FUNKTION-ONE®

VERO® Comprehensive User Guide

Loudspeakers, rigging, cabling,
networked amplification & control, prediction software

Version 0.96 29th April 2018

Contents

1 Introduction

Thank you for purchasing a Funktion One Vero system. All Funktion One loudspeaker systems are designed and built in England. Our design philosophy is to achieve outstanding sonic accuracy and efficiency through innovative acoustical design. This approach provides immediacy and involvement reminiscent of the finest musical instruments.

Reading this user guide will help you to achieve the best performance - just as Funktion One intended.

2 Receiving, unpacking & handling

Check the system components and quantities against your order to make sure your shipment is complete and in good condition before signing for the delivery.

2.1 Vero component descriptions and identification

Vero main array components

There are three types of Vero array loudspeakers that, together, form a comprehensive vertical array

- The Vero ***V60*** loudspeaker is a mid/high system that provides very narrow vertical coverage and 60° horizontal coverage
- The Vero ***V90*** loudspeaker is a mid/high system that provides narrow vertical coverage and 90° horizontal coverage
- The Vero ***V315*** loudspeaker is a mid-bass system that forms part of a Vero array

Note that each V60, V90 and V315 loudspeaker includes an integral Lambda® rigging system.

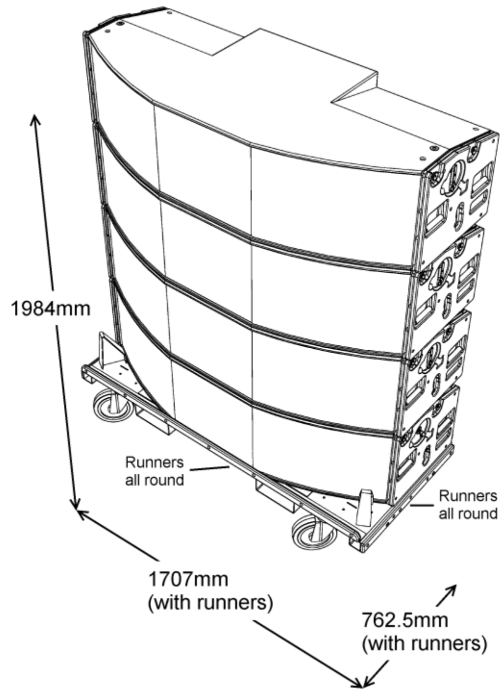
Vero ground-stack bass components

V60/V90/V315 systems may be augmented by:

- The ***V221*** dual 21” ground-stack bass system with ***V132*** low bass extension
  or…
- The ***V124*** single 24” ground-stack bass system

Vero stackable transport dolly *(See **Section 7.5** for dolly-mounted ground or stage-wing Vero stacking and **Section 9.3** for truck packing)*

- ***V-Dolly*** – The 4-cabinet V60/ V90/ V315 dolly with high performance 8" wheels and fork-lift provision
- Optional protective side and end ***Runners*** are available – see illustration to right ⇨
- ***Outriggers*** *are available for tilting dolly-mounted Vero stacks or for levelling during rigging operations (see **Section 7.5** for details)*
- *4-cabinet padded **Dolly Covers** are available. These feature index dial windows, handle flaps and a squaring lid to allow additional items to be transported on top*



4-cabinet V-dolly showing overall dimensions
(1677mm (w) x 732.5mm (d) without runners)

Vero amplifier rack

The ***V-RACK9U*** amplifier rack includes:

- 3-phase European AC mains panel providing 3 x 230v phase-to-neutral supplies – one per amplifier
  Optional USA AC mains panel providing three 208v phase-to-phase supplies – one per amplifier
- Up to 3 x Lab.gruppen PLM+ series 4-channel power amplifiers – type ***PLM20K44***
- Local *(managed, non-blocking, gigabit)* CISCO SG300-20 Ethernet switch for Dante networking

Each Lab.gruppen PLM+ Series ***PLM20K44*** power amplifier drives up to:

- 4 x V60 (or V90) + 2 x V315 top + mid-bass systems or…
- 6 x V221 bass cabinets* *(2.67 ohms per channel)* or 8 x V221 bass cabinets *(2 ohms per channel)*
  **Assumes two lots of 3 x parallel V221s driven in 2-channel (separate-driver) mode*

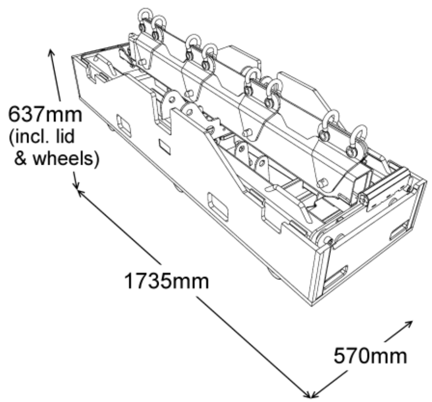
V60/ V90/ V315 loudspeaker cables

- ***VC8EX25M*** - 25m 8-core extension cable
- ***VC8EX10M*** - 10m 8-core extension cable
- ***VC8BB02M*** - 2m Bass break-out cable
- ***VC8LK1.5M*** - 1.5m link cable
- ***VC8LK02M*** - 2m link cable
- ***VC8LN01M*** – 500mm NL8-to-quarter-turn mil. spec. converter cable

V-FLYTRUNK1 rigging trunk – containing:

Lambda® flying system

- ***VBAR-MAIN*** – Main Vero 2-part loudspeaker system FlyGrid set *(for up to 24 enclosures)*
- Optional ***VBAR-S*** - Simple Vero loudspeaker system FlyGrid set *(for up to 6 out-fill enclosures)*
- Optional ***Outriggers*** *(for levelling the 4-cabinet dolly during rigging operations or for tilting Vero stacks)*
- ***V-YS1*** - Vero loudspeaker system rear pull-up system
  (Including 2 x Swing clips, plus spreader bar used when flying more than twelve Vero)
- 1-tonne, 3-metre lever hoist



FlyTrunk with Lambda® FlyGrid system

2.2 Unpacking and inspecting

All Funktion One products are tested and inspected thoroughly before being despatched.

- Inspect your shipment for any signs of abuse or transit damage as soon as you receive it
- If your shipment is incomplete or any of its contents are found to be damaged:
    - Inform the shipping company
    - Inform your dealer

We suggest that you keep some of the original packaging in case you have to return a unit for repair or replacement.

Funktion One Research Limited and its distributors cannot be held liable for product damaged through the use of non-approved packaging, shipping or handling methods.

2.3 Standard 24-box Vero touring system components supplied

Note that ***Projection*** software will generate "Kit lists". Click through ***Docs > Truck List*** to see a full list of components required for any other system configuration.

8 each	V60, V90, V315	Vero System Enclosure
16	V221	Vero System Bass Enclosure (or optional 4 x V132 + 12 x V221 Bass system)
4	VC8EX25M	Vero Cable - 25m extension
4	VC8EX10M	Vero Cable - 10m extension
4	VC8BB02M	Vero Cable – Mid-bass breakout
8	VC8LK1.5M	Vero Cable - Short Link
4	VC8LK02M	Vero Cable - Long Link
4	VC8NL01M	Vero Cable - NL8 to System Connector
6	V-DOLLY	Vero Transportation Dolly
2	V-RACK9U	Vero Amp Rack
2	Mains Distro	Vero Amp Rack Distro
6	PLM20K44	Lab.gruppen Amplifier PLM+
2	V-FLYTRUNK1	Vero FlyGrid Trunk
2	VBAR-MAIN	Vero FlyGrid
2	V-YS1	Tilt Strap Assembly
2	Lever Hoist	1-Tonne, 3-metre lever hoist for Tilt Strap

2.4 Handling

Please handle your Vero system safely to avoid injury.

- Palletised, wrapped and strapped shipments should only be moved using a fork-lift truck driven by a qualified forklift truck driver
- Vero systems are supplied with dollies making them very easy to move around.

We recommend providing staff with suitable manual handling training before they manoeuvre a Vero system. Helpful hints are available free of charge from the UK Health and Safety Executive. See *(**www.hse.gov.uk/pubns/indg143.pdf**)*.

If your Vero system is destined for an installation, keep it protected by its packaging until the installation site is clean, secure and ready to accept it.

3 Introduction & features

3.1 Vero® V60, V90, V315, V221, V124 and V132 systems

Vero V60, V90 and V315 loudspeaker systems

The Vero large-format touring loudspeaker system is in a class of its own. Its combination of unparalleled sonic performance and intelligent design provides easily deployed audio of the highest standard for arenas, concerts or festivals of any size.

The Vero system has all the physical attributes demanded by today's top-level productions with next-generation sound. Geometric Energy Summation *(GES)* enables natural tailoring of the coverage pattern and sound pressure levels to keep sound focused on the audience and minimise off site environmental impact. Vero's effortless power and clarity can often eliminate the need for inconvenient and costly delay positions where propagation conditions allow.

Vero's patented Lambda® rigging system offers rapid installation and features the unusual and useful capability of enclosure angle adjustment under load.

Vero is a complete system package with amplification, cabling, rigging, transportation dollies, racks and trunks - all optimised for practical and efficient touring.

Vero's industry-leading electro-acoustic efficiency gives more head-room for less power consumption. The system's natural tonal response and Funktion One's unique "out of the box" sound image projection is achieved with minimal frequency correction to maintain linearity and headroom. Vero gives artists, engineers and concert goers the unprecedented dynamic range, fidelity and satisfaction they deserve.

Vero V60 mid/high features

Electro-acoustic

- Advanced Andrews-Newsham tuning
- 200Hz – 18kHz frequency response
- 60^{o} horizontal coverage
- Up to 6^{o} vertical splay angle between V60s for contiguous coverage
  (large mechanical angles are available should deliberate mid and HF coverage gaps be required)
- Funktion One designed drivers with Neodymium magnets
  - 2 x 250W, 10" 16Ω mid drivers
  - 3 x 75W, 1.4" 32Ω HF drivers
- 2 x 8-pin ¼-turn Mil-spec Syntax connectors in parallel
  (A - D link through, 2 x parallel mids on E+/F-, 3 x parallel HFs on G+/H-)

Enclosure & fittings

- 15mm thick birch ply enclosure
- High durability Polyurea coating
- Integral Lambda® rigging system with colour-coded Index dial ***(blue)***
- Inter-cabinet locating cones for easy rigging
- Generous flip handles

Vero's family's common cabinet styling

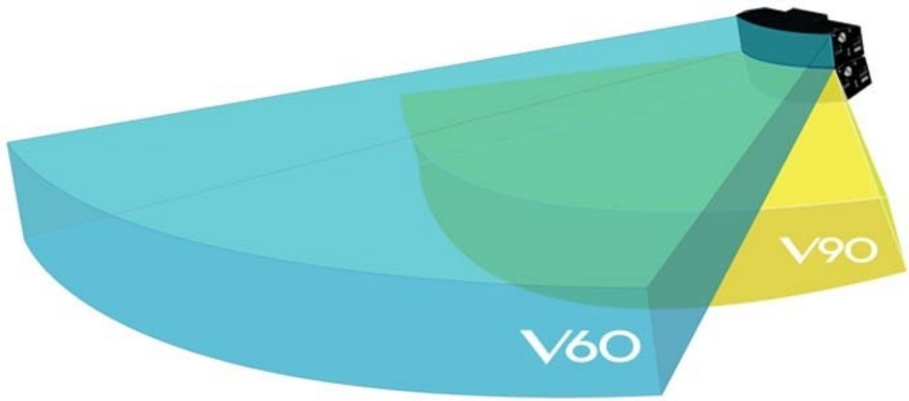
Prediction software

- ***Projection*** system design and rigging software

Vero V90 mid/high features

Electro-acoustic

- Advanced Andrews-Newsham tuning
- 200Hz – 18kHz frequency response
- 90^{o} horizontal coverage
- Up to 12.5^{o} vertical splay angle between V90s for contiguous coverage
- Funktion One designed drivers with Neodymium magnets
  - 2 x 250W, 10" 16Ω mid drivers
  - 2 x 75W, 1.4" 32Ω HF drivers
- 2 x 8-pin ¼-turn Mil-spec Syntax connectors in parallel.
  (A - D link through, 2 x parallel mids on E+/F-, 2 x parallel HFs on G+/H-)



V60's 60^{o} horizontal dispersion vs V90's 90^{o} horizontal dispersion

Enclosure & fittings

- 15mm thick birch ply enclosure
- High durability Polyurea coating
- Integral Lambda® rigging system with colour-coded Index dial ***(yellow)***
- Inter-cabinet locating cones for easy rigging
- Generous flip handles

Prediction software

- ***Projection*** system design and rigging software

Vero V315 mid-bass features

Electro-acoustic

- Advanced Andrews-Newsham tuning
- 50Hz – 250Hz frequency response
- 70^{o} horizontal coverage at 200Hz
- Up to 5^{o} vertical splay angle between V315 and adjacent enclosure
- Funktion One designed drivers with Neodymium magnets
- 3 x 400W, 15" 8Ω low mid/ bass drivers
- Single NL-4, 3 x parallel drivers on 1+/1-

Enclosure & fittings

- 15mm thick birch ply enclosure
- High durability Polyurea coating
- Integral Lambda® rigging system with colour-coded Index dial ***(red)***
- Inter-cabinet locating cones for easy rigging
- Generous flip handles

Prediction software

- ***Projection*** system design and rigging software

V315 Mid-bass

Vero V221, V124 and V132 bass systems

Vero V221 dual 21” bass system

In the 1980’s Tony Andrews began pioneering work on the development of the first 21” loudspeaker ever made. Continuous development has resulted in the high-intensity V221 bass system which benefits from two of these incredibly robust Funktion One designed 21” drivers with 6” voice coils for maximum power handling and efficiency.

- Advanced Andrews-Newsham tuning
- 40Hz-250Hz Frequency response
- Funktion One designed drivers with Neodymium magnets
- 2 x 21” drivers with 2 x 750W, 8Ω voice coils. *(4Ω when connected in parallel)*
- 107dB - 1W at 1m in half-space
- 2 x NL-4 *(pin-to-pin)*, one driver on 1+/1-, the other on 2+/2-
- 18mm thick birch ply enclosure
- High durability Polyurea coating
- Integral pocket handles
- Feet that locate into recesses when stacked - to prevent sliding or swivelling
- Rear wheels that nest inside flare of adjacent enclosure for efficient truck packing
- Optional throat grilles

Vero V124 single 24” bass system

Built to the same high power and efficiency standards and with the same practical features as the V221, the Vero V124 single 24”, dual voice coil bass system may also be used with Vero V60/V90/V315 systems. The V124 provides extended and defined low frequency bass performance in an efficient 992mm (w) x 702mm (h) x 1186mm (d) package.

- Advanced Andrews-Newsham tuning
- 35Hz-160Hz Frequency response
- Funktion One designed driver with FEA-optimised Neodymium magnets
- Single 24” drivers with 2 x 750W, 8Ω dual voice coils. (4Ω when connected in parallel)
- 107dB - 1W at 1m in half-space
- 2 x NL-4 (pin-to-pin), one voice coil on 1+/1-, the other on 2+/2-

Vero V132 single 32” low-bass system

Vero V132 single 32” 24Hz-70Hz low-bass systems may be used to extend a Vero systems low frequency response down to 24Hz for the ultimate in low frequency performance, high-power contemporary music applications and special effects. See ***Section 6.8*** for further details.

4 Connector pin-outs and cables

4.1 Vero connector pin-outs

VC8LN01M 1m NL8 *(rack end)* to Syntax converter cable then to…
10m/25m ***VC8EX10M/25M*** Syntax male to female *(cabinet end)* extension cable

VC8LN01M NL8 to female Syntax system link cable		**VC8EX10M or VC8EX25M male to female main extension cable**		**V60/90 cabinet wiring**		**Function**
NL8	**Syntax SPK socket**	**Syntax SPK pin**	**Syntax SPK socket**	**Syntax SPK male**	**Driver**	
1 +	**A**	**A**	**A**	**A**	**+**	***Through*** - to V315 #1 (via ***VC8BB02M*** mid-bass break-out cable - ***see below***)
1 -	**B**	**B**	**B**	**B**	**-**	
2 +	**C**	**C**	**C**	**C**	**+**	***Through*** - to V315 #2 (via ***VC8BB02M*** mid-bass break-out cable - ***see below***)
2 -	**D**	**D**	**D**	**D**	**-**	
3 +	**E**	**E**	**E**	**E**	**+**	2 x Mid per cab (up to 4 x V60/V90 via ***VC8LK1.5M/02M*** Syntax male-female links)
3 -	**F**	**F**	**F**	**F**	**-**	
4 +	**G**	**G**	**G**	**G**	**+**	2 (V90) or 3 (V60) x HF per cab (up to 4 x V60/V90 via ***VC8LK1.5M/02M*** Syntax male-female links)
4 -	**H**	**H**	**H**	**H**	**-**	

VC8BB02M – 2m Syntax male to 2 x NL4 mid-bass break-out cable for up to 2 x V315 mid-bass per block of 4 x V60/90

VC8BB02M Syntax SPK male to 2 x NL4 breakout			**2 x V315 cabinet wiring**			**Function**
Syntax SPK connector pin **A**	NL4 #1	**1 +**	NL4 #1	**1 +**	3 x V315 #1 drivers **+**	V315 #1
Syntax SPK connector pin **B**	NL4 #1	**1 -**	NL4 #1	**1 -**	3 x V315 #1 drivers **-**	
Syntax SPK connector pin **C**	NL4 #2	**1 +**	NL4 #2	**1 +**	3 x V315 #2 drivers **+**	V315 #2
Syntax SPK connector pin **D**	NL4 #2	**1 -**	NL4 #2	**1 -**	3 x V315 #2 drivers **-**	

Standard NL8 *(rack end)* to 2 x NL4 *(V221 end)* bass break-out cable for 6 or 8 x V221 bass *(6 shown in table)*

Standard NL8 to 2 x NL4 breakout			**3 x V221 cabinet wiring** *(paralleled via NL4 link cables)*		**Function**
NL8 **1 +**	NL4 #1	**1 +**	V221 #1, 2 & 3 NL4 socket **1 +**	V221 #1, 2 & 3 driver 1 **+**	V221 #1, 2 & 3 *(paralleled)*
NL8 **1 -**	NL4 #1	**1 -**	V221 #1, 2 & 3 NL4 socket **1 -**	V221 #1, 2 & 3 driver 1 **-**	
NL8 **2 +**	NL4 #1	**2 +**	V221 #1, 2 & 3 NL4 socket **2 +**	V221 #1, 2 & 3 driver 2 **+**	
NL8 **2 -**	NL4 #1	**2 -**	V221 #1, 2 & 3 NL4 socket **2 -**	V221 #1, 2 & 3 driver 2 **-**	
NL8 **3 +**	NL4 #2	**1 +**	V221 #4, 5 & 6 NL4 socket **1 +**	V221 #4, 5 & 6 driver 1 **+**	V221 #4, 5 & 6 *(paralleled)*
NL8 **3 -**	NL4 #2	**1 -**	V221 #4, 5 & 6 NL4 socket **1 -**	V221 #4, 5 & 6 driver 1 **-**	
NL8 **4 +**	NL4 #2	**2 +**	V221 #4, 5 & 6 NL4 socket **2 +**	V221 #4, 5 & 6 driver 2 **+**	
NL8 **4 -**	NL4 #2	**2 -**	V221 #4, 5 & 6 NL4 socket **2 -**	V221 #4, 5 & 6 driver 2 **-**	

8-pin Syntax SPK male cable connector (e.g. extension cable - rack end)

Speakon NL4 cable connector

Speakon NL8 cable connector

4.2 Recommended cable specifications for permanent installations

Vero systems are usually supplied complete with all loudspeaker cabling ***– see previous page for part numbers***. However, where permanent installation requires cabling to be routed through the building structure via intermediate junction boxes and local 8-pin Syntax SPK tails, the following cable recommendations may prove useful.

A good example of installation speaker cable is:

- Install grade loudspeaker cable LSZH
  (For installations in public buildings, clubs and cruise ships where Low Smoke Zero Halogen cables are specified. These cables are compliant with IEC60092, IEC60332.1, IEC60332.3C, IEC60754.1, IEC60754.2 and IEC60134.2)

Low loss cable runs

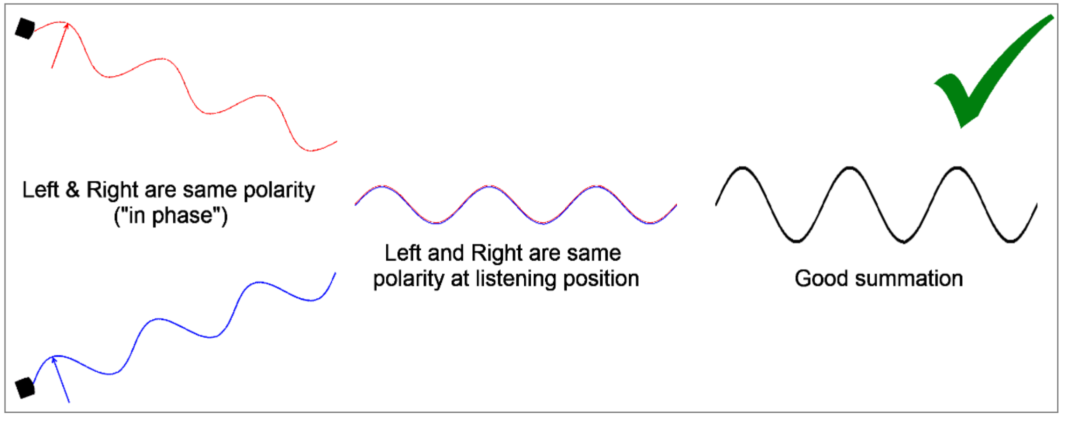
The following table gives the maximum cable lengths allowable to keep level losses below 0.6dB.

Copper core cross sectional area mm^2	Nearest suitable AWG size	8-ohm load metres *(ft)*	4-ohm load metres *(ft)*	2-ohm load metres *(ft)*
2.50 mm^2	13awg	36m *(118ft)*	18m *(59ft)*	9m *(29ft)*
4.00 mm^2	11awg	60m *(197ft)*	30m *(98ft)*	15m *(49ft)*

4.3 Loudspeaker polarity

Loudspeaker cones and diaphragms create sound by moving in and out to modulate air pressure and velocity. It is important that all loudspeakers in a sound system move in the same direction when driven in unison. In other words, they should all have the same polarity.

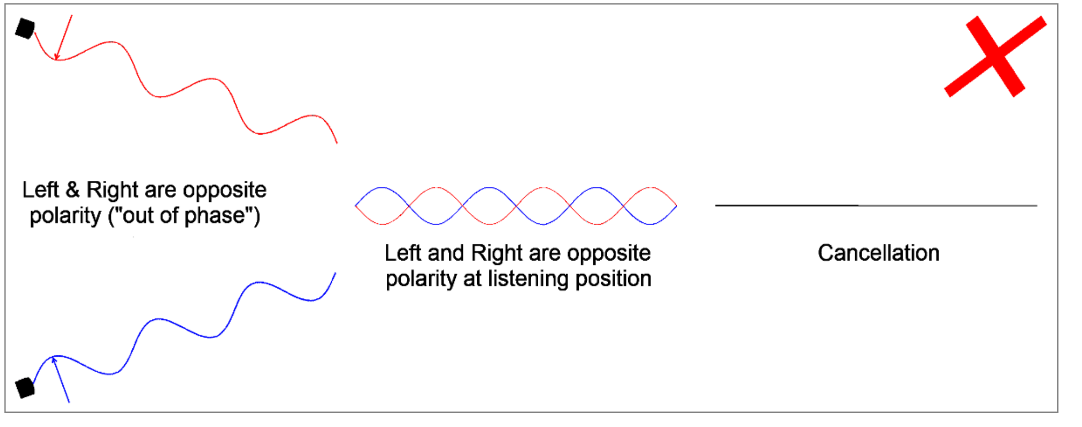
Same polarity *("in phase")*



Loudspeakers with the same polarity will give:

- Good bass extension
- Good bass impact
- Good "between the speakers" stereo imaging with a smooth central transition
- Solid centre-panned vocal imaging

Opposite polarity *("out of phase")*



Loudspeakers with opposing polarities will give the following, depending on which section *(bass, low-mid, high-mid or high)* is incorrectly set up:

- Poor bass extension
- Poor bass impact
- Poor "between the speakers" imaging - just hard left or hard right "distant" effects
- Vague and unstable vocal image. Centre-panned vocals will sound indistinct or thin and, again, the "distant" effect will be evident.

It is usually quite easy to ensure that your loudspeakers are wired to the same polarity if all your loudspeakers are being driven via identical signal paths and equipment. Simply check that the same loudspeaker cable core colours are wired to the same + or – pins at the amplifier end and at the loudspeaker connector end.

If you cannot be certain that your signal paths are identical *(you may be using a mix of old and new equipment or equipment from different manufacturers, for instance),* check out the system using a polarity checker.

A good polarity checker, such as the Funktion One ASPC1, will allow you to check not only your amplifier-loudspeaker combinations but also the entire signal path including mixers and crossover connections.

Introducing the Funktion One Audio Systems Polarity Checker *(ASPC1)*

Pulse generator

- XLR pulse generator output
- LED pulse indicator

Detector

- Built in microphone
- LED polarity indicators

General

- Belt pouch included
- Batteries included
- User Instruction Card

The ASPC1 is a two-part system comprising a pulse generator and a detector. The pulse generator allows you to inject a special test signal anywhere in the signal path whilst the detector microphone lets you check the resulting loudspeaker polarity acoustically. See the ASPC1 user guide for more information.

Absolute polarity and live sound

Maintaining absolute polarity through a sound system means not only making sure all your loudspeakers are "in phase" but also ensuring that your microphones give a positive-going output voltage for a positive-going pressure, that there are no polarity reversals anywhere in the signal path and that your loudspeaker systems provide a positive pressure in response to that positive-going signal.

Many natural sound sources, especially percussion and vocals, produce asymmetrical waveforms. It makes sense to ensure that a positive-going percussive impact creates a positive-going pressure for the audience to maintain that all-important bodily impact. Some recording engineers are also adamant that vocals overlay the band more clearly if absolute polarity is maintained.

Common sense, then, suggests that, if we're using polarity-matched microphones and well-documented signal paths, why not maintain absolute polarity – if only for consistency?

5 Amplification and control

5.1 Electrical and fire safety

Qualified and experienced system technicians only

Funktion One Vero series loudspeakers are high power systems and should only be powered from professionally assembled, electrically safety-tested amplifier racks designed and installed by fully experienced system technicians.

The following information is intended to be used by fully qualified personnel only.

Cabling

- Use the Funktion One loudspeaker cables supplied with your Vero system wherever possible
- Ensure that your cables are in good condition and free from damage
- Ensure that there are no loose conductor strands that could short out and create a fire hazard
- Keep connectors away from flammable furnishings as connectors can get hot under fault conditions

If you need to make up bespoke cables for an installation:

- Use the appropriate cable conductor gauge for the loudspeaker being driven. See ***Section 4.2***.
- Use fire retardant or low emission cables where these have been specified by the contract or by local safety regulations
- Ensure that your cables are properly assembled and free from damage
- Ensure that there are no loose conductor strands that could short out and create a fire hazard
- Again, keep connectors away from flammable furnishings as connectors can get hot under fault conditions

Amplification

Although your Vero system is supplied complete with fully assembled amplifier rack systems, the following points may still be relevant and should be adhered to:

- Remember that heavily clipped programme signals can double a power amplifier's output. See ***Appendix D***.
- Keep amplifier racks properly ventilated and well away from flammable furnishings

5.2 Power amplifier considerations

PLM20K44 power output and limiter settings

The PLM20K44 power amplifiers and preset libraries supplied with your Vero system fully comply with the AES power ratings of your Vero system drivers – assuming they are used as suggested in the user guide.

AES power ratings
Modern loudspeaker drivers are specified to an AES standard. This specifies their long-term *(typically 2 hour)* band-limited AES power rating.

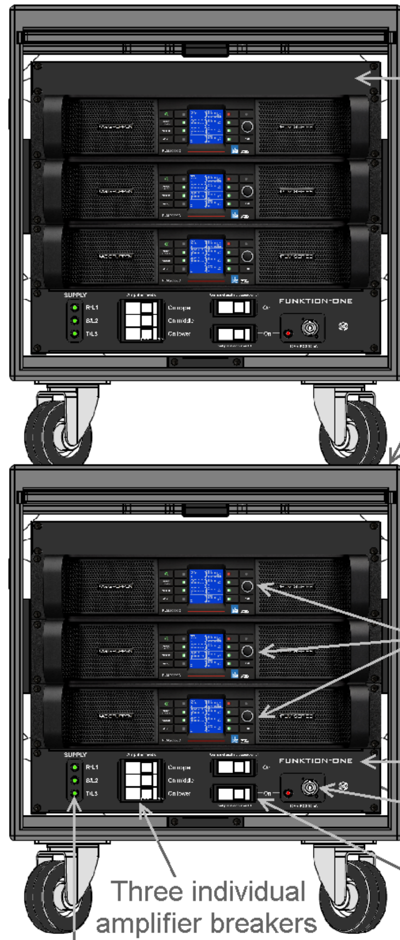
Peak power ratings
The AES band-limited test signal has a 6dB crest factor. This means that loudspeaker drivers can be specified with a peak power rating of 4 times their AES power rating - assuming suitable rms limiting is in place to protect the voice-coils from overheating on sustained signals.

5.3 Funktion One amplifier rack

Vero's amplifier rack is equipped for the challenges and rigours of the touring environment.

The amplifiers and power distribution unit are fully shock mounted within the rack.

It also features convenient retractable protective doors that store in the roof of the rack when not in use. Despite the extra shock mounting and the door stowage features, the Vero rack has a small footprint and may be flown where necessary.



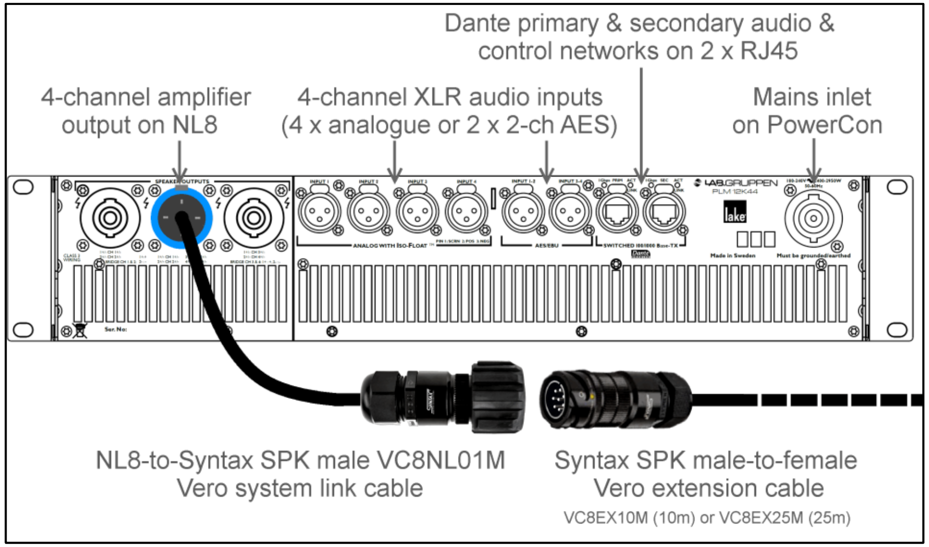
Each rack includes:

- A tour-grade suspended 19" rack system with sliding doors
- A 3-phase mains distribution panel – configured to either 230v phase-to-neutral (typical in Europe) or 208v phase-to-phase (typical in the USA), as required
- A 1-U rack space for user allocation *(fitted with blank panel)*
- 3 x Lab.gruppen PLM20K44 4-channel power amplifiers that include Dante network audio and Lake control facilities
- Two NL8-to-female Syntax SPK system link cables

A standard 3 x PLM20K44 amplifier rack will provide all the power and control to drive up to:

- 8 x V60/V90 mid/high loudspeakers
- 4 x V315 mid-bass loudspeakers
  Plus…
- 6 x V221 bass cabinets* *(2.67 ohms per channel)* or 8 x V221 bass cabinets** *(2 ohms per channel)*
  **Assumes two lots of 3 x parallel V221s driven in 2-channel (separate-driver) mode*
  ***Assumes two lots of 4 x parallel V221s driven in 2-channel (separate-driver) mode*
  (See illustration later)

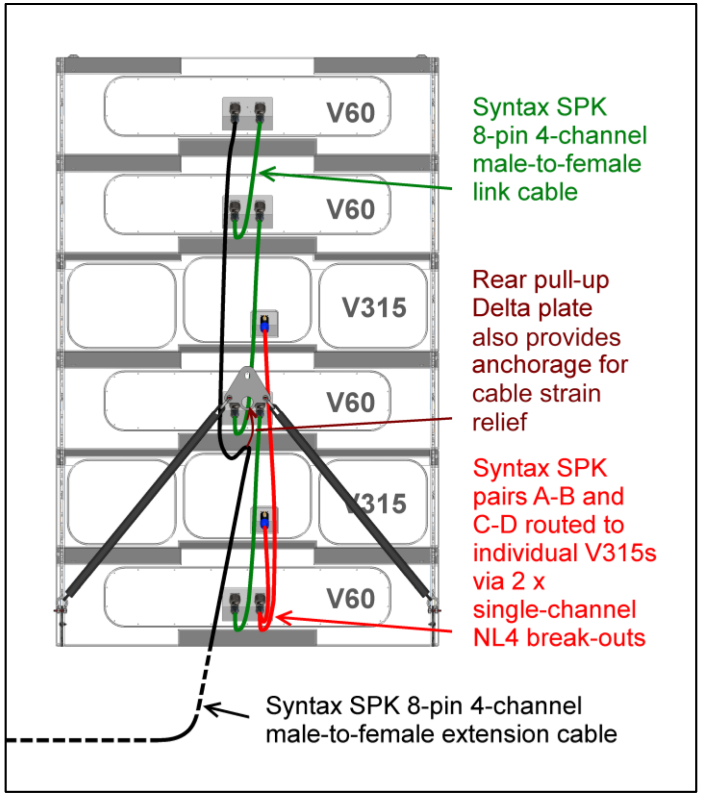
A single PLM20K44 amplifier will drive a 6-box array of 4 x V60 or V90 top cabinets plus 2 x V315 mid-bass cabinets via a single 4-pair Syntax SPK series cable.



PLM20K44 4-input/4-output amplifier/control with system link cable and extension cable.

The cabling *(shown left)* has been patched from the top (to follow the rigging sequence) but, as the V60/V90 connectors are fully paralleled pin-to-pin, any sequence is acceptable as long as the V315s are patched at the end of the cable run.

We strongly recommend cable-picking to the rear pull-up Delta plate for ground-run cabling – as shown on the left – or to the FlyGrid jib if cables are dropped in from a truss.



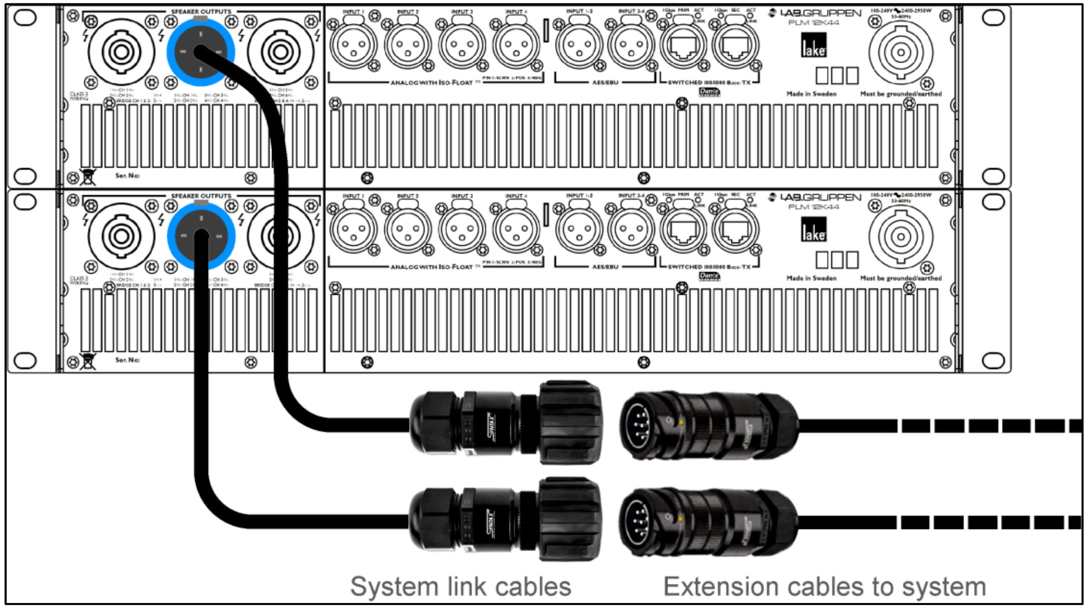
6-box *(4 x V60/V90, 2 x V315)* array patch. *(Different scale)*
Extension cable = ***black***, Link cables = ***green***, V315 break-out cable = ***red***

Extension cable and **Link cable** pin-outs – see ***Section 4.1*** **for full details**
NL8 pairs 1 -to- 4 are through-connected to Syntax pairs; A-B & C-D *(V315 mid-bass #1 & #2)*, E-F *(V60/V90 mids)*, G-H *(V60/V90 HFs)* respectively.

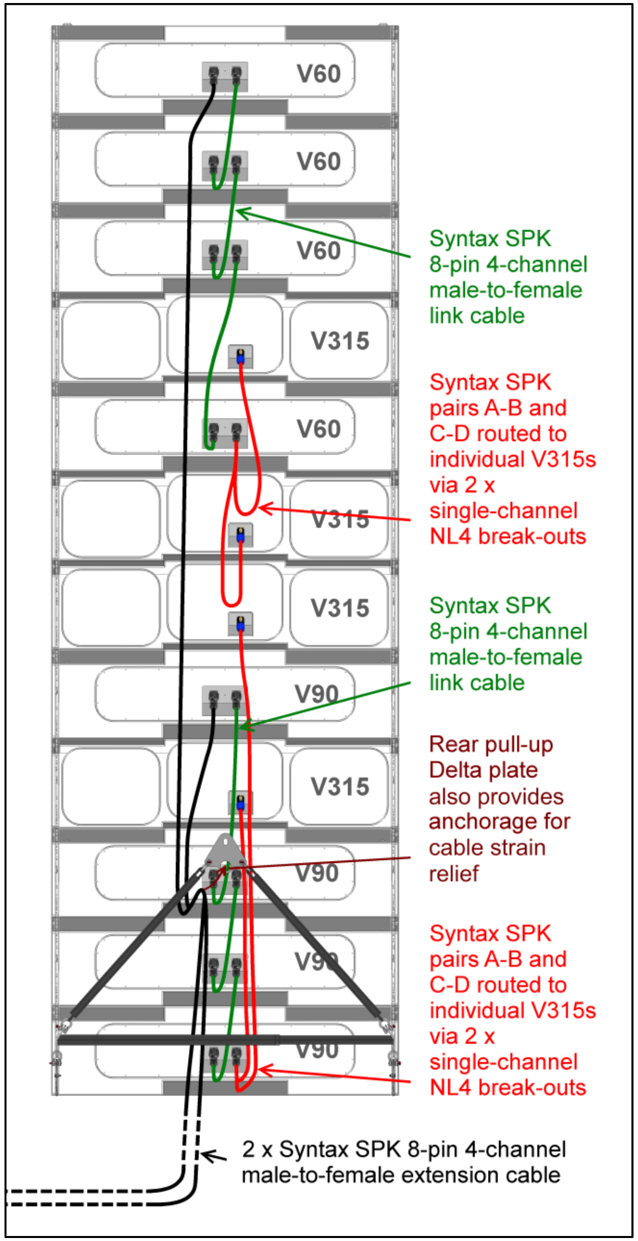
Syntax SPK-to-NL4 V315 break-out cable *(shown **red** on right illustration)*

VC8BB02M Syntax SPK male to 2 x NL4 breakout		2 x V315 cabinet wiring		Function
Syntax SPK connector pin **A**	NL4 #1 **1 +**	NL4 #1 **1 +**	3 x V315 #1 drivers **+**	V315 #1
Syntax SPK connector pin **B**	NL4 #1 **1 -**	NL4 #1 **1 -**	3 x V315 #1 drivers **-**	
Syntax SPK connector pin **C**	NL4 #2 **1 +**	NL4 #2 **1 +**	3 x V315 #2 drivers **+**	V315 #2
Syntax SPK connector pin **D**	NL4 #2 **1 -**	NL4 #2 **1 -**	3 x V315 #2 drivers **-**	

Each 2 x PLM20K44 amplifier system will drive a complete 12-box array of 4 x V60 and 4 x V90 top cabinets plus 4 x V315 mid-bass cabinets via two 4-pair Syntax SPK series cables.



2 x PLM20K44 4-input/4-output amplifier/control with system link and extension cables. Extension cables to array shown lower right. *(Not to scale)*



12-box *(4 x V60, 4 x V90, 4 x V315)* patch *(Different scale)*
Some cables shortened and false-coloured for clarity

The 12-box patch comprises two of the 6-box patches shown on the previous page.

Again, the array cabling *(shown right)* has been patched from the top *(to follow the rigging sequence)* but, any sequence is acceptable as long as the V315s are patched at the end of each 6-box cable run.

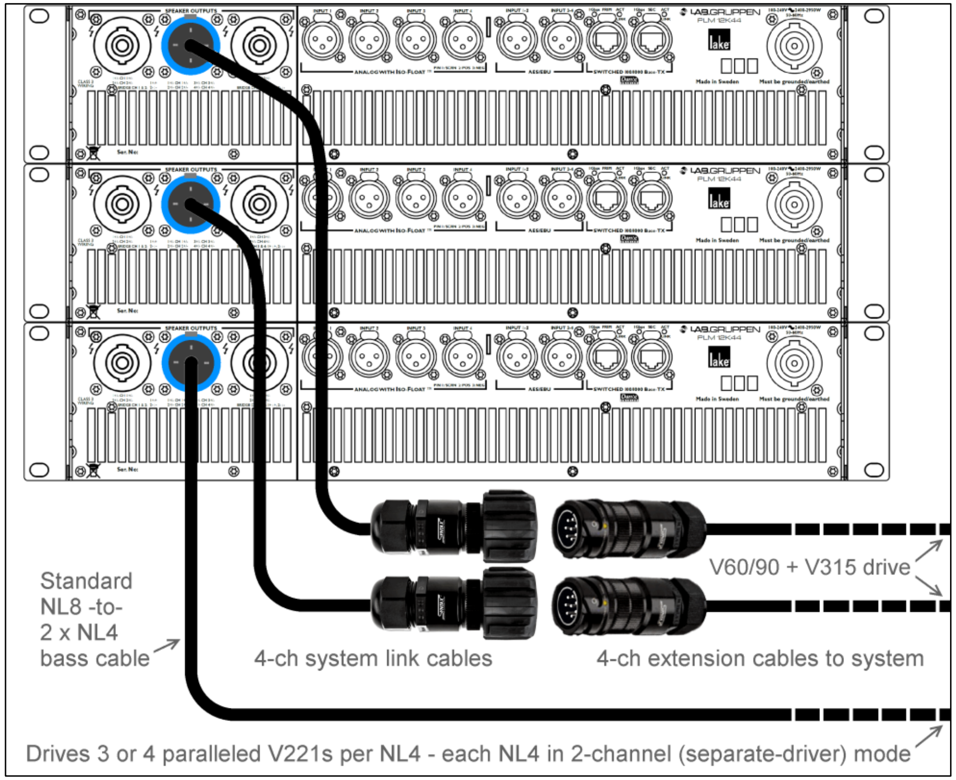
We strongly recommend cable-picking to the rear pull-up Delta plate for ground-run cabling – as shown on the left – or to the FlyGrid jib if cables are dropped in from a truss.

