adjustments may be helpful to artistically augment or emphasize En-Space for individual speakers or zones of speakers, without affecting other signals from En-Scene or matrix inputs. For example, you may choose to add extra En-Space level to a broadcast feed or to diminish the En-Space level to subwoofers or front fills.

To adjust the output levels of En-Space for individual speakers or speaker zones, in R1 navigate to the **Devices** page **>** **Devices** tab **>** select **DS100** **>** **En-Space Outputs** tab.



7.6.6. En-Space zones

The En-Space engine has four (4) inputs, or *zones*. The zones are labeled to represent different parts of the venue: (house) *Left*, *Center*, (house) *Right*, and *Audience*. Each zone will apply differing early reflections behavior, which mimics the acoustic behavior of sounds emanating from that part of the venue in the real world.

In other words: a mono signal sent into any one of the four En-Space zone inputs will result in reverberation out of every speaker in your system. The zones are not meant to represent left/center/right signals; they are meant to represent the differing reverberant behavior of an acoustic source on the left side of the stage vs. the reverberant behavior of a source from other parts of the venue. The differences are nuanced and primarily relate to early reflections as they would occur in the *En-Space room* which is being emulated.

The level of the zone faders represents the overall level of En-Space which is set to 0dB by default. While these parameters do not need to be changed often, they can be boosted or reduced to control overall En-Space level without affecting the level of speakers or En-Scene object sends.

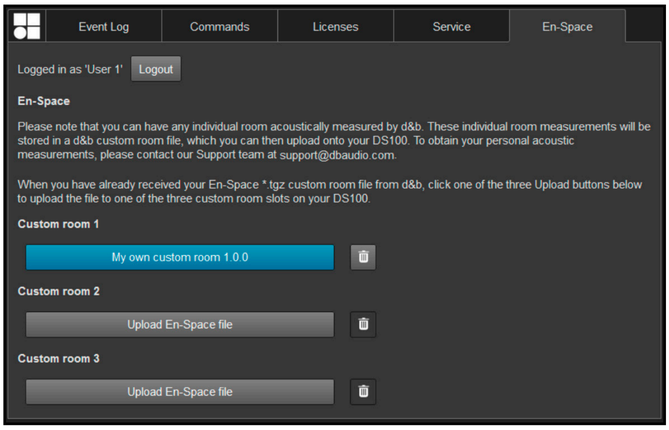
7.6.7. Loading En-Space Custom Rooms

The En-Space tab in the DS100 provides three storage slots for individually recorded room responses created using the proprietary d&b En-Space Custom Rooms measurement service. Only room files provided by d&b can be loaded to a DS100 and reproduced by En-Space.

After onsite capture, a Custom Room will be delivered in the form of a database file (Roomname.tgz) which needs to be loaded into your DS100.

Loading a Custom Room into a DS100/DS100M

1. When connected to a Soundscape processor on the network, enter the IP address of its control port into a web browser.
2. Click on the **En-Space** tab.
3. Log in using the default password < `dbaudio` >.
4. Click the **Upload En-Space file** button on one of three *Custom room* slots to open the file selection dialog.
5. Navigate to your Custom Room file (*.tgz) and select it.



To remove a Custom Room which has been loaded onto a Soundscape processor, simply follow steps 1-3 above then click the trashcan icon next to the slot you wish to free up.

See En-Space Custom Rooms for information on how to obtain a Custom Room.

7.6.8. Routing to En-Space

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

Zone routing by En-Scene (recommended)

For all DS100 inputs configured for En-Scene mode, the zone mixing is performed automatically according to the position of the object, relative to the Early Reflections Plane set in ArrayCalc.

For En-Scene inputs, only set the overall reverb level for the channel; the four zone levels will be automatically controlled by En-Scene positions.

Note: DS100 inputs in En-Scene mode will automatically overwrite and automate any user-adjusted En-Space Zone sends when they are moved. Therefore, don't worry about presetting the zone faders as they will change automatically.

Zone routing at matrix inputs
In R1, on the **Devices** page **>** **Devices** tab **>** **Matrix inputs** tab **>** **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 (1-4).

Each matrix-mode DS100 input will need the master En-Space fader adjusted as well as a single zone send. The active zone should be chosen based on the general location of the source within the venue (house-left, center stage, house-right, or audience).

Example:
- All mics on house-left side of the stage should be sent to *En-Space zone 1 - Left*.
- All mics center-stage should be sent to *En-Space zone 2 - Center*.
- All mics on house-right side of the stage should be sent to *En-Space zone 3 - Right*.
- All mics above, in, or around the audience area should be sent to *En-Space zone 4 - Audience*.

The above example is common with systems that do not use En-Scene such as Virtual Acoustics Shell (VAS) applications.

Note: Identical signals (including left and right of a stereo) should NOT be sent to two different 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 (recommended: Zone 2 "center"). In other words, the zones are not meant to represent left/center/right signals.

Zone routing at the mixing console
Alternatively, the mix to zones can be performed at the mixing console, perhaps using four AUX sends, one for each En-Space zone. The 4 signals should be routed to four DS100 inputs in Matrix mode, each preset to feed one En-Space zone.

Or, use one console aux send to feed a single En-Space zone (#2 - center). For most mix engineers, they are very happy to have a single aux which results in a 144-channel reverb with spatial awareness. The nuanced differences of the En-Space input zones may not be required.

7.7. Tips for customizing R1

7.7.1. Grouping channels

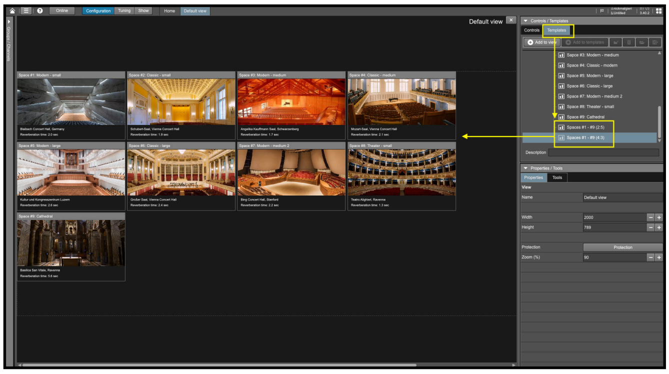
DS100 input and output channels, just like amplifier channels, and objects and their settings can be grouped in R1. The familiar controls such as EQ and level faders can be placed in a remote view and applied to a custom group of DS100 channels.

Example:
- Choir inputs (24-36) can be grouped within a user-created group named "Choir Inputs."
- On a remote view, a level fader can be added which is targeted to control the group "Choir Inputs" with the function: En-Space send level.
- The user now has a single fader which controls the En-Space level for all 12 choir inputs.

Note: Grouping does not link the channels. Instead, it allows for a control to be assigned to the group of channels to apply an offset (relative control) or an override of settings for all channels within the group (non-relative control). The individual channels can still be controlled independently, if desired.

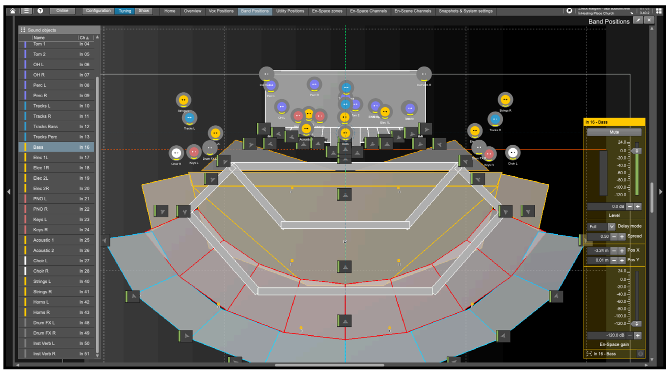
7.7.2. En-Space room graphics

For added visual interest and artistic understanding, it is easy to add photos of En-Space rooms onto a remote view. This makes the use of En-Space much clearer and more exciting compared to a drop-down list alone. While it is easy to create your own graphics, R1 already includes two templates which fulfill most users needs. They can be found when R1 is in **Configuration** mode **>** **any remote view** **>** **Controls/Templates** **>** **Templates** tab **>** **d&b Templates** category **>** **Soundscape** **>** **En-Space** **>** **Visuals**.



7.7.3. Positioning View graphics

The Positioning Views in R1 have options to show objects, speakers, and the outline of a single Positioning Plane. It is often helpful, and exciting for the user, to add graphics to the positioning views to help with understanding object orientation within the venue. You can add multiple graphics (.png, .jpeg, and .jpg) to a remote view, such as a stage plot, seating chart, or even satellite imagery from Google Maps for outdoor events.



Shown here: A screen capture from ArrayCalc which was imported onto a remote view as a picture to add visual clarity to sound object positioning.

💡 **Tip**: To properly scale and position the imported image, make sure all loudspeakers are shown on the positioning view for reference.

7.7.4. Deleting Remote Views

The auto-create feature of R1 will generate a number of Remote Views, including two views per amplifier/speaker zone, a page for Soundscape Outputs, and a page called DS100 Outputs. In some situations, all four of those pages might be considered redundant for day-to-day operation. Feel free to delete Views to simplify the use of the system. All controls are still active, even if they're not presented on a remote view. All the controls can still be found within the *Devices* page, or they can be re-added to remote views later.