Each 6-box cable patch follows the same sequence as before. The incoming 4-channel extension cables are shown in ***black***, the 4-ch link cables in ***green*** and the V315 break-out cable are shown in ***red***.

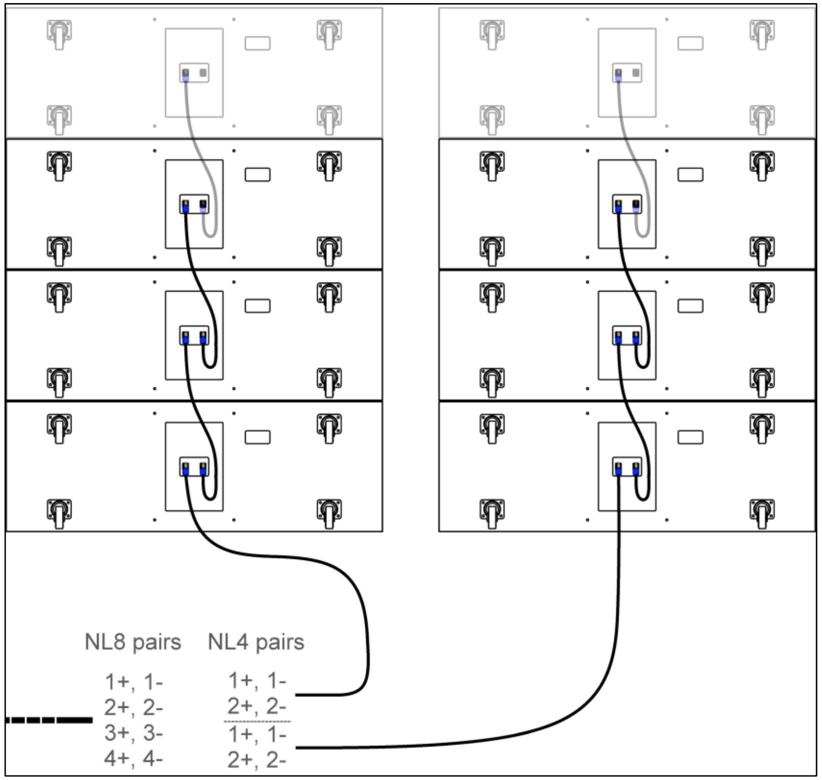
A typical Funktion One 3 x PLM20K44 amplifier rack will drive a complete 12-box array of 4 x V60, 4 x V90 and 4 x V315 plus a compatible bass system comprising 6 x V221 (2.67 ohms per channel) or 8 x V221 (2 ohms per channel) via:

- 2 x 1m NL8 -to- 4-ch Syntax SPK series female System Link cables (left) then…
- 2 x 10m or 25m 4-ch Syntax SPK series male -to- female Extension cables (right – male end shown)
- Local 2m 4-ch Syntax SPK series male -to- 2 x NL4 break-out cables daisy-chain from each block of four V60/90 cabinets to drive 2 x V315 cabinets per block (via NL4 pins 1+ & 1-)
- Standard NL8 -to- 2 x NL4 cables drive, typically, 6 or 8 x V221 cabinets

See ***Section 4.1*** for cable and pin-out details.



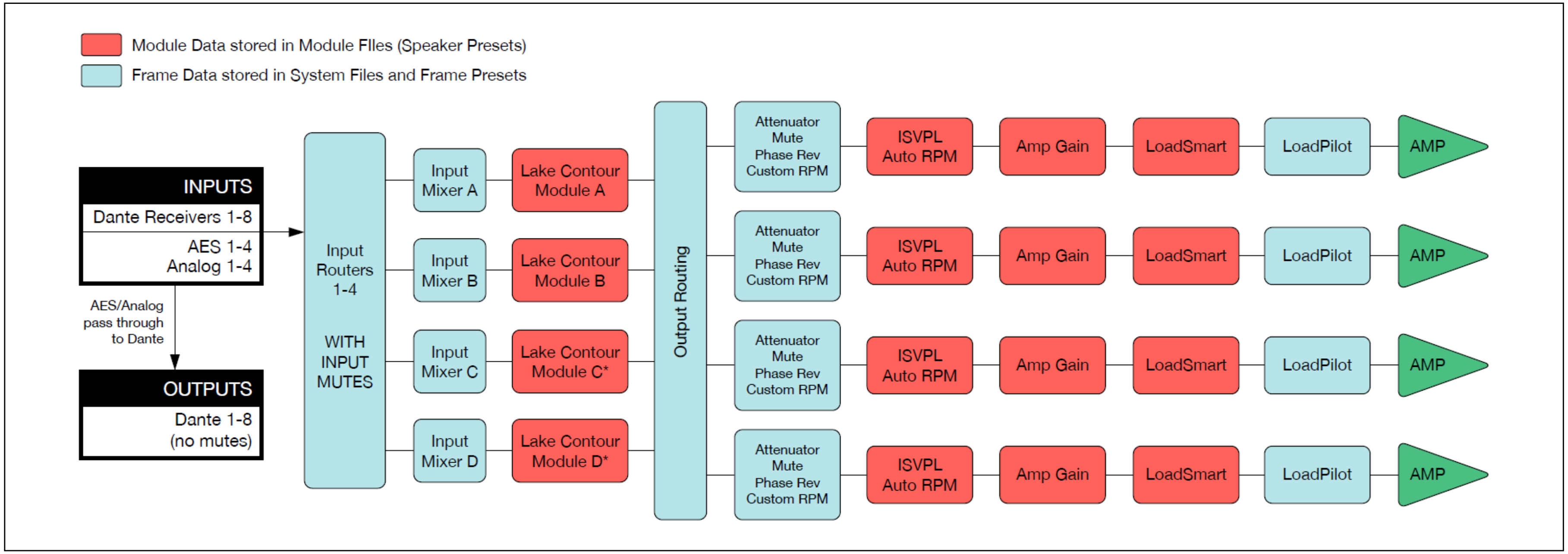
3 x PLM20K44 4-input/4-output amplifier/control with system link, extension and bass cables. Cables to array shown lower right. *(Not to scale)*



Bass cable -to- 6 *(or 8)* x V221 stack patch showing NL8 -to- 2 x NL4 scheme *(Different scale)*

5.4 PLM20K44 amplifier and Dante™ networking overview

The PLM20K44 is a ***P**owered **L**oudspeaker **M**anagement* system that includes all the necessary DSP functions to run a Funktion One Vero system to full specification. For further PLM20K44 4-channel amplifier details go to: ***http://labgruppen.com/view-model/plm-plus-series/plm-20k44***



Lab.gruppen PLM20K44 signal path

The PLM20K44 amplifier is designed to be controlled by ***Lake® Controller*** software over a Dante™ network - an Ethernet-based digital audio and control technology – using standard Cat-5e or Cat-6 *(recommended)* cabling. Fibre-optic cabling can also be used for long site runs.

For information on Lab.gruppen PLM+ series amplifiers, Lake Controller and Café software, manuals and other documentation, go to: ***www.labgruppen.com/Categories/c/Labgruppen/Downloads*** and click through the appropriate categories and sub-pages.

We also strongly advise keeping your PLM20K44 amplifier firmware up to date to ensure compatibility with other Lab.gruppen & Lake updates. To do this, follow the instructions in the ***Lake Controller Operation Manual*** *(address above).*

5.4.1 Dante™ network

Unless you have completed Audinate-approved Dante training and are fully qualified to ***Dante Certification Level 2***, we recommend that you leave Dante network design to the experts.

Audinate training will ensure that you configure networks – often linking a wide range of pro-audio products – in a reliable and fail-safe manner.

For further information about training see: ***www.audinate.com/resources/training-and-tutorials/dante-certification-training.***

Direct PLM Series amplifier control without switches

Although it is possible to patch a laptop directly into several daisy-chained PLM Series power amplifiers by using each amplifier's secondary Ethernet port as a "loop thru", Lab.gruppen don't recommend this method for Dante operation. Daisy-chaining can also cause latency build-up.

Dante network routing is, therefore, usually performed via multiple switches – at front-of-house, to handle Dante audio from the console and control from a laptop, and in each amplifier rack so that the audio and/or control may be starred to the individual PLM Series amplifiers. The FOH laptop is often augmented by a tablet via a wireless access point *(WAP)* for in-field listening adjustments. See the ***following page*** for a network example featuring dual modular redundancy.

Switch considerations

Dante technology is based on standard network design techniques. A confusing number of network switches are available so here are some switch recommendations:

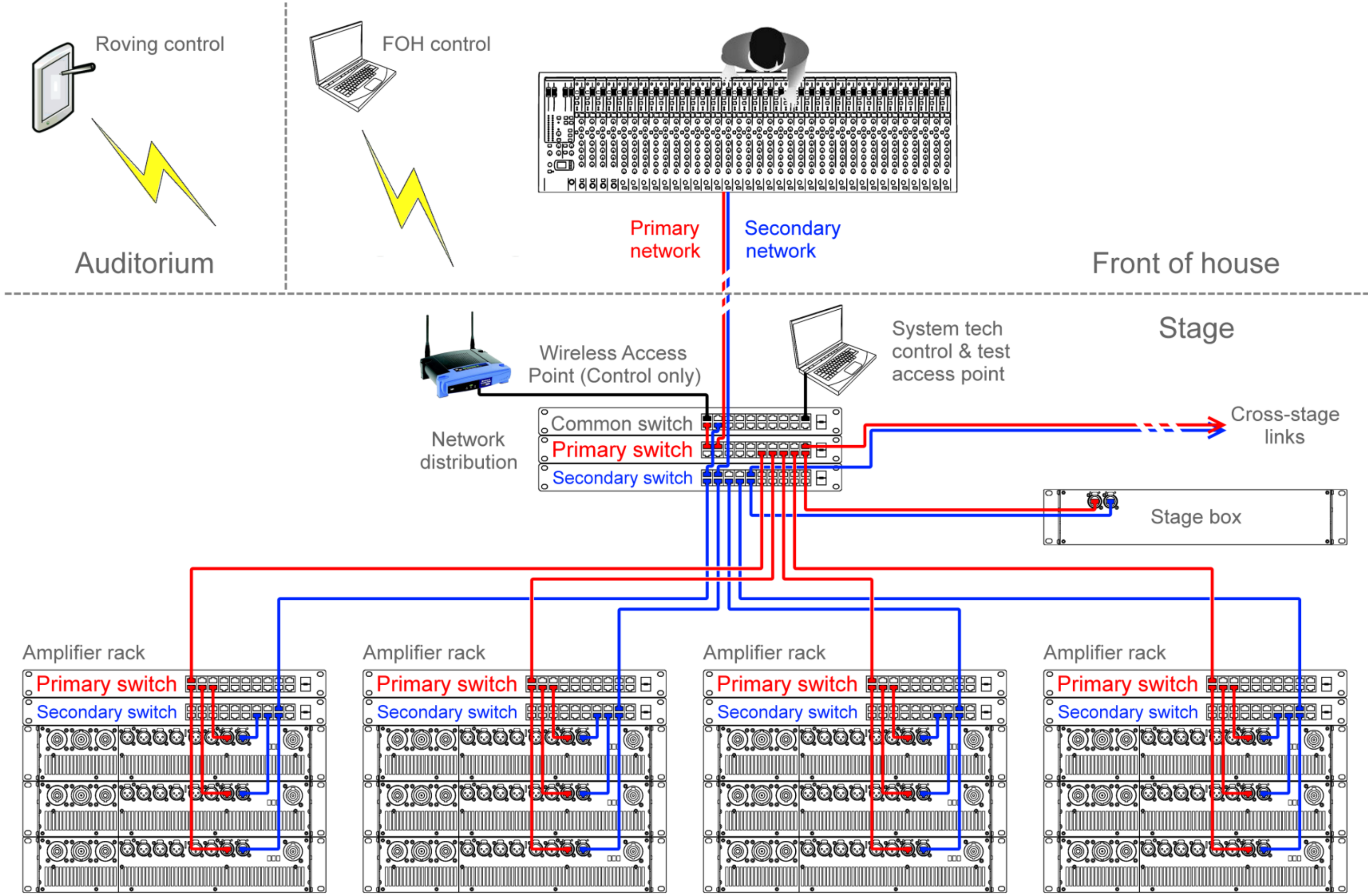
Choose switches *(e.g. **CISCO SG300-20** or direct replacements)* that provide:

- Robust, rack-mount construction
- Professional IEC mains connectors. Switches with just a few ports often use "wall-wart" domestic-style power units that are easy to lose or damage
- 1000BASE-T *(or better)* switch-to-switch operation
- Non-blocking layer-2 – so that all ports will continue to work reliably at 1.488Mpps packet throughput ***per port***
- SFP module support – to allow fibre connections to be used for long-distance links
- Managed facilities for configuration and monitoring. This will enable switches to be optimised for Dante use by implementing or disabling various features
- Ability to disable any Energy Efficient Ethernet *(EEE)* or power-saving feature – to avoid the switch "falling asleep"
- DSCP *(DiffServe Code Point)* QoS *(Quality of Service)* to ensure that Dante's clock, then audio take strict priority
- RSTP *(Rapid Spanning Tree Protocol)* capability – so that multiple switches can be placed in a "fail-safe" ring topology

Note that PLM Series power amplifier Ethernet ports must never be placed *within* a ring as they don't support RSTP

For further Dante network advice, please see: ***www.audinate.com***.

Dante network example with dual modular redundancy



Dual modular redundancy networks take advantage of the PLM+ power amplifiers' primary and secondary network ports by employing two networks comprising separate cabling and switches. The system is configured so that secondary network cabling and switches take over if a primary network element fails.

For further information and current thinking on network design, see: ***www.audinate.com/resources/training-and-tutorials/dante-certification-training.***

6 System design

6.1 Hearing safety

Enjoy your system responsibly

Funktion One loudspeaker systems are designed to provide extensive audience coverage at low distortion and are capable of producing very high near-field sound pressure levels.

The richness and impact of a powerful sound system can be great fun and very exhilarating; but please consider others by following the advice below.

- Design your system for good projection and coverage without overexposing specific audience areas to damaging levels. Individual audience members should not be exposed to levels significantly greater than levels at the mix position.

- Don’t place your ears too close to high power loudspeaker systems during system set-ups. Erroneous patching or un-muting by others could generate unexpected and damaging sound pressure levels during set-up.

- Wearing ear plugs may be a sensible precaution during prolonged system set-ups but remember that you have a duty of care towards other people who may be working in the area. Operators should always be fully aware of the sound levels they are producing and should not wear ear plugs whilst staff or audience members are present.

- Hearing loss is cumulative and can result from long-term exposure to sound pressure levels as low as 85dBA. Installers and get-in crew should consider long-term staff exposure when positioning high power loudspeaker. Managers should rotate staff - especially security staff at the front of the audience - to minimise each staff member’s cumulative exposure.

- Check the relevant noise exposure/noise at work regulations and comply with them. If in doubt, seek expert advice.

If no noise exposure or noise at work regulation is available, we suggest the following LAeq* *(A-weighted Equivalent Continuous spl)* and peak spl limits for staff:

**LAeq is referred to as dBA Leq on some meters*

85dB LAeq - 8hrs

88dB LAeq - 4hrs

91dB LAeq - 2hrs

94dB LAeq - 1hr

97dB LAeq - 30mins

100dB LAeq - 15mins

Instantaneous/peak sound pressure levels must never exceed 137dB *(C-weighted).*

Leq is the average spl over a defined time period. We suggest averaging and logging LAeq, and logging peak spl, over consecutive industry-standard 15 minute periods.

LAeq may be logged with an integrating spl meter or by using a calibrated microphone and IO with the appropriate sound measurement software.

Note!
Unweighted instantaneous sound pressure levels *(especially where there's a solid bass presence)* will be considerably higher than the dBA LEQ figures imply. This is because LEQ measurements are averaged over a considerable period of time and the applied A-weighting curve significantly under-represents the LF and HF content.

6.2 Typical array design using Funktion One Projection software

Projection

Funktion One's Projection design and prediction software allows Vero users to optimise array designs for smooth audience coverage and impact.

Vero users may request Projection and a user-licence directly from Funktion One or from your local Funktion One distributor.

Projection also alerts you if your proposed array design is not feasible or is unsafe. Genuine Projection software will always require a Funktion One user-licence.

Windows OS

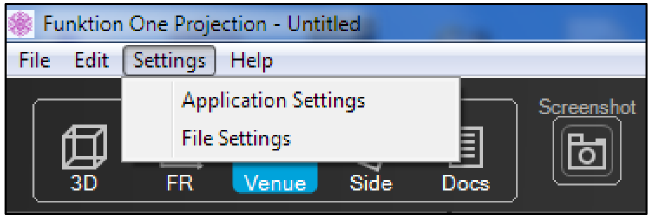
Projection is designed for Microsoft Windows operating system and has been thoroughly tested on all versions from Windows 7 onwards.

Mac *(iOS)*

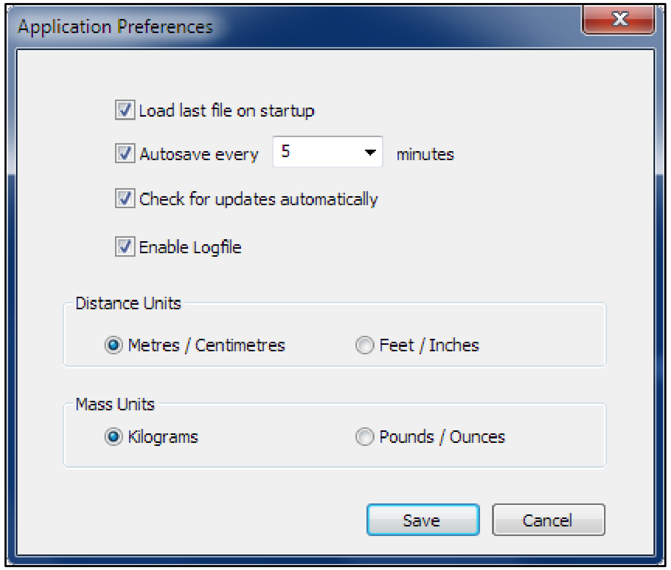
If you're a Mac *(iOS)* user, you can still run Projection on your Mac by installing Apple's ***Bootcamp***.
See www.apple.com/support/bootcamp/ for further information or contact your local Mac expert.

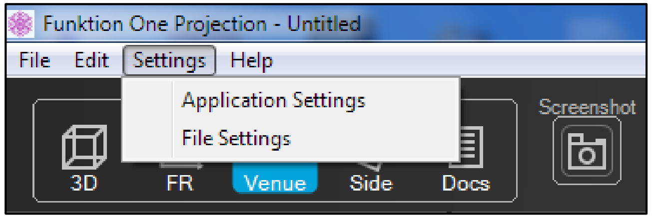
Projection Preferences

Once Projection is installed and booted up, some of its parameters may be tailored to your personal preferences as follows:

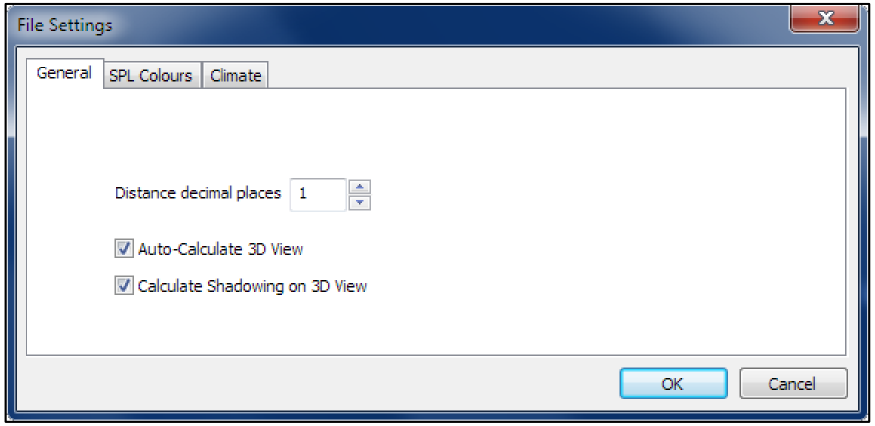


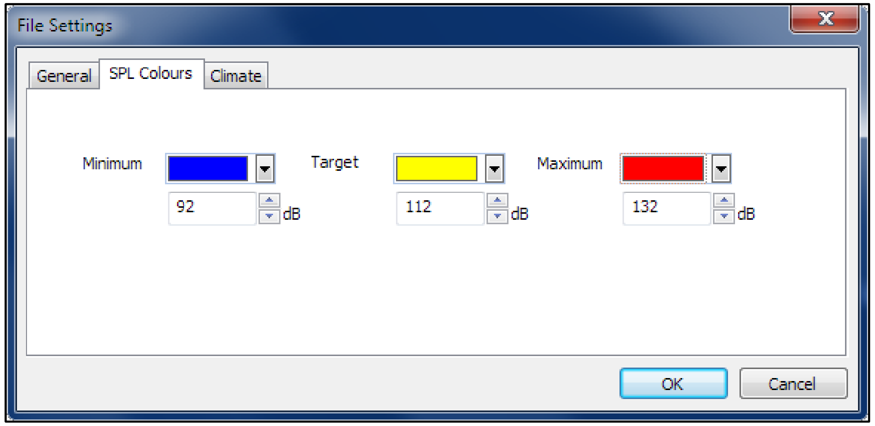
Click through ***Settings > Application Settings*** to select your Startup, Autosave, Update, Logfile, Distance Unit and Mass Unit preferences. ⇨





Click through ***Settings > File Settings > General*** to select your initial 3D coverage prediction preferences. If in doubt, tick both boxes. ⇨

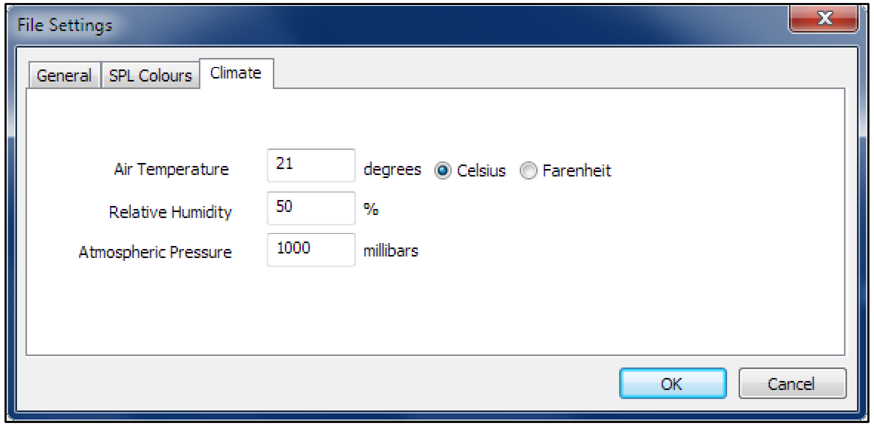




Click through ***Settings > File Settings > SPL Colours*** to select your initial 3D coverage SPL preferences.

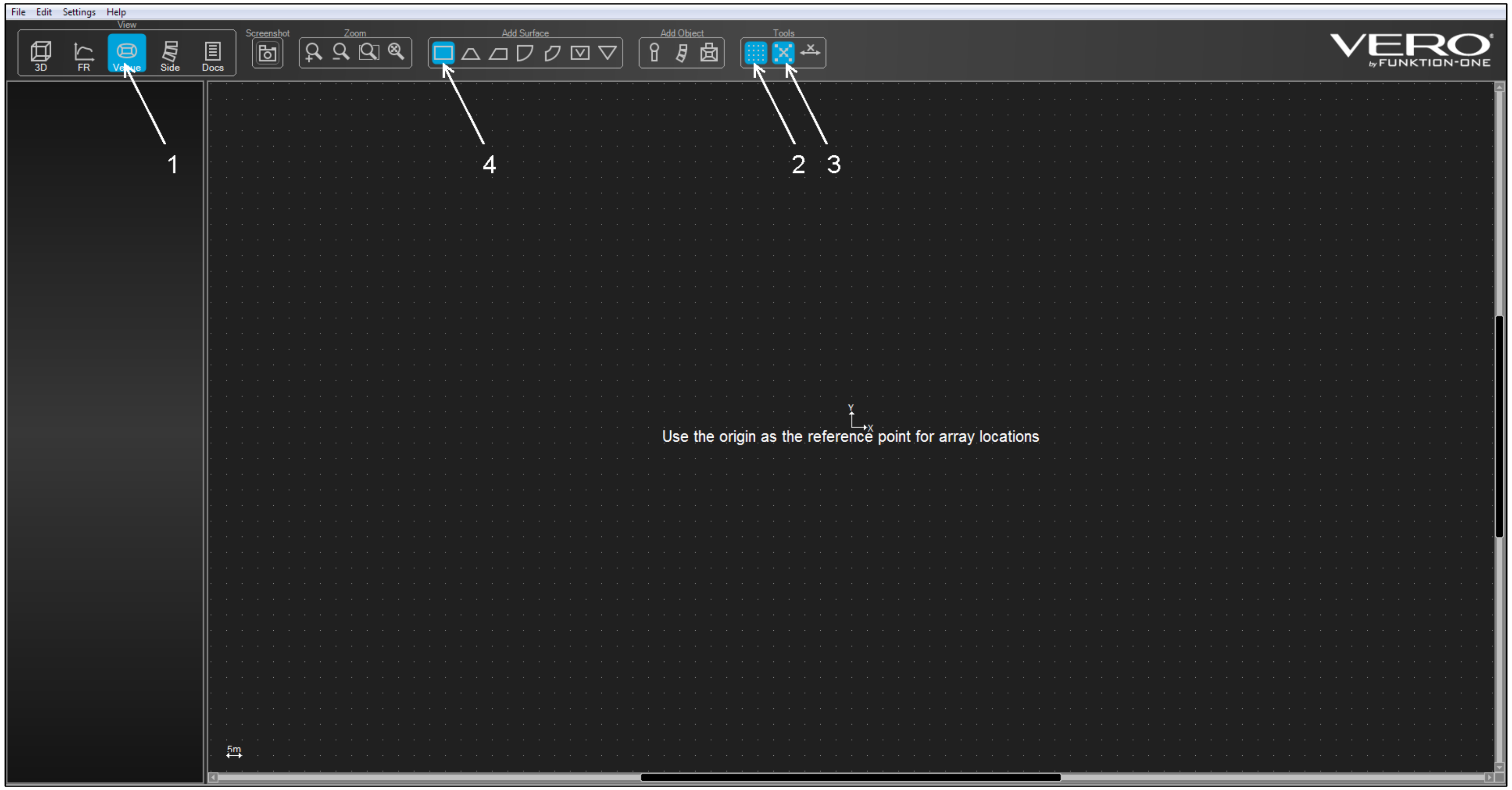
⇦ The settings shown here are a good starting point.

Click through ***Settings > File Settings > Climate*** to select your initial environmental parameters. Air absorption and, therefore, long-distance high frequency attenuation, varies with temperature and relative humidity. ⇨



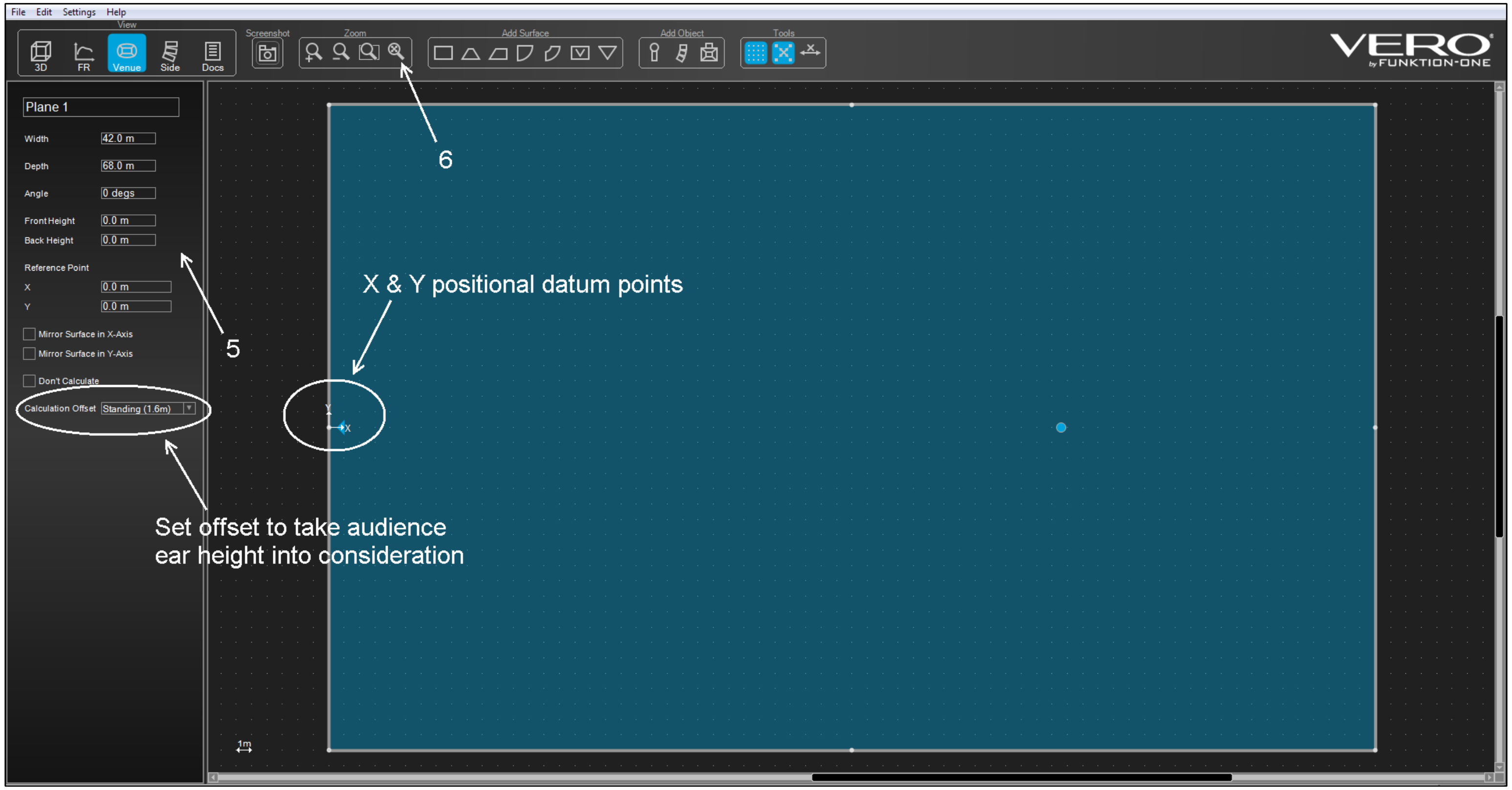
Array design example

The easiest way to learn to use Projection is to practise designing a system from scratch. You'll find this easier if you have Projection running on a laptop while you follow this work-through for a typical 12-cabinet sports arena system. *(You can always read this guide on an extension screen, tablet or smart phone)*



The arrow numbers refer to the points below. All mouse clicks are left button clicks unless noted otherwise

1) Select ***Venue*** from the ***View*** menu box if it is not already highlighted – Projection usually defaults to this at start-up
2) A ***Grid on/off*** switch is available in the ***Tools*** menu *(top centre)*. *(Metric or imperial parameters may be chosen in **Settings > Applications settings** - top left)*
3) A ***Snap to grid*** function is also available in the ***Tools*** menu. This automatically aligns your venue surfaces with the nearest grid point if required
4) Click on the appropriate audience section shape in the ***Add Surface*** menu box and then click in the main display area of the screen to add it



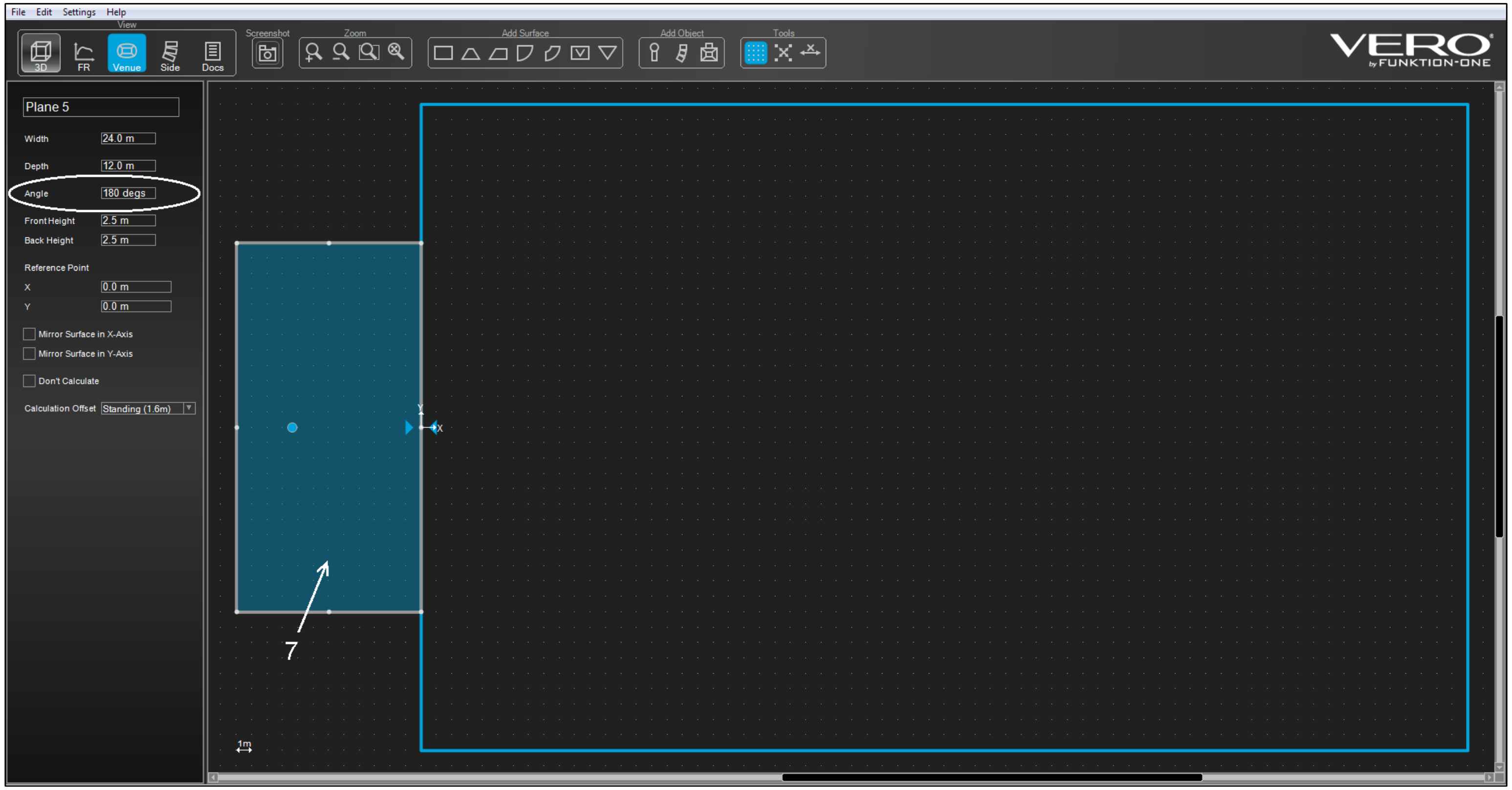
5) The new surface will appear with a properties strip on the left of the screen This allows you to adjust the surface's ***Width***, ***Depth***, orientation *(**Angle**)* plus ***Front*** and ***Back heights*** with respect to 0m. The main shape parameters can also be altered by dragging the nodes *(white dots)*.

   The X & Y ***Reference Point*** settings *(to left of 5 arrow)* position the centre-left of the surface with respect to the ***X*** *(0m)* & ***Y*** *(0m)* datum point *(circled).*

   The ***Calculation Offset*** *(circled)* allows average audience ear heights to be taken into consideration. This ear height is used for coverage predictions and will appear as a broken line just above the relevant surface when side views are selected later. A standing audience has been assumed for the floor.

   The properties strip *(screen left)* also allows surfaces to be ***Mirrored*** – i.e. mirror-image paired – in the X or Y axes. This feature is more likely to be used when a complex, symmetrical venue is being modelled – see later in this work-through

6) You can centre your plot on the screen at any time by clicking the ***Zoom to fit*** button in the Zoom menu



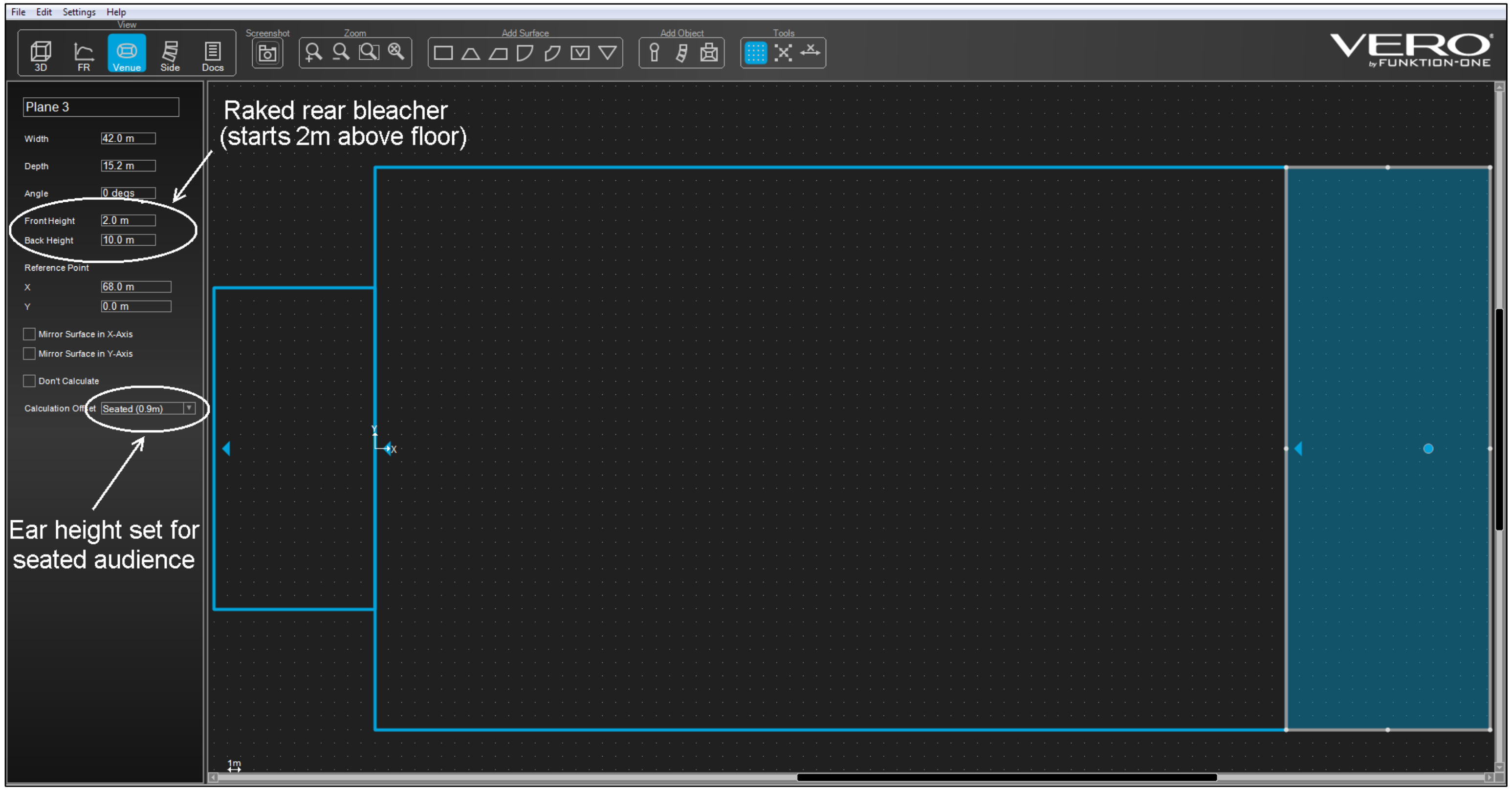
7) Extra venue surfaces may be added by clicking the required shape in the ***Add Surface*** menu and then clicking in the approximate position on the venue plan. The surface may then be dimensioned and precisely positioned by typing in the properties strip *(to left of main screen)* or by dragging the *(white dot)* nodes. A 2.5m high rectangular stage has been added. Surface heights are relative to 0m in Projection – i.e. the floor level in this example.

Surfaces or objects can be dragged and dropped to a new position. You can use your ***Delete*** key to delete a selected *(highlighted)* surface or object.

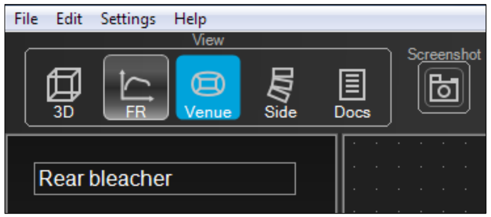
If you have ***Snap to grid*** selected in the ***Tools*** menu, your surface will move in grid-marker increments. Grid-marker spacing varies with zoom – zoom in or out for smaller or larger increments by using the ***Zoom*** menu **+** and **–** buttons . . . or by clicking in the plot area and then using your mouse wheel.

A rectangular surface's reference point is usually centre-left unless the surface has been rotated from its default ***0 degs*** using the ***Angle*** text box. In the example shown, we've rotated the stage surface *(7)* by 180º *(circled)* so that down-stage-centre and audience front centre coincide with the ***X** (0m)*, ***Y** (0m)* datum point.

We'll continue to build up our arena example by adding raked bleacher seating at the rear *(highlighted)*.



- The bleacher's ***Front Height*** is set to 2m above the floor. It rises to a ***Back Height*** of 10m above the floor
- The ***Calculation Offset*** is set at 0.9m for the average ear height above the bleacher surface

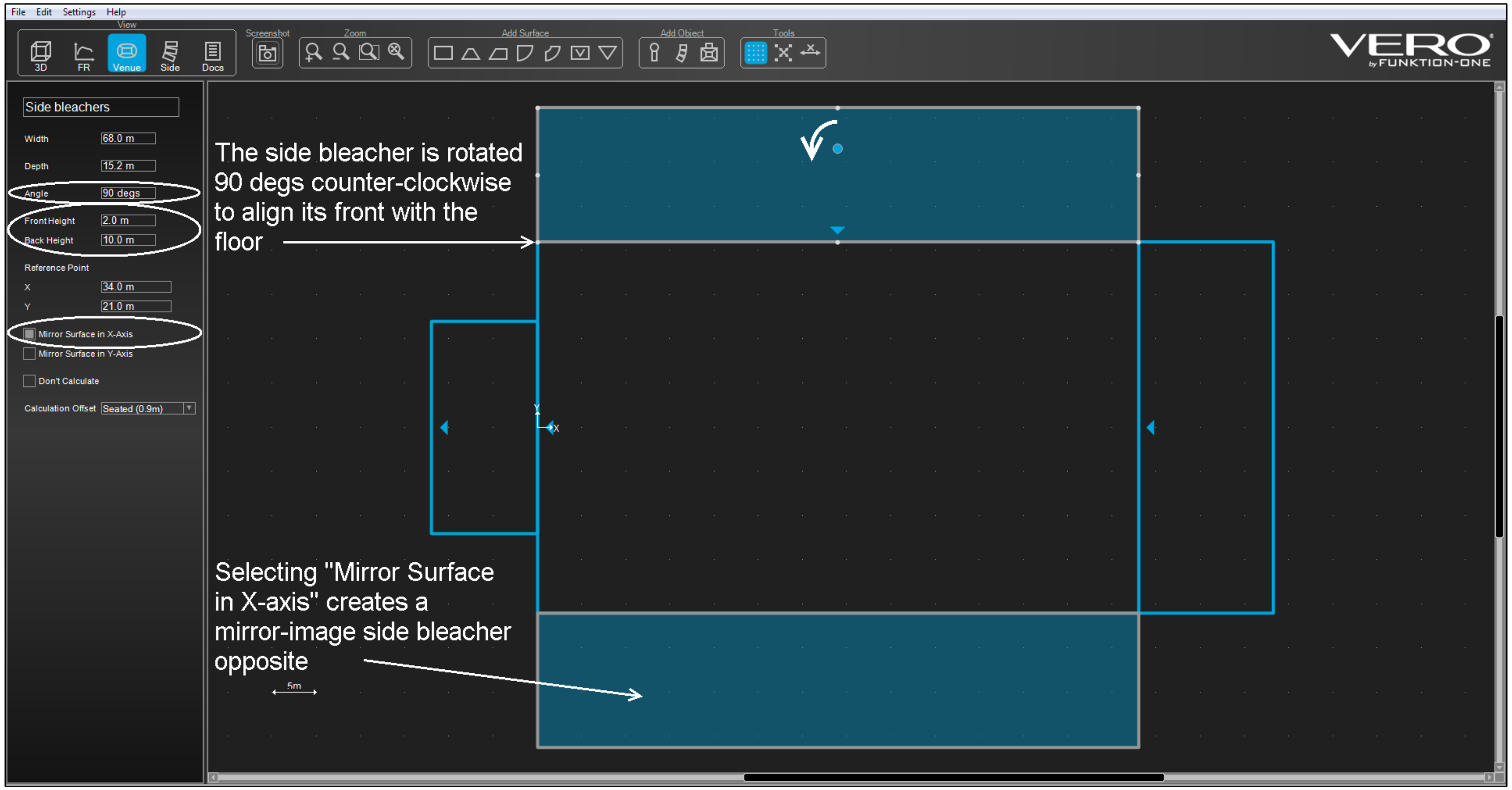


Contextual *Plane, Venue* or *Project* text boxes

The properties strip ***Plane*** number text box *(**Plane 3** in the above illustration)* may be double-clicked and edited for any selected area at any time to give the surface a friendly name; for example, "***Rear bleacher***" ⇨

When no surface is selected, a ***Venue*** name may be typed into the empty text box. The venue also determines the ***Project*** name for Projection's .pdf exports.

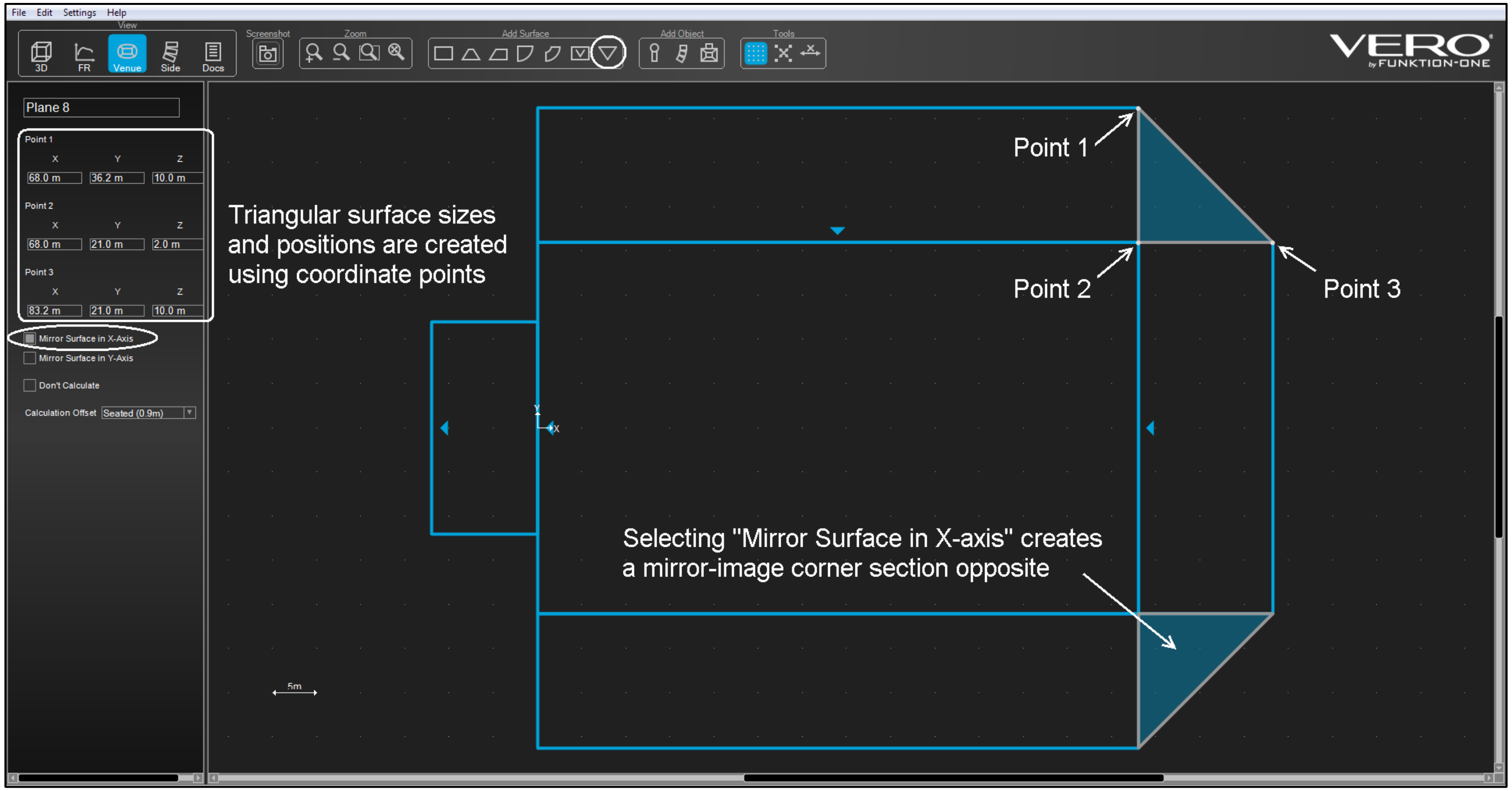
Now we'll add some arena side bleachers *(highlighted).*



Notes:

- The *(upper)* side bleacher is added as a rectangular surface – but it's rotated 90^o counter-clockwise *(see arrow in upper highlighted area and the properties strip* ***Angle*** *text box - circled)* and positioned to align its front ***Reference Point*** half-way along the *(upper)* side of the floor *(X = 34m, Y = 21m)*
- The bleacher starts 2m above the floor level and rises to 10m above floor height at the back *(see* ***Front Height*** *and* ***Back Height*** *- circled)*
- The ***Calculation Offset*** is set at 0.9m above the bleacher surface
- Once the upper side bleacher is correctly dimensioned and positioned, the ***Mirror Surface in X-axis*** button is selected to create a mirror-image bleacher on the opposite side of the venue's X datum line. The mirrored *(house left)* surface then tracks the original *(adjustable)* house-right surface.

And, finally, we'll add the corner seating areas *(highlighted).*

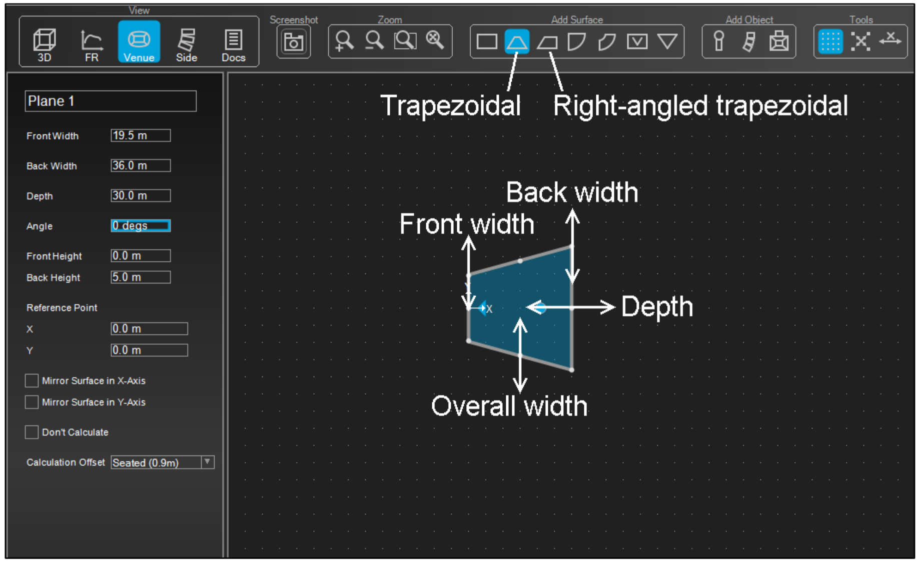


Notes:

- Triangular surfaces may be added by left-clicking the triangular shape *(circled)* in the ***Add Surface*** menu and then clicking on the venue plan as before
- The triangular surface may then be accurately dimensioned and positioned by entering X, Y and Z *(height)* positional coordinates into the appropriate properties strip text boxes *(screen left – shown boxed)*. Each corner of the triangle is allocated a point number as illustrated above
- The shape's nodes *(the small white dots)* may also be dragged to adjust the appropriate X and Y point positions if you wish
- The upper corners *(**Points 1** and **3**)* are 10m above floor height and the lower corner *(**Point 2**)* aligns with the side and rear bleachers 2m above the floor
- Again, the ***Mirror Surface in X-axis*** button *(circled)* has been selected to create a mirror-image surface on the opposite side of the venue's X datum line. And again, the original house-right corner surface remains the reference surface for further adjustments.

Other surface shapes

The ***Add Surface*** menu offers additional surface shapes *(not used in our design example)*.



Trapezoidal example

Trapezoidal surface

A trapezoidal surface may be selected from the ***Add Surface*** menu and clicked into position. *(A standard trapezoidal shape is shown above)*.

The shape's main dimensions – in this example, ***Front*** and ***Back Width***, ***Depth*** and orientation ***Angle*** - may be typed into the appropriate property strip boxes.

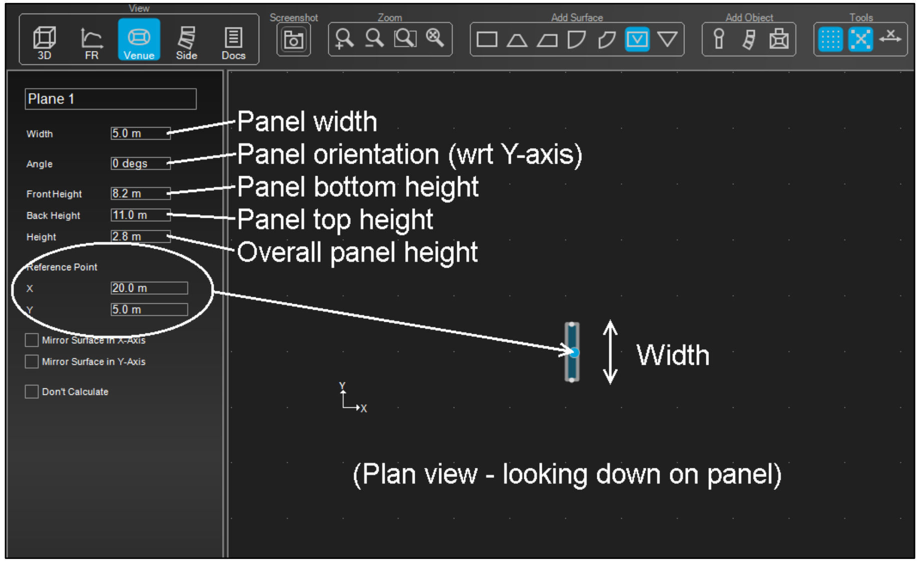
The front width, back width, overall width or depth may also be adjusted by dragging the appropriate *(white dot)* nodes. Note that the trapezoidal shape remains symmetrical, so it doesn't matter whether you use upper or lower width nodes.

Right-angled trapezoidal surface

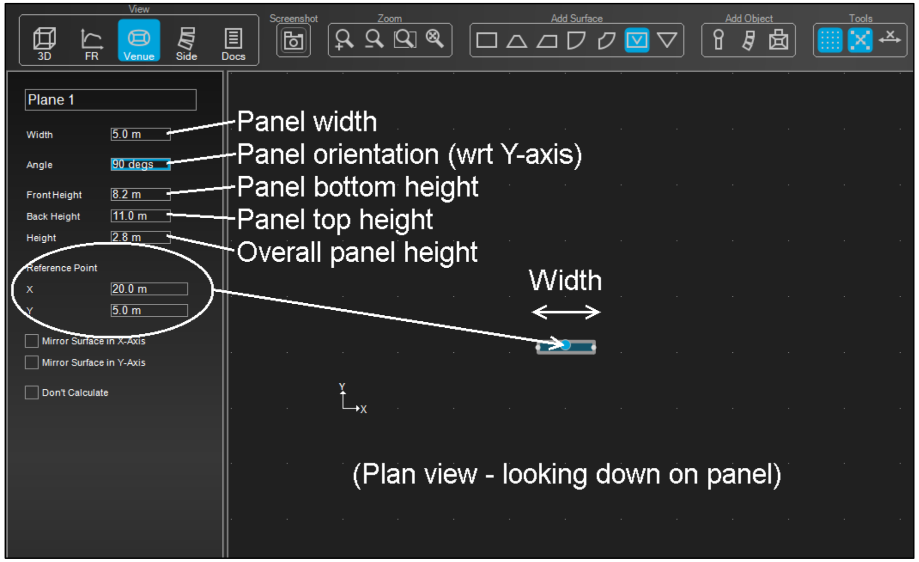
A right-angled trapezoidal surface may be placed, dimensioned and orientated in a similar way. The only difference is that the right-angled trapezoidal shape is asymmetrical – angled on one side and squared off on the other – as illustrated in the ***Add Surface*** menu above.

Vertical rectangular panels

Vertical rectangular surfaces may be added by left-clicking the menu item *(highlighted in blue in the **Add Surface** menu)* and then left-clicking on the venue plan as before. We'll use these later to simulate video screens to demonstrate sound shadowing…



Panel at 0 degs

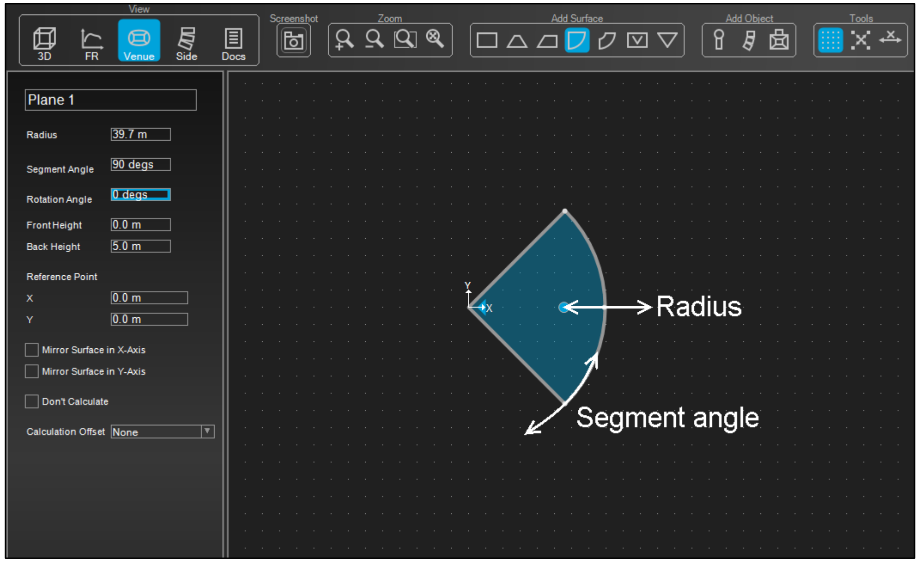


Panel at 90 degs

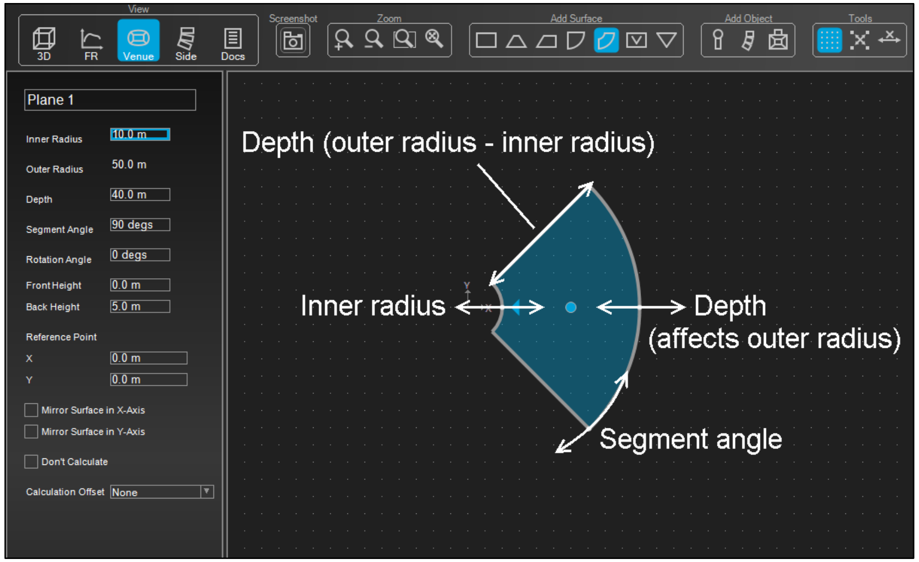
Notes:

- The surface may then be accurately dimensioned and vertically positioned by entering its ***Width***, ***Back** (top)* and ***Front** (bottom)* ***Heights***
  *(Note that the **Venue** view is always a plan view, so vertical sizes and positions will only be visible as text box figures . . . until we switch to 3D view later)*
- The surface may then be accurately positioned horizontally by entering its ***X*** & ***Y Reference Points*** *(circled)* or dragging and dropping it into position
- The surface's nodes *(the small white dots)* may also be dragged to adjust the width symmetrically about its X/Y reference point
- The surface may be orientated, with respect to the Y-axis, by entering the required figure into the ***Angle*** text box *(shown 0º on the left illustration and 90º on right)*
- Again, the ***Mirror Surface in X or Y-axis*** buttons may be selected to create a mirror-image surface on the opposite side of the venue's respective datum lines

Segments



Circular segment



Annular segment

Circular segment

Once a circular segment has been selected from the ***Add Surface*** menu and clicked into its initial position, its main dimensions – ***Radius*** and ***Segment Angle*** - may be typed into the appropriate property strip boxes. Again, the main dimensions may be adjusted by dragging the appropriate nodes. Overall orientation is set using the ***Rotation Angle*** text box.

Note that a complete circle may be obtained by setting or dragging the segment angle to 360°.

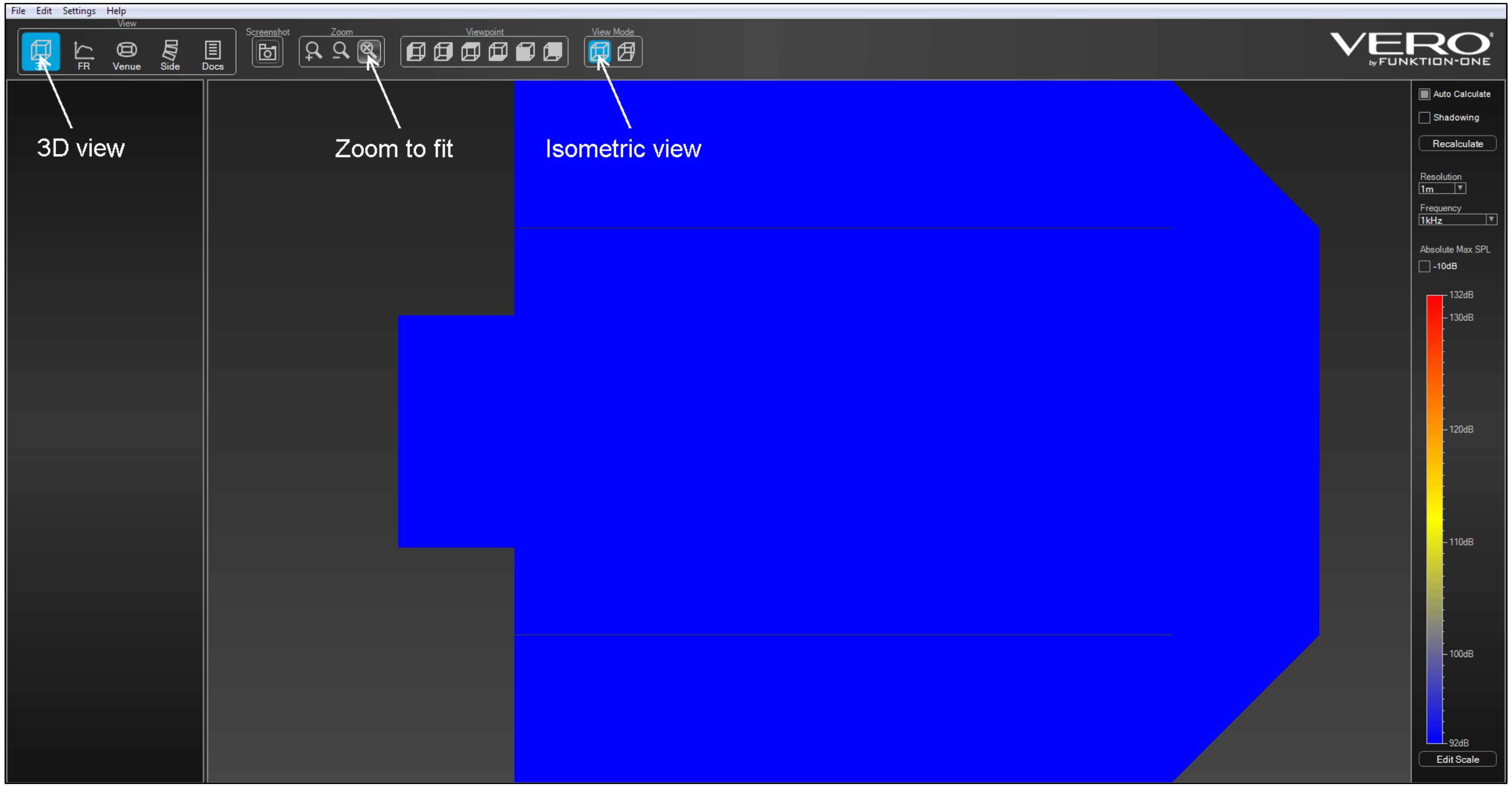
Annular segment

An annular segment's main dimensional properties are set by the ***Inner Radius***, the ***Depth*** *(the space between the inner and outer radii)* and the ***Segment Angle***. Overall orientation is set using the ***Rotation Angle*** text box. Again, dimensional parameters may be adjusted by dragging the appropriate nodes.

Note that a complete annular ring may be obtained by setting or dragging the segment angle to 360°.

Check your venue design

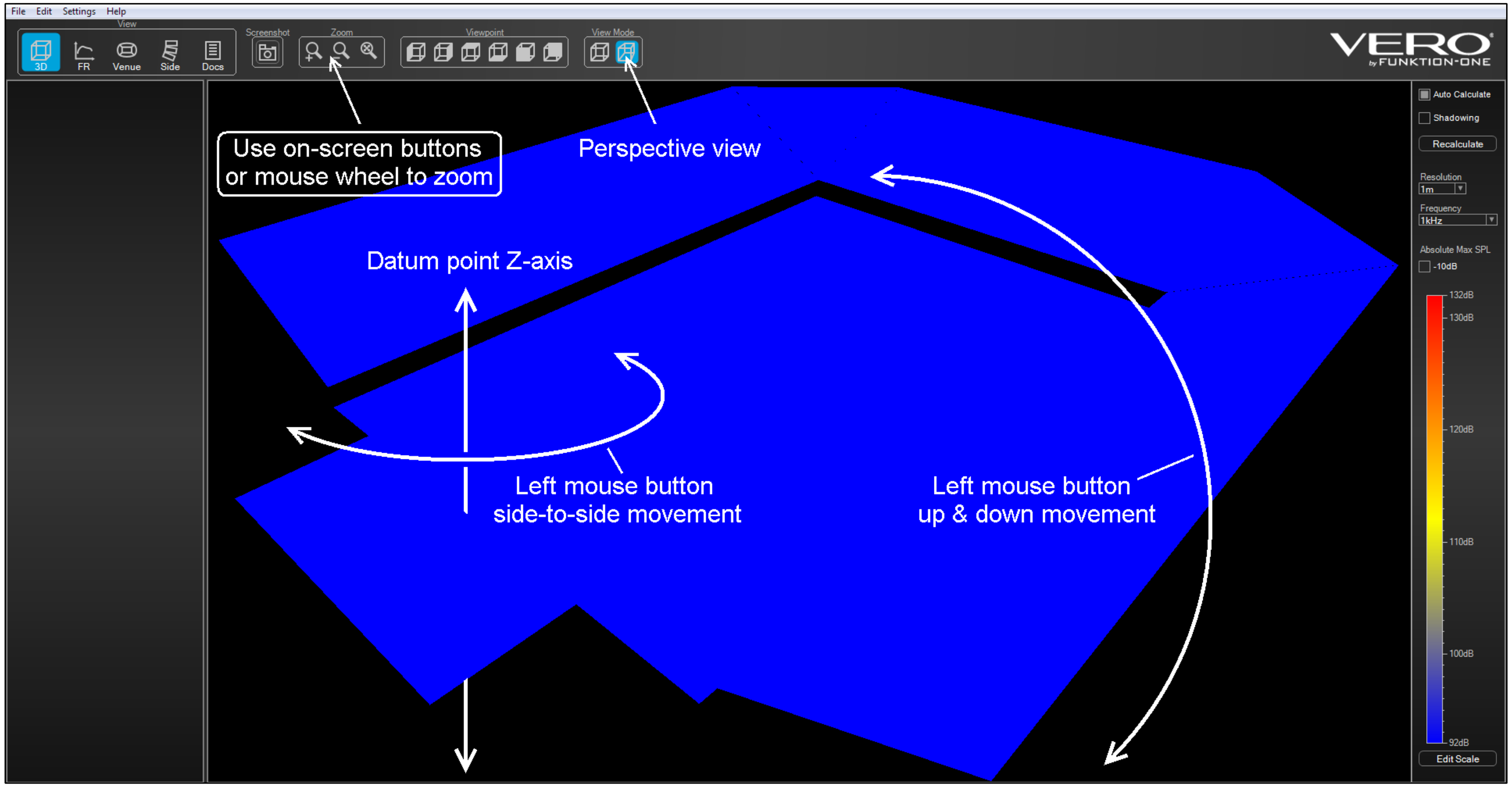
The easiest way to check that your venue design is the correct shape is to briefly switch Projection to ***3D*** view.



Start with a simple isometric plan view to optimise the venue model's position:

- Click the ***3D*** button *(arrowed)* in the top left ***View*** menu to view your venue model
- New models usually default to an isometric plan view – if yours doesn't, select ***Isometric View*** *(arrowed)* in the top centre ***View Mode*** menu
- Centre and expand the view to fill the display area by clicking on the ***Zoom to fit*** button *(arrowed)* in the ***Zoom*** menu. You're now in a good position to view your venue in 3D.

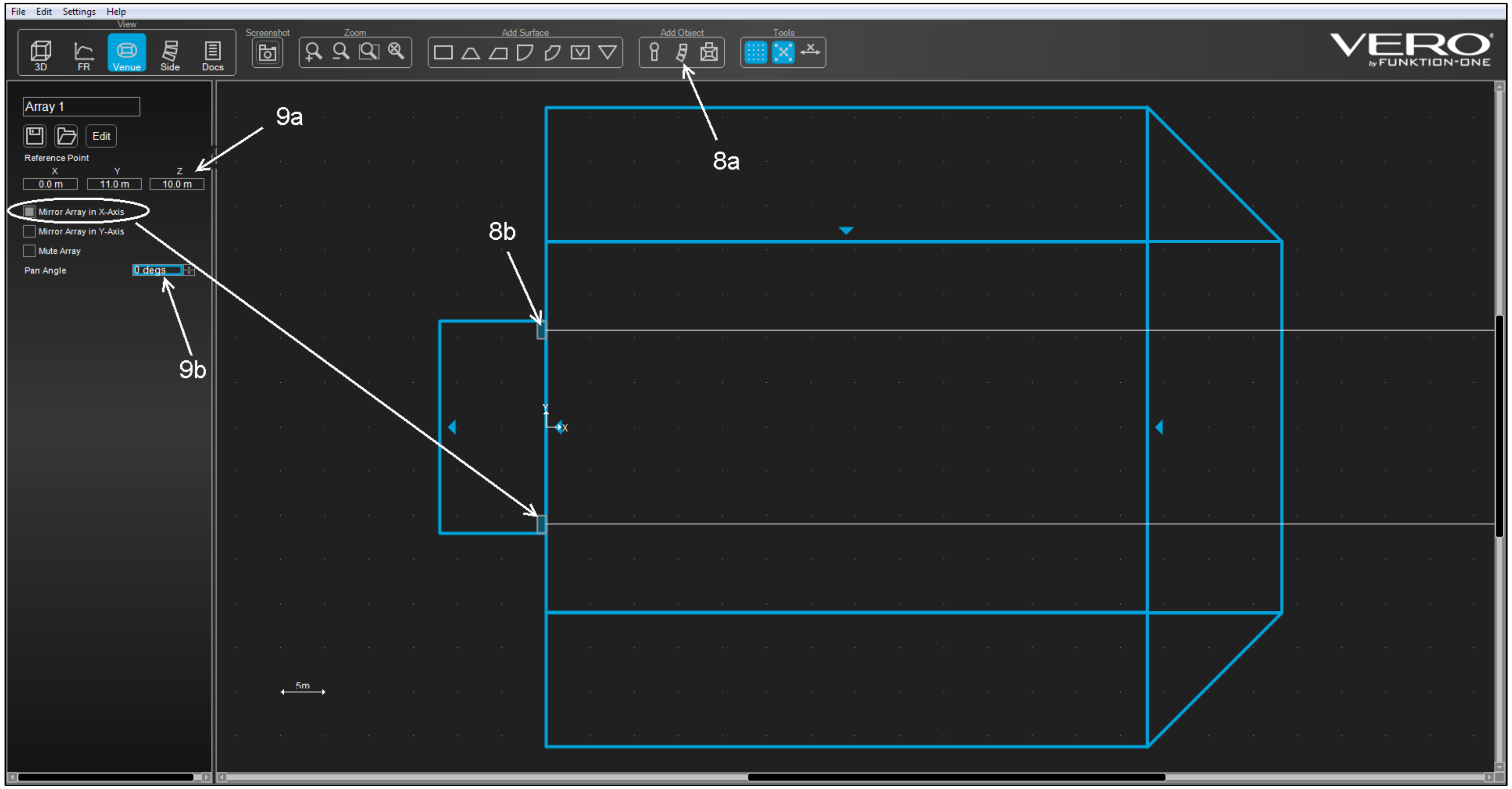
You can use your mouse to rotate and roll your venue model to inspect for any venue modelling errors or omissions in whichever ***View mode*** you prefer. Some engineers and draftsmen prefer the more purist ***Isometric view*** whereas many users prefer ***Perspective view*** *(arrowed)* as it looks more natural. It's your choice.



A 2-button mouse with a centre-wheel *(for zoom functions)* is ideal – but, if you don't have a centre wheel or equivalent, you can use the on-screen ***Zoom*** buttons.

- Left click and drag from side-to-side to rotate your venue model about its vertical datum point Z-axis – see above
- Left click and drag up and down to roll your venue model about its current horizontal centre line – see above
- Right click and drag from side-to-side to pan your venue model horizontally
- Right click and drag up and down to pan your venue model vertically
- Push your mouse wheel forward to push your model away from you *(zoom out)*
- Pull your mouse wheel backwards to pull your model towards you *(zoom in)*

Once you're happy with your venue design, you can go back to ***Venue*** view, zoom to fit if necessary and then add loudspeaker positions and design the arrays.



8) Go to the ***Add Object*** box, select the ***Add an Array*** symbol *(8a)*, then click the required array position, in this case house right *(8b)* in the main display area

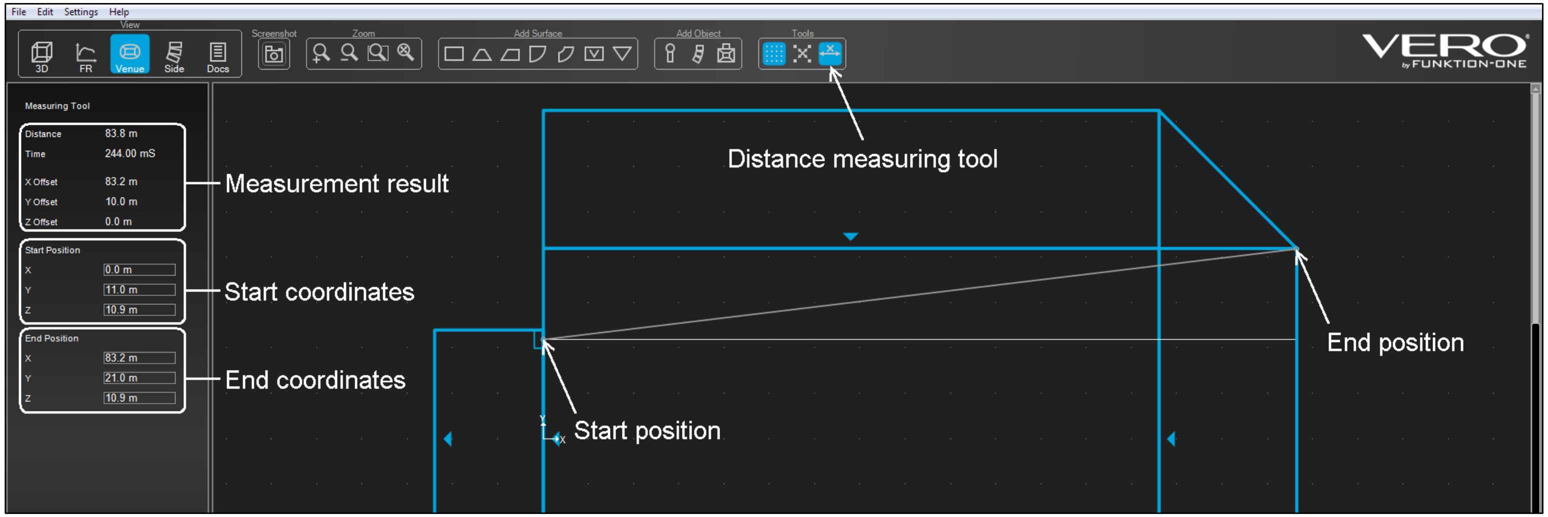
   In the above example, the ***Mirror Array in X-axis*** button has been selected to form a stereo pair *(button circled, and mirror-image array arrowed)*. Note that the original house-right array remains the reference array for any array adjustments.

9) Array position and grid height may be edited using the ***X, Y & Z Reference Point*** text boxes *(9a)* and orientation may be set using the ***Pan Angle*** text box *(9b)*. Stereo toe-in/out is usually unnecessary as each V60/V90 already incorporates inner and outer mid sections for excellent stereo image width.

Tip: You may find yourself selecting an adjacent surface when trying to left-click on an object *(an array in this case)*. If this happens, simply switch to a right-click to select the object.

A quick note about the distance measuring tool

A distance measuring tool is available in the ***Tools*** menu if required. Propagation time is accurately calculated for the distance being measured. As the speed of sound varies with temperature, the propagation time is calculated at the temperature entered via ***Settings > File settings > Climate*** – see earlier in this section.



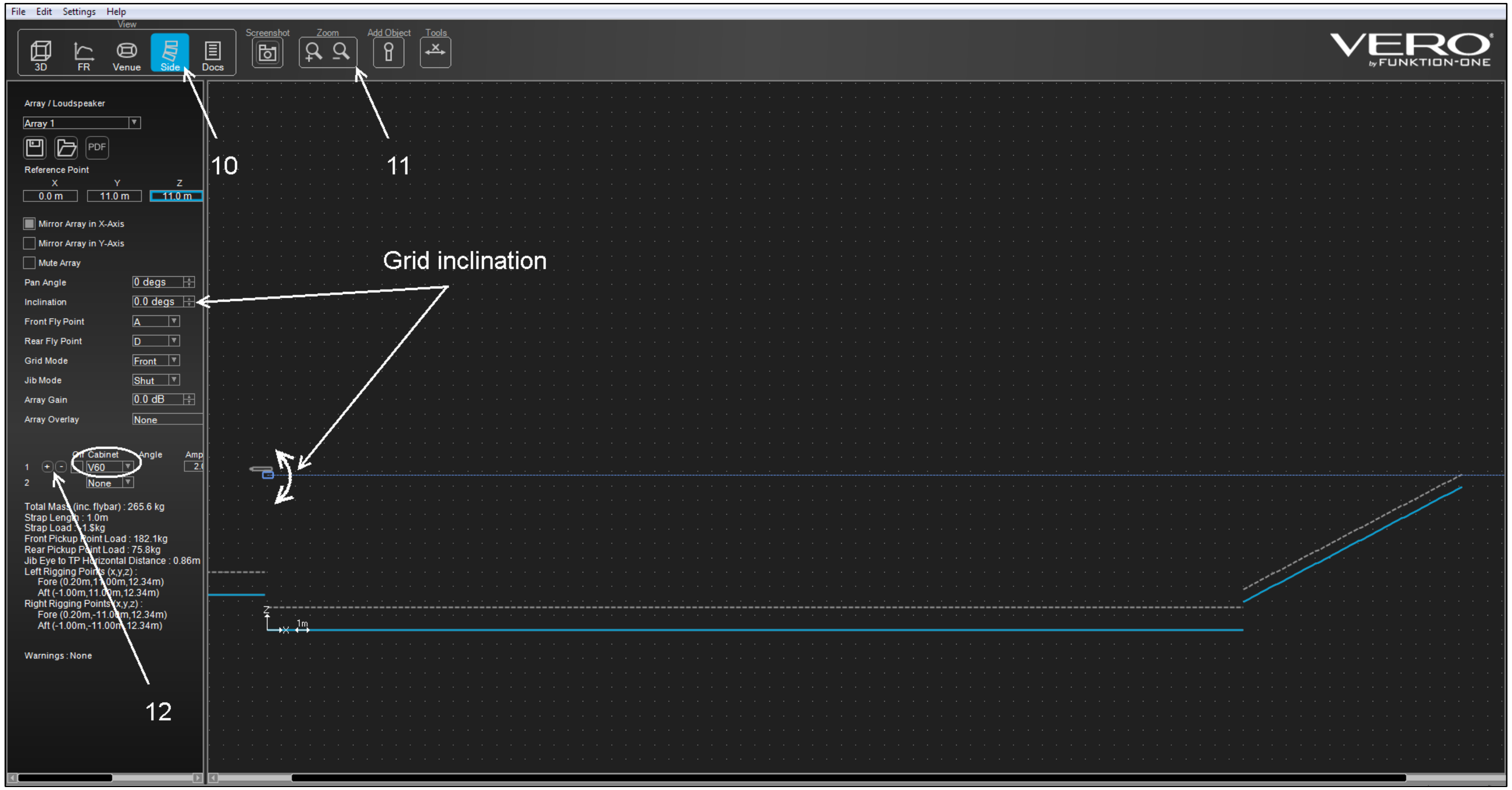
- Click on the distance measuring tool icon - - in the ***Tools*** menu *(shown highlighted in blue above)* to activate the tool
- Left-click at your required start position. In the above example, we've started at the top of our proposed array position
- Now left-click at your required end position. In the above example, we've ended at the top house-right corner of the rear bleacher
- Inspect the ***Start*** and ***End Position*** coordinates and make fine adjustments by editing the appropriate ***X, Y*** or ***Z*** coordinate text boxes – paying attention to the height *(**Z**)* coordinate to ensure accuracy
- Measurement results will be shown at the top of the ***Measuring Tool*** strip *(labelled **Measurement result** in the above illustration)*:
  - ***Distance*** is the direct path length from your current start position to your current end position
  - ***Time*** is the time it takes sound to travel that direct path length at the temperature you entered via ***Settings > File settings > Climate***
  - The ***X, Y*** and ***Z Offsets*** are the differences between the start and end position X, Y and Z coordinates

Note that the distance measuring tool will remain active all the time its icon - - is highlighted in the ***Tools*** menu so a third click will create a new measurement start position and so on *(CAD users will be familiar with this)*. The tool may be toggled on and off by clicking on the distance measuring tool icon.

Now back to our system design process. We'll continue with an array design …

10) The ***Side*** button *(arrowed 10)* may now be clicked in the ***View*** menu to display a side view of the venue with the grid in place. The first cabinet may now be added by selecting the appropriate V60, V90 or V315 from the ***Cabinet*** drop-down box *(circled)*. The grid ***Inclination*** *(arrowed)* should be set, initially, to aim towards the furthest/ highest listener position – see the thin blue line on the illustration.