7.7.5. Multiple Positioning Views

For a number of reasons, it may be helpful to create multiple positioning views. Here are some:

Examples:

A) Using one Positioning view for moving the groups and additional positioning views for arranging objects within each group.
B) All vocals on one Positioning view and all musicians on another.
C) The opening band on a Positioning view, the headliner on a second Positioning view. (In this case, all the objects are added to both views, but each view represents the objects with differing colors and channel names.)

7.8. Storing and recalling settings

‼️ **A very important note:** Storing your R1 file does not save any of your audio settings by default. This behavior is intentional as it allows a newly opened file to read all settings from the system without overwriting active settings. However, a user may be disappointed when they open a particular file and see old settings without a restore point available.

For this reason, it is recommended to store a System Settings and/or a Snapshot to your .dbpr file to provide a restore point for later. This is particularly true before a firmware update is performed, as all updated devices will have their settings cleared.

Method of storing	Includes all amplifier settings?	Includes all Soundscape mix settings?	Recallable by OSC?	Recalls device network settings?
R1 Snapshots	Depending on user defined scope	Depending on user defined scope	No	No
R1 System Settings	Yes	Yes	No	No
Device Scenes	No	Depending on user defined scope	Yes	Yes
Create.Control snapshots	No	Yes (except Function Group and matrix routing)	No	No
Device backup file	Yes	Yes	No	Yes

Shown here: A table which compares the most basic differences between the methods of storing settings.

7.8.1. Storing Snapshots

The Snapshot option of R1 can store all DS100 and amplifier control settings, so long as they are located on a remote view. Make sure all parameters to be captured in the Snapshot are represented by controls on a Remote view.

Snapshots are useful for quick, limited-scope recall during a show or between acts. Additionally, snapshots can be created offline and recalled later, if desired.

When storing a snapshot, select any remote view(s) which contain controls that you want to store.

To store a Snapshot in R1, navigate to **Snapshots & system Settings** page **>** **Snapshot** tab **>** **+ Snapshot >** **select one or more Remote views >** click **Store**.

7.8.2. Storing System Settings

The System Settings option in R1 is simpler than Snapshots for basic system backup requirements. A stored system setting will store all parameters of all connected network devices (DS100s and d&b amplifiers, except DS10/DS20), not only the controls visible on Remote views. This should be considered the fail-safe option, which is great as a backup of an installed system or a restore point which is required before a firmware update.

The store System settings function is disabled when R1 is not connected to network devices. When storing, a pop-up will appear letting you know which devices are connected for saving System settings. If a device is offline, its System settings cannot be saved.

In R1, navigate to **Snapshots & system Settings** page **>** **System settings** tab **>** **+ System settings >** ensure all devices are connected and checked **>** click **Store**.

Note: It is recommended that all systems have at least one System setting stored. This can be helpful in situations of unexpected equipment failure, firmware updates which clear settings, and other unforeseen situations. Additionally, it can be immensely helpful when sending your .dbpr file to d&b support, so they can see your settings by recalling them remotely.

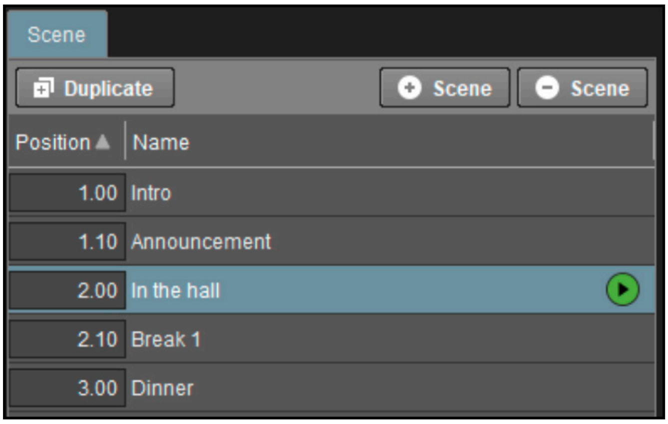
💡 **Tip:** Storing many System settings can drastically increase the size of your .dbpr file, impacting the opening and closing speed of R1. For this reason, it is recommend to only store a limited number of System Settings. Snapshots are more efficient for various day-to-day storing.

7.8.3. Device scenes

Device scenes operate exactly the same as R1 Snapshots except, that they are stored into the DS100 directly. They can then be recalled either by R1 or by a remote device (such as a Stream Deck or QLab) OSC command, with R1 online or not.

The DS100 provides local memory for Device scenes. They are organized numerically in a range from 0.01 to 999.99 and contain a user-selectable set of parameters which may include DS100 inputs/outputs, En-Scene, En-Space, and matrix settings.

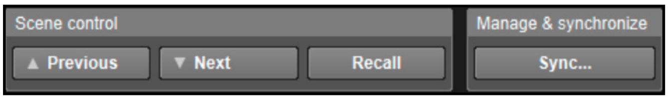
Note: Device scenes cannot store nor recall amplifier parameters.



To create a Scene in R1, navigate to **Home** page **> Device Scenes** page. This is done by choosing the Positioning and/or Remote views that contain the desired DS100 control elements, and assigning a name and a scene number (xxx.xx).

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 Scene 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). 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.

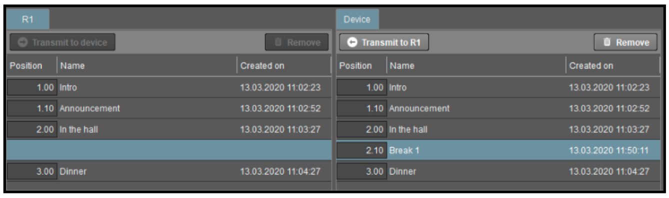


Scenes can be recalled using the **Recall** button or by stepping through the DS100 scene memory using the **Previous** or **Next** buttons. All three functions can also be assigned to switches on a Remote view or triggered by OSC commands. A direct Scene recall by OSC is done on the basis of the scene number. See Third-party control for more details.

While the DS100 is 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** and **Next** commands are not available when offline.

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: A Scene does not include the DS100 channel input modes (En-Scene/Matrix). Scenes only contain parameters of the selected DS100 and no other settings of R1 or other connected devices such as amplifiers.

7.8.4. Create.Control project file

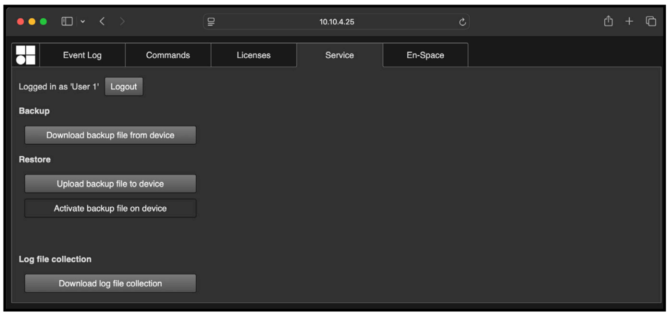
If Create.Control is used to control the spatialization of an event, Snapshots can be used to store the settings and state of the Soundscape processor. The settings which will be stored include all mix settings which are controllable by Create.Control—object positions, object names, object spread, object level, En-Space send level, object Delay Mode, and En-Space master settings.

As Create.Control cannot control amplifiers, none of those settings are stored within its Snapshots.

7.8.5. Device backup file

All d&b amplifiers (except 5D) and all variants of the DS100 have the ability to create a Device Backup File via the web browser. A device backup file will reset all functions of the device, including networking settings like IP address and remote ID.

The backup file can work well as a commissioning backup as it can be loaded into the same device or directly into a replacement unit, in the case of device failure.



Creating a Device backup

1. Enter the control IP address (not Dante) of the device into any web browser.
2. Click on the **Service** tab (see above) and enter the default password < `dbaudio` >.

3. Click **Download backup file from device**
4. Navigate to the location on your computer where you would like to save the Device backup and click save.

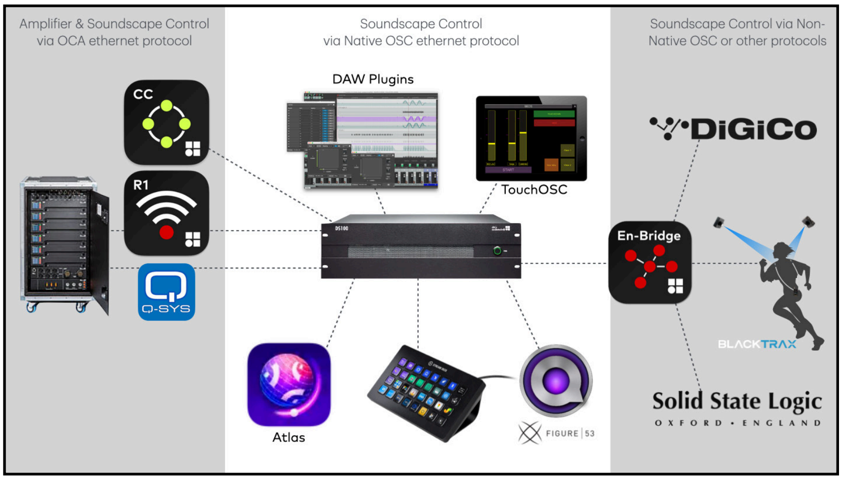
Restoring from a Device backup

1. Follow steps 1 & 2 from above.
2. Click **Upload backup file to device**.
3. Navigate to the Device backup file on your computer and select it.
4. You will be presented a window which will show the network settings as they were saved in the file and as they will be applied to the device.
5. Click **Activate backup file on device** to recall the settings from the backup.

The reason you need to click *Activate backup file on device* after uploading is because the control network settings stored in the backup could differ from the current network settings of the device, which may cause you to lose the active network connection to the device. This way you have an opportunity to notate the necessary information to get you re-connected to the device after the settings go into effect.

8. Third-party control

8.1. Intro to OSC control



Most parameters within the DS100 Signal Engine are controllable not only by R1 but also via OSC messages. For example, level and delay of matrix inputs/outputs, matrix crosspoints, as well as Device Scenes can all be controlled by external devices through a network connection. En-Space presets can be recalled by touch screen devices for easy operation by non-technical operators. Or, anything your imagination can conceive of is theoretically possible with some creative programming.

Any number of third-party OSC-compatible controllers can be used, even at the same time. In the event simultaneous messages are received by the DS100, the last message will always prevail.

While R1 is required during the initial setup of a DS100, it does not need to be connected for ongoing operation nor for OSC control. OSC controllers can typically communicate with the DS100 directly, regardless if R1 or Create.Control are online or not.

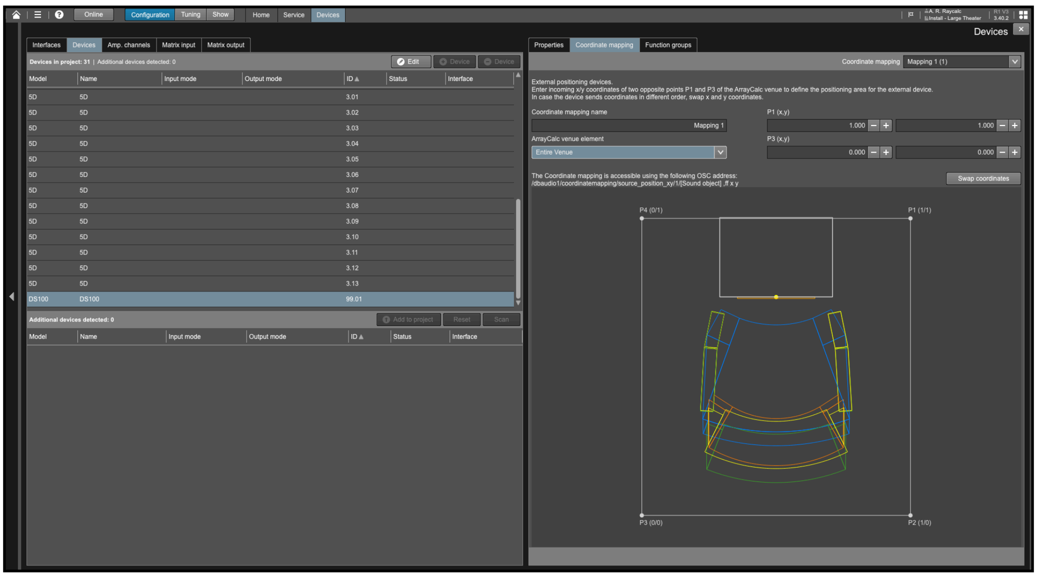
Note: The OSC protocol does not confirm receipt of messages or provide any form of verification. Therefore, sending a large number of OSC messages at the same time could result in some of the packets being missed or dropped. Using Device Scenes as a method to recall many settings via a single OSC command can help avoid this issue. In the case of commands for dynamically moving objects, a constant stream of movement is not as vulnerable to these issues. This is because a single missed packet is indistinguishable among the large number of commands being sent.

8.1.1. Coordinate mapping object positions

En-Scene object positions can be controlled by external devices like mixing consoles, show control systems, DAWs, or tracking systems. 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. Up to four external devices can be scaled differently at the same time through the use of up to four coordinate mapping functions.

The mapping is done when R1 is in Configuration mode within the **Devices** view **>** **Devices** tab **> select DS100 > Coordinate mapping** tab. The OSC message to address sound objects using this mapping is displayed on the tab. In the screenshot below, this DS100 is set to scale all incoming OSC data which is targeted to Coordinate Mapping #1 to a Positioning Plane named “Entire Venue.” This assumes that the Positioning Plane was added to the project file when working in ArrayCalc.

Note: Coordinate mapping is only required if third-party control of object positioning is needed.



Show above: This DS100 is set to scale all incoming OSC positional data to a Positioning Plane entitled “Entire Venue” under the dropdown *ArrayCalc venue element* which encompasses an area of roughly 200% the venue size. The assigned Positioning Plane now corresponds with the quad-panner within a third-party device such as a DiGiCo console or QLab.

8.2. Integration products list

8.2.1. Supported integrations

d&b Software and resources

En-Bridge: Free d&b software for Mac and Windows which works as a network bridge to enable:

- Network control across subnets.
- Per-object enabling/disabling of various third-party controllers.
- Protocol translation for BlackTrax, DiGiCo consoles, and SSL consoles.
- Forwarding messages to multiple DS100 processors for redundancy or output expansion.

En-Space DAW Plugin: a free control plugin which can be automated within a DAW software which enables network control of En-Space emulated room acoustics. Note: requires a My d&b account for download.
En-Scene DAW Plugin: a free control plugin which can be automated within a DAW software which enables network control of object positions and object En-Space send levels. The plugin is available VST, AU, and AAX formats to support all available DAWs. Note: requires a My d&b account for download.
Q-SYS Plugin: A plugin thats runs within the Q-Sys platform to enable network control over DS100 settings. Note: there is also an amplifier Q-Sys plugin which is specifically for controlling d&b amplifier DSP. Both types of plugin can be installed on Q-SYS at the same time.
En-Snap: A list-based object positioning software developed in partnership with Gareth Owen Sound. Not available via public download but can be obtained for free by contacting support@dbaudio.com.

Note: Most users will prefer to use Create.Control once it is released.

DS100 OSC protocol: a text library PDF document which shows all available OSC messages that can be used to program a third-party controller (software or hardware) to enable network control over any DS100 parameter.
ArrayCalc Viewer App: A free mobile app for iOS and Android which allows for viewing of speaker deployment data which is handy when building a system onsite.
Create.Control: Free d&b software for Mac and Windows which allows for the programming and recalling of Soundscape. It is planned for release in the second half of 2025.

Show control software

QLab: Mac software for cueing audio, video, midi, OSC, lighting and other messages. Very common in theater and corporate events.

Note: Figure53, the makers of QLab, are Soundscape owners at their own theater.

Mixing consoles

DiGiCo: All SD-Series and Quantum Series consoles support control of Soundscape objects from within the console surface (via network connection to a DS100). Object positions are recalled with snapshots. No download required, available within current console firmware.
AVID: By installing the d&b AAX format control plugin, any AVID S6-Series desk can control a DS100 via network connection. Object positions are recalled with snapshots.
SSL: All SSL live consoles support control of Soundscape objects from within the console surface (via network connection to a DS100). Object positions are recalled with snapshots. No download required, available within current console firmware.
Lawo: All Lawo live consoles support control of Soundscape objects from within the console surface.

Hardware surfaces

Stream Deck: A range of various hardware controllers which can be programmed to send OSC messages to the DS100 to recall various mix settings or presets using DS100 Device Scenes or to recall various En-Space Rooms, as examples. There are some pre-made templates available for Companion software which are built to control amplifiers and the DS100 - they are not supported by d&b.

Tracking systems

BlackTrax: Infrared-based real-time tracking system for monitoring the position of actors, props and scenery Note: requires En-Bridge to translate network protocols between the tracking system and Soundscape (more info above under "d&b Software")
Zactrack: Radio-based real-time tracking system for monitoring the position of actors, props and scenery
TiMax Tracker D4: Radio-based real-time tracking system for monitoring the position of actors, props and scenery. Note: requires "Tracker Translate" software for Windows to translate network protocols between tracking and Soundscape.
Stagetracker II by TTA: Radio-based real-time tracking system for monitoring the position of actors, props and scenery.

8.2.2. DIY integrations

❗ **Note:** These options are not officially supported by d&b, but feel free to check them out and have fun on your own!