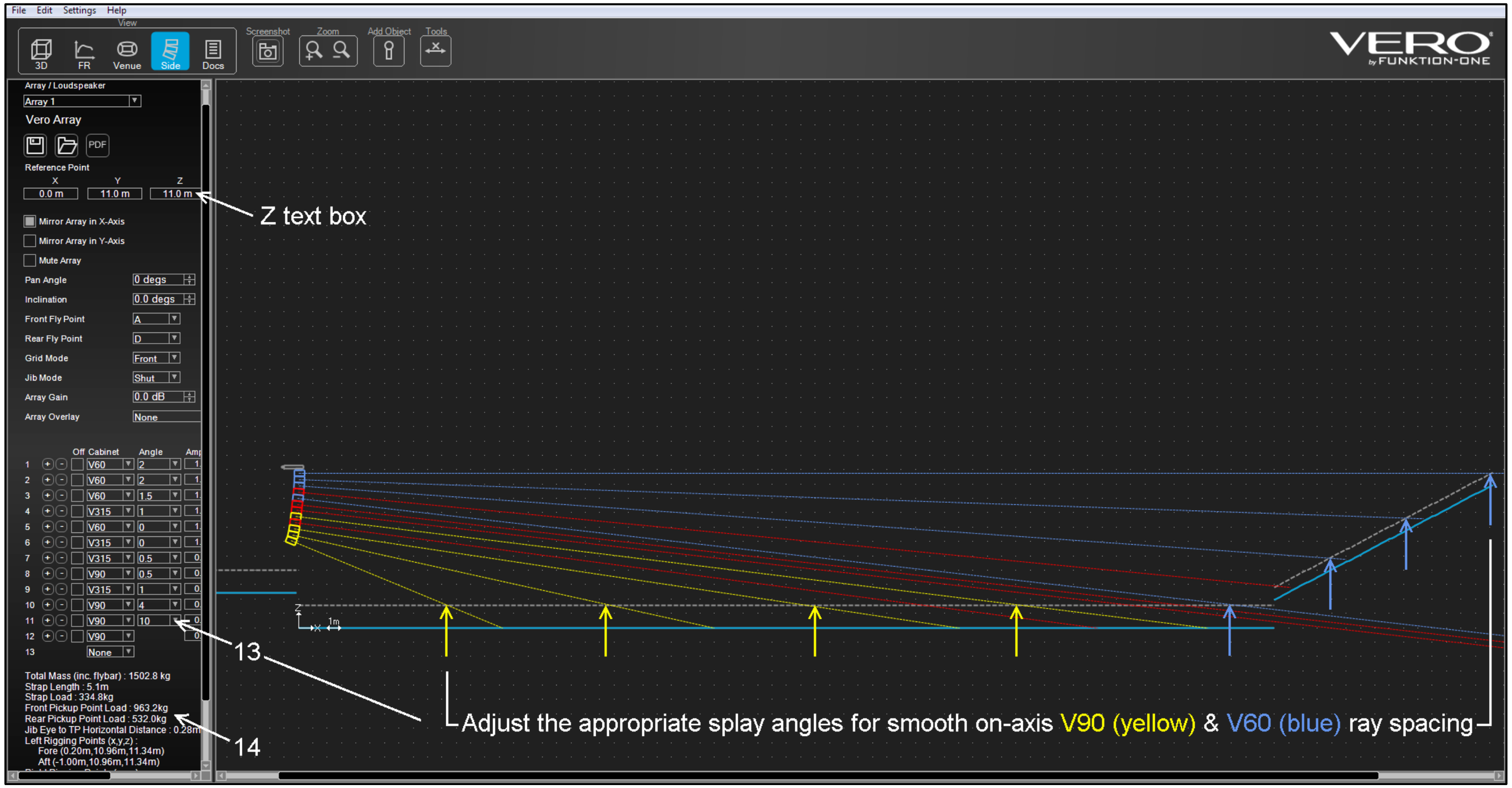
    When working outdoors, a slight rear-audience overshoot is sometimes sensible to hedge against temperature and wind gradients and to ensure that the furthest/highest audience members benefit from the upper array section's maximum GES (Geometric Energy Summation) – see later…



11) If necessary, use the ***Zoom + or - buttons*** *(arrowed 11)* – or click in the plot area and use your mouse wheel. You may need to use the horizontal scroll bar *(if present)*. Also, remember that the left property strip-to-main plot area divider *(under the left Grid inclination arrow above)* may be moved if required.

12) Extra Vero loudspeakers may now be added by clicking the appropriate **+** button *(arrowed 12 above)* and then selecting the required V60, V90 or V315 cabinet from the extra drop-down box.

The inter-cabinet ***Angle*** parameter will also be functional once subsequent loudspeakers are added. See below…

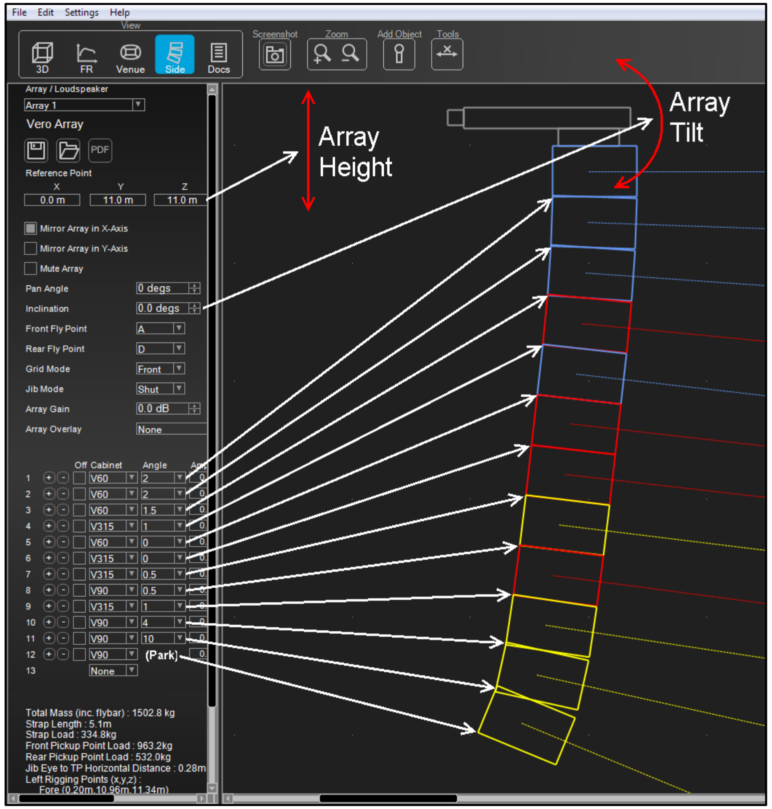


13) Start with an array length of approximately 6% of the distance to be covered. Set the array height for good sight line compliance by editing the grid height figure in the Reference Point's ***Z*** text box *(arrowed top left)*.

Adjust the inter-cabinet angles, initially, for equally-spaced ray positions *(see yellow (V90) and blue (V60) arrows above)* at ear height for each audience area. Each angle *(arrowed 13)* sets the splay angle between a cabinet and the one beneath it. These angles may be adjusted to optimise coverage later

14) Projection will indicate ***Front and Rear Pickup Point*** loading *(arrowed 14)*. An on-screen safety warning will appear if rebalancing is required – see later

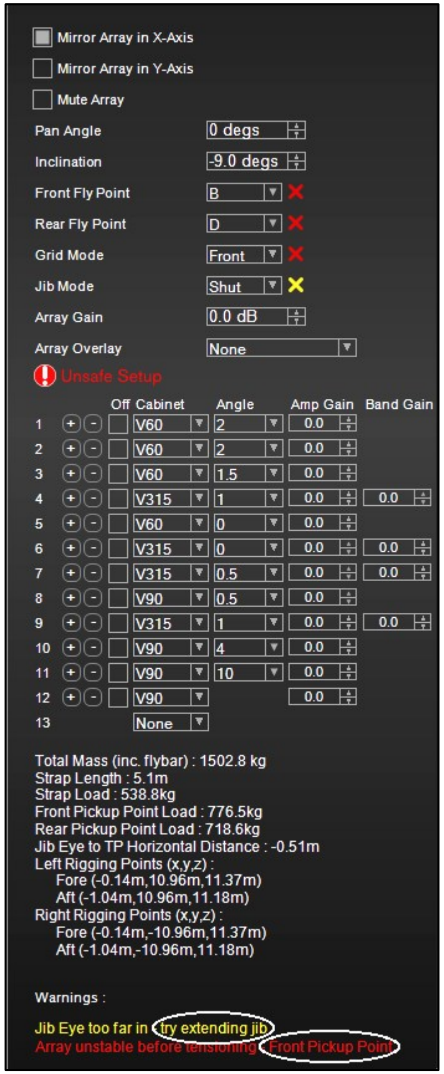
The V315s are placed towards the centre of the array to provide optimal pattern control for good audience impact combined with excellent mid-bass attenuation beneath the array. See ***Appendix C*** for more about under-array bass attenuation.



- Each ***Angle*** setting refers to the Lambda mechanisms at the bottom of the cabinet and sets the splay angle between the cabinet and the one below
- The ***Inclination*** angle sets the grid up-tilt *(+ve)* or down-tilt *(-ve)*. The top cabinet is always parallel to the grid so the grid defines the overall array tilt
- The bottom cabinet's Lambda mechanism is set to ***Park*** – i.e. it is left concealed to keep things looking neat and tidy

Fine adjustments may be required once the coverage predictions have been checked. Rigging properties are updated as you design the array.

Projection rigging safety warnings and solutions



Unsafe

> **Important safety advice**
>
> **Always use Funktion One Projection® software to design your Vero arrays.**
>
> **Projection will alert you to any undesirable (yellow) or unsafe (red) rigging properties, highlight any parameters that require attention and hint at possible solutions.**

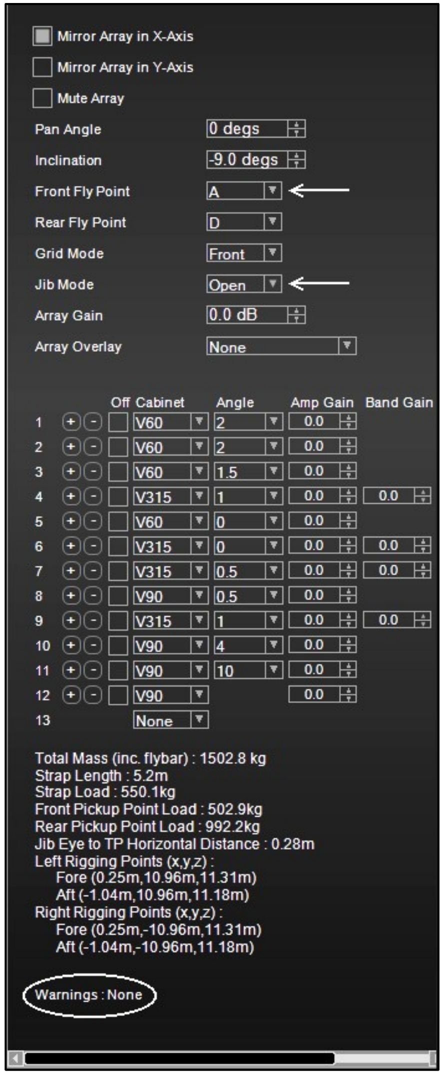
⇦ Projection will display an **Unsafe Setup** warning - with hints and information at the bottom left of your screen - if the design needs attention.

"Unsafe" setups can usually be fixed by adjusting the ***Front and Rear Fly Point*** positions and/or the ***Grid*** and ***Jib Modes*** *(the jib's rear pull-up extension)*.

In this example *(see left)* both ***Front Fly Points*** *(**Red X**)*, ***Grid mode*** *(**Red X**)* and the ***Jib Mode*** *(**Yellow X**)* have been flagged up for possible attention.

Changing the ***Jib Mode*** from ***Shut*** to ***Open*** fixes the jib problem and changing the ***Front Fly Point*** from ***B*** to ***A*** fixes the lift point problem *(see right)*.

Ideally, the array should be flown at a height and inclination that gives a reasonable balance between front and rear pickup point loads. In practice, of course, sight line considerations influence the choice of array height, and the array often needs to be tilted to avoid echoes from behind the audience. This makes it difficult to balance the front and rear pickup point loads. If a well-balanced hang is unachievable, try not to exceed a 4:1 imbalance – and ***never***, under any circumstances, ignore safety warnings.

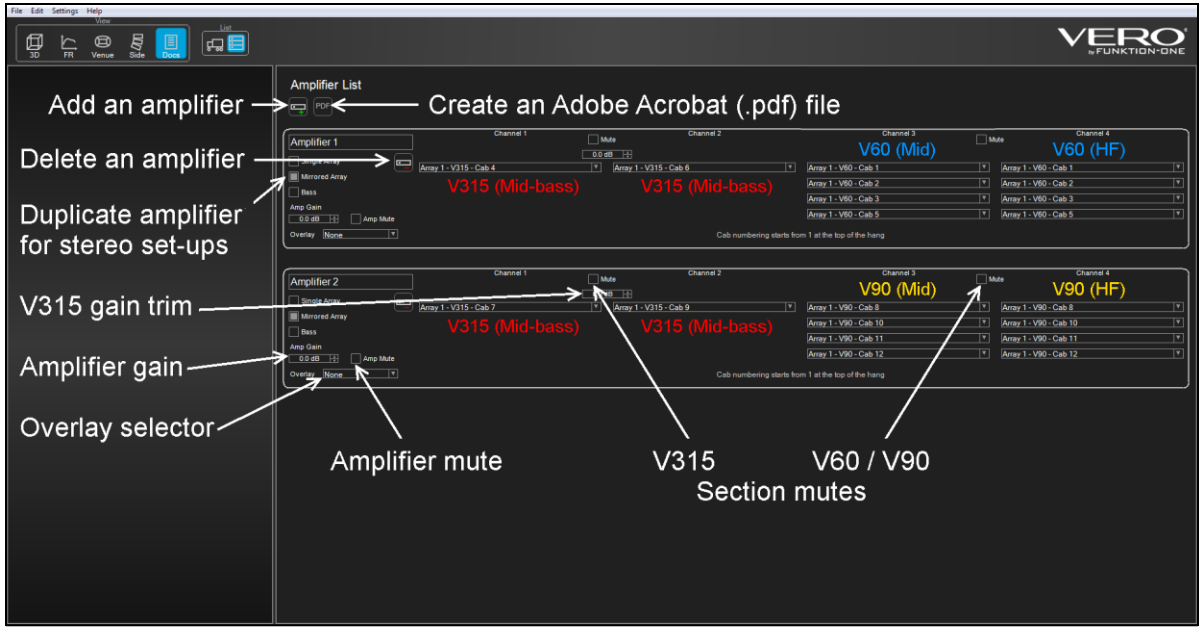


Safe

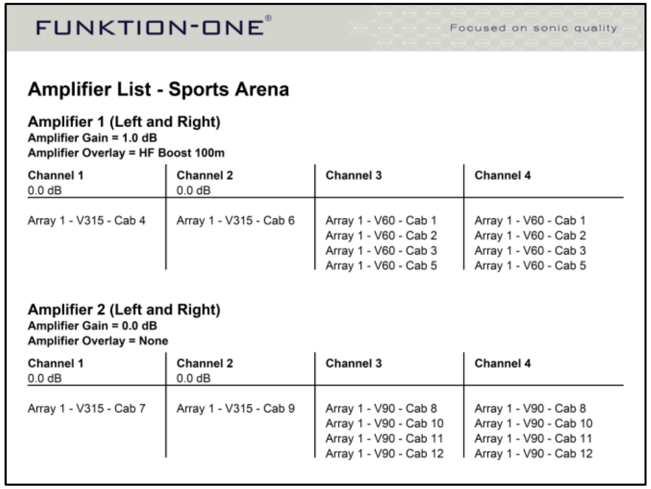
6.2.1 Initial amplifier patch and coverage predictions using Projection

Once you've completed your venue and array designs, you'll need to work out an initial amplifier patch so that ***Projection*** receives the correct power amplifier gain information for its coverage predictions.

A ***Mirrored Array*** button is provided on each amplifier for stereo systems. Only one side of the PA needs to be set for symmetrical systems.



- Click on ***Docs*** then the ***Rack*** symbol *(highlighted in blue)* to view the ***Amplifier List***.
- Amplifiers may be added or deleted using the buttons shown above
- Channels 1 & 2 are normally used for individual V315s
- Channels 3 & 4 are allocated for up to four V60s or up to four V90s per channel
- Drop-down menus allow channels to be allocated and gains to be set
- Initially leave the overall amplifier gains and the V315 *(Ch1 & Ch2)* band gains set to 0dB
- Initially set the amplifier ***Overlay*** selector to ***None***. *(More about this later…)*
- Press the ***PDF*** button to save the ***Amplifier List*** if you wish to save your settings as a pdf at any time


FUNKTION-ONE® Focused on sonic quality

Amplifier List - Sports Arena

Amplifier 1 (Left and Right)
Amplifier Gain = 1.0 dB
Amplifier Overlay = HF Boost 100m

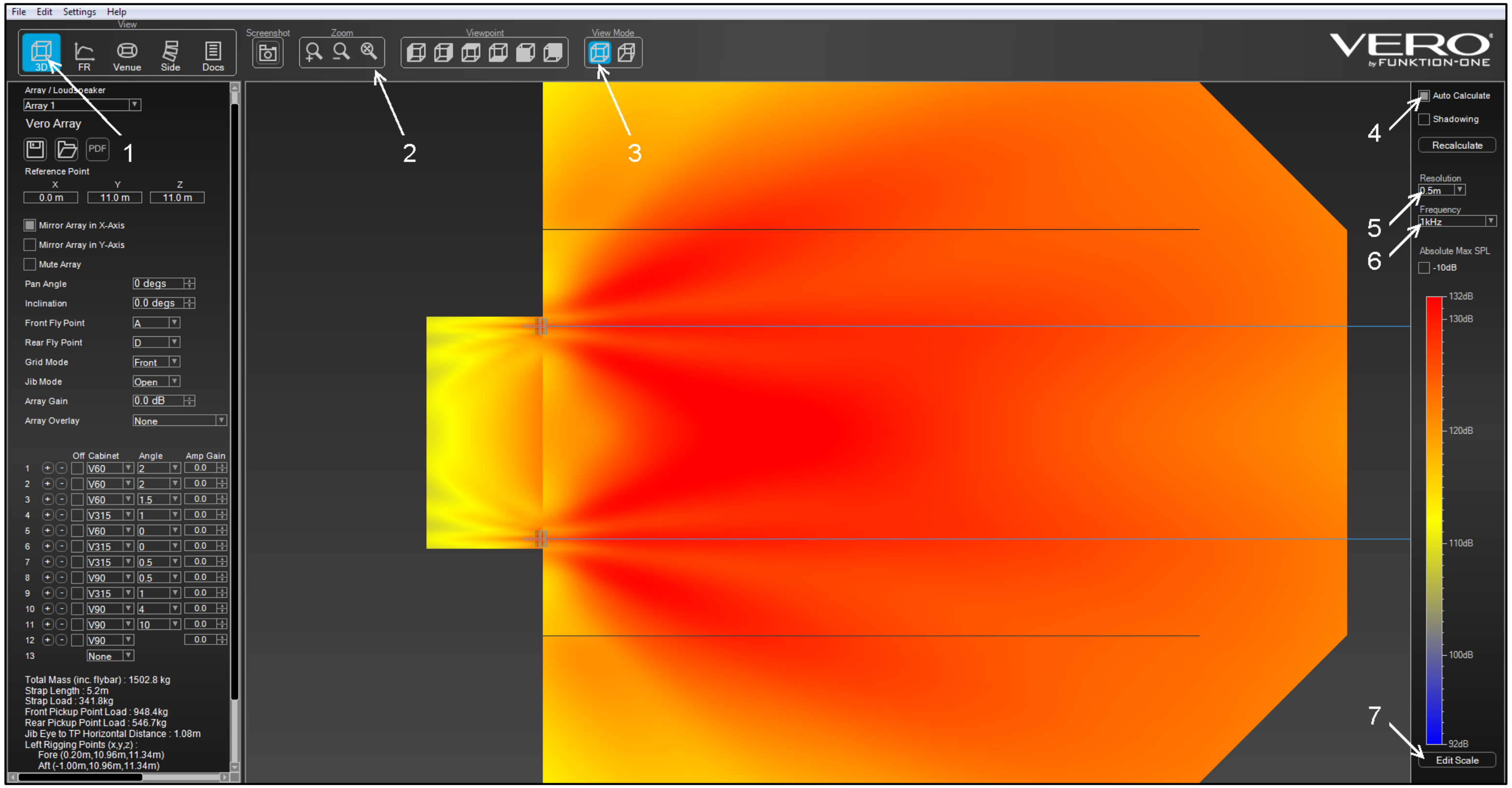
Channel 1 0.0 dB	Channel 2 0.0 dB	Channel 3	Channel 4
Array 1 - V315 - Cab 4	Array 1 - V315 - Cab 6	Array 1 - V60 - Cab 1 Array 1 - V60 - Cab 2 Array 1 - V60 - Cab 3 Array 1 - V60 - Cab 5	Array 1 - V60 - Cab 1 Array 1 - V60 - Cab 2 Array 1 - V60 - Cab 3 Array 1 - V60 - Cab 5

Amplifier 2 (Left and Right)
Amplifier Gain = 0.0 dB
Amplifier Overlay = None

Channel 1 0.0 dB	Channel 2 0.0 dB	Channel 3	Channel 4
Array 1 - V315 - Cab 7	Array 1 - V315 - Cab 9	Array 1 - V90 - Cab 8 Array 1 - V90 - Cab 10 Array 1 - V90 - Cab 11 Array 1 - V90 - Cab 12	Array 1 - V90 - Cab 8 Array 1 - V90 - Cab 10 Array 1 - V90 - Cab 11 Array 1 - V90 - Cab 12

Projection amplifier list pdf – *more about this later…*

You are now ready to predict your Vero system's broadband or 1/3rd octave spl coverage by using Projection's GES *(Geometric Energy Summing)* simulation.



1) Click the ***3D*** button *(arrowed 1)* to view your venue model - initially in plan-view*.
2) Centre and expand the view to fill the display area by clicking on the ***Zoom to fit*** button *(2)*
3) *If your model doesn't appear flat in plan-view, select ***Isometric View*** *(3)* in the ***View Mode***
4) Ensure that coverage calculations are updated for every array change made by selecting ***Auto Calculate*** *(4)*
5) Select the required coverage plot resolution from the ***Resolution*** dropdown menu *(5).* 0.5m *(maximum resolution)* has been chosen in this example
6) Select the required broadband or 1/3rd octave centre frequency from the ***Frequency*** dropdown menu *(6)*. 1kHz 1/3rd octave is shown above
7) Edit the spl amplitude colour scale by clicking on the ***Edit Scale*** button *(7)*. Note that broadband plots will show higher spl figures than 1/3rd octave plots

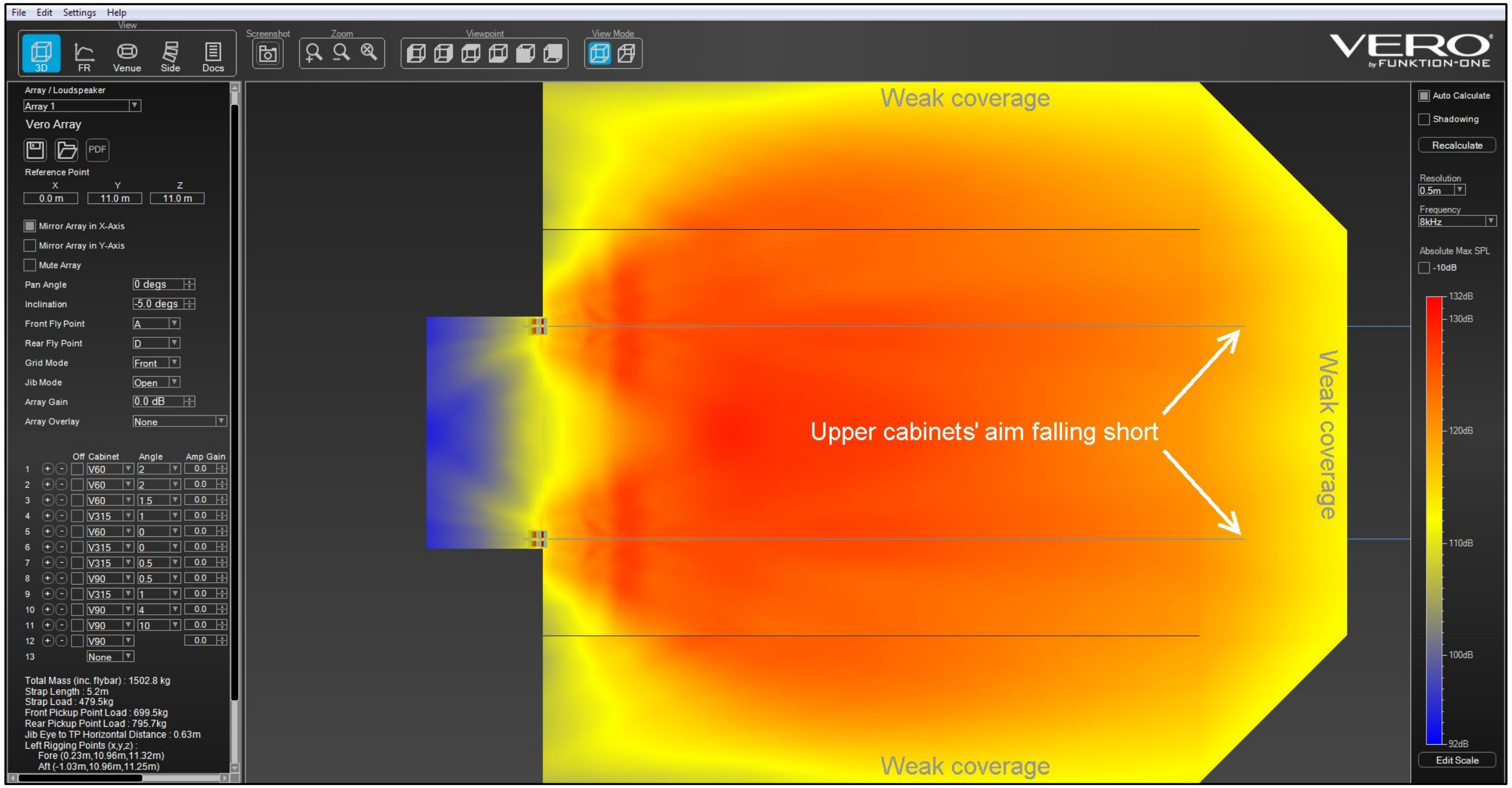
Geometric adjustments

Coverage should be reasonably good from the outset if your array is long enough *(6% of max distance)* to provide good low-mid -to- high-mid spectral balance vs distance and if you've designed your array so that the V60 or V90 on-axis rays are equally spaced through the audience area in Projection's ***Side*** view – see earlier.

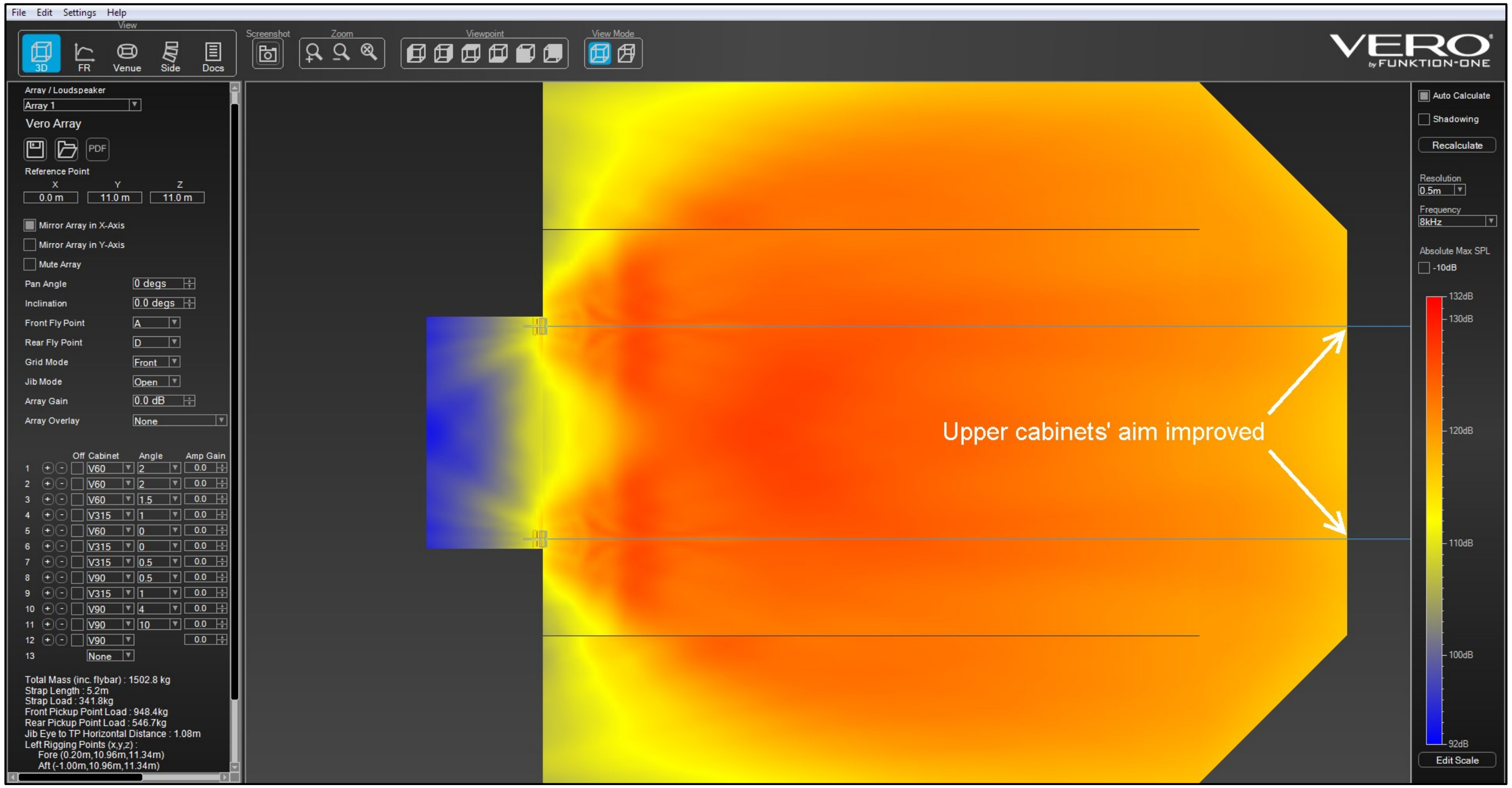
However, coverage may be further improved by fine-adjusting the array inclination and the inter-cabinet splay angles.

Inclination

Array inclination may be used to fine adjust far-seat HF coverage whilst avoiding excessive rear wall reflections. In this example, HF *(8kHz)* coverage is weak *(yellow rather than the required red/orange)* for the upper side and rear seat rows because the array has been tilted down to much:



Array tilted too much - HF coverage weak in the upper side and upper rear seat rows



Array's aim improved - better HF coverage in the upper side and upper rear seat rows

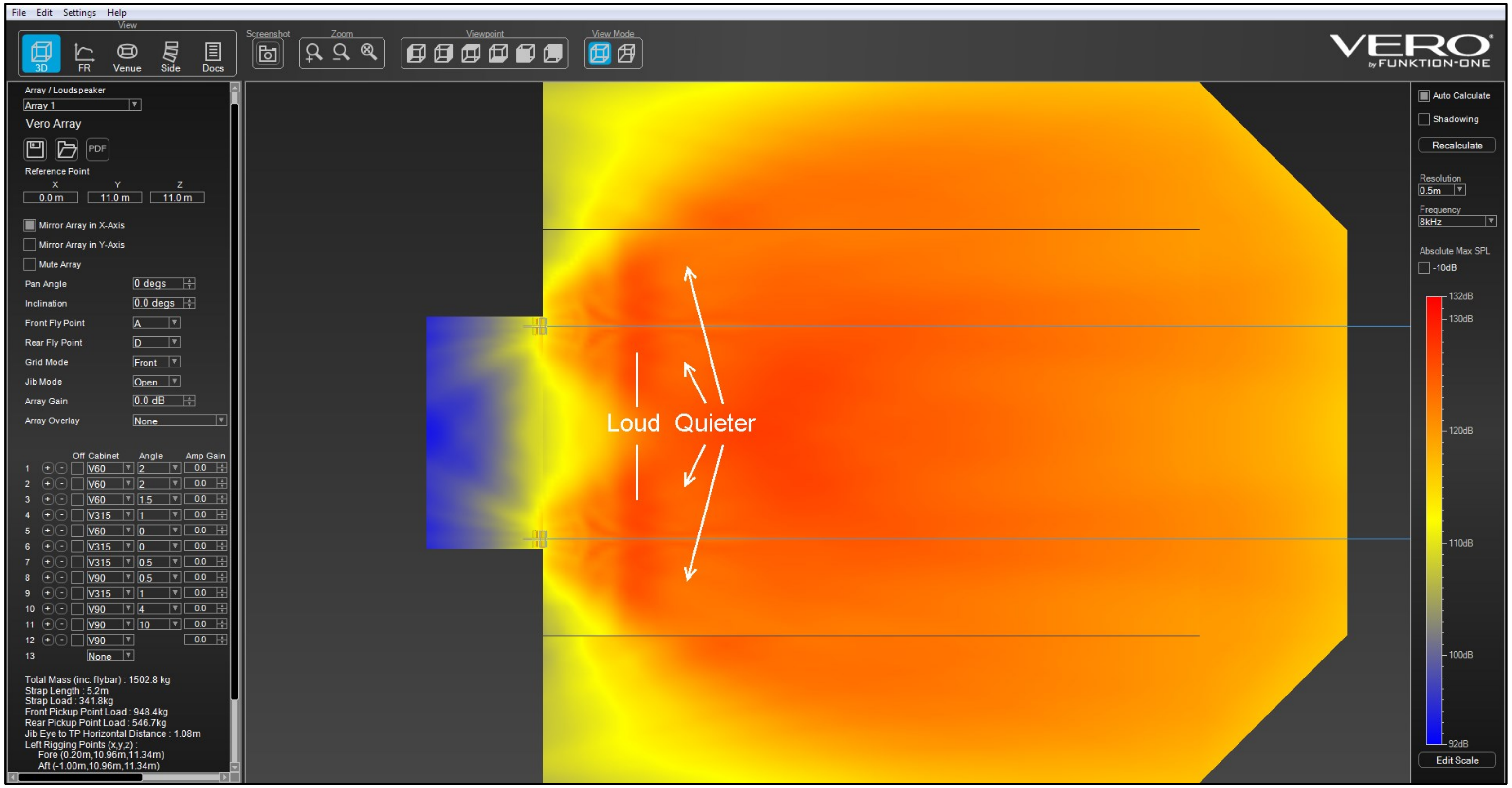
Aiming the upper cabinets towards the highest seats improves far field coverage. If sight line considerations allow, it's best to avoid flying your array too high and then using excessive down-tilt as this can cause unacceptable slap-back and echoes from hard surfaces above the seating.

(Some decrease in level vs distance – e.g. +3dB at the front and -3dB at the back with respect to mid-audience levels - is perfectly acceptable and will sound natural for audiences in the furthest seats. Don't be tempted to overcompensate overall spl and HF levels with distance as this may exacerbate room anomalies)

Coverage smoothing using splay angle adjustments

Inter-cabinet splay angle adjustments may be used to smooth out mid-hi and HF coverage without resorting to excessive amplitude shading or equalisation. Decrease inter-cabinet splay angles to increase high-mid and HF spl in the target audience area or increase inter-cabinet splay angles to decrease high-mid and HF.

HF coverage example

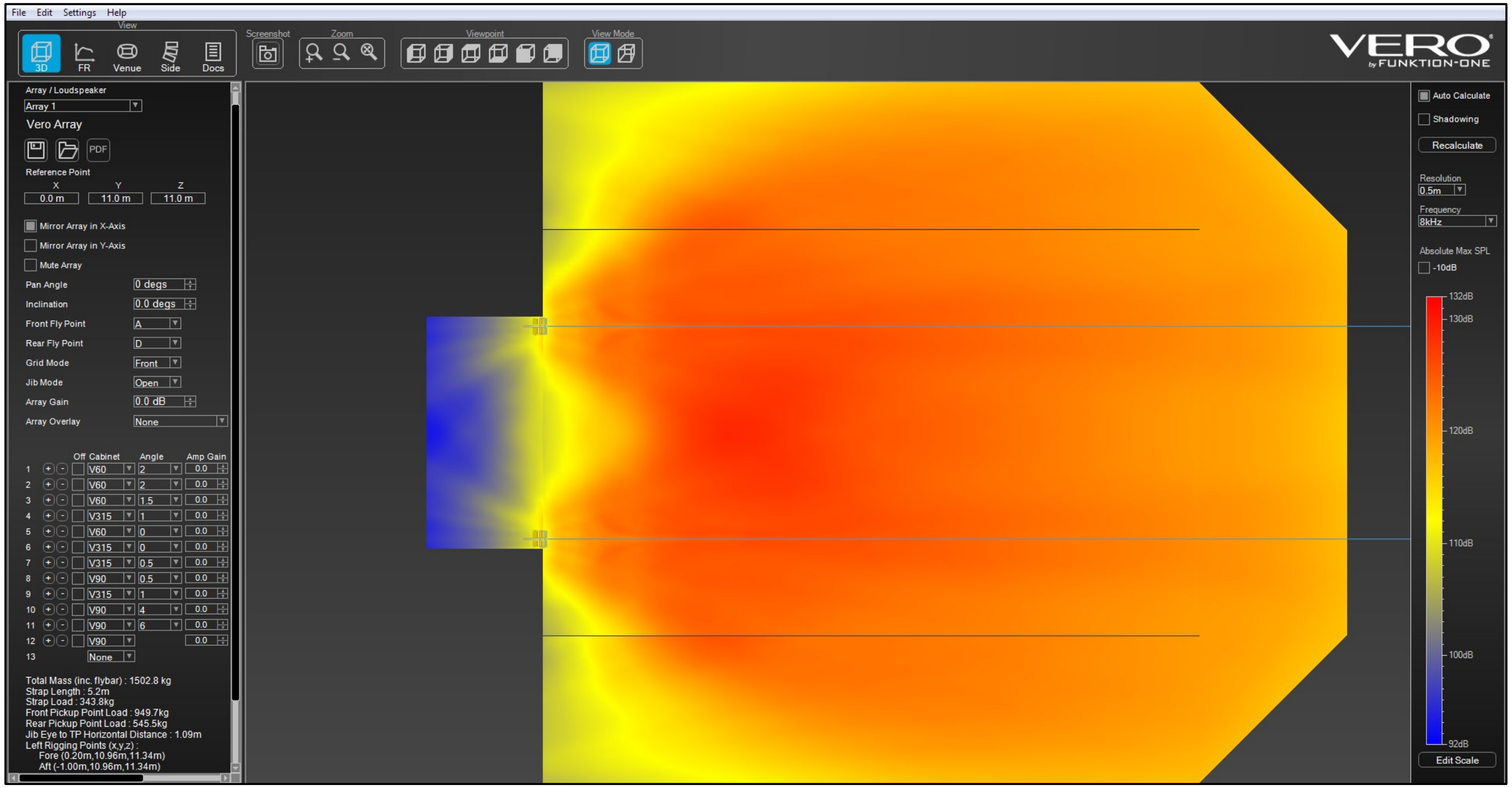


Excessive splay at the bottom of the array causing irregular coverage near the front of the audience

Whilst it is perfectly acceptable to use 12.5° splay angles between V90 cabinets at the bottom of smaller arrays, a tighter splay angle may be required for a smooth transition between the bottom V90 of a large array and the geometrically summed cabinets above it.

The above example illustrates what can happen. Most of the array is providing incredibly smooth, step-free geometric energy summation vs distance but the bottom V90 is splayed a little too much to provide a smooth spl transition between itself and the seven geometrically summing V90/V60 cabinets above it. The bottom V90 isn't following that smooth array curve and is acting like a separate down-fill with its own on-axis hot-spot followed by a quieter inter-cabinet drop in coverage.

Here's the result after we've readjusted the splay angles at the bottom of the array:



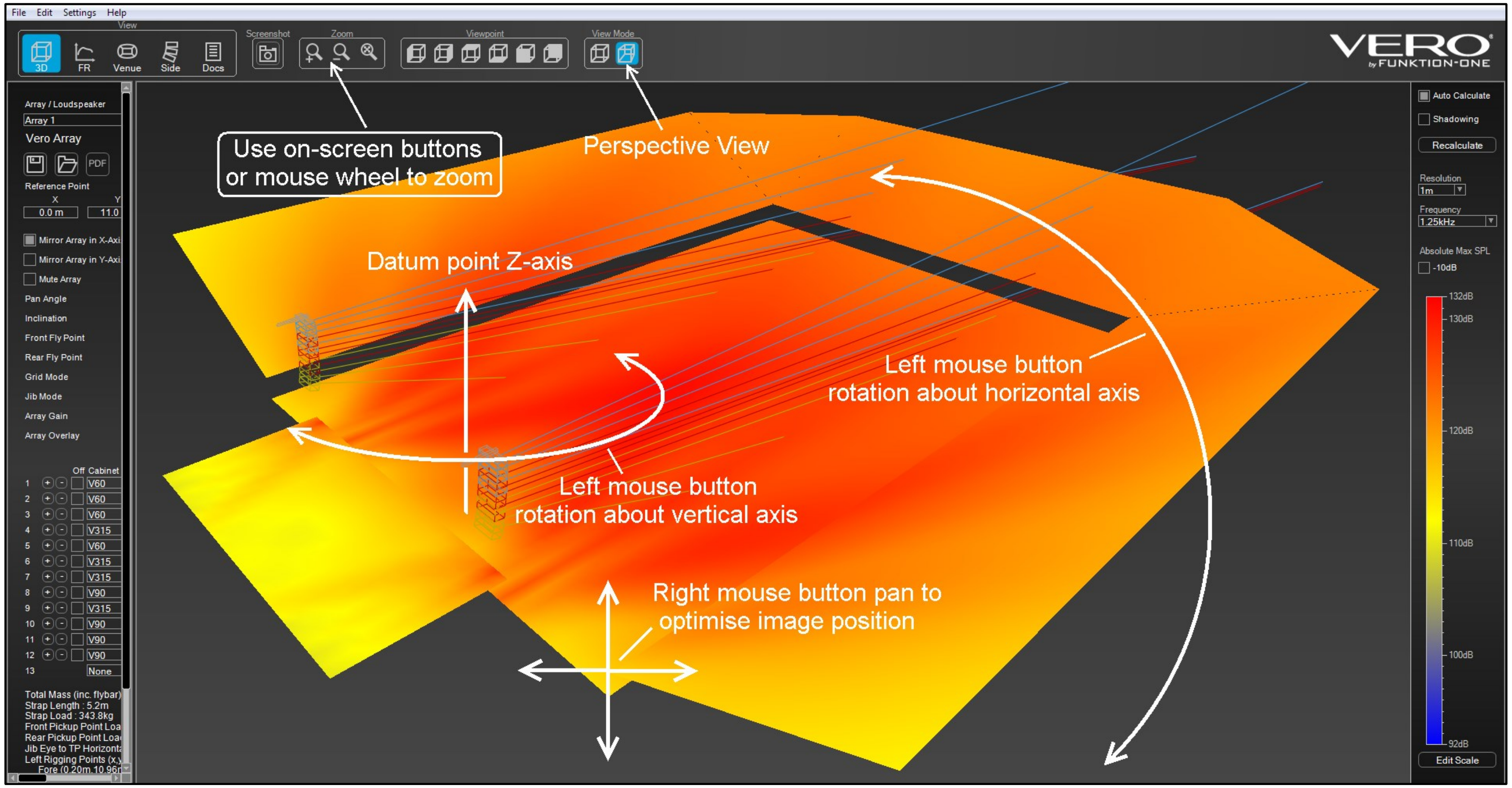
Smoother coverage provided by readjusting the splay angles at the bottom of the array

As HF cut-off is not as sharp at the bottom, more curved, part of an array, we were able to reduce the splay angle between the lower two V90s to smooth out coverage without losing front row HF coverage.