Atlas: A macOS-only standalone software for advanced programming of Soundscape shows, made by a Soundscape user. Very useful for advanced theatrical applications.
TouchOSC & TouchOSC 2: Build custom control pages for any iOS device to control any parameter of a DS100. Great for wireless control of the DS100, stage manager panels, or a limited/focused artist GUI.
OSCar plugin: a clever DAW plugin by ircam that allows for custom OSC strings to be automated within DAW software. Available in VST and AAX formats. Might be useful for automating parameters not supported by the d&b plugin such as Object routing or Function Group Delay.
Unreal Engine: Clever programming can send OSC messages to a DS100 to represent the virtual position of the camera and/or elements within the Unreal model. This allows for the integration of open-world Unreal models to deliver live-rendered spatial audio across multiple types of speaker zones.
MiMU Gloves: Gloves filled with sensors created by Imogen Heap (who is a Soundscape user) that track hand position and gestures, which can then be configured to control a sound object's position or other parameters via OSC.
Grapes: A macOS-only standalone software which provides a bank of programmable buttons used to trigger sound object movements. Good for on-the-fly MIDI or OSC triggered spatilzation of sound objects for DJs and live music performance applications.
Leap Motion Sensor: A hardware sensor for computers which tracks hand position and gestures which can be programmed to send OSC messages to a DS100, enabling gestural controls of object positions or other parameters.
OSC / PAR: An AU/VST3 plugin that allows for musical events to be translated and transmitted via OSC messages. This may be helpful for applications where object parameters should be synced with reproduced musical content in a DAW.
SpaceBox: A simple macOS-only software for programming object movements, designed specifically for Soundscape.
Mrmr: A simple iOS app which allows a mobile device such as iPhone or iPad to become an object controller. Example: iPhone accelerometer used to control an objects position via WIFI.
X-Keys: Manufacture of hardware controllers which can be programmed to transmit OSC messages to a DS100.
PrePosition: A free Max4Live device for Ableton Live which allows for control of sound object position both in the studio and live, with or without a DS100.
Naostage: Camera-based tracking system for real-time tracking of actors and presenters.
FOLLOW-ME: Manual or automated tracking systems.
Lemur: Controller software for iOS and iPadOS devices, allowing wireless control via OSC of DS100 parameters.
Max/MSP: Highly customizable software interfaces and protocol translation.
Grass Valley - Make Pro X: An alliance of solutions manufacturers to gain better integrations within broadcast markets.
NUGEN Audio - Halo Upmix: An AU/VST3/AAX plugin for stereo-to-surround up mix.
Chataigne: A clever Mac app which allows for custom scaling and conversation of many protocols. If you search for "d&b" within the module library, you will find pre-made control modules for d&b amplifiers, and the DS100 (here).

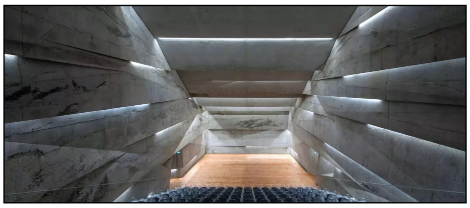
- Example 1: Convert a MIDI signal to OSC messages for sound object positioning and movement in Soundscape.
- Example 2: Take a DMX signal and convert it to an En-Space acoustics level via OSC command.
- Example 3: Take the OSC output from another manufacturer's spatial control software and covert it to OSC strings for the DS100.

Note: Most of the integrations listed above require manual programming of OSC messages. For this reason, you will need the list of OSC strings which are compatible with the DS100, located **here**. This PDF document shows all available OSC messages that can be used to program a third-party software or hardware controllers, enabling network control over any DS100 parameter.

9. Quick references

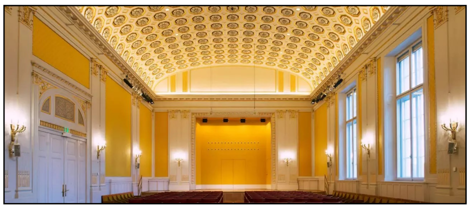
9.1. En-Space venue library

Space #1: Modern - Small



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

Space #2: Classic - Small

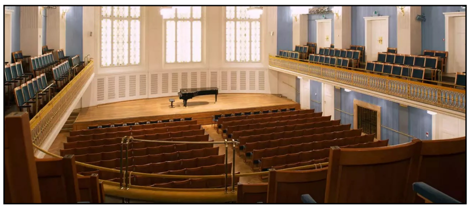


Schubert Saal,
Vienna Concert Hall
Capacity: 350 seats
Reverberation time: 1.9s
(T40: 200 Hz - 2 kHz)

Space #3: Modern - Medium

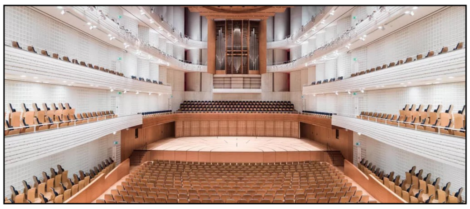
Angelika-Kauffmann
-Saal, Schwarzenberg
Capacity: 600 seats
Reverberation time: 1.7s
(T40: 200 Hz - 2 kHz)

Space #4: Classic - Medium



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

Space #5: Modern - Large



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

Space #6: Classic - Large

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

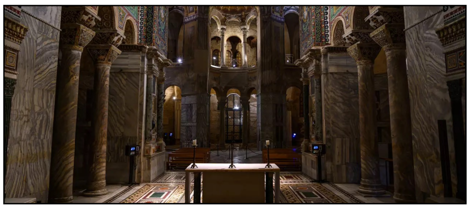
Space #7: Modern - Medium 2

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

Space #8: Theater - Small

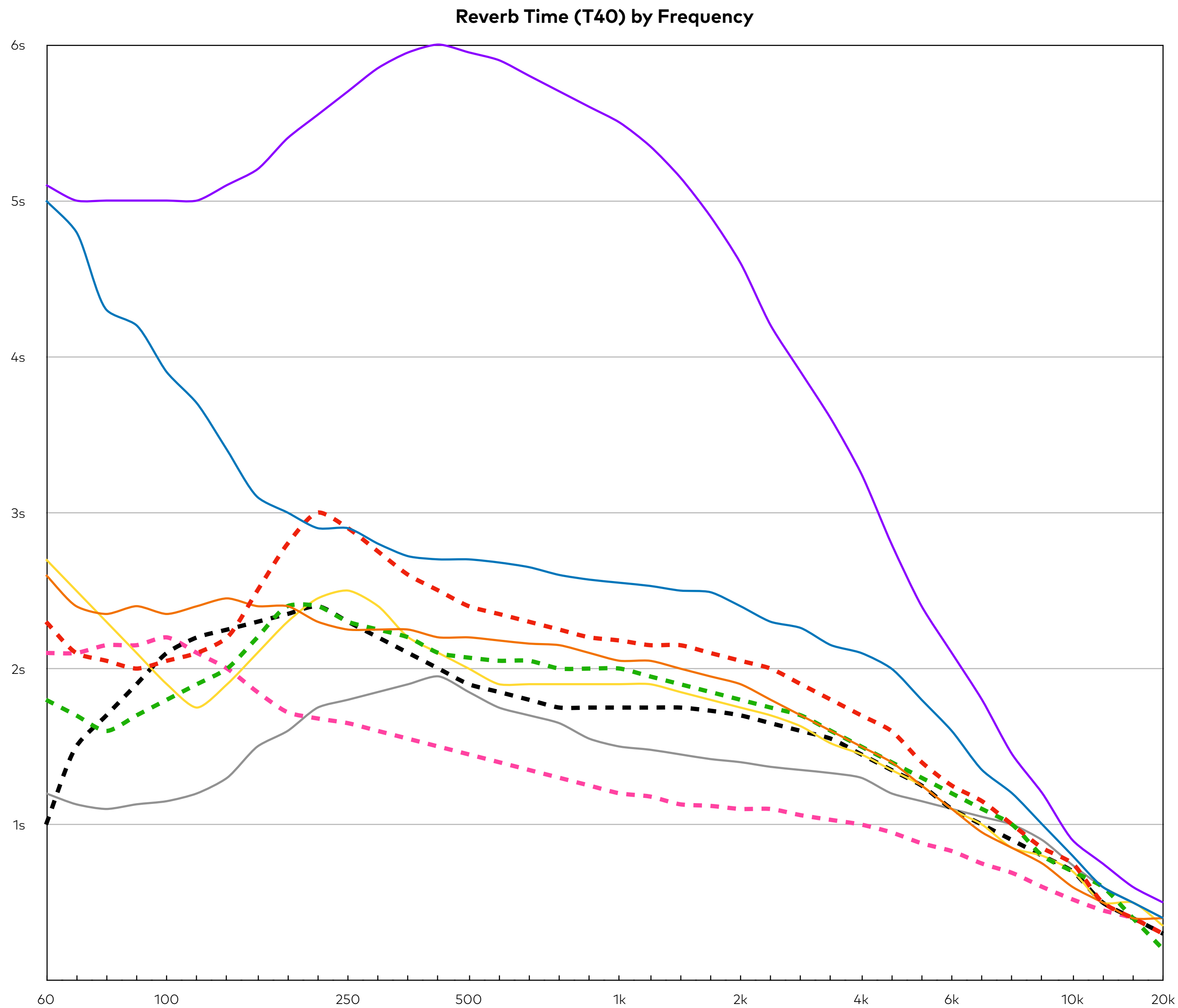
Teatro Alighieri,
Ravenna
Capacity: 830 seats
Reverberation time: 1.3s
(T40: 200 Hz - 2 kHz)

Space #9: Cathedral



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

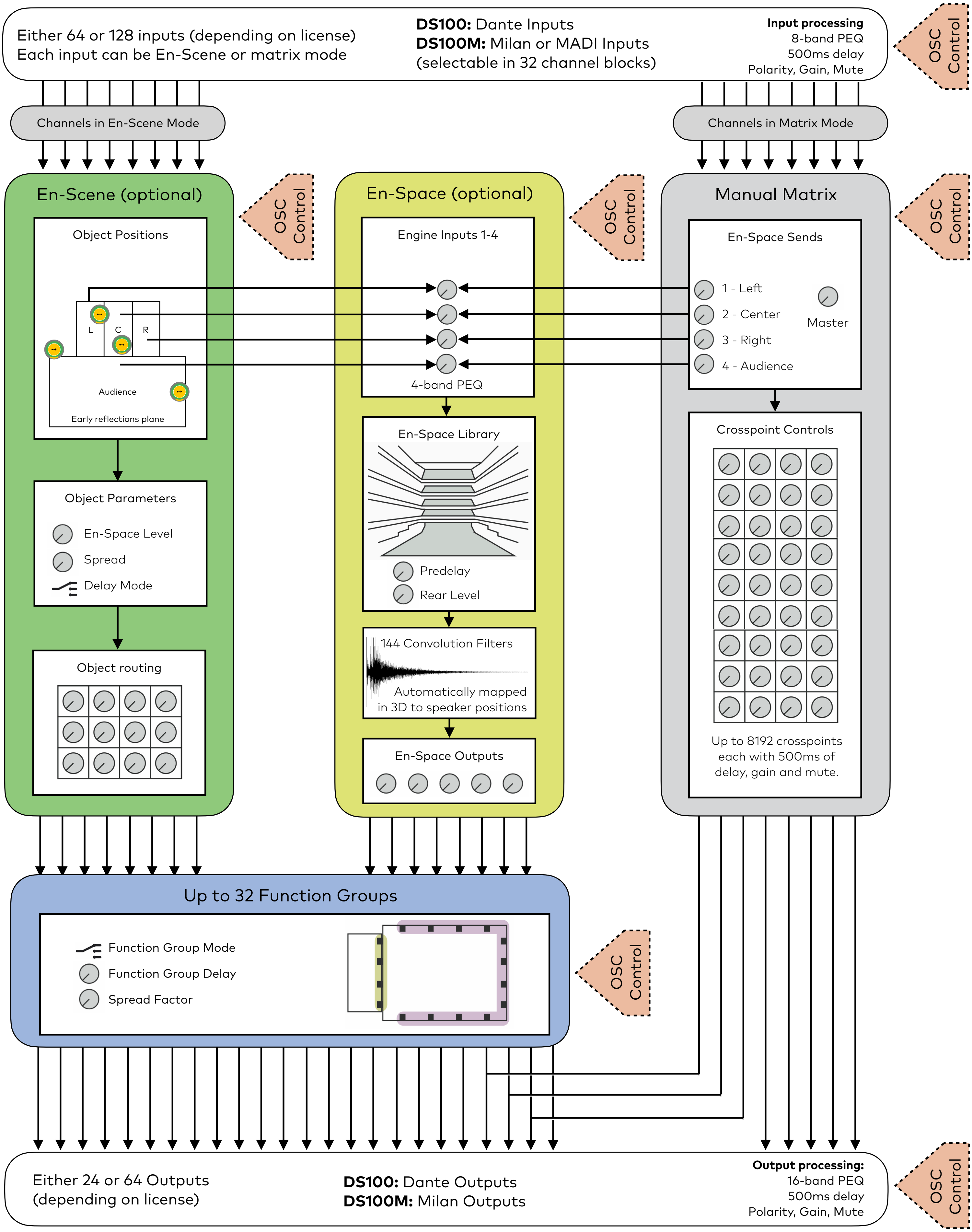
9.2. En-Space venues compared



This graph visualizes the amount of time a sound will reverberate, depending on the frequency.

- "Cathedral" | San Vitale Cathedral, Ravenna, Italy
- "Modern - Large" | Great Hall, Vienna, Austria
- "Classic - Large" | Großer Hall, Vienna, Austria
- "Modern Medium 2" | Bing Concert Hall, Stanford, USA
- "Classic - Medium" | Mozart Hall, Vienna, Austria
- "Modern - Small" | Blaibach Hall, Blaibach, Germany
- "Classic - Small" | Schubert Hall, Vienna, Austria
- "Modern - Medium" | Angelika Kauffmann Hall, Austria
- "Theater" | Alighieri Theater, Ravenna, Italy