Further tweaking may be possible if you have lots of time, but our main objective is to iron out any obvious coverage anomalies first.

(Obviously, out-fill and centre-fill systems may be required. These could be shorter Vero arrays - or Funktion One Evolution series enclosures, for example, as these employ the same driver components for seamless main-to-fill transitions)

You can now use mouse commands to rotate and roll your venue model in either isometric or perspective view mode to inspect for coverage anomalies.



Mouse operation reminder

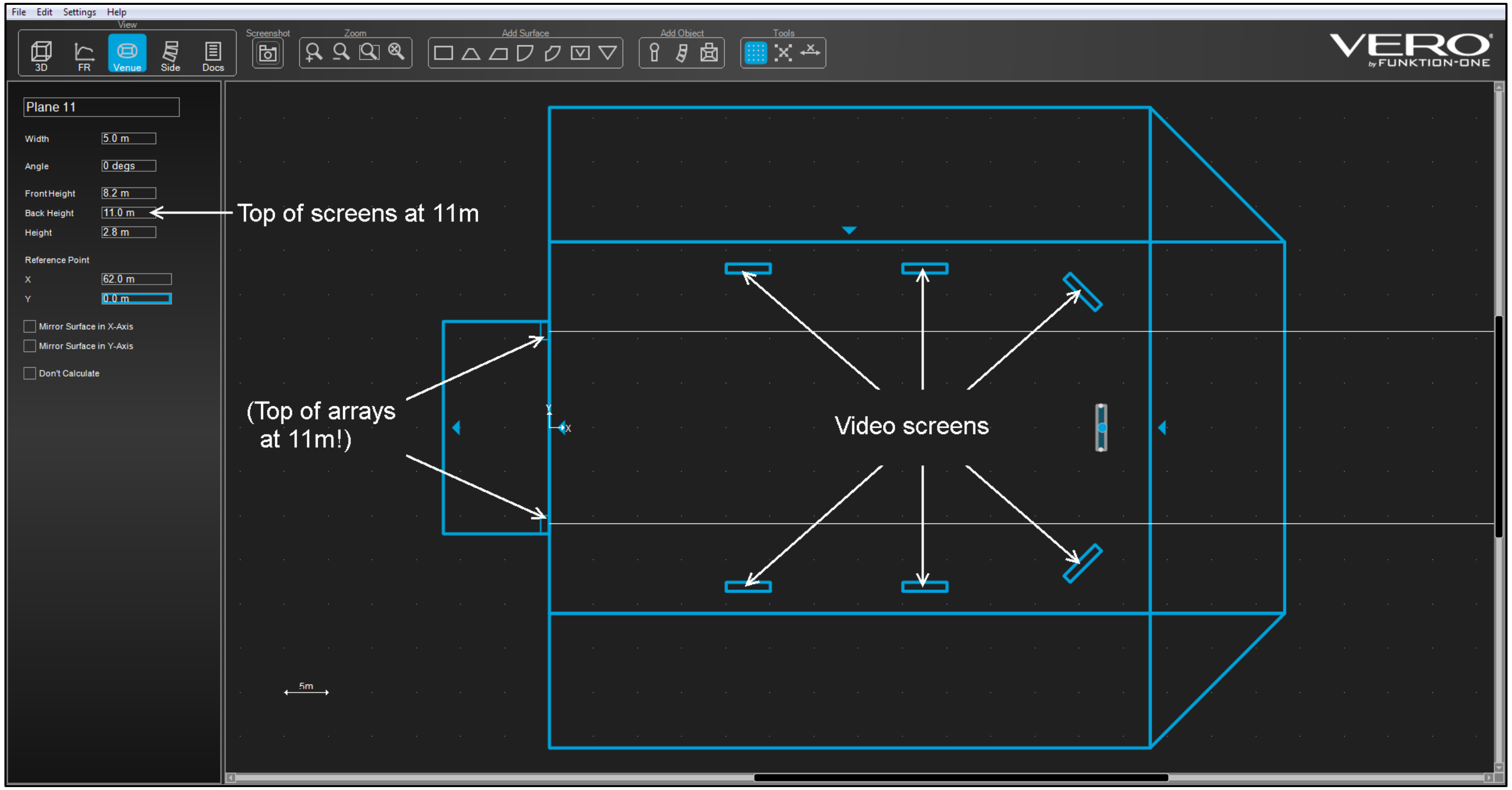
- *Left click and drag from side-to-side to rotate your venue model about its vertical datum point Z-axis – see above*
- *Left click and drag up and down to roll your venue model about its current horizontal centre line – see above*
- *Right click and drag from side-to-side to pan your venue model horizontally*
- *Right click and drag up and down to pan your venue model vertically*
- *Push your mouse wheel forward to push your model away from you (zoom out)*
- *Pull your mouse wheel backwards to pull your model towards you (zoom in)*

Coverage shadowing

Projection's coverage predictions can also take sound shadowing into account. This allows the sonic shadowing of architectural features such as protruding balcony edges or technical facilities like lighting, video screens or score boards to be indicated clearly on the 3D plot.

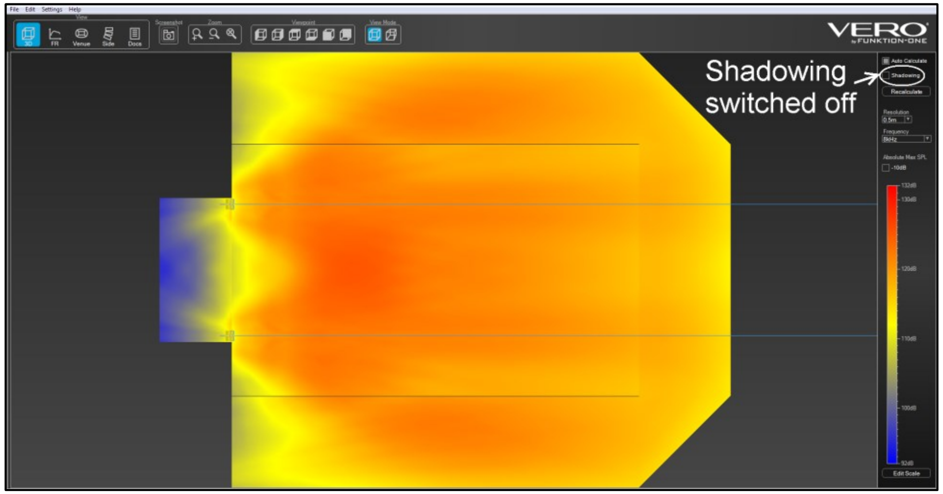
Shadowing predictions also help when designing both balcony and under-balcony seating from single arrays. See "dual-curvature" array suggestions later.

Here's our previous venue example, but with some proposed video screens flown at the same height as our loudspeakers:

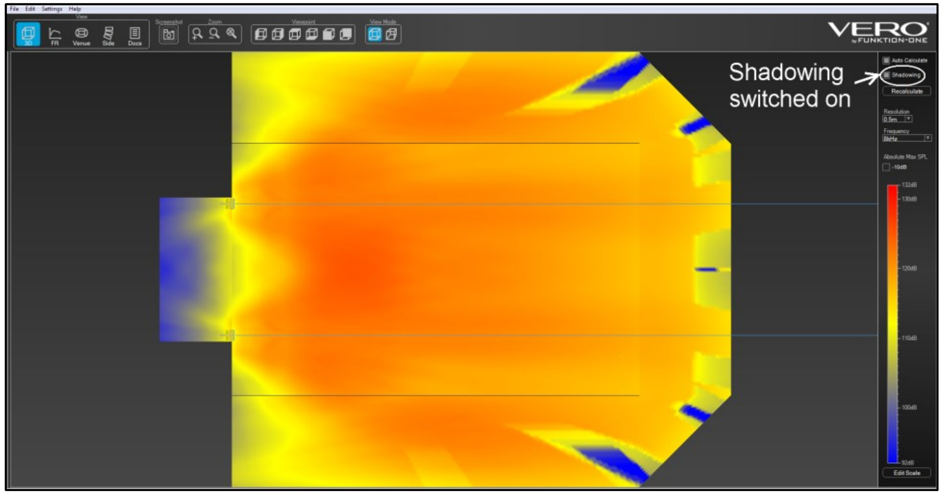


When such proposals are received from a production team, it's useful to be able to show their effects on audience coverage when negotiating to have them raised slightly.

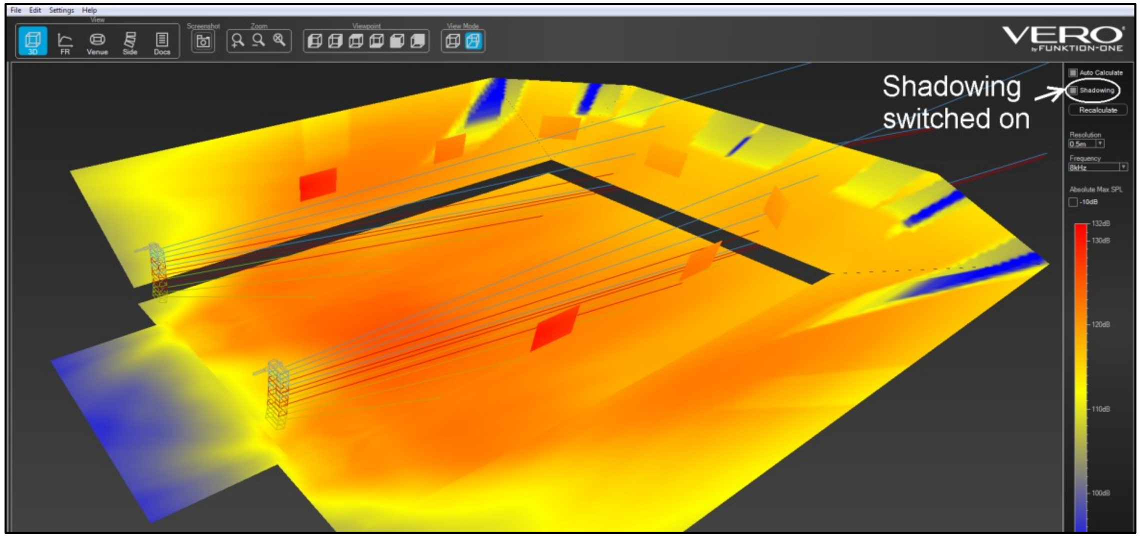
Here are some 8kHz 1/3rd octave coverage plots; the top left view shows normal coverage whilst the top right and bottom views show shadowing predictions:



Normal coverage – isometric plan view



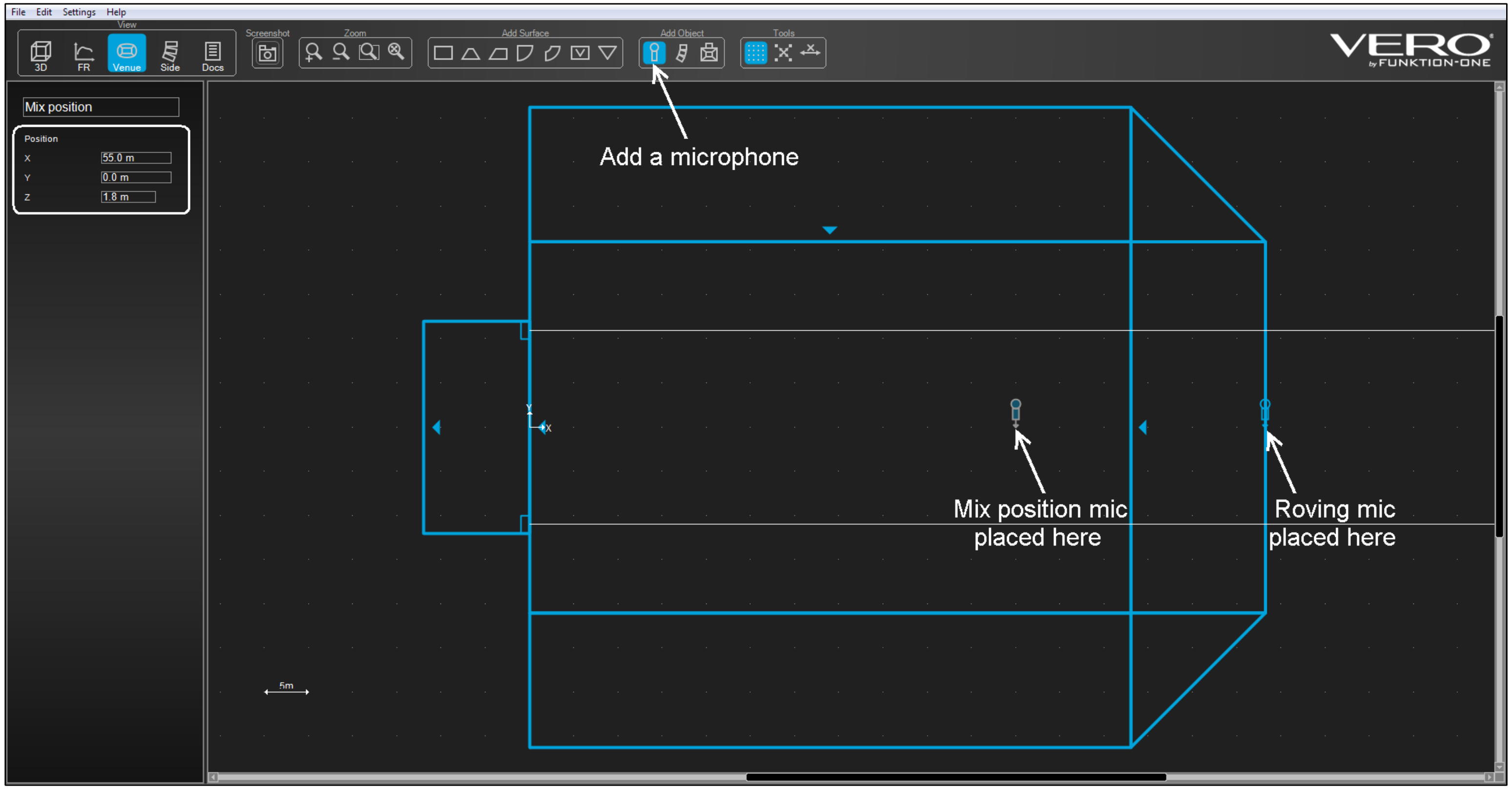
Predicted shadowing effects – isometric plan view



Predicted shadowing effects with offending screen just visible – 3D perspective plot

Spot amplitude response assessments

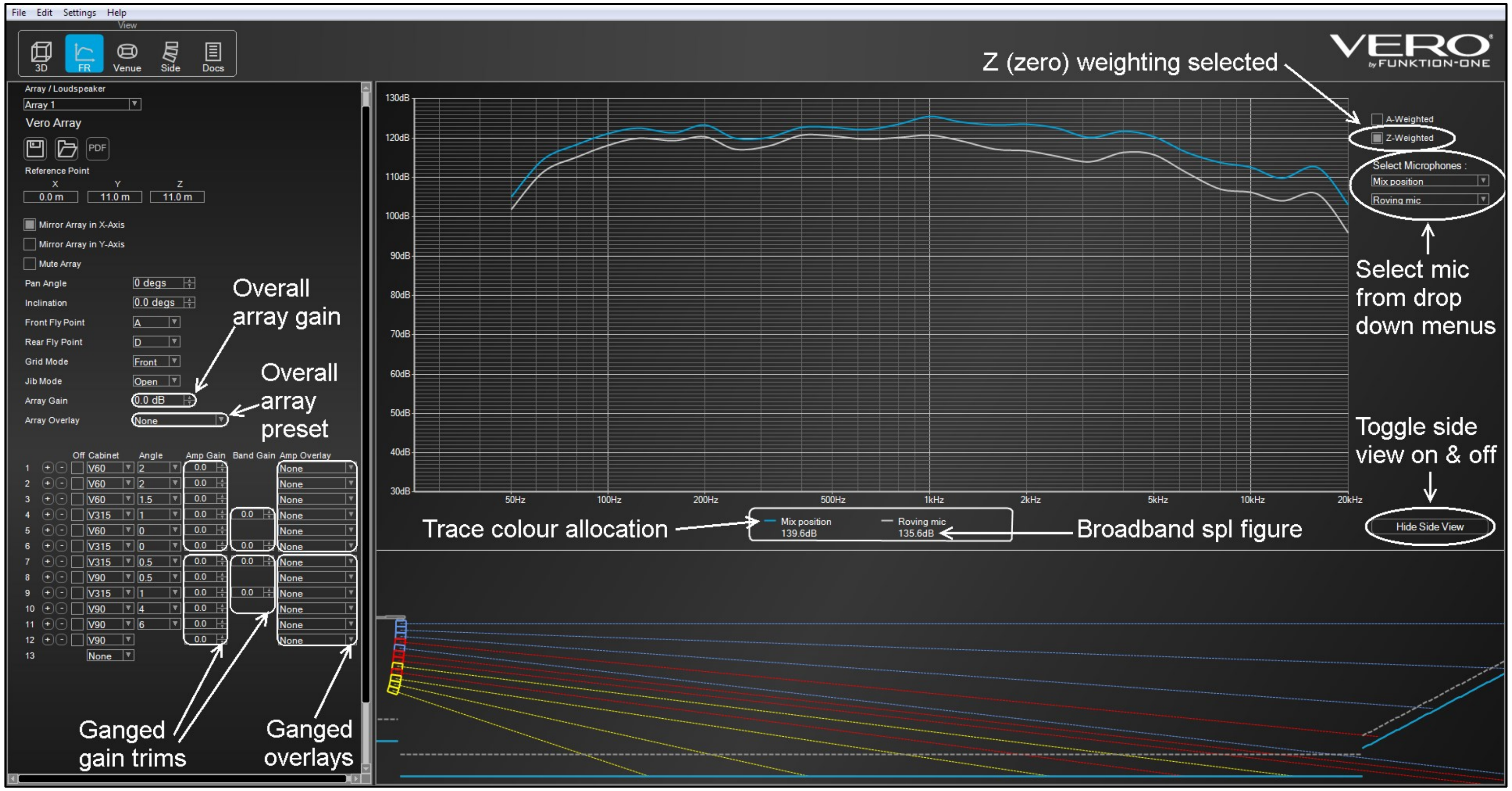
Once you've achieved smooth coverage using geometric adjustments, you can go back to the 2D ***Venue*** view, add "microphones" and look at spot amplitude vs frequency responses in various venue positions.



To add microphones:

- Select the microphone icon - - *(highlighted in blue in the **Add Object** menu)* and click the required venue position. Do this for each microphone.
  *(In this example we've placed microphones on the centre line so that we can assess left-plus-right coverage at the **Mix position** and at the back of the rear bleacher. We've designated the second mic a **Roving mic** as we may wish to move it around the venue (off the centre line) for single-array predictions)*
- Click on each microphone *(right-click if the surface highlights)* and type the required ***X, Y*** and ***Z*** coordinates into the appropriate text boxes for accuracy.
  *(Note that each microphone's **Z** coordinate is its height above the venue's 0m datum, so you need to factor in the relevant surface height plus ear height)*

To plot amplitude responses, select ***FR*** in the ***View*** menu *(highlighted in blue)*.



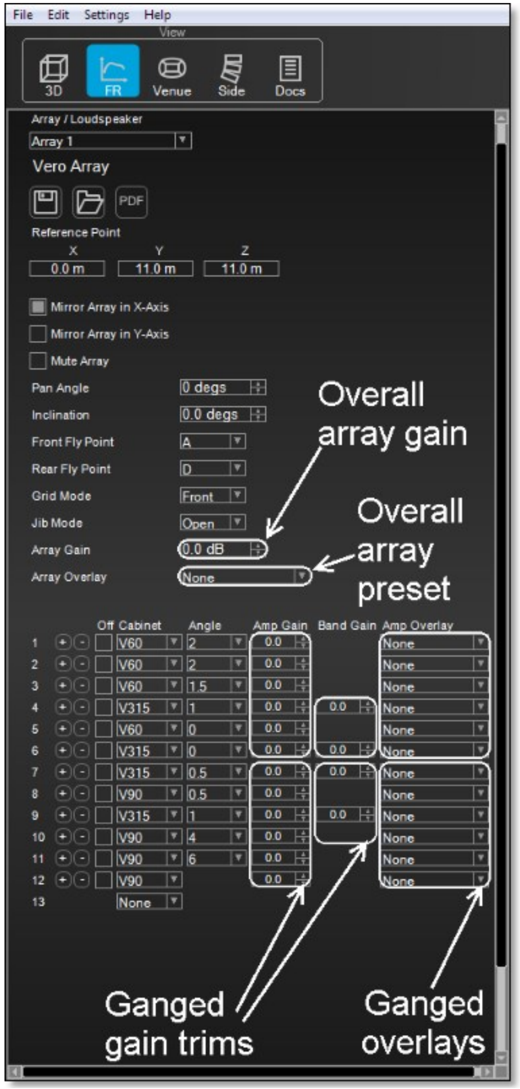
FR View showing typical screen layout *(the left on-screen controls are described later)*

Upper right on-screen controls – from top to bottom

- ***A-weighted*** or ***Z-weighted*** selection. Use ***A-Weighted*** to see A-weighted broadband spl below plot. *(**Z-weighted** - effectively unweighted – selected here)*
- ***Select Microphones*** dropdown menus *(**Mix position** and **Roving mic** selected)*
- ***Show/Hide Side View*** toggle button *(**Show Side View** selected, **Hide Side View** available)*

At bottom of FR plot

- Trace colour allocation and broadband spl figures for each microphone position



Left on-screen controls – top to bottom, left to right *(for mirrored array in this example)*

Geometric array controls

- ***Array/Loudspeaker*** dropdown menu to select relevant array *(or mirrored array pairs)* for fine adjustments
- Array ***Reference Point*** – text boxes for fine positional adjustments
- ***Pan Angle*** and ***Inclination*** controls available for fine horizontal and vertical aim adjustments
- ***Front*** and ***Rear Fly Points, Grid Mode*** and ***Jib Mode*** dropdown menus for fine adjustments *(See alert notes earlier)*
- Cabinet ***Angle*** dropdown menu *(in cabinet list at bottom of screen)* available for fine splay adjustments

Amplifier controls – per array

- ***Array Gain*** – global gain control for the whole array *(or mirrored array pairs)* selected
- ***Array Overlay*** – dropdown overlay *(preset)* menu for the whole array *(or mirror pair of arrays)* selected

Amplifier controls – per cabinet

- Cabinet ***Off*** – allows individual cabinets or groups of cabinets to be muted if required. *(Note that muting a cabinet will remove its on-axis ray line from the side view)*

Amplifier controls – per amplifier

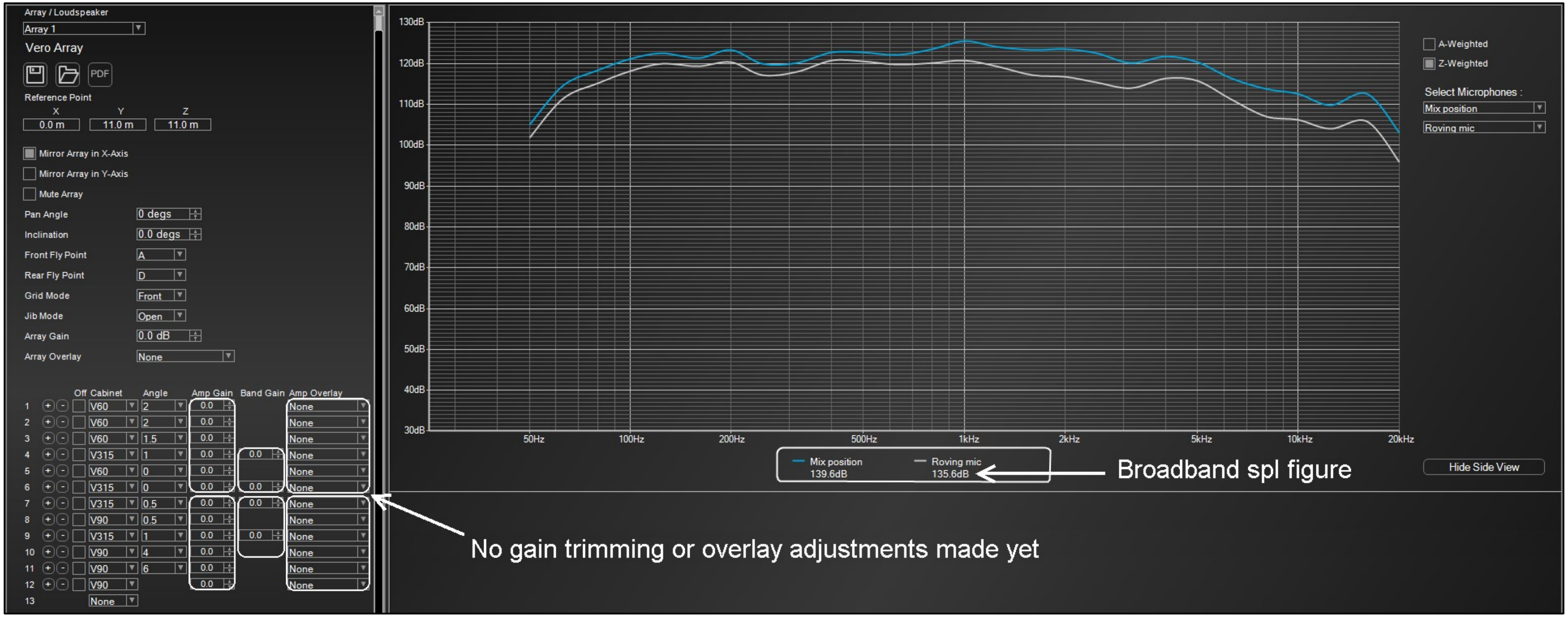
- ***Amp Gain*** – ganged 4-ch amplifier gain trims. Each group adjusts up to 2 x V315 and up to 4 x V60 or V90
- ***Band Gain*** – ganged pairs of channel gain trims. Each pair provides additional mid-bass *(V315)* adjustment
- ***Amp Overlay*** – ganged 4-ch amplifier presets. Each group adjusts preset characteristics for up to 2 x V315 and up to 4 x V60 or V90

If you're designing a large-scale system *(to cover more than 50m distance)* you may wish to run predictions for the most likely range of climatic conditions – accessed via ***Settings > File Settings > Climate***. A little experimentation will give you a feel for the fine amplifier gain and overlay settings you may wish to consider once you're on site.

Note that these direct sound predictions are provided to alert you to *possible* on-site responses. Final decisions must always be made on site during careful pre-sound-check tests, taking all real-world variables into account. For instance, whilst it may seem a good idea to ***fully*** compensate for HF air absorption, splitting the difference *(e.g. compensating by 3dB for a 6dB loss at 8kHz)* sounds far more natural to distant audience members whose ear-brain systems will already have a pre-conceived idea of what sounds acceptable.

See below. ➪

Gain trim and overlay adjustment example…



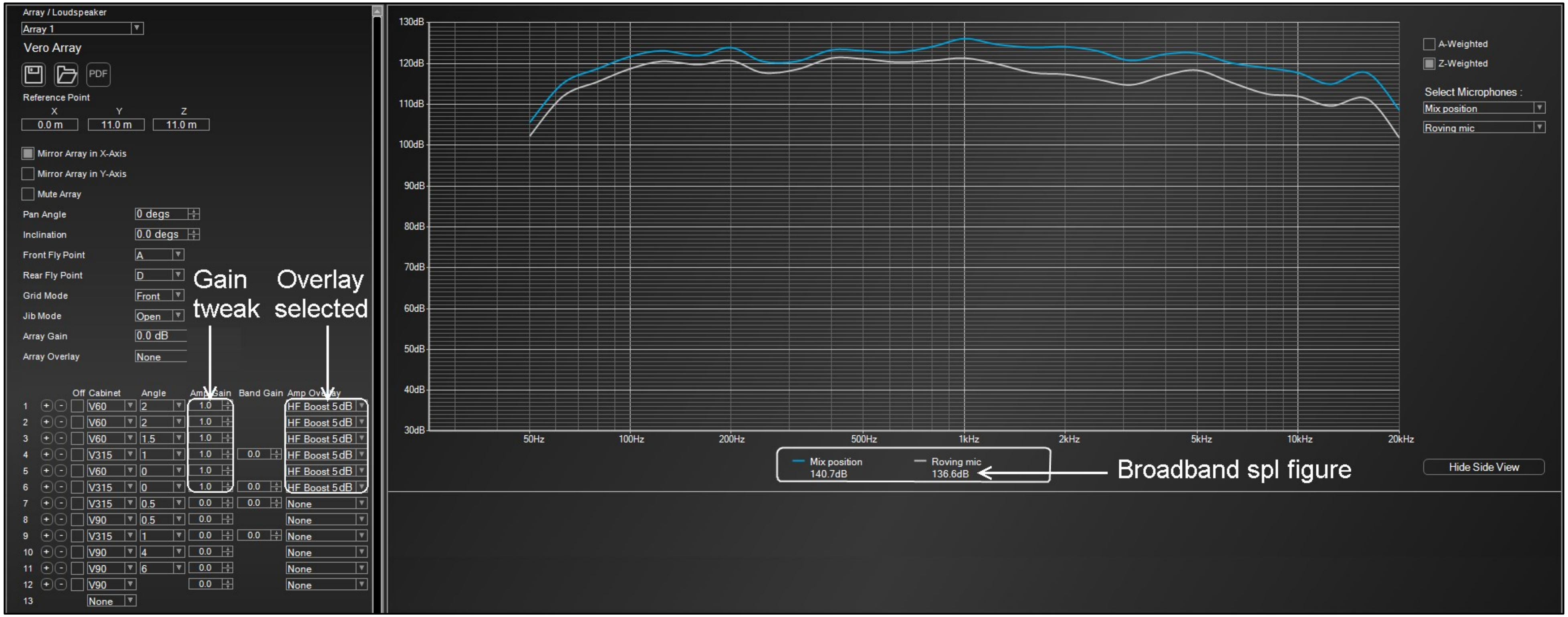
<u>Before</u> - Initial amplitude vs frequency *(FR)* prediction at mix position *(blue curve)* and at furthest centre seat *(white curve)* at 25°C, 60% RH

The plot above shows a typical example of an initial large-scale system measurement with all controls set flat – i.e. No gain trimming and no overlays employed. Note the falling HF response.

A significant portion of the drop at 10kHz will be due to natural air absorption which is 0.112dB per metre at 25°C with a relative humidity figure of 60%.

Although Vero systems have plenty of HF headroom, don't be tempted to use excessive HF boost in a vain attempt to fully compensate for air absorption. Your Vero system will sound far more natural to distant listeners if you only partially compensate. For instance, if the plot shows a 10dB droop at 10kHz, only use about 5dB of boost. Don't try to make the system look flat as it will sound unnatural and will use up too much HF headroom.

Here’s our result after ***Amp Gain*** and ***Amp Overlay*** adjustments:



<u>After</u> - Amplitude vs frequency *(FR)* prediction at mix position and at furthest centre seat with *Amp Gain* and *Amp Overlay* tweaks

Note that we’ve used just 1dB of ***Amp Gain*** trim plus an ***Amp Overlay*** preset on the upper part of the array only. This provides:

- 1dB broadband boost to the upper part of the array. *(Keep these Amp Gain adjustments small to avoid sudden coverage level changes with distance)*
- An HF lift with a gentle slope rising at approximately 3dB per octave above 4kHz – reaching a maximum of +5dB boost at 12kHz.

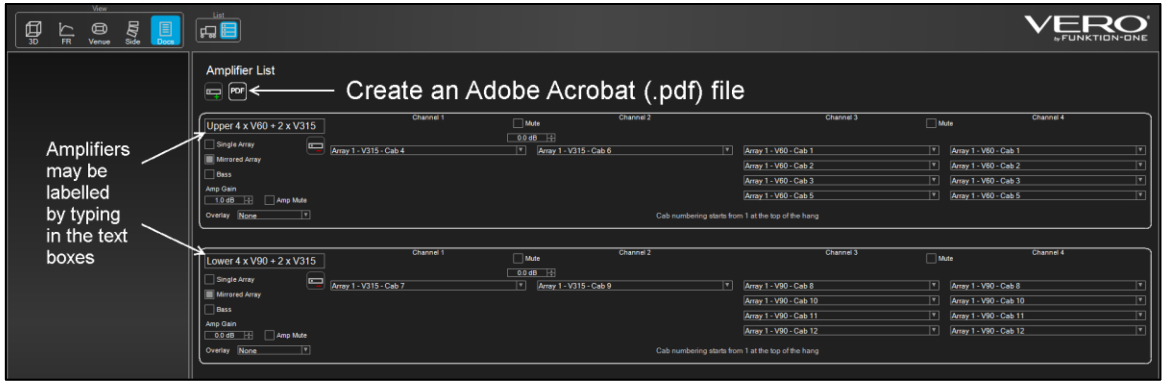
General advice
Using about half of the compensation that predictions and/or on-site measurements would suggest will provide a good starting point for further on-site listening tests and adjustments. Humidity levels change quite slowly so significant in-show adjustments are unlikely to be required.

6.2.2 Projection documents

Amplifier List

When you've finished using Projection for coverage and frequency response prediction work and you feel that your patch and settings are optimised, you can save the ***Amplifier List*** as a printer-friendly .pdf by clicking through ***Docs > Amplifier List*** *(buttons highlighted in blue – see right)* and then clicking the ***PDF*** button *(arrowed)*.

Note that amplifiers may be named by typing in their text boxes *(arrowed)*.



FUNKTION-ONE® Focused on sonic quality

Amplifier List - Sports Arena

Upper 4 x V60 + 2 x V315 (Left and Right)
Amplifier Gain = 1.0 dB
Amplifier Overlay = None

Channel 1 0.0 dB	**Channel 2** 0.0 dB	**Channel 3**	**Channel 4**
Array 1 - V315 - Cab 4	Array 1 - V315 - Cab 6	Array 1 - V60 - Cab 1 Array 1 - V60 - Cab 2 Array 1 - V60 - Cab 3 Array 1 - V60 - Cab 5	Array 1 - V60 - Cab 1 Array 1 - V60 - Cab 2 Array 1 - V60 - Cab 3 Array 1 - V60 - Cab 5

Lower 4 x V90 + 2 x V315 (Left and Right)
Amplifier Gain = 0.0 dB
Amplifier Overlay = None

Channel 1 0.0 dB	**Channel 2** 0.0 dB	**Channel 3**	**Channel 4**
Array 1 - V315 - Cab 7	Array 1 - V315 - Cab 9	Array 1 - V90 - Cab 8 Array 1 - V90 - Cab 10 Array 1 - V90 - Cab 11 Array 1 - V90 - Cab 12	Array 1 - V90 - Cab 8 Array 1 - V90 - Cab 10 Array 1 - V90 - Cab 11 Array 1 - V90 - Cab 12

Printer-friendly amplifier list pdf

The ***Amplifier List's*** project name is the name you gave the venue at the initial design stage. If you didn't name the venue, it will come up as ***Untitled Project***. *(This example was called **Sports Arena**)*.

Note that the project name is limited to the 28 characters *(including spaces etc)* that will fit into the venue text box.

*The **Amplifier List pdf** includes the following information on each amplifier (left):*

- ***Amplifier name*** *(or number, if you haven't named it)*
- ***Amplifier gain***
- ***Amplifier overlay***
- ***Channels 1 & 2 band gain*** *(for V315 mid-bass level trim)*
- ***Channels 1 to 4 loudspeaker allocation**. Typical channel allocations are:*

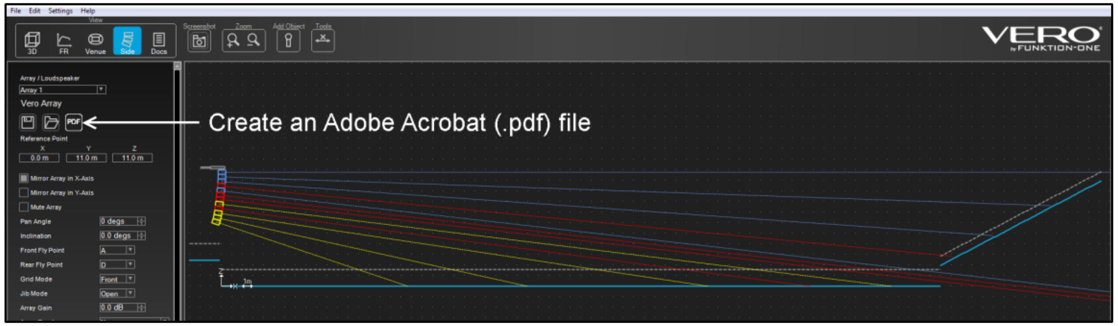
Ch1 = 1 x V315, Ch2 = 1 x V315, Ch3 = 4 x V60 Mids or 4 x V90 Mids in parallel, Ch4 = 4 x V60 HFs or 4 x V90 HFs in parallel.

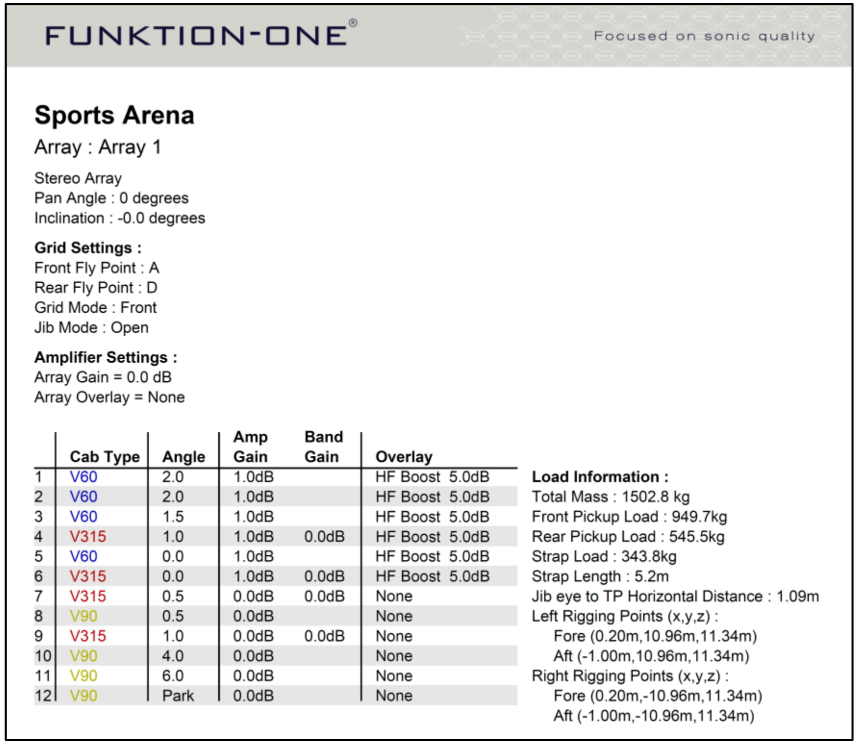
V60 and V90 elements must not be mixed on the same amplifier channel.

Array details

You can save the array details as a 2-page printer-friendly ***.pdf*** by clicking on the ***Side*** button *(highlighted in blue – see right)* in the ***View*** menu and then clicking on the ***PDF*** button *(arrowed)*.

Note that you may need to scroll up to see the ***PDF*** button once the left screen strip is fully populated.





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Sports Arena

Array : Array 1

Stereo Array
Pan Angle : 0 degrees
Inclination : -0.0 degrees

Grid Settings :
Front Fly Point : A
Rear Fly Point : D
Grid Mode : Front
Jib Mode : Open

Amplifier Settings :
Array Gain = 0.0 dB
Array Overlay = None

	Cab Type	Angle	Amp Gain	Band Gain	Overlay
1	V60	2.0	1.0dB		HF Boost 5.0dB
2	V60	2.0	1.0dB		HF Boost 5.0dB
3	V60	1.5	1.0dB		HF Boost 5.0dB
4	V315	1.0	1.0dB	0.0dB	HF Boost 5.0dB
5	V60	0.0	1.0dB		HF Boost 5.0dB
6	V315	0.0	1.0dB	0.0dB	HF Boost 5.0dB
7	V315	0.5	0.0dB	0.0dB	None
8	V90	0.5	0.0dB		None
9	V315	1.0	0.0dB	0.0dB	None
10	V90	4.0	0.0dB		None
11	V90	6.0	0.0dB		None
12	V90	Park	0.0dB		None

Load Information :
Total Mass : 1502.8 kg
Front Pickup Load : 949.7kg
Rear Pickup Load : 545.5kg
Strap Load : 343.8kg
Strap Length : 5.2m
Jib eye to TP Horizontal Distance : 1.09m
Left Rigging Points (x,y,z) :
Fore (0.20m,10.96m,11.34m)
Aft (-1.00m,10.96m,11.34m)
Right Rigging Points (x,y,z) :
Fore (0.20m,-10.96m,11.34m)
Aft (-1.00m,-10.96m,11.34m)

Printer-friendly array pdf – Page 1

The ***Array pdf - page 1*** includes the following information *(see left)*:

- ***Venue/project name*** *(**Untitled Project**, if you haven't named the venue)*
- ***Array*** *(**Stereo Array** if it's mirrored) for:*
  Pan Angle (toe-in/out)
  Inclination (tilt)
- ***Grid Settings*** *for:*
  Front Fly Point (A or B), Rear Fly Point (C or D)
  Grid Mode (Front, Mid or Rear)
  Jib Mode (Open, Half or Shut)
- ***Amplifier Settings*** *for:*
  Array Gain
  Array Overlay
- ***Array table*** *(from top to bottom of array) showing:*
  Cabinet type
  Inter-cabinet angle (between cabinet and its neighbour below)
  Amplifier gain (per 6 x cabinets)
  Band Gain (per pair of mid-bass cabinets where applicable)
  6 x cabinet amplifier Overlay (preset)
- ***Load Information*** *(also illustrated on the **Array pdf – Page 2** drawing – next page)*
  Including Left and Right Rigging Point X, Y and Z positional coordinates

Design Notes

<u>G</u>eometric <u>E</u>nergy <u>S</u>ummation relies on the appropriate summation at the listening position – usually provided by regular on-axis mid and HF ray spacing throughout the audience area.