9.3. DS100 block diagram

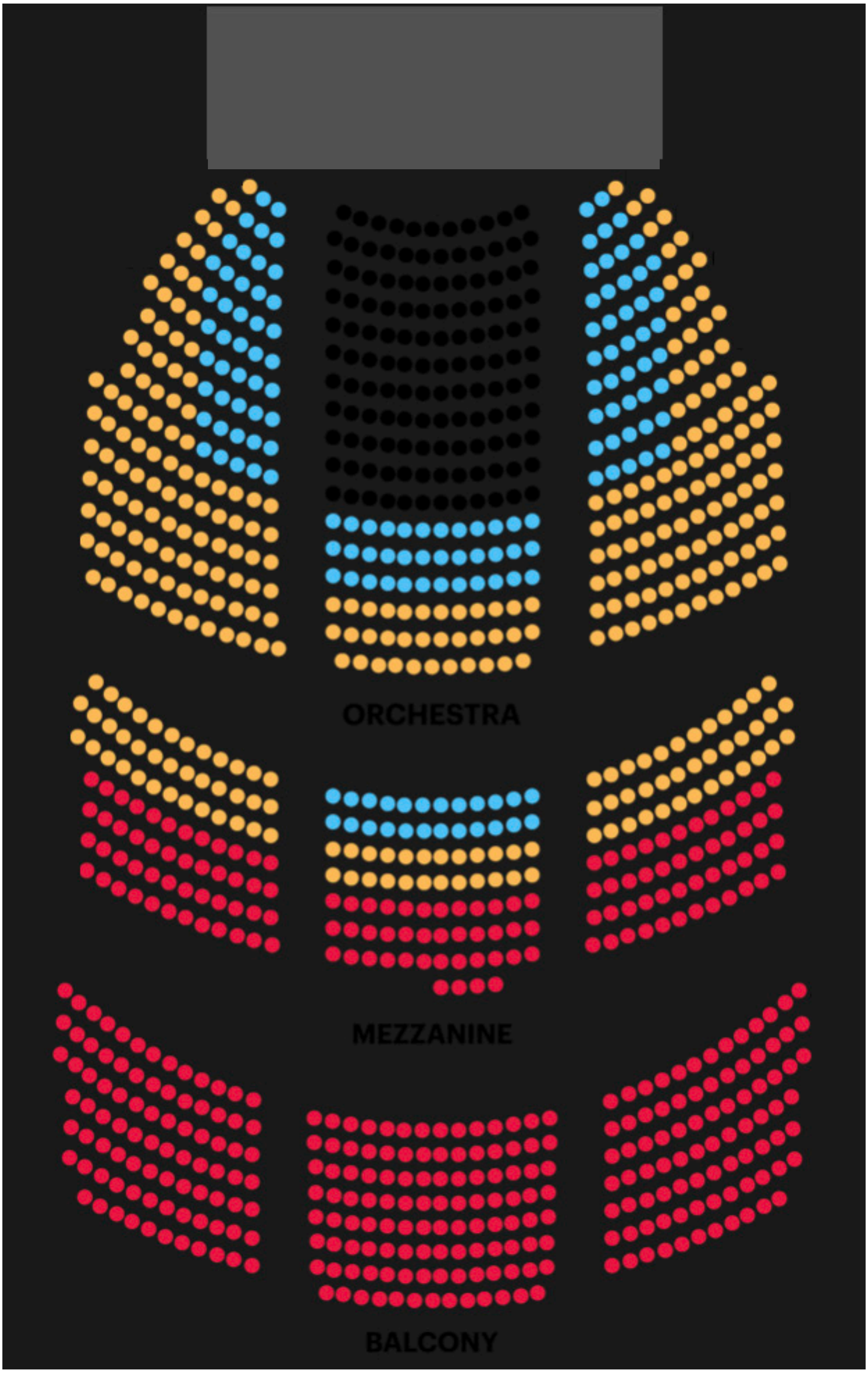


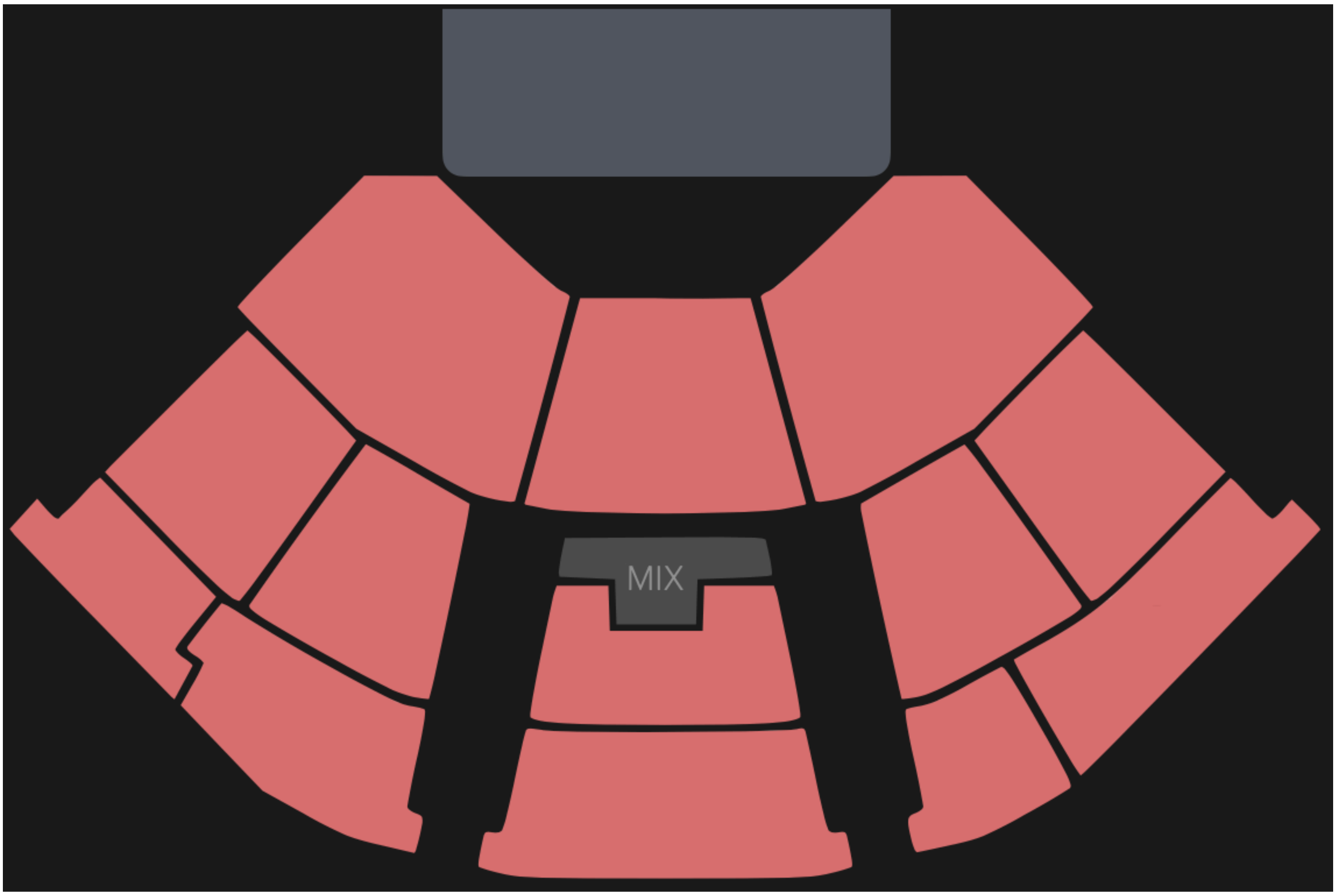
9.4. Function Groups overview table

	Mode —>>	Main	Surround	Frontfill	Delay Line	Delay Line Embedded	SUBs group	SUB array	Outfill	Outfill Embedded	Mono Out	Ceiling	Unassigned processor outputs
Matrix	**Can DS100 inputs in Matrix mode be manually routed and delayed using**	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓
Matrix	**Can DS100 inputs in En-Scene mode be manually routed using matrix**	No. Object position and object spread dictates level distribution. All matrix crosspoint controls between En-Scene inputs and function group outputs are disabled. 'Object Routing' feature is available for this purpose and can be adjusted per object and per function group.											✓
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 routing from matrix inputs.	Not applicable - does not support En-Scene. Matrix crosspoints must be manually configured for desired level and delay processing. Any DS100 input (En-Scene mode or Matrix mode) can be manually routed to these outputs without limitation.
En-Scene	**Supports dynamic time alignment per Sound Object?**	✓	✓	✓	✓	✓	✓	✓	✓	✓	No		
En-Scene	**'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	✓		
En-Scene	**Yields object to other function groups?**	✓	✓	✓	No		No	No	No		No		
En-Scene	**Requires FG delay for objects in Full mode?**	No. Time alignment is handled automatically in both X and Y axes. Objects in Full mode ignore 'Function Group Delay' parameter.									✓		
En-Scene	**Requires FG delay for objects in Tight mode?**	✓	✓	✓	✓	✓	✓	✓	No. Automatically time-aligned to first FG with the mode "Mains"		✓		
En-Scene	**Requires FG delay for objects in Off mode?**	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓		
En-Scene	**Supports 'Sound Object Routing' customized mixes?**	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓		
En-Space	**Number of impulse response files available.**	7 x 4 zones	40 x 2 zones	9 x 2 zones	7 x 2 zones	7 x 2 zones	7 x 2 zones	1 x 2 zones	1 x 4 zones	1 x 4 zones	Not supported	7 x 2 zones	Does not support En-Space.
En-Space	**Supports En-Space for 'Matrix' mode DS100 input?**	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	✓	

9.5. Example backgrounds for use in R1

Screenshot one of the attached images to be used as the background of a positioning view in R1. This may be helpful when understanding how Soundscape works, with some additional visual aid, even if the system is not deployed in a venue at the time (demo room, small practice system, etc.). The image background color matches the background color of a positioning view. On Mac, use Command+Shift+4 and use the crosshairs to select the area around the image to create a screenshot. For Windows, use the Snipping Tool. Adding a Soundscape logo to your remote views guarantees that the system sounds its best.



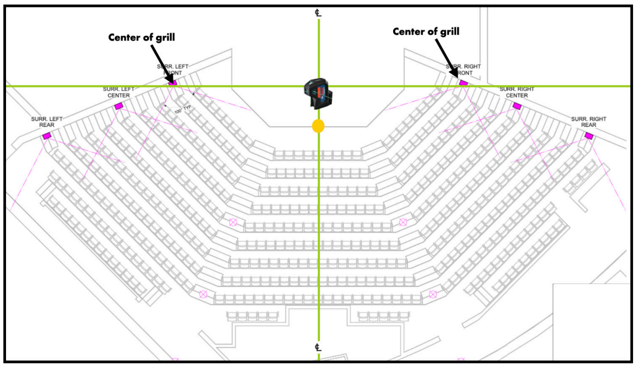

MIX

9.6. Commissioning checklist

This resource can be handy for first-time Soundscape providers such as rental/hire companies and integrators. For more information, clarification, or guidance, please email support@dbaudio.com.

ArrayCalc prediction software (required for programming Soundscape!)

- **Venue tab:**
  - If En-Scene is used: Ensure that the venue file has at least one Positioning Plane. The Positioning Plane should encapsulate all areas where a sound object will be placed. Up to four positioning planes can be added although only one is required.
  - If En-Space is used: Ensure that the file has an Early Reflections Plane. This plane should generally match the stage location.
- **Sources tab:**
  - Because it is common for speakers to be installed slightly askew from their intended position due to rigging and structural constraints, verify all speaker positions to ensure the software's representation of their position matches the real-world deployment. X/Y position & horizontal angle is all that matters for En-Scene. Z axis is only important when En-Space is used. It is also recommended that all speakers be listed in left-to-right order within the file which is backwards from the ArrayCalc default.
  - Observe where the global point of origin is within the ArrayCalc file. You may consider changing it for easy measurements. You will be using a laser to measure to that position so make sure the position is something you can point a laser to, perhaps a road box or music stand.



Green laser used to pinpoint the X and Y axis of surround speakers

  - In many venues, it may be difficult to exactly measure the X and Y axes because of curved seating rows, varying speaker heights, and other optical illusions. This is when a laser-square is helpful.
  - Place the laser-square so that the main laser axis (front to back: X axis) points towards the global point of origin and the second axis (left-to-right: Y axis) symmetrically points at a loudspeaker position.
  - Now that you have an exact X/Y measurement position indicated by the location of the laser-square, use a laser Disto to measure both X/Y axis of the speaker position. You should strive to be accurate within 0.3m (1ft) for mid sized venues. Measurements are to be done in reference to the center of the speaker grill for point source speakers and to the center/top of the rigging frame for arrays.
  - Update each speaker position in ArrayCalc so the DS100 will have the correct positional data when the file is migrated to R1.
- **Alignment tab:**
  - Ensure that no delay has been added to any speakers within the amp channels (except a SUB array). Yes, there are sometimes reasons why you may use amp delay but this is less common.
- **Devices tab:**
  - Ensure that the patching between DS100/DS100M, DS10/DS20, and amps is completed and correct. The file must say "Ready for R1" before you can continue to migrate this file into R1. All amp inputs should be set as discrete as every speaker position requires dedicated processing from the DS100/DS100M.
  - Specify which DS100 inputs will be used as sound objects (En-Scene mode) or used to manually matrix signals in a traditional manner (Matrix Mode). If no guidance is available, switch the first 48 channels to En-Scene mode and leave the rest as matrix mode. This can also be changed later when online with R1.
  - Export Dante Preset file from the Devices tab in ArrayCalc. This is a good opportunity to adjust ArrayCalc's automatic naming structure of channels to your personal preferences as these labels will transfer into Dante Controller.