Our design example provides regular on-axis ray spacing.

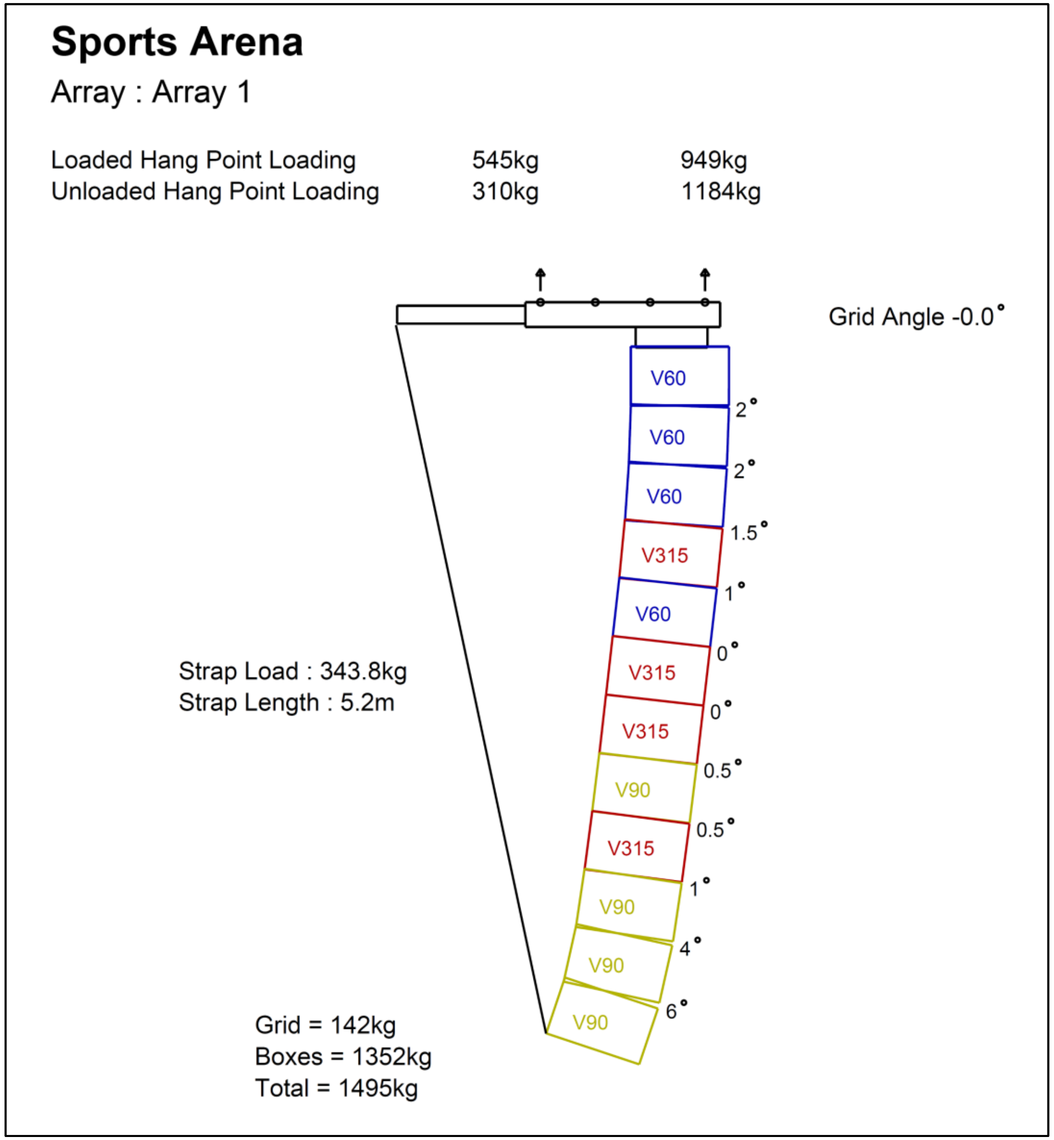
From top to bottom, our mid and HF coverage is catered for by:

- ***4 x V60s** at 2°, 2°, 3° and 0° inter-cabinet angles*
- ***3 x V315** mid-bass opening at 0°, 0° and 1° inter-cabinet angles**
- ***2 x V90s** at 2° and 1° inter-cabinet angles*
- ***1 x V315** mid-bass opening at 1° inter-cabinet angle**
- ***2 x V90s** at a further 5° inter-cabinet angle (the bottom cabinet is set to park)*

**The resultant ray spacing is often greater than a casual glance at the angles would indicate – due to V315 mid-bass cabinet openings within the array.*

Rigging Notes

- The ***Loaded Hang Point Loading*** indicates the flying point loads with the cabinets fully tensioned with the rear ratchet strap
- The ***Unloaded Hang Point Loading*** indicates the flying point loads during rigging before any rear tensioning - when the cabinets are hanging free
- The top cabinet is always flown parallel *(at 0°)* to the grid
- Note that the load figures do not include cable weights

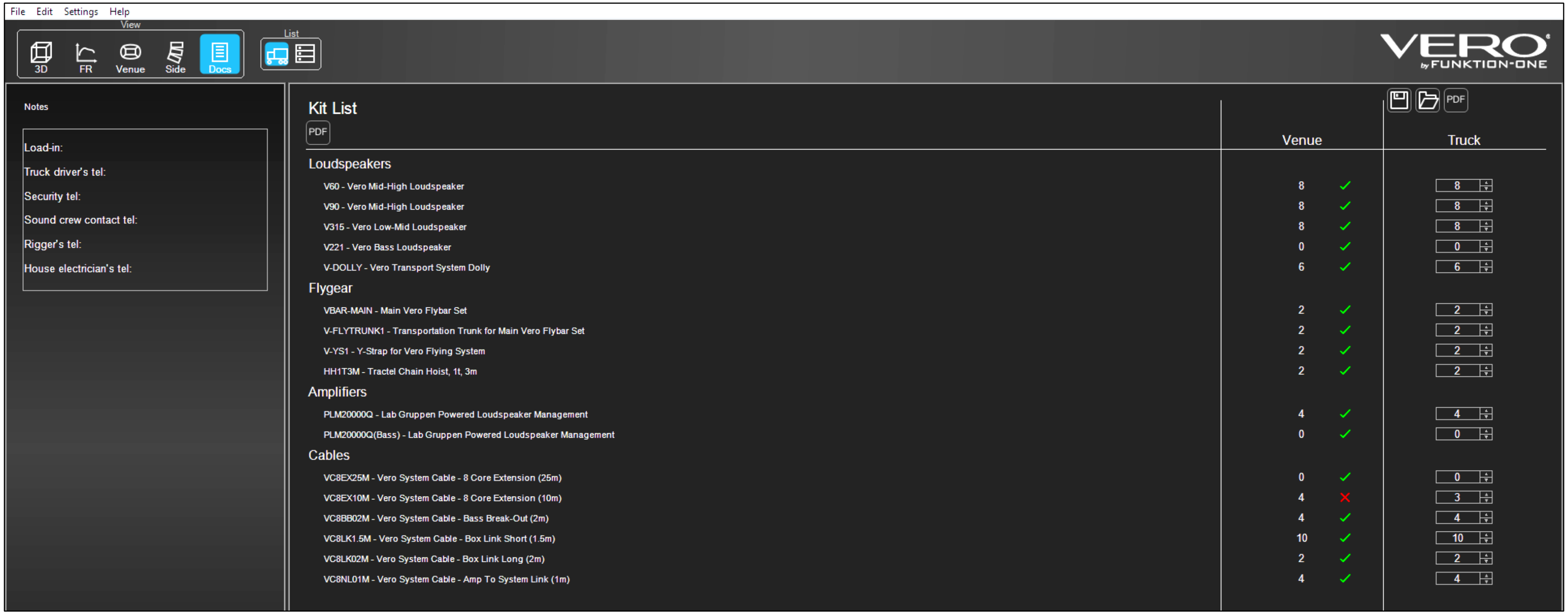


Printer-friendly array pdf – Page 2

Kit List

And finally, if you click on ***Docs*** *(highlighted in blue in the* ***View*** *menu)*, then the ***Truck icon*** *(highlighted in blue in the* ***List*** *menu)*, ***Projection*** will display a ***Kit List*** based on your system design. The required bill of materials is listed in the ***Venue*** column.

Projection allows you to update the ***Truck*** column to confirm that the kit is available or has been loaded onto the truck. Any shortfalls are indicated by a red cross beside the relevant item – note that Projection has flagged up that one ***8-core Extension (10m)*** is missing from ***Cables*** section below.



Kit List screen

A printer-friendly .pdf file may be generated from Projection's ***Kit List*** *(see next page)*. Simply click on the ***PDF*** button.

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Venue Kit List - Sports Arena

Loudspeakers

V60 - Vero Mid-High Loudspeaker	8
V90 - Vero Mid-High Loudspeaker	8
V315 - Vero Low-Mid Loudspeaker	8
V221 - Vero Bass Loudspeaker	0
V-DOLLY - Vero Transport System Dolly	6

Flygear

VBAR-MAIN - Main Vero Flybar Set	2
V-FLYTRUNK1 - Transportation Trunk for Main Vero Flybar Set	2
V-YS1 - Y-Strap for Vero Flying System	2
HH1T3M - Tractel Chain Hoist, 1t, 3m	2

Amplifiers

PLM20000Q - Lab Gruppen Powered Loudspeaker Management	4
PLM20000Q(Bass) - Lab Gruppen Powered Loudspeaker Management	0

Cables

VC8EX25M - Vero System Cable - 8 Core Extension (25m)	0
VC8EX10M - Vero System Cable - 8 Core Extension (10m)	4
VC8BB02M - Vero System Cable - Bass Break-Out (2m)	4
VC8LK1.5M - Vero System Cable - Box Link Short (1.5m)	10
VC8LK02M - Vero System Cable - Box Link Long (2m)	2
VC8NL01M - Vero System Cable - Amp To System Link (1m)	4

Notes :

Load-in:

Truck driver’s tel:

Security tel:

Sound crew contact tel:

Rigger’s tel:

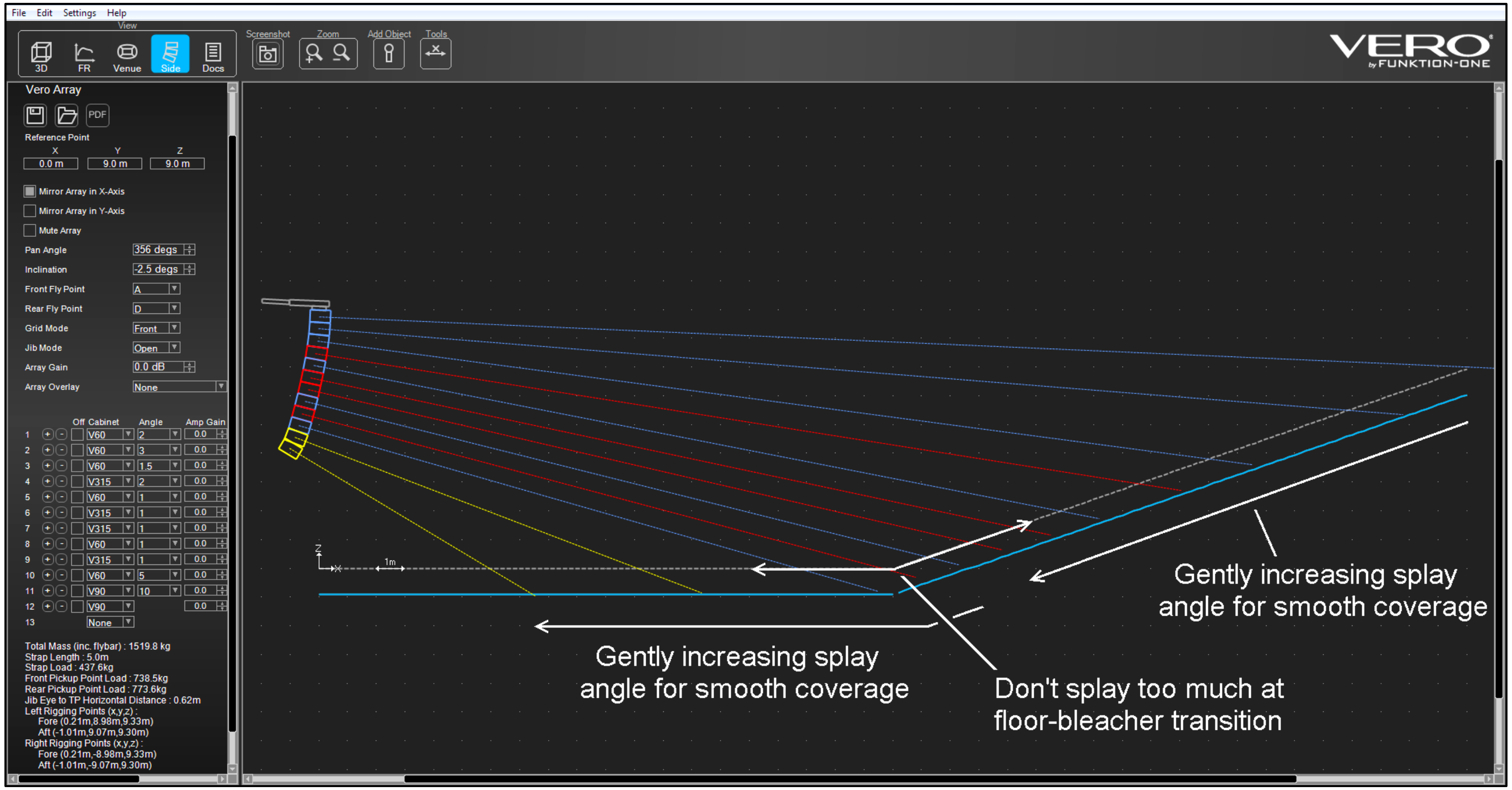
House electrician’s tel:

Printer-friendly kit list pdf

6.3 Multiple plane venues

Venue with bleacher seating

When designing arrays for venues with a bleacher it's easy to make the mistake of starting with a very small curvature for the top of the array and increasingly curving the array from top to bottom. This can cause the coverage to be too strong at bleacher height and too weak at the floor-bleacher transition.



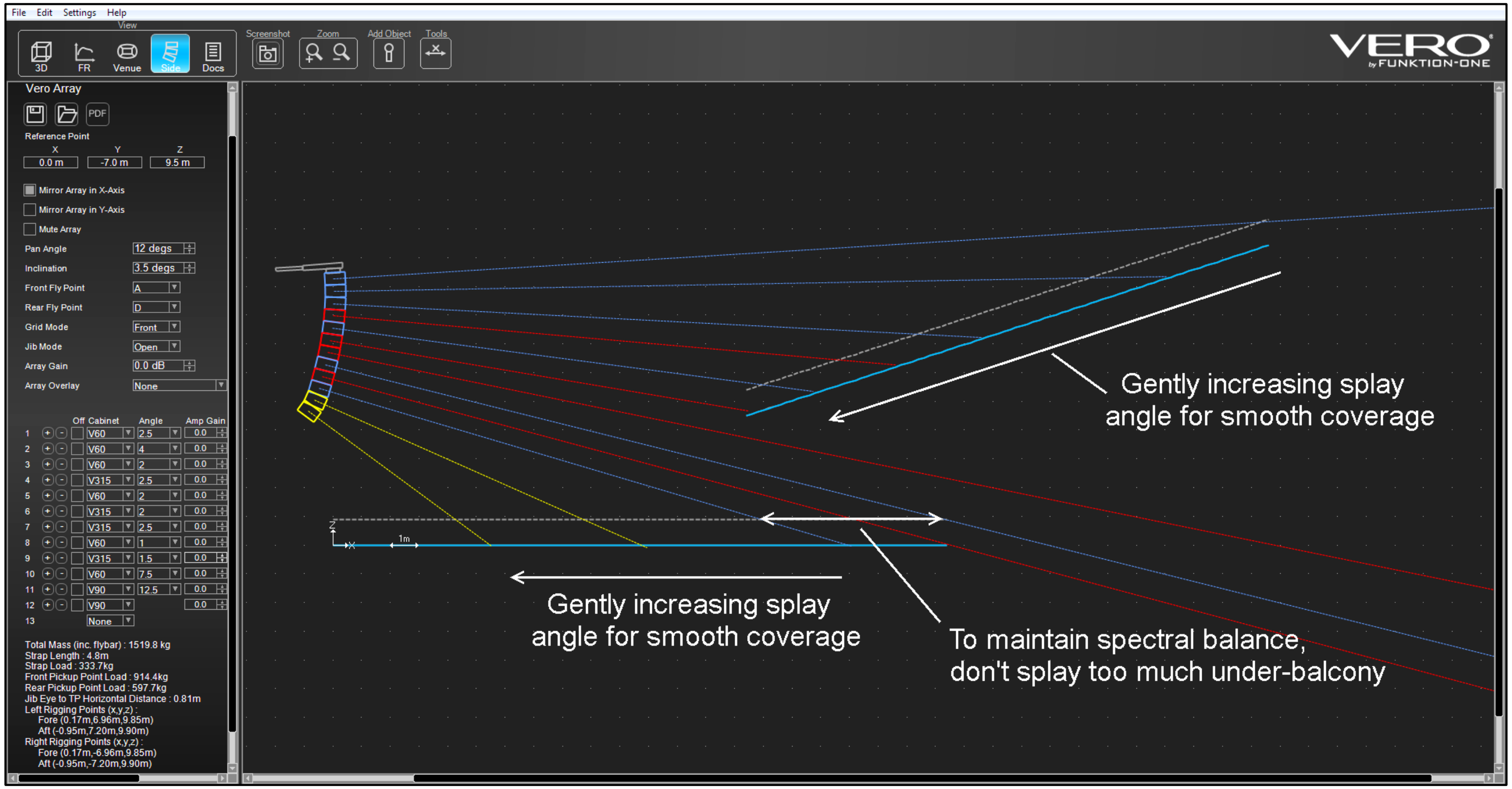
Venue with bleacher seating

When designing Vero systems for venues with bleachers, your design needs to cater for two audience areas – the bleacher and the floor - each with subtly different acoustical characteristics. A simple, single-curvature approach will often result in sound level imbalances, with very strong bleacher coverage accompanied by a weak area at the rear of the floor.

You can think of the array as needing two curvatures; one to provide smooth coverage for the bleacher and the other to provide smooth coverage from the bleacher-floor transition forwards towards the stage.

Venue with a deep balcony

Again, when designing arrays for deep-balcony venues people often forget that they're designing for two different audience areas; the balcony and the floor.



Venue with a deep balcony

In this example, the array is designed with a very subtle double curvature; a curvature for the balcony and another curvature for the floor, with each section having its own subtly increasing top-to-bottom splay curvature. The array is also tilted up slightly for improved imaging.

Note that V60s are used to maintain intelligibility under the balcony and to provide spectral balance between the under-balcony and mid-floor audience areas.

The 6.5° "hinge" between the balcony and floor curvatures is augmented by two V315s between the front-of-balcony and under-balcony V60s. This weakens the mid-high sound level projected to the balcony's front edge, minimising balcony-edge slap-back. Balcony edges tend to be acoustically small compared with V315 wavelengths so making use of the space between the balcony and floor curvatures for two V315s doesn't cause audible slap-back.

6.4 V315 mid-bass considerations

The V315 mid-bass advantage

Most vertically arrayed systems compromise mid-bass performance by squeezing the low frequency section into the main cabinet. This results in inefficient mid-bass response that lacks impact and slam. The Vero system has a dedicated, full-spec., triple-horn-loaded mid-bass section for the kind of no-compromise performance you've come to expect from Funktion One's high-efficiency designs.

V315s are usually flown with the main V60s and V90s in a ratio of one V315 for every two V60/90 *(4 x V315, 4 x V60 plus 4 x V90 is typical for a 12-cabinet array)*.

Mid-bass coverage

A typical 12-cabinet arena array may comprise (from the top): 4 x V60, 3 x V315, 2 x V90, 1 x V315, 2 x V90 for good spectral balance throughout the audience.

Vertical coverage angle vs number of adjacent V315s for large-scale arrays

Larger blocks of V315 *(greater than 3)* provide improved pattern control – especially at upper mid-bass frequencies. The appropriate number of adjacent V315s may be chosen to achieve the required vertical coverage. V315 mid-bass directivity increases with frequency and with the number of adjacent V315s.

See ***Appendix B*** for explanatory polar simulations. See ***Appendix C*** for information on achieving mid-bass attenuation under your array if required.

Frequency	4 x V315 block	5 x V315 block	6 x V315 block	7 x V315 block	8 x V315 block
80Hz	*See note 1*	*See note 1*	*See note 1*	**119°**	**98°**
100Hz	*See note 1*	**149°**	**107°**	**87°**	**74°**
125Hz	**149°**	**101°**	**80°**	**67°**	**58°**
160Hz	**98°**	**74°**	**70°**	**51°**	**44°**

Vertical coverage angle vs number of V315s per block

Notes

1) Where acoustically small blocks of V315s are placed mid-array *(i.e. blocks of less than 4 x V315 – or larger blocks at lower mid-bass frequencies)*, adjacent V60 or V90 cabinets will still limit lower mid-bass vertical coverage to around 180°, reducing rear leakage and increasing forward lower mid-bass amplitude by up to 6dB
2) Multiple V315 blocks are often positioned along the array so that the upper mid-bass coverage follows the V60/V90 curvature

Mid-bass-to-top crossover alignment

Note, of course, that V315–to-V60/V90 crossover phase alignment is maintained wherever the V315s are placed in the array as the V315s, V60s and V90s are running in unison at the 200Hz crossover frequency. Stand-alone V315 figures at 200Hz are, therefore, not included above.

6.5 Multiple horizontal V60/V90 arrays

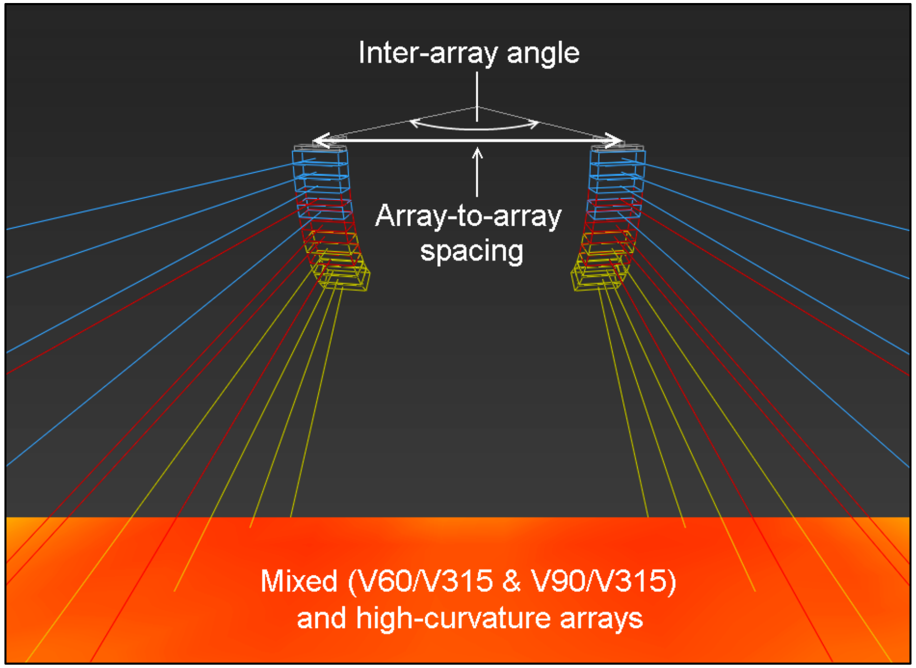
Multiple Vero arrays should be optimally positioned and splayed to provide smooth coverage.

Close arrays

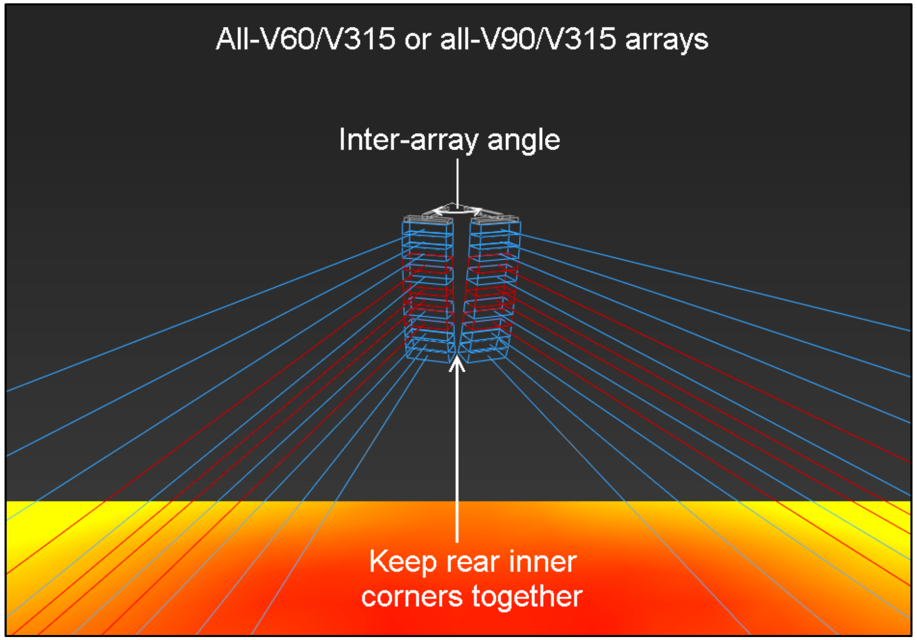
- You may place all-V60/V315 or all-V90/V315 arrays close together
  But…
- Try to keep the lower rear inner corners together ⇨

Mixed arrays *(i.e. systems that include V60s and V90s in the same array)*

- Don't place mixed arrays close to each other
- If V60s & V90s are deployed in the same array, the arrays must be spaced 5-15m[1] apart (centre-to-centre) as shown below



Mixed arrays - V60/V315/V90 arrays shown above



Close arrays - V60/V315 arrays shown above
(Don't use this closely-spaced configuration for mixed arrays)
⇧

Close arrays - Don't use for mixed arrays - *(See notes top left)*	**Recommended Inter-array angle *(between upper cabinets)***	**Overall mid-high coverage**
All V60-to-all V60	60° closely-splayed or spaced	120°
All V90-to-all V90	90° closely-splayed or spaced	180°
All V60-to-all V90	75° closely-splayed or spaced	150°

Estimated horizontal coverage[2] for arrays placed close together

[1] The optimum mixed-array spacing will depend on the array heights *(determined by point positions, sight lines, screen heights, TV production requirements etc)*.
[1 & 2] Always use projection to check the predicted coverage before issuing rigging plots.

6.6 V221 bass system and alignment

How many?

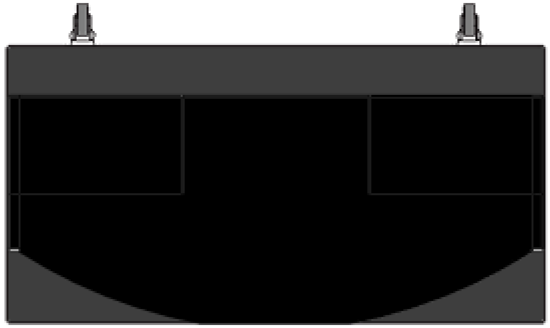
Funktion One bass cabinets and top cabinets are typically deployed in a 1 x V221:2 x V60/V90 ratio, i.e. one V221 for every two Vero main cabinets.

A ratio of 1:2 will keep your V221 bass system within a dB or two of your Vero's low-mid section sensitivity so the preset crossover settings will be optimum.

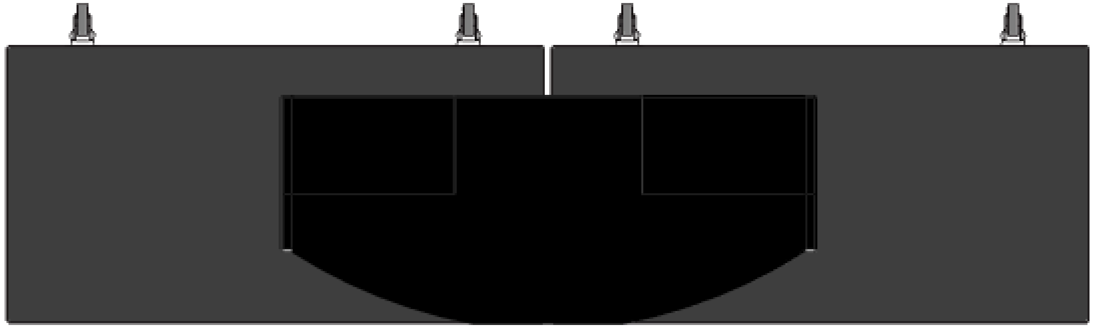
Bass array shapes

For smooth low-mid-to-bass coverage alignment at crossover, it makes sense to:

- Place your bass systems as close as possible to your main arrays - directly beneath the main flown arrays whenever you can
- If safety considerations *(for example, uneven ground)* don't allow bass arrays to be stacked high, you may need to place them side-by-side, making the bass array wider than the main array. If your bass array ends up significantly wider than the Vero array, arrange the bass cabinets in an arc so that they follow the curvature of Vero's front grille – i.e. on a radius with the Vero to a common rear point to provide a similar horizontal dispersion through the crossover region. Your bass system can, of course, be time and phase aligned with the Vero array - see ***Section 6.7.2*** later…



Single-width V221 stack under flown Vero array

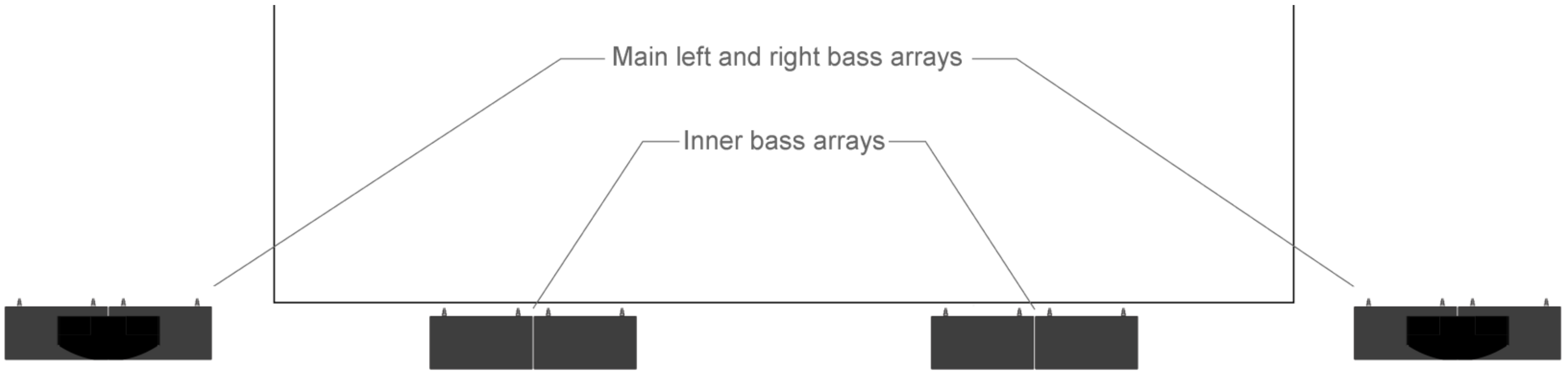


Double-width V221 stack under flown Vero array

Smoothing bass coverage

It is worth allowing for some extra bass cabinets between left and right systems to smooth coverage and avoid a "bass alley" effect up the centre of the venue.

There isn't often a budget for a multitude of closely spaced bass cabinets right along the stage apron – so one solution for smaller stages is to retain some of the bass cabinets from each side and space these inner 2-wide bass arrays equidistant along the stage apron as shown below…



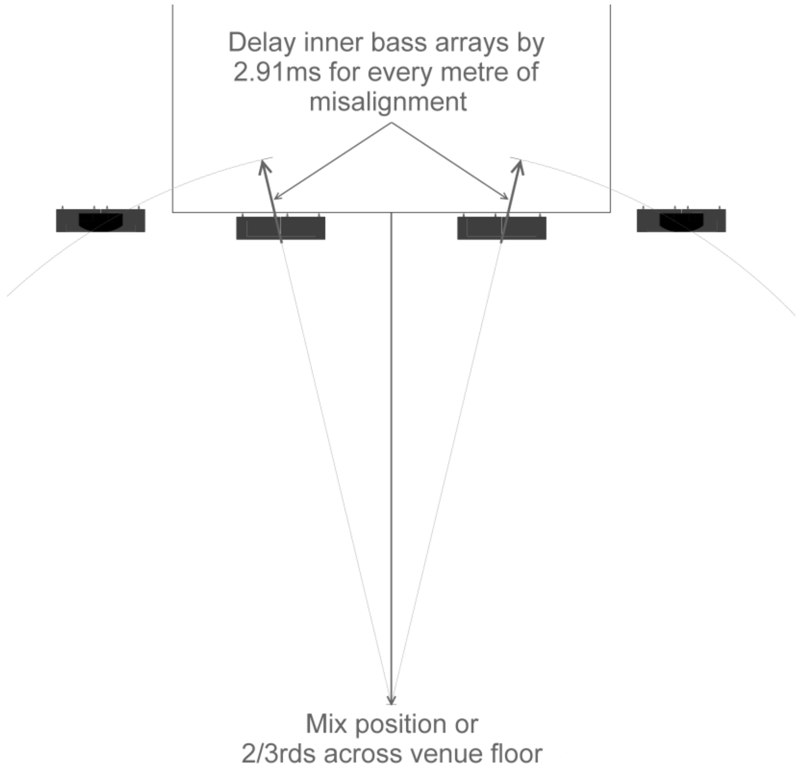
These inner bass arrays also create handy locations for front fills.

Bass alignment options

It is possible to electronically control the main and inner bass arrays for various coverage and impact options depending on the time alignment applied.

- **Point destination alignment**
  Choosing the mix position as the time alignment reference point gives maximum impact at the mix position for bass-loving mix engineers
- **Planar alignment**
  Choosing left and right off-centre reference points produces smoother audience coverage and reduces the central peak

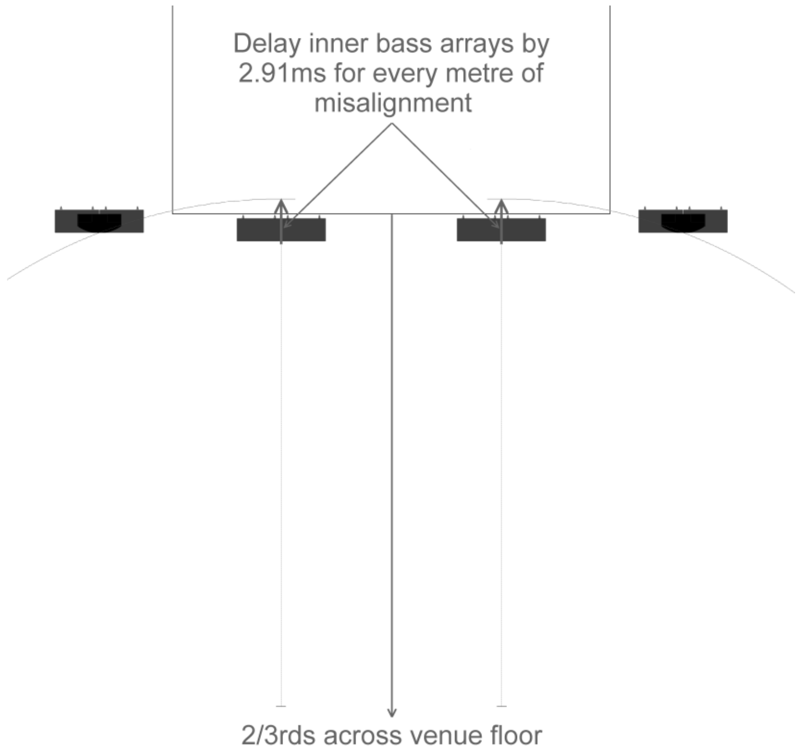
Point destination alignment



Mix position alignment providing maximum mix position impact

For larger stages you should probably think about adding a 2-wide splayed block every 6-10m or so. These also provide a handy support for front fills.

Planar alignment



Planar alignment providing smooth bass coverage and reduces the central bass peak

6.7 Directional bass arrays and why you may not need to use them

Directional bass arrays are rarely required with Funktion One bass systems because Funktion One bass systems are designed for maximum broadband efficiency.

Q. Why is broadband efficiency important?

A. Unfortunately, the answer is a bit drawn out – but please bear with us. We experience the most realistic impact from a sound system when a sound wave effects our whole body, not just our ears. The most effective way of doing this is to design systems to convert as much amplifier power into sound intensity (sound power per unit area) as possible – and over as wide a bandwidth as possible.

Although sound engineers and manufacturers tend to specify loudspeaker performance in terms of the resulting sound ***pressure****, sound power is made up of two components; particle velocity and sound pressure. It's analogous to electrical power being the product of current and voltage. Sound power is important; whether you're trying to excite your audience with a larger than life kick drum sound or you're trying to convey plucked strings without running your orchestral reinforcement system into low frequency feedback.*

To get maximum power, the two components (velocity and pressure) need to be in phase over as wide a bandwidth as possible. "One-note" resonant bass systems should be avoided. And, to achieve maximum power transfer, the source impedance needs to match the load impedance.

There is usually a significant mismatch between the acoustic impedances of direct radiating loudspeakers and the all-important air that conveys the loudspeaker signal to the audience. But the match can be improved with careful loudspeaker design, especially if horn-loading is used, as the horn will act like an impedance-matching transformer.

The increasing availability of high power amplifiers and improvements in loudspeaker materials has led many manufacturers to abandon their quest for efficiency. Unfortunately, these inefficiencies can't simply be overcome by applying more power, so the current trend isn't very clever.

We hear sound pressure, but that sound pressure is caused by the sound power that's transferred to us. If the power transfer is inefficient, that important pressure-velocity relationship is compromised, and the resultant sound pressure will lack physical impact and slam. Inferior power transfer can often be observed as excessive phase vs frequency changes over the system bandwidth – over and above the usual boundary and distance effects.

Horn loaded bass systems tend to project more impact than direct radiating systems because horn loading provides a better driver-air impedance match, conveying power more effectively.

Most sound engineers will attempt to get the impact they want by driving an inefficient bass system even harder – either by turning it up, tweaking the eq., or a combination of both. This causes more leakage behind the system and has led to the current popularity of cardioid bass configurations.

Cardioid bass configurations are obviously going to be popular with a lot of loudspeaker manufacturers as they can generate extra profits by selling you more speakers to deal with the adverse effects of having to drive their systems so hard in the first place. However, if local conditions – a monitor mix position or a camera tracks close to the bass system, for instance – dictate, please feel free to try the following V221 set-ups:

6.7.1 Types of cardioid/end-fired systems & their pros and cons

There are two main types of directional array, but few users seem to understand their pros and cons:

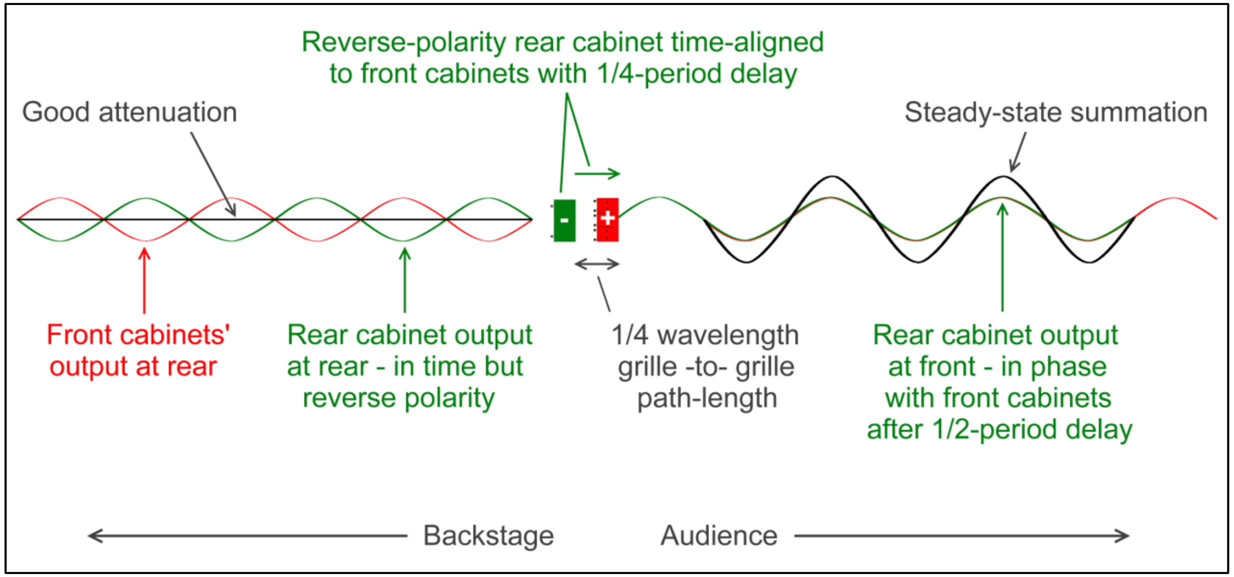
- The reverse-polarity cardioid array *(although end-fired arrays can exhibit a cardioid pattern as well)*
  - Pros: The typical 15-20dB rear cancellation is maintained over most of a typical bass bandwidth
  - Cons: A loss of initial impact – due to half-cycle rear-to-front summation delay if used with small low-directivity arrays or in single cabinets*
- The in-phase ***end-fired*** array
  - Pros: Excellent forward gain and impact – especially when used low-profile *(not stacked too high)* for maximum summation, or when used 4-deep for long, narrow street festivals
  - Cons: The typical 15-20dB rear cancellation is limited to middle of the bass bandwidth covered - dropping to 5-6dB near the upper and lower cut-off frequencies

Reverse-polarity cardioid arrays

Reverse-polarity ***cardioid*** configurations are popular because they provide excellent signal cancellation at the rear – keeping monitor engineers and back-stage crew happy.

The illustration *(right)* shows a reverse-polarity rear element *(**green cabinet**)*, time-aligned to the front system *(**red**)* using ¼-period electronic delay. The resulting out-of-phase conditions provide the required signal attenuation *(**black**)* in the backstage direction.

In the audience direction, the rear cabinet's ¼-period delay combined with its extra ¼-wavelength sound path-length to the front results in an overall phase reversal, phase aligning it to the front system for steady-state summation after an initial ½-period delay.



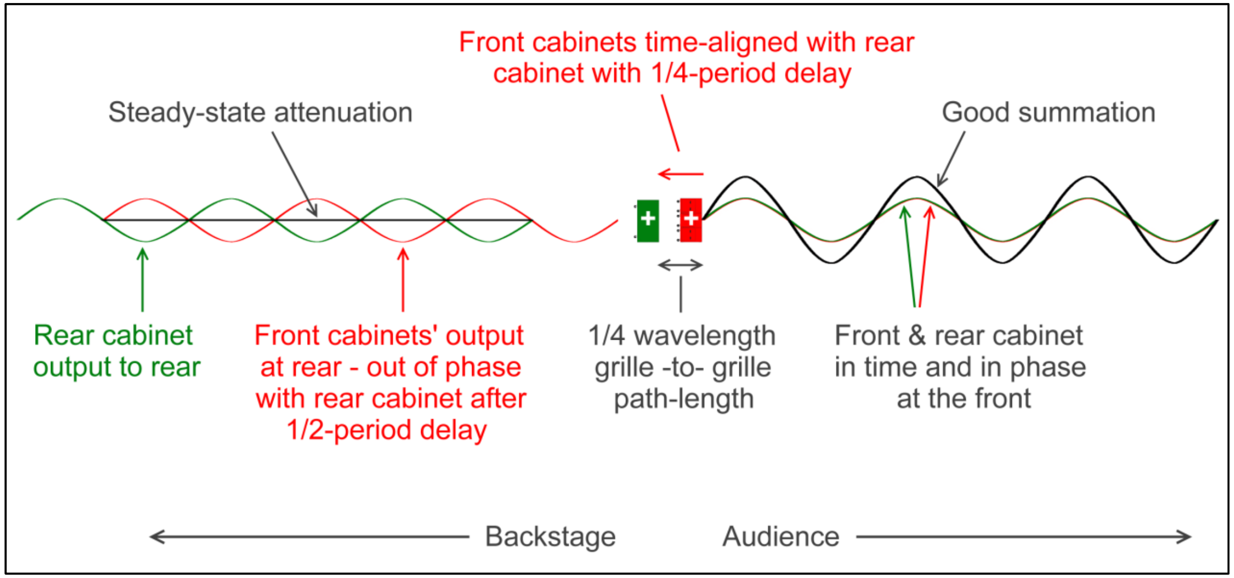
Reverse-polarity cardioid operation – wavelengths & cabinets to scale but rear levels exaggerated for clarity*

Note that Funktion One V221 horn-loaded bass systems have excellent rear attenuation in practice so a 3:1 (front-to-rear) ratio is feasible. V221 T-arrays (see later) provide excellent forward impact, even when used in the popular reverse-polarity cardioid configuration.

See following pages for typical ¼-period delays and ¼-wavelength grille -to- grille spacing…

In-phase end-fired arrays

Many live sound engineers prefer an in-phase ***end-fired*** configuration as it doesn't exhibit the ½-period rear-to-front summation delay and, therefore, provides more accurate summation and impact for the audience. Unfortunately, the ½-period delay now occurs behind the array and can cause inferior rear cancellation.



In-phase end-fired operation – wavelengths & cabinets to scale but rear levels exaggerated for clarity

End-fired configurations are analogous to a delay loudspeaker system. The illustration above shows the front ***(red)*** system time-aligned to the rear ***(green)*** cabinet using ¼-period electronic delay. This provides excellent in-time and in-phase summation ***(black)*** for maximum impact in the audience direction.

In the backstage direction, the front system's ¼-period delay combined with its extra ¼-wavelength sound path-length to the rear results in an overall phase reversal. The resulting out-of-phase conditions provide a steady-state attenuation after an initial ½-period delay.

Note that "time zero" is located at the rear cabinet of end-fired systems – so the main arrays need to be time-aligned to that position.

Typical sub-bass alignment parameters would be:

- 1.717m* grille -to- grille path-length *(includes sound path around cabinet)* for a ¼-wavelength alignment at 50Hz combined with…
- 5ms* delay *(1/4-period at 50Hz)*

*The final figures may be considerably smaller for broadband bass systems – where a higher alignment frequency may be preferred.

Funktion One T-array

T-array - for “end fired” or “cardioid” configuration

Funktion One T-arrays offer a choice of operating configurations and deployments, depending on the required directional characteristics and practical considerations.

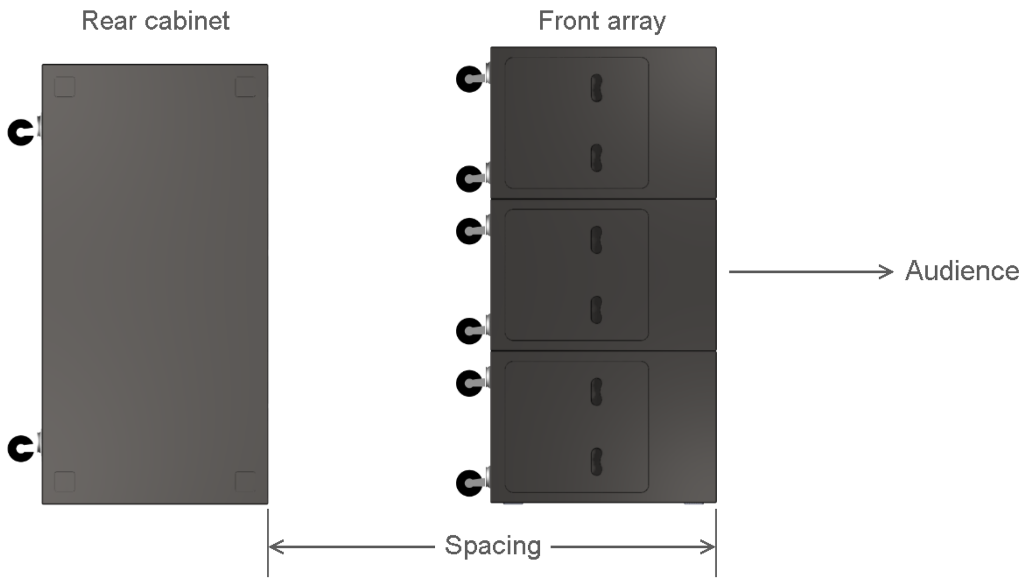
They may be used:

- “End fired” - with the rear element in-phase for maximum forward summation

  Or

- “Cardioid” - with the rear element polarity-reversed for maximum rear cancellation.

T-array settings



T-array - for “end fired” or “cardioid” configurations

Configuration		T-array				
			Front array		Rear cabinet	
		Straight-line spacing – allowing for round-cabinet path-length *(metres)*	Polarity	Delay *(ms)*	Polarity	Delay *(ms)*
“Cardioid” *(Best rear cancellation)*		1.2m	Normal	0ms	Reversed	5.2ms*
“End fired” *(Best audience impact)*		1.2m	Normal	5.2ms*	Normal	0ms

(*Adjust for max cancellation at preferred distance behind the array)

*Note, again, that the final delay figures may be considerably lower for broadband bass systems – where a higher alignment frequency may be preferred.

If a higher frequency (mid-bass) rear cancellation characteristic is required but space is restricted - a ***Compact cardioid*** *array is possible. See later…*

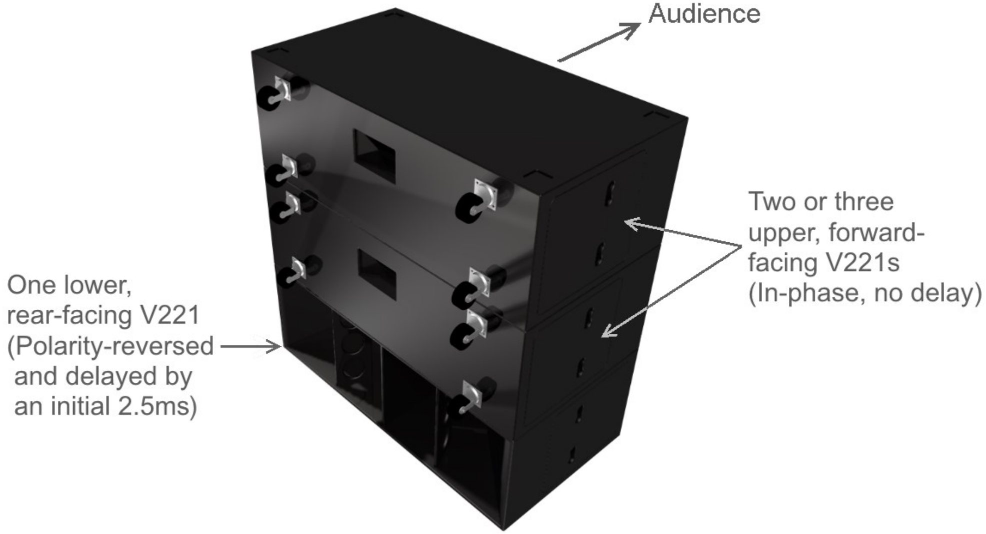
Compact cardioid array

A compact reverse-polarity “cardioid” array may be deployed where there isn’t enough space for the back-to-back T-array. In this configuration, a rear-facing V221 is placed at the bottom of a V221 stack.

This compact deployment trades a slight decrease in power and forward summation for the very compact footprint and a more mid-bass rear cancellation characteristic.

Compact cardioid array				
	Upper cabinets		**Lower cabinet**	
Fixed placement	**Polarity**	**Delay *(ms)***	**Polarity**	**Delay *(ms)***
Vertical stack with bottom element facing backwards	**Normal**	**0ms**	**Reversed**	**2.5ms***

(*Adjust for max cancellation at preferred distance behind the array)



Compact cardioid array viewed from behind

*Again, the final figures may be considerably lower for broadband bass systems – where a higher alignment frequency may be preferred.

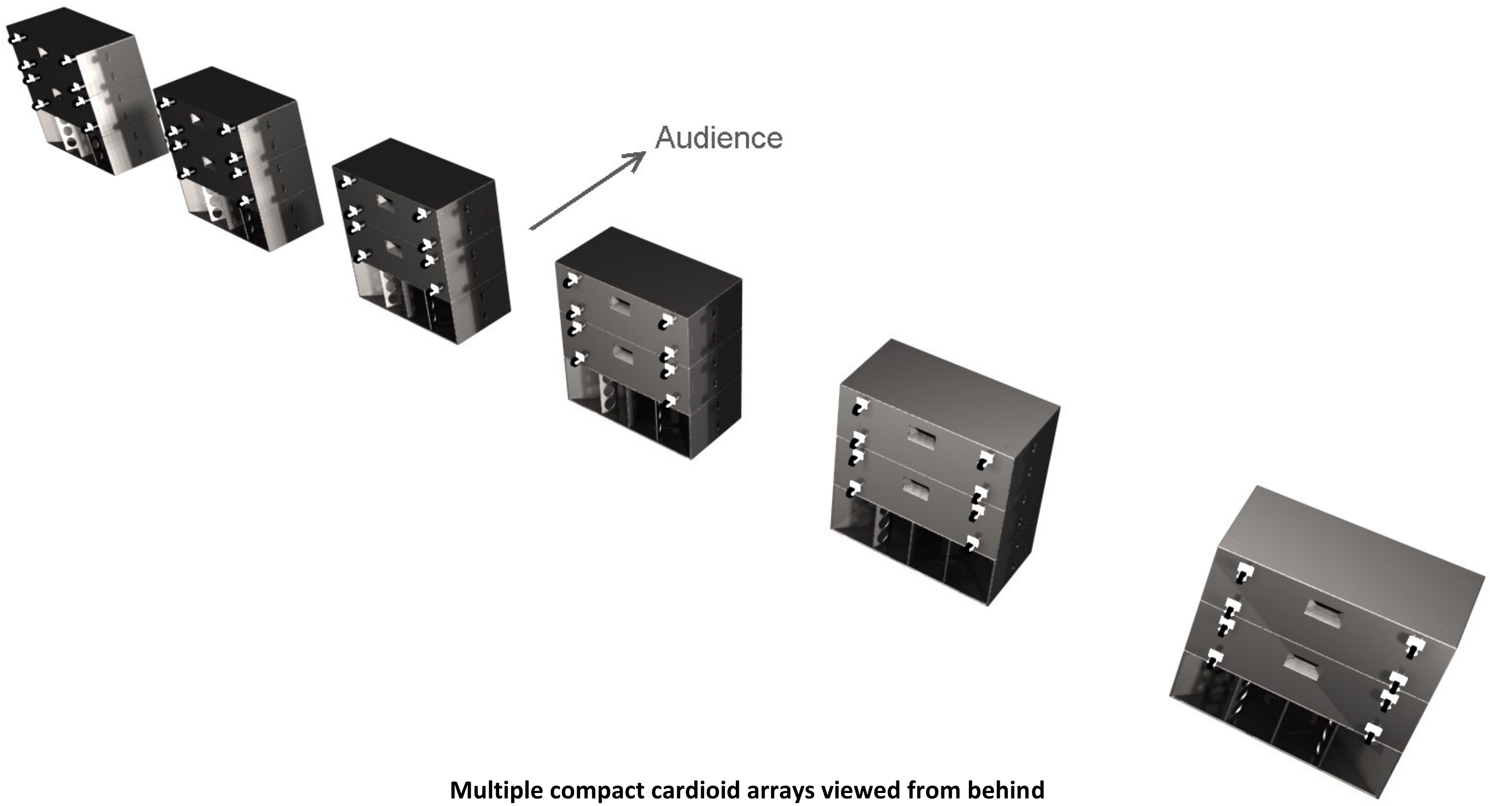
When V221s are placed like this, their mouth-to-mouth sound path-lengths are slightly shorter than the traditional mid-band quarter-wavelength, but rear cancellation is still achievable if we delay the reversed-polarity rear-facing V221 by around 2.5ms. The delay setting may need to be fine adjusted to achieve complete cancellation at a specific distance behind the array – at the monitor mix position, for example. Local boundaries can also affect the optimal delay setting.

These compact cardioid blocks may be used individually under the main arrays, or in multiples to form stage apron arrays for large-scale events *(see next page)*.

When multiple cardioid blocks are placed close together, the front-to-rear path length can sometimes increase due to the height of the upper, forward-facing V221s. The best solution is to leave gaps between the compact cardioid blocks as follows…

Multiple compact cardioid arrays

Large-scale, very high power V221 compact cardioid arrays may be set up along stage aprons in a 1:1 block-to-gap configuration.

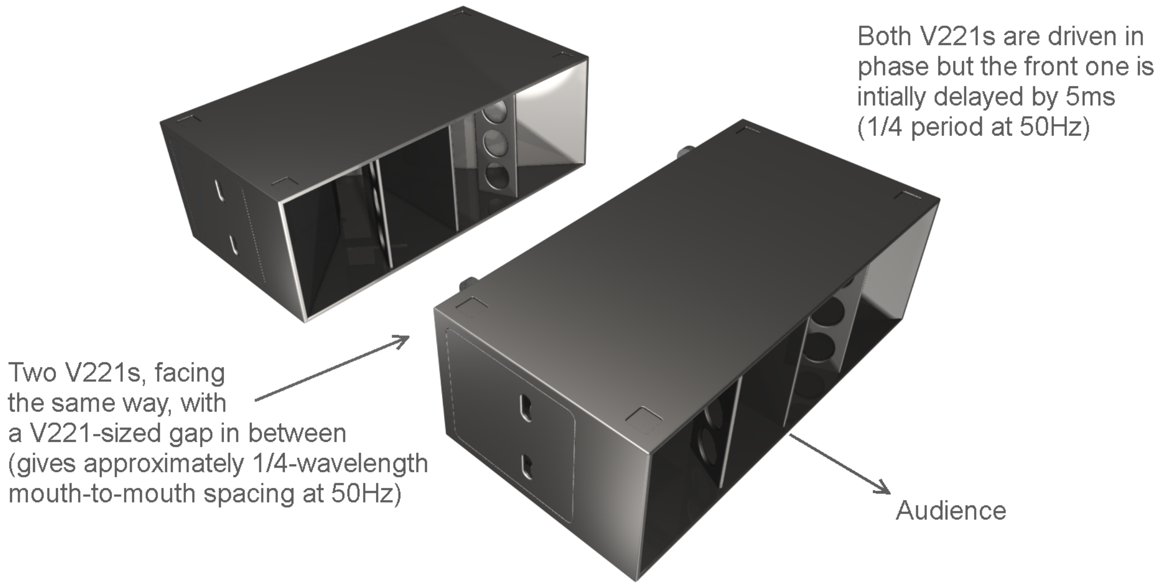


Multiple compact cardioid arrays viewed from behind

Again, the lower, reversed polarity V221's delay setting may need to be fine adjusted to achieve complete cancellation at a specific distance behind the array – at a camera track, for example. Local boundaries can also affect the optimal rear element delay setting.

These large planar arrays are often "electronically curved", section-by-section, by applying progressively more overall middle-to-outer delay whilst keeping each section's rear-to-front delay fixed for maximum rear cancellation per block. Check with Funktion One for further details.

Low profile V221 in-phase, end-fire configuration



Low profile V221 end-fired array

A couple of practical issues need to be borne in mind:

1) End-fired systems cover a larger footprint as there needs to be a significant gap between front and rear systems to make up the quarter-wavelength grille-to-grille *(or horn mouth-to-horn mouth)* spacing. See large-scale system on next page…

2) In order to vector-sum with the front system efficiently, the rear system's output must have an unimpeded route forward. This works fine for low-profile *(single storey)* arrays but large, stacked systems are less efficient as the front stacks tend to partially block the rear ones - unless the front and rear systems are staggered, which unfortunately, looks very messy.

Multi-layer, low profile V221 end-fire array

Multi-layer low profile V221 end-fired array

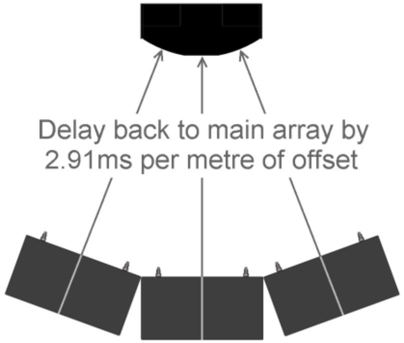
This type of end-fire array can be placed either side of a stage and can be very effective for urban carnivals and narrow street events as it produces a well-defined “long-throw” forward beam with good rear cancellation.

Each cabinet is delayed to the one behind but nearer to the main-to-bass crossover frequency this time:

- The rear cabinet is at zero delay *(and this is the time the main PA should be delayed to)*
- The cabinet in front of that is delayed a quarter period *(e.g. 3.6ms for 69Hz)*
- The next one forward is delayed a half period *(7.2ms for 69Hz)*
- And, finally, the front cabinet is delayed three quarters of a period *(10.8ms at 69Hz)*

Again, some delay adjustments may be required for optimal summation – depending on local conditions.

6.7.2 Compensating for positional offsets



Extreme example for illustration only!

There are two distinct stages to delay-aligning bass and main systems successfully.

1) **Physical offset/time alignment -** Measure the relative distances to the bass and main systems from a representative position in the room - usually the mix position, but, if there is no auditorium mix position, 2/3rd across the audience or dance floor area. Then delay the required section by 2.91ms for every metre of physical offset.

2) **Phase alignment - at the representative listening position *(usually the mix position)* -** This fine adjusts for phase shifts and their resultant group delays. This can be caused by bass enclosure physics, crossover topologies and, of course, phase lags caused by local boundaries.

> **Note that the above time alignment must be done first. If you jump straight in with phase alignment, before you've compensated for physical offsets, you could end up a whole cycle out at crossover!**
>
> **It's also worth checking your signal paths *(amplifier, controller settings etc.)* to make sure everything is in-phase before you start.**

a) Set up a pink noise or musical signal through a graphic equaliser with all its bands fully attenuated except the bass-main crossover frequency - set the crossover frequency slider to full boost. Make sure you use this 1/3 octave-filtered signal or noise. Do not use a sine wave oscillator…

b) Using this crossover-tuned signal, switch between your main and bass systems and balance their levels.

c) Now ***temporarily*** switch the bass system out of phase and, with both main and bass systems on, fine adjust the delay for a null in level.

(Note that you are unlikely to hear a complete cancellation in a real room)

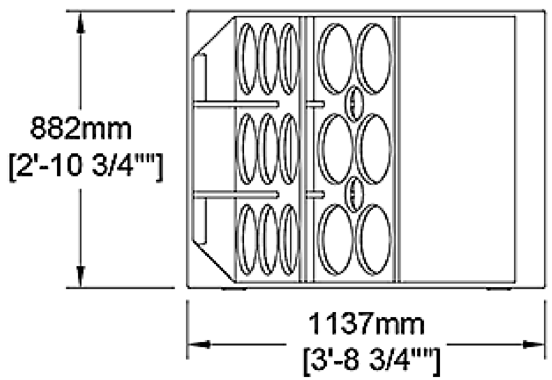
d) Now ***- and this is most important*** - switch the bass system back to the correct polarity.

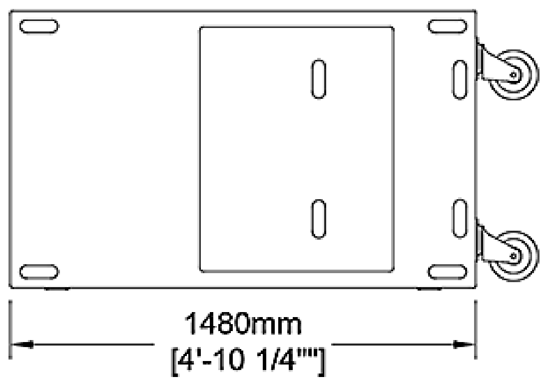
6.8 V132 low-bass system

The Vero V132 is an extraordinarily powerful single 32” horn-loaded low-bass enclosure that may be used to augment Vero systems for the ultimate in low frequency performance, high-power contemporary music applications and special effects. The V132 features Powersoft's patented M-Force moving magnet 10kW linear transducer and Funktion One's cone and enclosure technology to extend low-bass response to 24Hz.

The V132’s moving magnet linear transducer provides unprecedented levels of electromagnetic conversion and reliability while Funktion One’s acclaimed horn-loading provides high levels of sound intensity (i.e. the product of both particle velocity and sound pressure). This provides excellent directional impact without having to resort to cardioid configurations.

To quote Funktion One’s Tony Andrews: *“The motor allowed us to make a big enough cone so that we could scale up our horn-loaded bass technique. The near half a ton of push it gave us meant that we could scale up to a 32in driver and still keep it dynamic. Scaling the size up meant we could load down to 24Hz.”*





V132 dimensions in millimetres *(and feet & inches)*

Contact Funktion One Research Limited for the latest V132 applications examples, hints and tips.

7 Patented Lambda® flying and stacking system

Important safety advice

1) However tight the deadline, always ensure that rigging and installation work is completed by fully qualified and experienced personnel

   Remember that people's LIVES depend on safe rigging practice

2) All rigging personnel – and staff near the rigging operation – should wear high visibility jackets, safety boots and hard hats for their own protection
3) The rigging system is designed for a maximum of 24 Vero cabinets – subject to Funktion One's ***Projection*** software confirmation
4) Always use Funktion One's ***Projection*** software to design your array. Projection will alert you to potentially unbalanced or unsafe FlyGrid lifting point, frame position or Jib mode selections
5) Vero systems are designed to be lifted from a VERTICAL position straight from their transport dolly.

   Vero arrays must NEVER be assembled horizontally on the floor and then lifted diagonally!

6) Use only Funktion One manufactured or approved flying accessories with your Vero system
7) Always read the user guide and heed any warning labels before attempting to rig your Vero system
8) However well trained you are, remember that equipment and techniques vary from system to system. If in doubt, ask.
9) For added safety, always invite other rigging staff to cross-check your work
10) Install safety steels at the earliest opportunity and comply with the appropriate national and international laws and directives
11) Ensure that FlyPlate angle settings are identical at both ends of each cabinet
12) Ensure that all pip-pins are locked in place – with their buttons out - making sure their steel lanyards are correctly positioned and not snagged
13) Vero cabinets should be transported in dedicated Funktion One dollies in vertical stacks of four
14) Always ensure that Vero cabinets are transported in perfect show-ready condition with ***all*** their pins in place

And finally,

Never attempt to rig the Vero cabinets with their FlyPlate covers removed. As stylish as they look, the covers are not just cosmetic embellishments. They're there to protect your fingers and, of course, the mechanism.

7.1 FlyGrid and pod introduction

Overview

Funktion One has always favoured inter-enclosure connection at the centre of gravity, which allows easy angle adjustment under load. The desired angles are set with the Lambda Dial and fixed by pulling **up** the bottom Cabinet to the extendable boom on the flying frame, creating a strong and rigid triangular distribution of loads and forces that can be elevated or declinated as a whole.

Vero's ingenious patented flying system also positions the pivot point exactly equidistant between the cabinets such that the physical alignment and therefore the acoustic summation of enclosures is naturally correct, regardless of angle (GES).

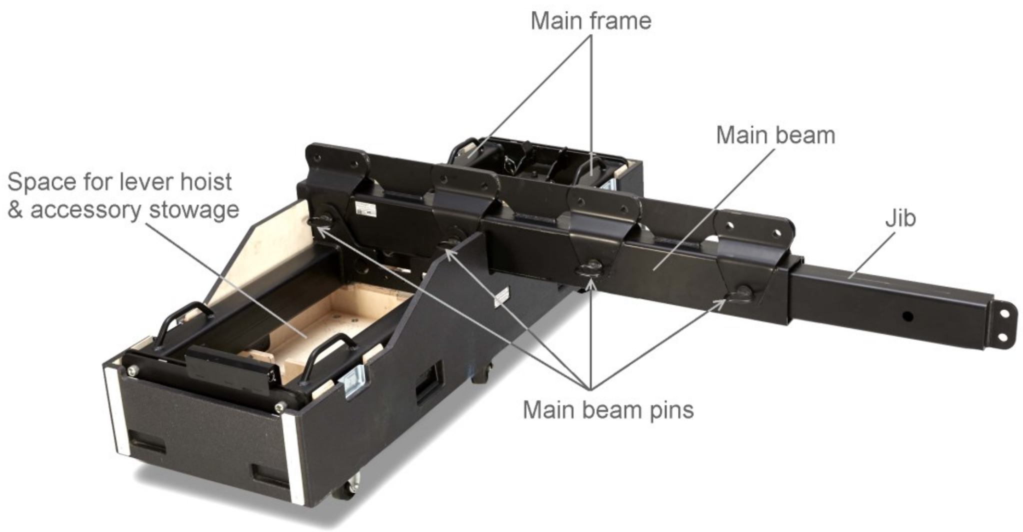
As the sound sources in each enclosure have a high vertical directivity control, summation occurs in the far field. This gives many advantages, including adjustment between vertical coverage at larger angles for near and midfield and near perfect summation at smaller angles for high-intensity long throw. Vero's design provides even coverage by natural acoustic means, providing greater coherency and resolution than electronically processed approaches.

The Lambda® rigging system also allows Vero arrays to be adjusted with the system in suspension; a major advantage when last-minute adjustments are required.

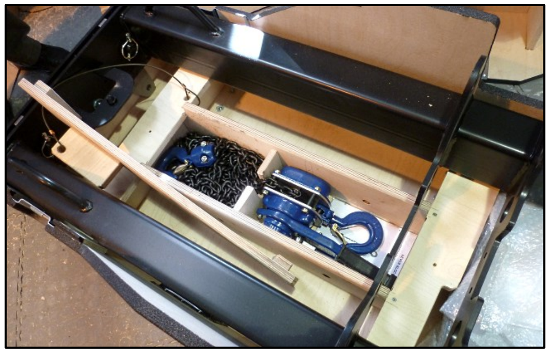
FlyGrid

The FlyGrid assembly comprises a main beam with rear jib extension, plus a main frame *(shown in its trunk)* which connects to the top cabinet of the array.

The rear jib extension and the main frame -to- main beam attachment positions may be optimised for best load balance using your ***Projection*** software.



FlyGrid in preparation on its trunk



Lever hoist stowed in trunk

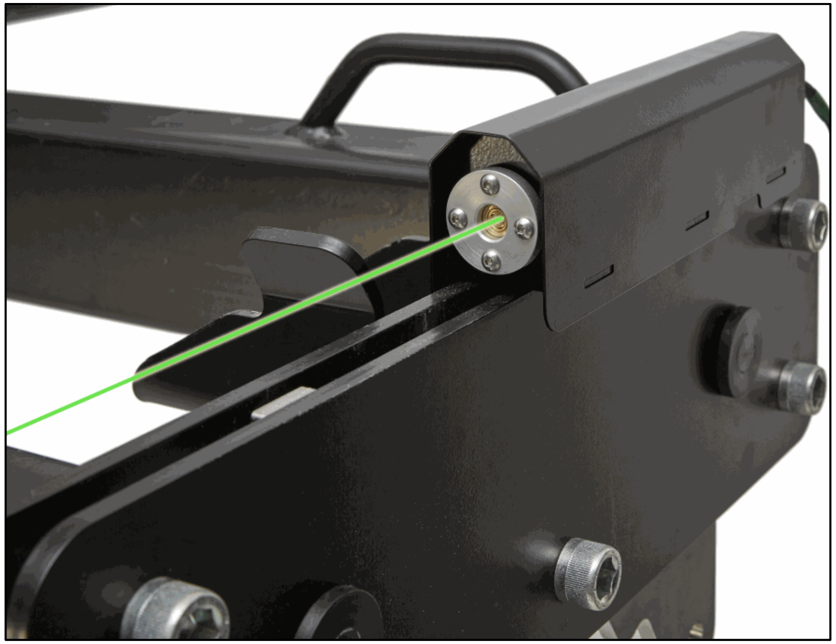
FlyGrid Pod

To avoid laser hazards:

1) **Never look directly into the laser**
2) **Always warn others in the venue that a laser is about to be used**
3) **Never point the laser where other people are located**
4) **Never use the laser when the public have access to the venue.**

The multi-function Vero FlyGrid Pod is mounted on the grid and contains 3 indication systems.

A green laser for sighting, an inclinometer for grid angling and flashing white LEDs to indicate the correct tension of the rear pull-up between the jib and the bottom box of the Vero hang.



FlyGrid-mounted laser, inclinometer and pull-up limit indicator sensor

Hand-held inclinometer system

The Pod component *(above right)* which contains an inclinometer sensor, laser pointer and load-pin microswitch, is permanently attached to the VERO FlyGrid. It is connected to the hand-held display unit *(shown left)* using a suitable length of CAT5e or CAT6 ethernet. Make sure you check functionality before the grid is lifted.

Pull up tension indicator

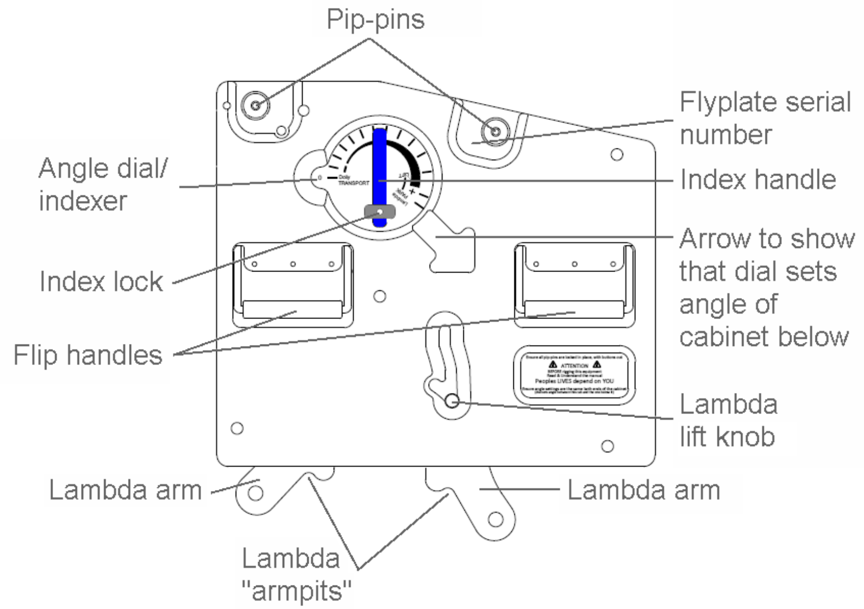
Vero's unique "Lambda" flying method allows the individual adjustment of inter-box angles, if necessary, whilst hanging. The dialled-in curvature is achieved by adding tension with the lever hoist which is connected to the bottom box via a bridle and the extendable jib via steel. This pull-up tension is monitored by the FlyGrid pod and indicated on the hand-held inclinometer shown on the left. See ***Section 7.3.1*** for remote hand-held inclinometer instructions.

FlyPlate system

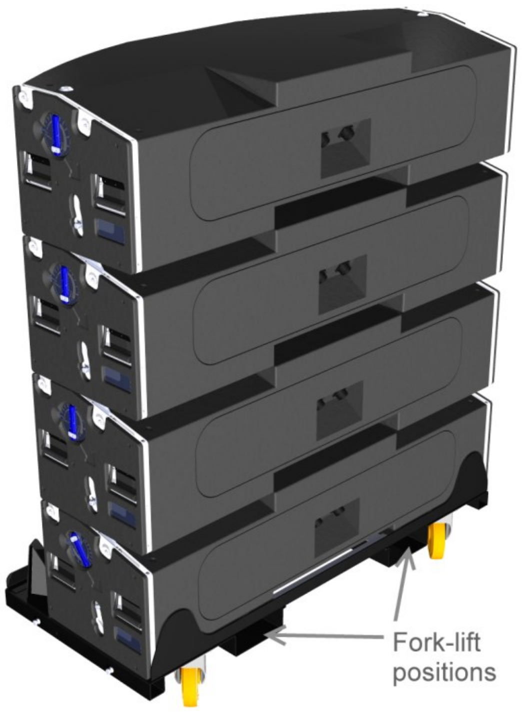
Vero V60, V90 and V315 cabinets include FlyPlate systems that provide secure upper pip-pin attachment points and adjustable lower ***Lambda*** connector arms.

Inter-cabinet splay angles are pre-set using an indexer which features a very clear angle dial. The system stays as a straight hang – making rigging operations much easier – and is then tensioned to its pre-set splay angles using a lever hoist and bridle arrangement in the traditional Funktion One way.

This makes it easy to lift cabinets from the dolly during initial rigging operations and facilitates fine inter-cabinet angle adjustments with the system off the ground. It is also much easier to land cabinets back on the dolly during derigging.



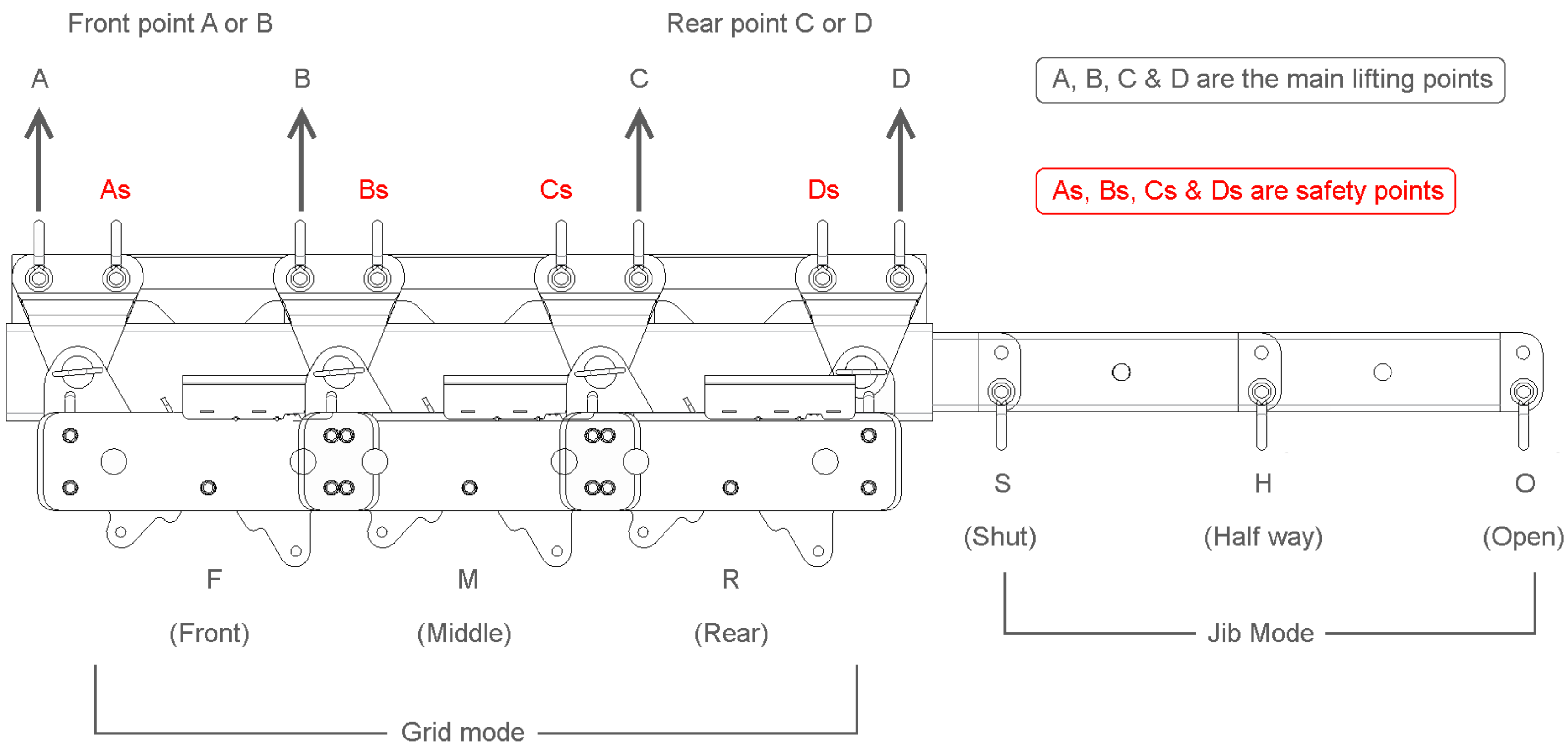
FlyPlate system



Transportation

- The rigging system has its own transport trunk *(shown earlier)*.
- The fork-lift equipped ***V-dolly*** system *(left)* houses four pre-rigged Vero V60, V90 or V315 cabinets – with the upper three cabinet indexers set to the to the ***0° Dolly TRANSPORT*** position and the bottom cabinet indexer set to its ***Lambda Park*** position. *(See **Section 7.4** later…)*
- Always ensure that Vero cabinets are transported in perfect show-ready condition with <u>*all*</u> their pins in place

7.2 FlyGrid preparation



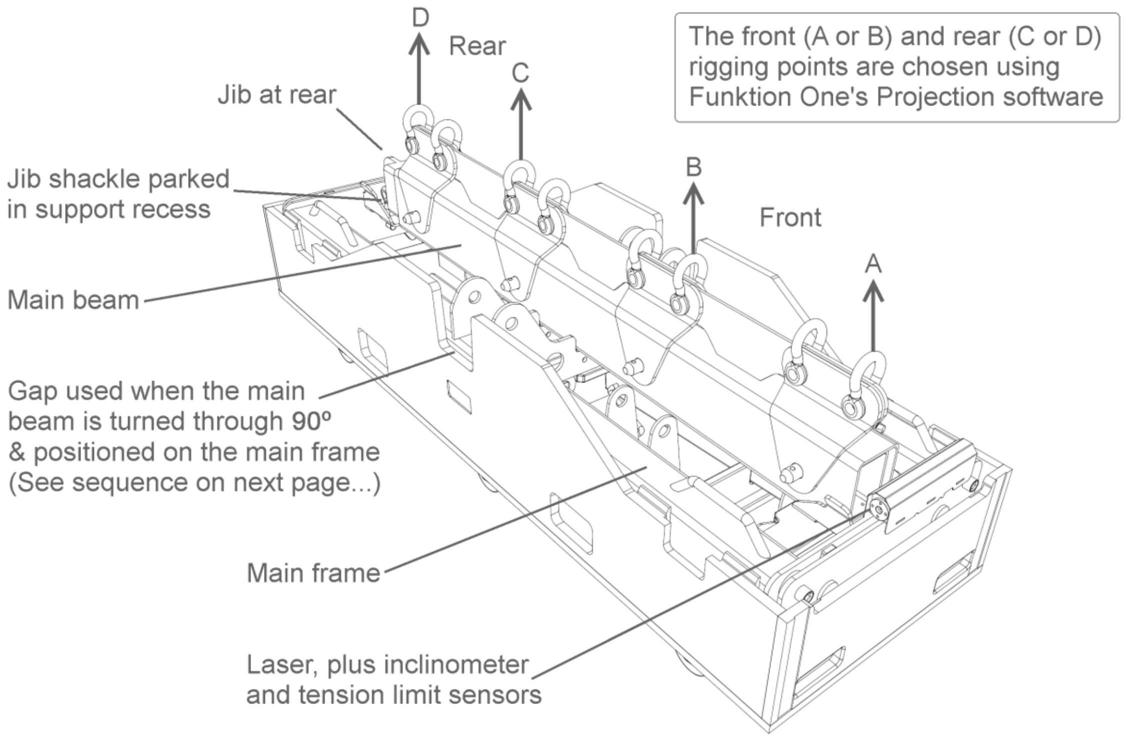
Projection software print-out

The front *(A or B)* and rear *(C or D)* main lifting points will be determined at the design stage using your ***Projection*** software.

If you are not the system designer, check with your colleagues for a print-out of the lifting point positions, the optimal frame position *(F, M or R)* and the optimal jib extension *(S, H or O)*. See the following pages for further instructions…

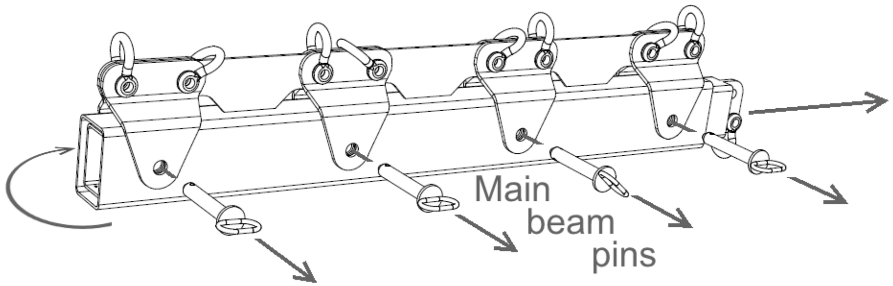
Caution!

The FlyGrid trunk contains the main frame, the main beam and the extendable jib and is heavy! Get others to help move it – especially across rough terrain.