Dante Controller (can be downloaded here. This section assumes a DS100 and not a DS100M with Milan)

- Open Dante Controller and click the *Launch Dante Updater* icon from the top menu.
- Update the firmware on all Dante devices which need it, including DS10s and the Dante card in all DS100s.
- Load the previously exported Dante preset file into Dante Controller. This will label your devices, label channels, and patch crosspoints.
- This may also be a good time to label all remaining Dante channels for the ease of use for end-users. For example, the DS100 shows 128 inputs and outputs regardless of the installed I/O license. You may consider labeling the superfluous inputs/outputs as "not used."
  - ✴ **Note:** It is recommended to minimize the latency of all Dante devices as much as possible. This will help the Soundscape delay calculations to best match acoustic sources on stage. Generally, latency times below 1ms are perfectly stable.
  - ✴ **Note:** Milan systems utilizing a DS100M have a similar process. A Milan Manager preset file can be exported from ArrayCalc. Milan Manager can be downloaded here.

R1 control software

- Open R1 and go *Online* from the Service tab, verify R1 is connected to all devices and update the firmware on all amplifiers and DS100s that need it.
  - ✳ **Note:** DS10s cannot be controlled by R1, only Dante Controller.
- Open the .dbpr file.
  - AutoCreate will automatically prompt you to create groups and remote views. Click "Yes" If the file has already been through the AutoCreate process this step can be skipped.
- Go *Online* and recall the ArrayCalc snapshot from the Overview remote view.
  - ✳ **Note:** The Delay mode for all sound objects will be set to *Tight* by default. However, it is recommended to switch the Delay mode for all sound objects from *Tight* to *Full*.
- Run *System Check* and *ArrayVerification* on all speakers and arrays.
- Generate signal from the DS100 to test the entire signal path from DS100, through network bridges, to amplifiers, and finally speakers. This is done on the *System Check and Array verification* page on the Matrix tab.
- Make noise, do show... have fun!

Important:

- Once commissioning is completed, save a System settings to backup all audio settings, then save the R1 file. This is required because R1 does not automatically save all your settings. By default, saving in R1 only saves the layout of your controls and remote views. Therefore, saving a System settings provides a restore point for you audio settings in case of user-error, device failure, or firmware update.

Notes on acoustic measurement & advanced tuning

- If En-Scene is used, tuning criteria can only be observed with an En-Scene input which also applies Soundscape processing. Therefore, send pink noise into an En-Scene input, place the sound object in a representative position (ie: center/mid-stage). Now you can tune each Function Group to taste (not individual loudspeakers).
- Make sure all loudspeakers have the same SPL and voicing to a relative listener area. This will be needed for smooth 'panning' of sound objects. By default, most d&b loudspeakers are gain-matched and voiced similarly so this should be fast and easy. Put your measurement mic (or ears) in the center of the audience area. Capture SPL and frequency response measurements of the sound object in every representative position (center stage, side, rear, etc.). Adjust level, CPL, and EQ for all speaker zones so that all the Function Groups have similar behavior. Generally, adjusting individual loudspeakers is not recommended.
  - Note: because there will always be multiple speakers working to amplify a sound object, multiple arrival times will be observed by a measurement mic. This makes it nearly impossible to use a transfer-function measurement. For this reason, an RTA may be preferred.
- If ArrayProcessing is used on some sources such as main arrays, you may consider adding 5.9ms of delay to be entered into the amp DSP (6.2ms total on all amp channels which matches the inherent latency of ArrayProcessing). However, keep in mind: the goal of Soundscape is to make all speaker time-align to the performer's position on stage, not to the other loudspeakers. For this reason, you may prefer to keep the ArrayProcessed mains delayed a little more than the rest of the speakers.

- Time-Alignment measurements:
  - If all objects are set to *Full* Delay mode, speaker timing should not be required except to compensate for differing speaker heights (if desired). In this case, delay could be added to the amp DSP. In the majority of cases when Full is used, measuring and adding delay is not required.
  - If objects are set to Delay Mode *Off* or *Tight*, Function Group delay may be needed to time-align Function Groups to each other. This delay parameter is entered by going to the **Devices** page **>** **Devices** tab **> select the DS100 > select the Function Groups** tab **> Function Group delay**. This parameter is only applied to sound objects in Tight and Off modes, but is ignored by objects in Full mode.
  - Full mode is strongly recommended to avoid these considerations and compromises.

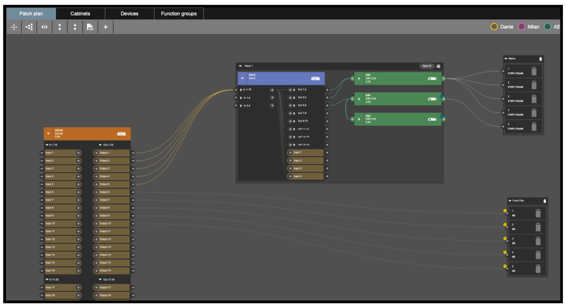
Optional: Third-party OSC positioning control (QLab, TouchOSC, Console control, tracking system, etc.):

- Coordinate mapping must be enabled for one or more positioning planes. This is done while R1 is in Configuration mode, **Devices** page **>** **Devices** tab **> select the DS100 > coordinate mapping tab** (on the right). At least one coordinate map should be assigned a corresponding Positioning Plane which will inform the processor how to scale incoming positioning messages. This step can be skipped if external control is not used although it is still recommended in case a user wants to implement third-party control at a later time.
  - ✳ **Note:** Coordinate mapping planes are added to the project as positioning planes within ArrayCalc. If no positioning planes are available for selection within R1, the .dbpr file must be opened in ArrayCalc, the plane(s) added, and the file re-saved. Only then can the file be re-opened in R1 and the process can be completed.

9.7. FAQ

Q: Can Soundscape be used with speakers not made by d&b?
A: Yes. However, the spatial algorithms are highly dependent on loudspeakers which provide spectrally consistent and symmetrical off-axis dispersion characteristics, such as d&b loudspeakers. Biaxial speakers, which are more common than d&b designs, will generally result in localization errors due to their asymmetrical off-axis response.

The required software workflow of ArrayCalc and R1 only supports d&b loudspeakers. For this reason, it is required to create a system design in ArrayCalc using d&b speakers which reflects the positions and orientations of the real loudspeakers as closely as possible. For this purpose, it may be helpful to create "Draft Patch Lines" in ArrayCalc (DS100 patched directly to loudspeakers without an amp in between). This provides the DS100 with the required speaker positioning information without the software requiring a d&b amplifier which may not be used with non-d&b loudspeakers.



Shown here: 5x main speakers wired through an amp rack containing 3x D80s, as well as 5x front fills, with draft patches directly to the speaker cabinets.

Q: Can speakers be paired on an amp channel when using Soundscape?
A: Yes, speakers can be paired on a single amp channel, although it is not recommended as it results in diminished spatial resolution. When speakers are paired, the DS100 adapts its algorithms to treat the speakers like a single source, located in an average position of all the actual sources. Line arrays do not count—they are considered a single speaker position.

A: Yes, a DS100 output can be sent to multiple speaker positions. However, only one of the speakers can be set to a Function Group (this will dictate the virtual speaker position within the DS100 for the En-Scene and En-Space algorithms). The same signal can then be patched to any number of additional speakers that are not associated with a Function Group (simply don't select a Function Group for these additional speaker groups in ArrayCalc—leave the drop-down list blank). This approach may be helpful to replicate a surround speaker deployment on multiple levels of a venue or to patch individually to upper-and-lower mains speakers without needing separate Function Groups.

For En-Space Custom Rooms, can d&b measure any venue?
The owner of the venue will have to provide permission to d&b. The customer will coordinate access to the venue with our clever German engineers. d&b does not negotiate access and permissions to third-party venues on behalf of a customer.

Extremely large venues with extremely long reverb times may present challenges. This would include venues like arenas or stadiums. But, we'd like to chat about the possibilities!

What can you say about time-alignment? What about comb filtering?
These are common questions from system designers, and we have a couple of thoughts:

Soundscape works by intentionally 'breaking' the delay times between speaker positions in a way that more accurately portrays your event. Once this concept is fully understood, a completely different approach to system alignment is better appreciated. No two speakers should actually be aligned with each other. This is true in the horizontal axis as well as the vertical plane. Generally speaking, speakers which are mounted lower (i.e., front fills) should arrive at the listener before the flown system to improve the vertical localization for the audience.

Comb filtering is natural and unavoidable in the real-world. Traditionally, we strive to avoid comb filtering with standard speaker systems where every speaker amplifies the same signal. In those traditional applications, the comb filtering is extremely damaging as its happening in a way that is totally unrelated to the actual show, just as a byproduct of the speaker system. However, with Soundscape, the comb filtering now matches the positioning of our event parts. Each performer and individual sound is now behaving closer to how sounds behave in the real-world (including comb filtering). The result is a system which does not exhibit any audible effects of comb filtering.

Can En-Space emulate more than one acoustic environment at the same time?
No, the processor's ability to run 144 convolution filters in less than 1.3ms is already a small miracle. Running 288 is not possible. However, you can use your mixer's FX or outboard FX and route the output channels to sound objects. This allows for the use of your FX engines and En-Space at the same time. This now allows you to have the short snare reverb located near the actual snare, while your vocal delays might be placed to the sides or rear of the venue, all while having the backup vocals in an En-Space hall.

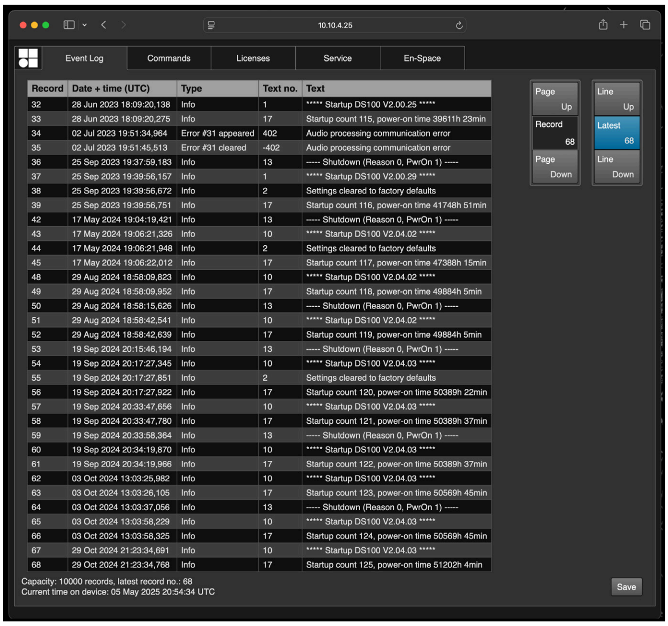
Can Soundscape work with Dolby ATMOS?
Yes! The Soundscape processor can receive any type of audio signal. Running Surround or ATMOS content is a great way to enable cinema content while also supporting real-time audio for live events.

Note: The Soundscape processor does <u>not</u> provide ATMOS decoding.

10. Troubleshooting

10.1. Event log

Enter the DS100 IP address into any web browser to access the event log. This will be helpful to diagnose failures or unexpected behaviors.



Record	Date + time (UTC)	Type	Text no.	Text
32	28 Jun 2023 18:09:20,138	Info	1	***** Startup DS100 V2.00.25 *****
33	28 Jun 2023 18:09:20,275	Info	17	Startup count 115, power-on time 39611h 23min
34	02 Jul 2023 19:51:34,964	Error #31 appeared	402	Audio processing communication error
35	02 Jul 2023 19:51:45,513	Error #31 cleared	-402	Audio processing communication error
36	25 Sep 2023 19:37:59,183	Info	13	----- Shutdown (Reason 0, PwrOn 1) -----
37	25 Sep 2023 19:39:56,157	Info	1	***** Startup DS100 V2.00.29 *****
38	25 Sep 2023 19:39:56,672	Info	2	Settings cleared to factory defaults
39	25 Sep 2023 19:39:56,751	Info	17	Startup count 116, power-on time 41748h 51min
42	17 May 2024 19:04:19,421	Info	13	----- Shutdown (Reason 0, PwrOn 1) -----
43	17 May 2024 19:06:21,326	Info	10	***** Startup DS100 V2.04.02 *****
44	17 May 2024 19:06:21,948	Info	2	Settings cleared to factory defaults
45	17 May 2024 19:06:22,012	Info	17	Startup count 117, power-on time 47388h 15min
48	29 Aug 2024 18:58:09,823	Info	10	***** Startup DS100 V2.04.02 *****
49	29 Aug 2024 18:58:09,952	Info	17	Startup count 118, power-on time 49884h 5min
50	29 Aug 2024 18:58:15,626	Info	13	----- Shutdown (Reason 0, PwrOn 1) -----
51	29 Aug 2024 18:58:42,541	Info	10	***** Startup DS100 V2.04.02 *****
52	29 Aug 2024 18:58:42,639	Info	17	Startup count 119, power-on time 49884h 5min
53	19 Sep 2024 20:15:46,194	Info	13	----- Shutdown (Reason 0, PwrOn 1) -----
54	19 Sep 2024 20:17:27,345	Info	10	***** Startup DS100 V2.04.03 *****
55	19 Sep 2024 20:17:27,851	Info	2	Settings cleared to factory defaults
56	19 Sep 2024 20:17:27,922	Info	17	Startup count 120, power-on time 50389h 22min
57	19 Sep 2024 20:33:47,656	Info	10	***** Startup DS100 V2.04.03 *****
58	19 Sep 2024 20:33:47,780	Info	17	Startup count 121, power-on time 50389h 37min
59	19 Sep 2024 20:33:58,364	Info	13	----- Shutdown (Reason 0, PwrOn 1) -----
60	19 Sep 2024 20:34:19,870	Info	10	***** Startup DS100 V2.04.03 *****
61	19 Sep 2024 20:34:19,966	Info	17	Startup count 122, power-on time 50389h 37min
62	03 Oct 2024 13:03:25,982	Info	10	***** Startup DS100 V2.04.03 *****
63	03 Oct 2024 13:03:26,105	Info	17	Startup count 123, power-on time 50569h 45min
64	03 Oct 2024 13:03:37,056	Info	13	----- Shutdown (Reason 0, PwrOn 1) -----
65	03 Oct 2024 13:03:58,229	Info	10	***** Startup DS100 V2.04.03 *****
66	03 Oct 2024 13:03:58,325	Info	17	Startup count 124, power-on time 50569h 45min
67	29 Oct 2024 21:23:34,691	Info	10	***** Startup DS100 V2.04.03 *****
68	29 Oct 2024 21:23:34,768	Info	17	Startup count 125, power-on time 51202h 4min

Capacity: 10000 records, latest record no.: 68
Current time on device: 05 May 2025 20:54:34 UTC

Tip: All d&b amps (except the 5D) also have a built-in event log.

10.2. DS100 not connecting to R1

In the majority of cases, failure of the device to connect to R1 is the result of incorrect networking configuration. It is recommended to double-check all network settings or to reset the processor to its default network settings.

However, this can also be caused if the processor is not completing its boot sequence. In this case, there is generally a series of tones which can be heard from the motherboard of the device. The number of tones indicates the type of error.

Note: The tones are fairly quiet and high-pitched, so they may be easy to miss in noisy environments, even with your ear up against the chassis.

If you experience these motherboard tones, please email support@dbaudio.com immediately.

10.3. Device reset

The front panel button is a status indicator and has various functions, depending on the number of times it is pressed within short sequence. This includes variations of full reset or networking configuration reset among other choices.

Device status	Sequence	Result
Power down	Press power button 2x within 4 seconds	Normal shut down
Network to default settings	Press power button 4x within 4 seconds	• IPmode: DHCP+FB (Fallback) • Fallback IPaddress: 192.168.1.100 • IPmask: 255.255.255.0 • RemID: Set to 0.01
Network to DHCP+LL settings	Press power button 5x within 4 seconds	DHCP+LL (LinkLocal) for better integration with Dante network IP schemes
Reset device name and audio path	Press power button 7x within 4 seconds	• Device name is reset to factory default • All inputs are reverted to Matrix mode • All input, matrix nodes and output • processing is reset • Device Scenes and network settings unaffected

For a full list of all functions and more details, download the DS100 or DS100M hardware manual here.

11. Document change log

v2.00 - June 2025 - Nick Malgieri and Andrew Rahman
- Converted from TI501 and reformatted to entire document to consider future products
- Added Create.Control information
- Added DS100 I/O sizes information
- Added "Applications of Soundscape"
- Added "Components of Soundscape"
- Added more Custom Rooms details
- Added "Commissioning Checklist"
- Added Third-party integrations list
- Added "Helpful Links"
- Added screenshots for use in R1
- Added more details to the Function Group Reference Table
- Added mentions of Milan-equipped amplifiers
- Added support of 32 Function Groups
- Added mention of ACv12 patch Plan
- Added En-Space reverberation time graph
- Integrated information from TI503 - DS100 Device Redundancy

v1.12 - Dec 2024 - Frank Bothe and Nick Malgieri
- Reworked to include all new features and to reduce complexity

General information

The Soundscape Guide Book - TI501
Previously: *TI 501: d&b Soundscape System design and operation*

Version: 2.00 en, 06/2025

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d&b audiotechnik GmbH & Co. KG
Eugen-Adolff-Str. 134
D-71522 Backnang
Germany

T +49-7191-9669-0,
F +49-7191-95 00 00
www.dbaudio.com

Written in Asheville, NC