Use your motors to lift from the main A or B *(Front)* and C or D *(Rear)* points as directed by the *Projection* software

(Note orientation! Jib must be to the rear – i.e. upstage)

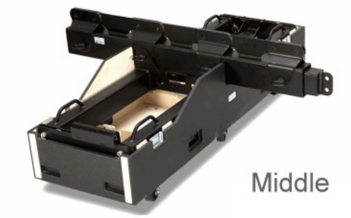


Raise the main beam out of the trunk, rotate the trunk through 90° and float the beam just above the main frame. The jib beam must be to the rear *(to right in illustrations).*

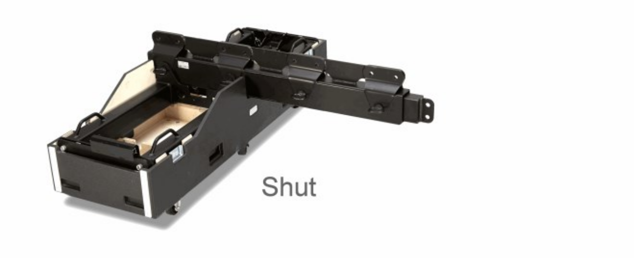
- Release the four ***main beam pins*** *(above left)* by removing their lynch pins.
  (Note that the main beam and lifting bracket are welded together so the beam won't drop)
- Lower the ***main beam*** to the ***main frame*** position *(found using Projection's **Grid mode** adjustments)* ⇨
- Extend the jib to the position found using Projection's ***Jib mode*** adjustments *(see below)*

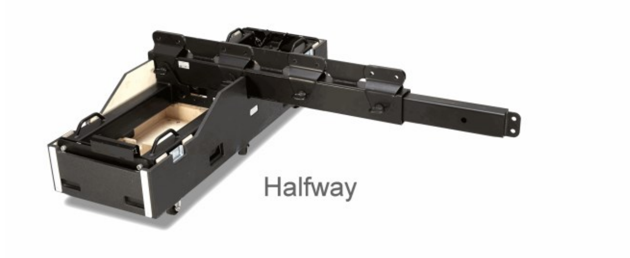
Important! After checking that the assembly is exactly as determined by the ***Projection*** software:

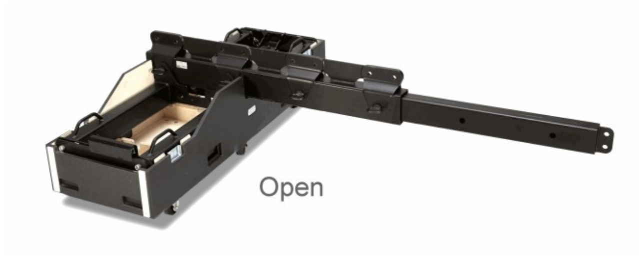
- Secure the beam to the main frame and lock the jib in place by re-inserting all four main beam pins
- Secure the main beam pins with their lynch pins
- Get another rigging colleague to cross-check the system is safe



Grid modes

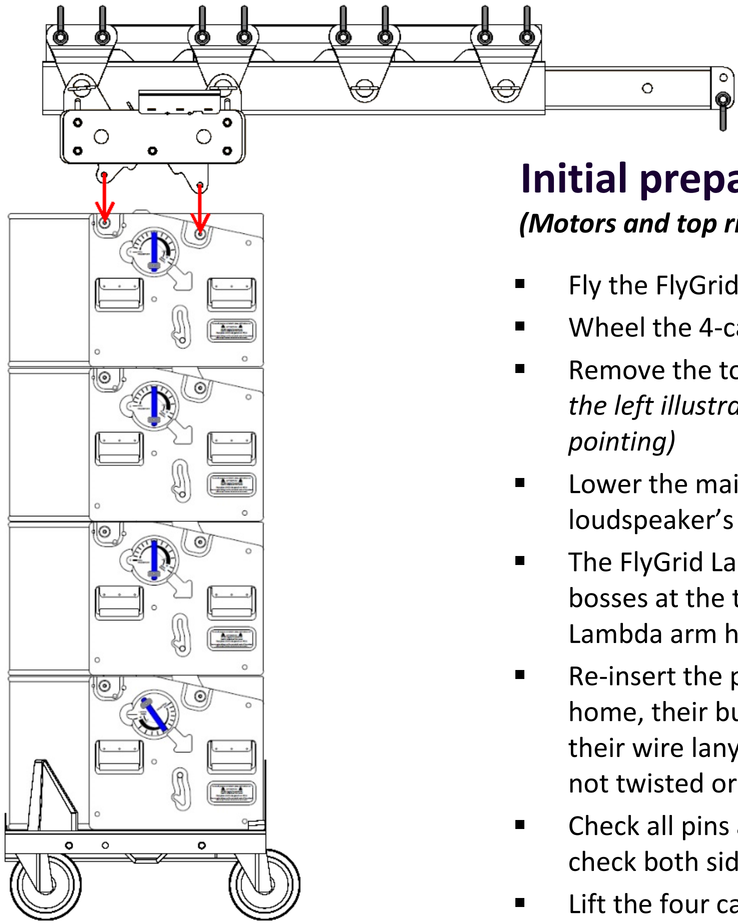






Jib modes

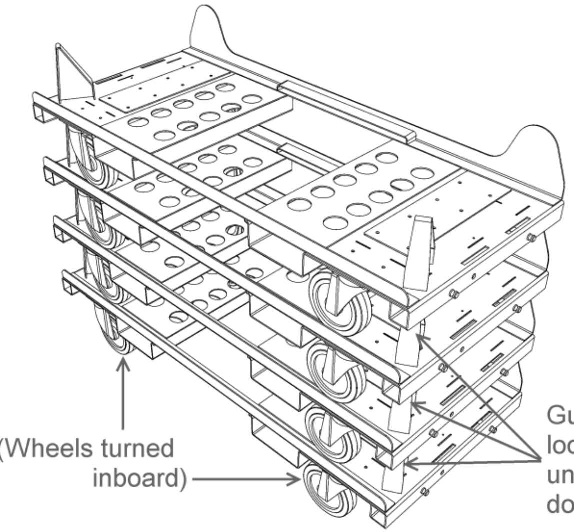
7.3 Vero cabinet preparation and flying



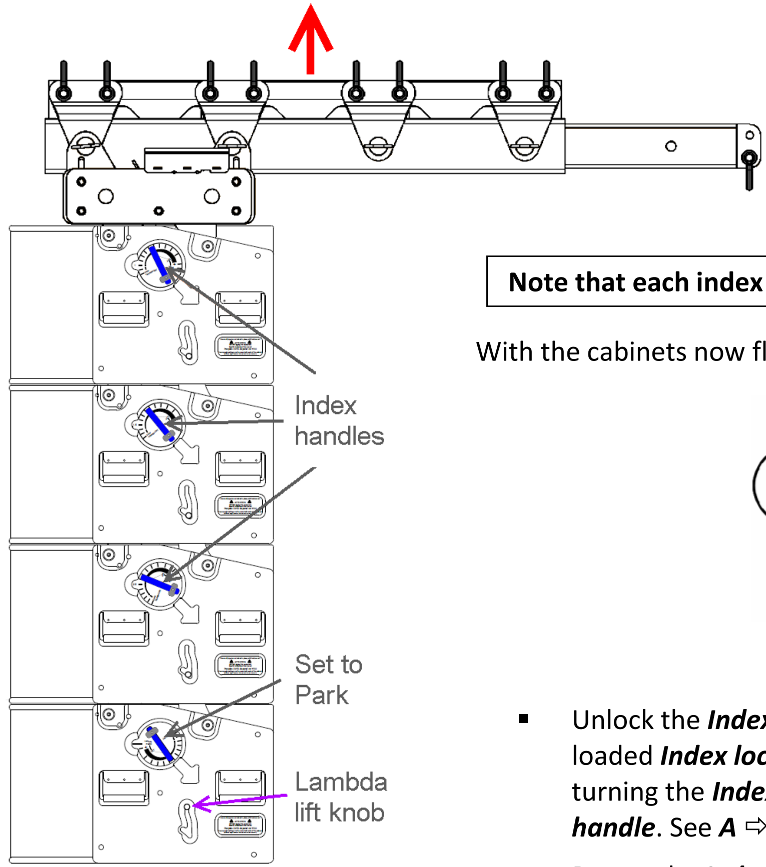
Initial preparation and first 4-cabinet lift

(Motors and top rigging not shown)

- Fly the FlyGrid assembly to just above a 4-cabinet dolly height
- Wheel the 4-cabinet dolly into place beneath the FlyGrid assembly
- Remove the top loudspeaker's pip-pins *(where the left illustration's coloured arrows are pointing)*
- Lower the main frame to line up with the top loudspeaker's pip-pin positions.
- The FlyGrid Lambda arm "armpits" locate on bosses at the top of the FlyPlate to line up the Lambda arm holes with the pip-pin holes
- Re-insert the pip-pins; making sure they are fully home, their buttons have popped out and that their wire lanyards are correctly positioned and not twisted or snagged
- Check all pins are securely in place and cross-check both sides before proceeding
- Lift the four cabinets so that they float free from the dolly.
- ***Move the empty dolly to a safe distance ready for stacking***



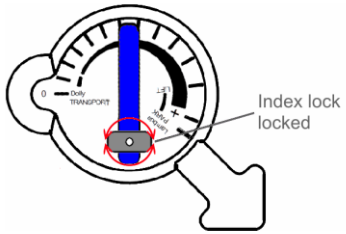
Dolly stowage



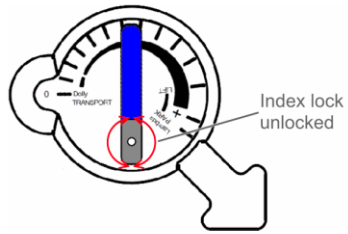
Although inter-cabinet angles can be set by one person working alone, we recommend two people – one on each side of the cabinet. Two people tend to be safer as they can provide mutual support and cross-check each other's work.

Note that each index handle sets the vertical splay angle between its cabinet and the one *below*.

With the cabinets now floating just off the ground, you can set the inter-cabinet splay angles as follows:

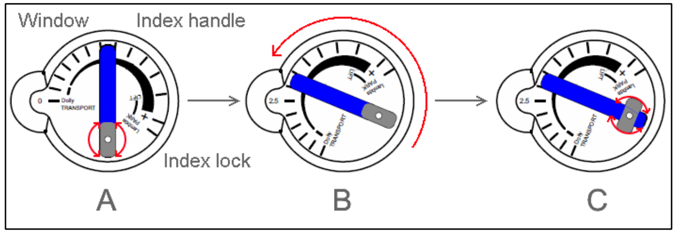


Index handle locked

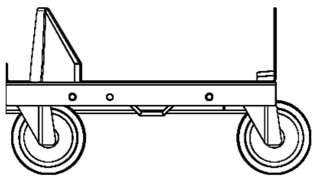


Index handle unlocked

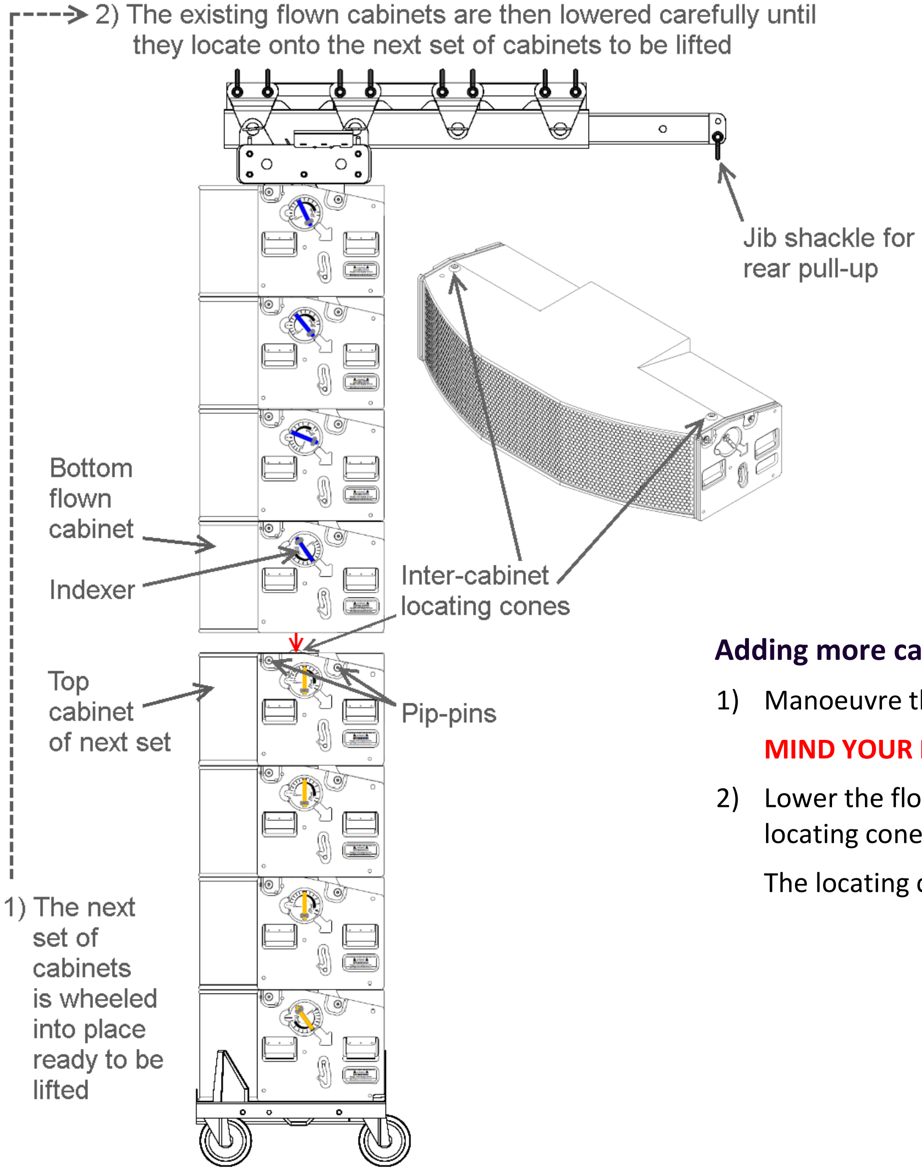
- Unlock the ***Index handle*** by gently pulling the spring-loaded ***Index lock*** away from the ***Index handle*** and turning the ***Index lock*** so that it is in line with the ***Index handle***. See ***A*** ⇨
- Rotate the ***Index handle*** to bring the required angle into the ***Window*** *(e.g.* ***2.5°*** *shown on* ***B****)* ⇨



Don't forget to confirm the angle on both sides of the enclosure!



- Lock the index handle by lifting and turning the ***Index lock*** so that it is at 90° to the ***Index handle*** *(see* ***C*** *above)*⇗
Make sure it clicks back into place!



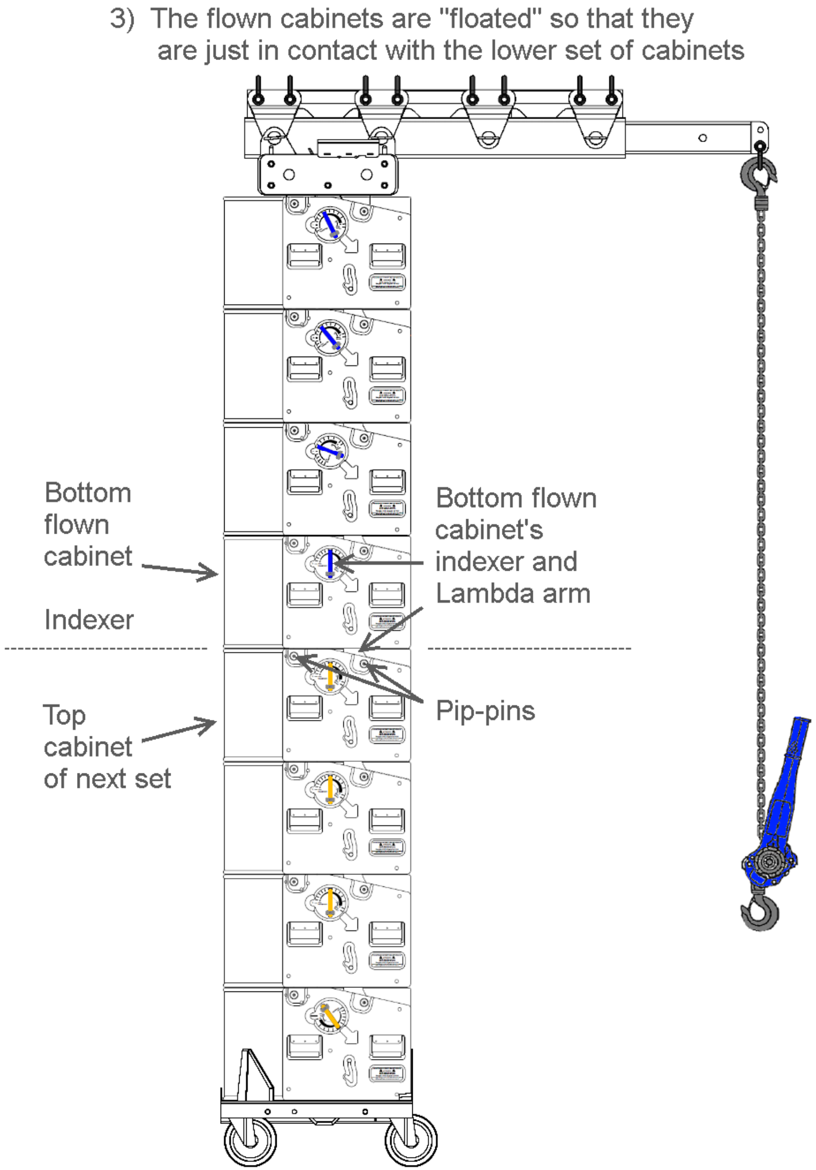
Adding more cabinets

1) Manoeuvre the dollied cabinets beneath the previously flown cabinets

   MIND YOUR FINGERS!

2) Lower the flown cabinets so that the bottom flown cabinet locates onto the top dollied cabinet's locating cones – but with the cabinets only just touching.

   The locating cones will optimise the cabinet positioning before the Lambda arms are deployed.



Before adding more cabinets

A) *Clip the lever hoist's lifting swivel hook to the rear jib shackle so that the lever hoist mechanism is at the bottom (see left)*

B) *Add speaker cables and links (not shown) as you go along*

Once the lower flown cabinet has located onto the upper dollied cabinet's cones:

3a) Remove the ***top pip-pins*** from the top dollied loudspeaker

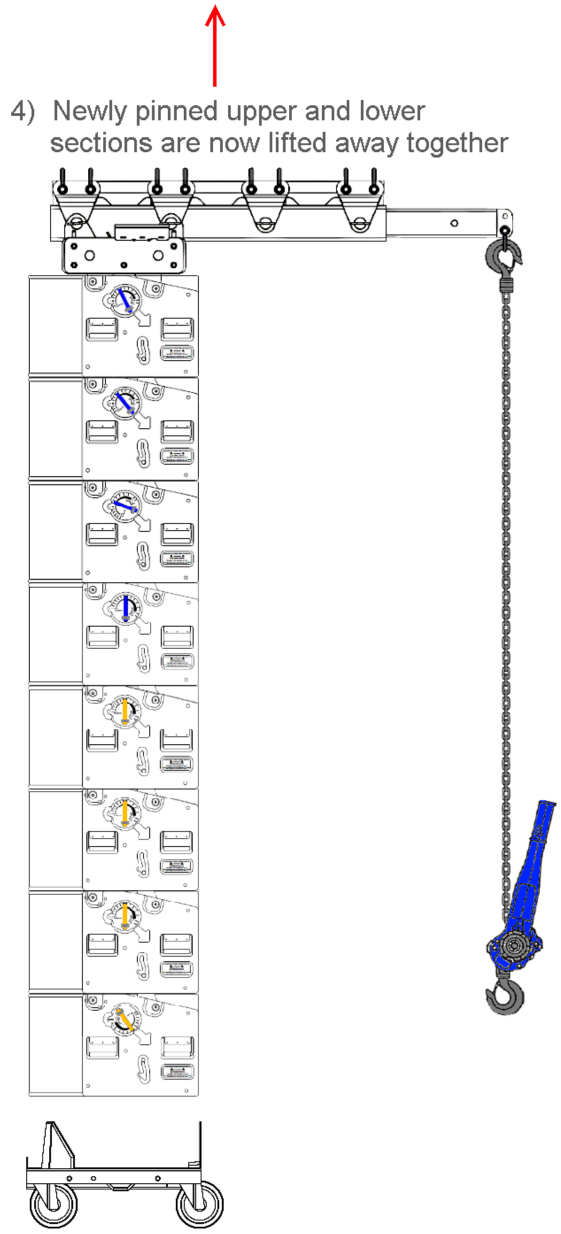
b) Release the bottom flown cabinet's ***Lambda Arms*** by rotating the ***indexer*** on the lower flown loudspeaker to the ***10° Lift*** position *(or any point past this)*

MIND YOUR FINGERS!

c) The Lambda arms will drop into the FlyPlate gaps below, guided by the FlyPlate's Lambda "armpits"

d) Secure the Lambda arms by replacing the pip pins

Again, make sure that the pip-pins are fully home, their buttons have popped out and that their wire lanyards are not twisted or snagged.



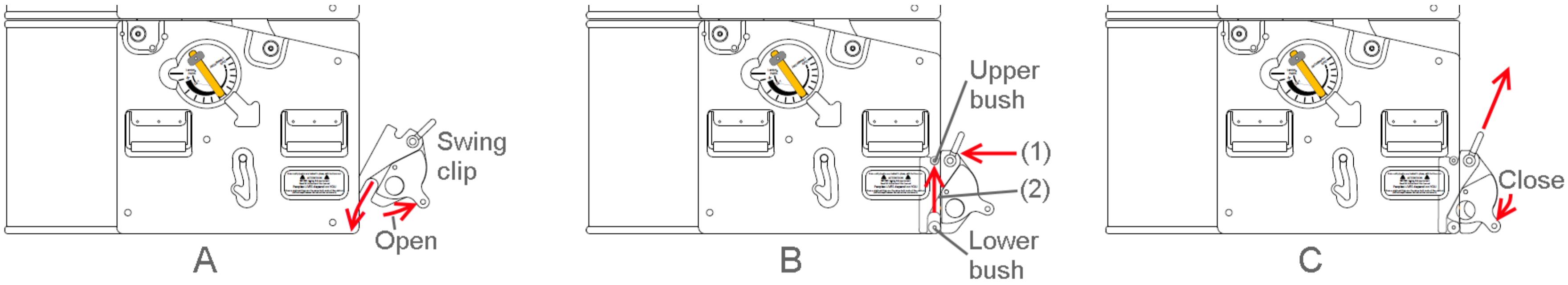
Once you've cross-checked that both sides of the array are secured:

4a) Lift the whole system clear of the dolly *(see left illustration)*

b) Select the remaining angles by unlocking the relevant ***Index locks*** and turning the ***Index handles*** to the required positions. *(Leave the bottom cabinet in the **Lambda Park** position)*

c) Return the ***Index locks*** to their locked position *(90° to the **Index handles**)* on all the cabinets where angles have recently been selected

d) Wheel in a new dolly of loudspeakers and repeat the process until your array is complete.



Rear pull-up points are provided by ***Swing clips*** that are fitted to the lower rear corners of the bottom loudspeaker's FlyPlate assemblies as shown

To fit these *Swing clips* to the array:

Illustration A

With the Swing clip orientated as show:

- With the Spring clip held ***open*** *(see illustration for handle direction)* slot the Swing clip's lower gap in and down over the ***lower bush***

Illustration B

- Whilst keeping the Swing clip ***open***, push the Swing clip in against the cabinet FlyPlate ***(1)***
- Still keeping the Swing clip ***open***, push the Swing clip up so that it engages the ***upper bush (2)***

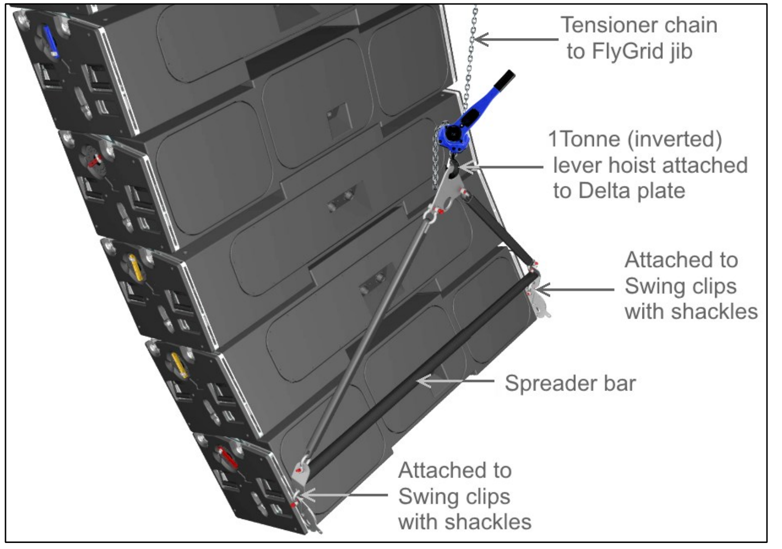
Illustration C

- Let the Swing clip close *(see illustration)*. *This usually occurs under the influence of gravity but make sure by pushing the handle in the "Close" direction*
- Check that the Swing clip is securely attached to the cabinet before fitting the rear ***Spreader bar*/V-YS1 Y-strap/Delta plate assembly*** in place between the Swing clips and the rear jib shackle.

  See next page…

  (*Note that the Spreader bar is only required when flying twelve Vero cabinets or more)

Spreader bar/V-YS1 Y-strap/Delta plate assembly



Spreader bar/V-YS1 Y-strap/Delta plate assembly in place between the Swing clips
(The Spreader bar is only required when flying more than 12 cabinets)

Overview

The *(inverted)* lever hoist is attached to the Delta plate at the bottom, with the tensioner/load chain attached to the FlyGrid jib at the top. It is used to pull the cabinets to their pre-set splay angles. The applied tension is monitored by the Pod on the FlyGrid as follows…

- The FlyGrid should be level *(at zero degrees on the inclinometer)* whilst the array is being set.
- As the lever hoist is used to apply rear tension, the array curvature develops from the bottom to the top.
- When the rear tension is correct, a micro switch activates - causing white LEDs to flash on the handheld display and the grid itself.

Do not over-tighten the lever hoist!

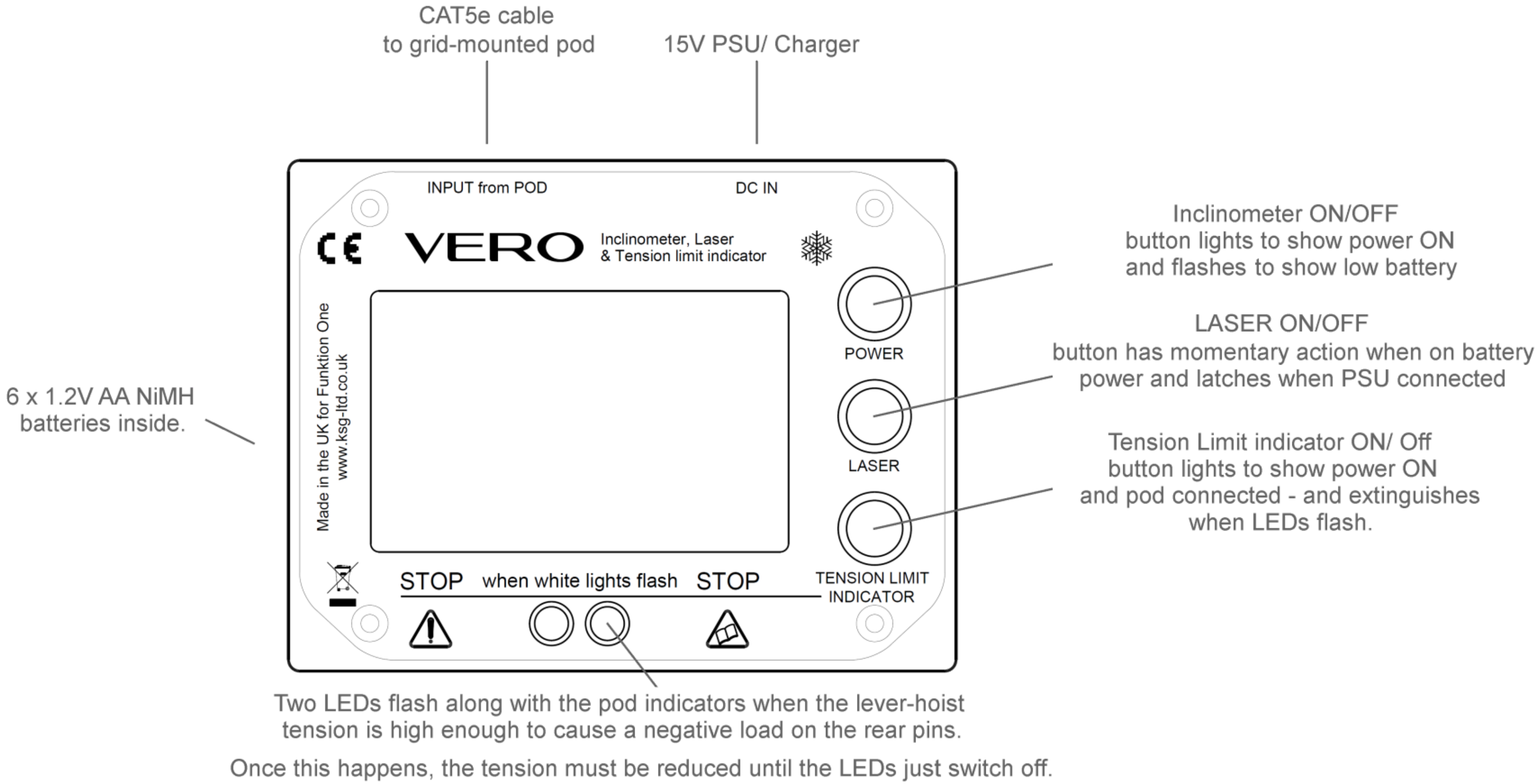
Stop tightening and back off slightly as soon as the FlyGrid-mounted pull-up limit indicator lights up.

Over-tightening could damage the flying hardware.

- The desired inclination or declination can subsequently be added by adjusting the front or rear motors.

(See the next page for further instructions…)

7.3.1 Using the FlyGrid Pod functions

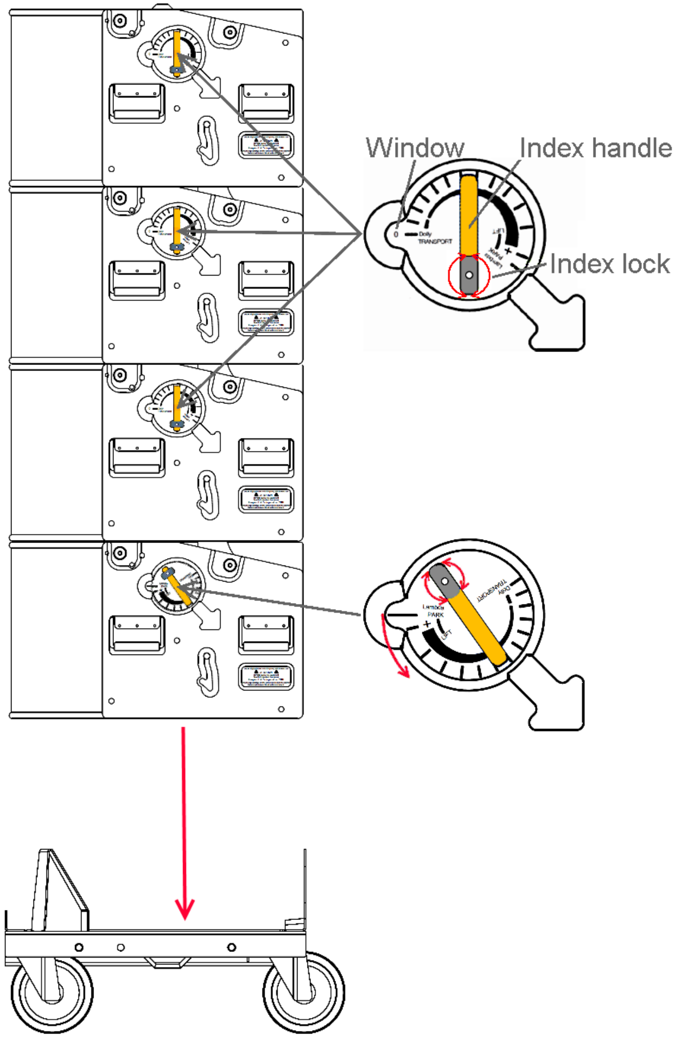


After the grid and cabinets are in place and the tilt chain and lever-hoist are attached…

1) Level the grid using the inclinometer as your guide
2) Once the grid is level, switch on the Tension Limit Indicator
3) Operate the lever-hoist gently and STOP! as soon as the white LEDs start to flash
   (Keep an eye on the inclinometer and adjust the front or rear motors to keep it level)
4) Now reverse the lever-hoist and let off just enough tension to extinguish flashing LEDs
5) With grid now level and Tension limit LEDs just extinguished, the LASER can be turned ON and the array lifted and tilted, as a whole, to its final position
6) The display unit can now be disconnected and used on another grid.

7.4 De-rigging procedure

Preparation



> For your own safety, and as a duty of care to those around you, please observe the strict rigging safety procedures outlined earlier.

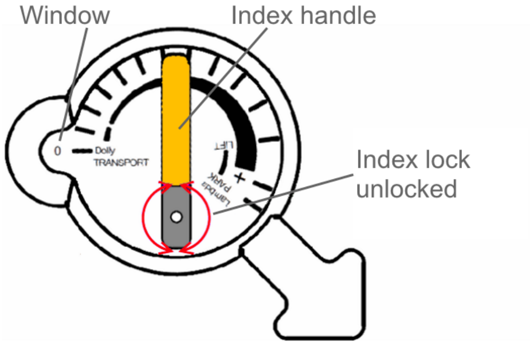
Level the grid

- Warn anyone near the array to stand clear before gradually releasing the pull-up tension using the lever hoist to enable the rear Y-strap and Swing clips to be removed from the lowest loudspeaker
- Lower the array so that the bottom loudspeaker is just above the transport dolly

With the Y-strap and Swing clips removed and stowed:

- Whilst the array is still suspended just above the transport dolly, unlock the ***Index locks***
- Set the **<u>upper three cabinets'</u>** ***Index handles*** to ***0° Dolly TRANSPORT*** ⇦ *(See left illustration)*

  Note that the angles are far easier to adjust when the array is still suspended.



Upper three cabinets' Index handle setting

- **Note that the <u>bottom cabinet</u> settings are different!**
  Follow the instructions overleaf to retract the lower cabinet's Lambda arms *(using the Lambda lift knob)* and to set its ***Index handles*** to ***Park*** – *see next page…*

Parking the bottom loudspeaker's Lambda arms

To avoid damage, the bottom cabinet's Lambda arms must be retracted before lowering the block of Vero cabinets onto a dolly.

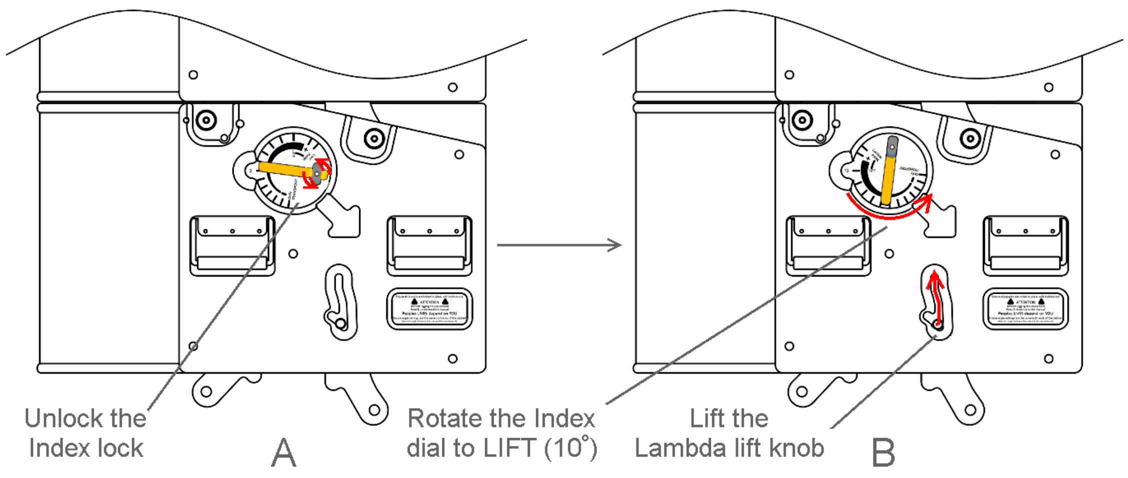
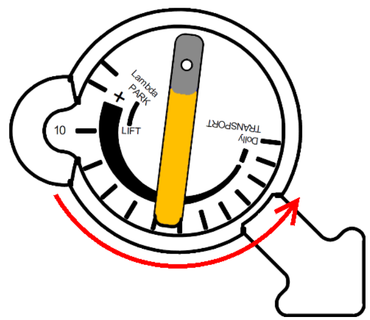


Illustration A

- Unlock the FlyPlate ***Index locks***

Illustration B

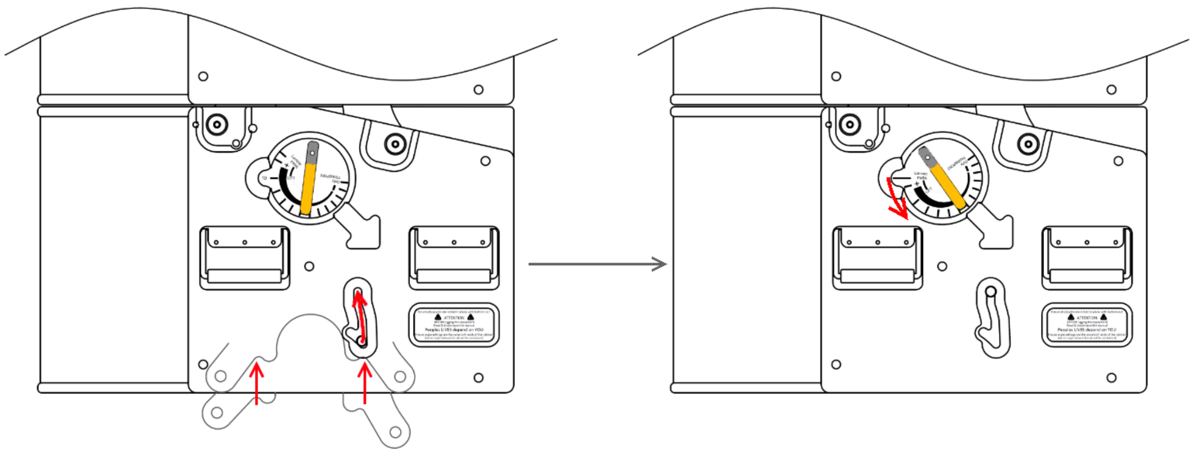
- Rotate the index dial to the ***LIFT*** *(10°)* position whilst moving the Lambda lift knob to its highest position to retract the Lambda arm



Rotate Index dial to *LIFT* *(10°)*

Illustrations C & D *(overleaf)*

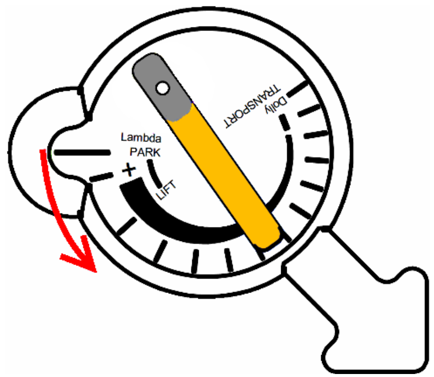
- Continue to hold the Lambda lift knob at its highest position whilst rotating the index dial counter-clockwise to the ***Lambda PARK*** position. *(See next page for details)*



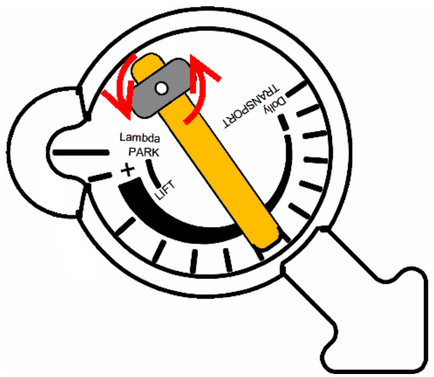
C D

Illustrations C *(before)* & D *(after)*

- Rotating the Index dial to the ***Lambda PARK*** position *(Illustration D)* engages the internal retaining mechanism that keeps the Lambda arm safely retracted inside the FlyPlate ⇨

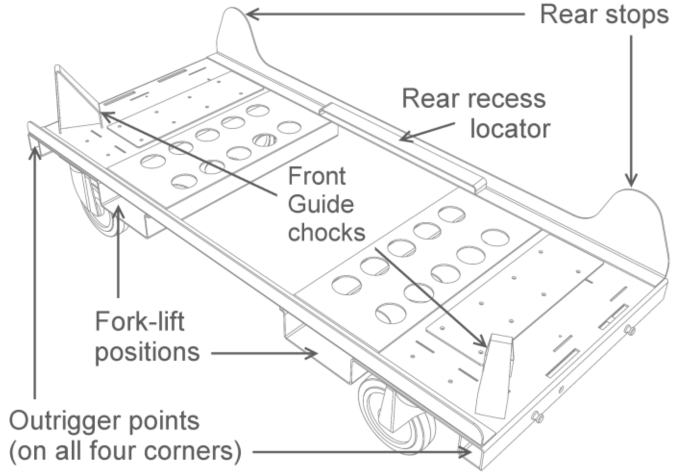


Rotate Index dial to Lambda Park



Then re-lock the Index locks

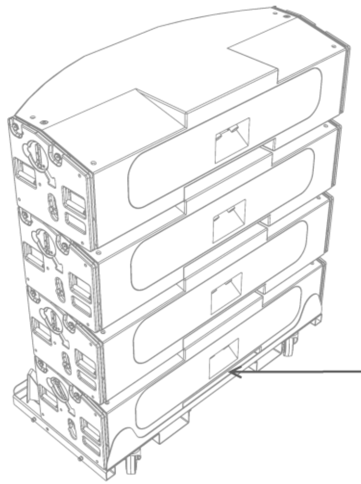
- Re-lock the ***Index locks*** by turning them 90° to the index handle. Make sure they click into place . . . .



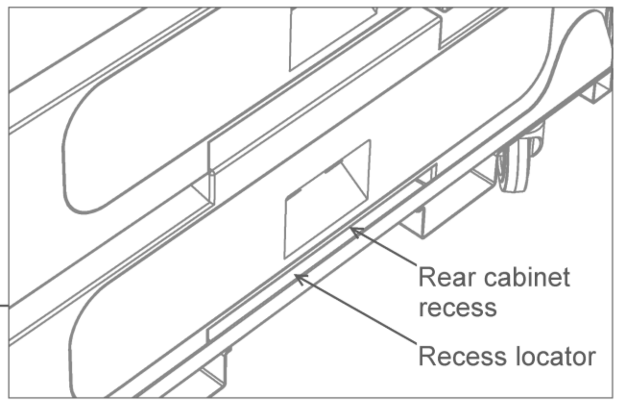
V-dolly features

Lower the bottom four cabinets onto the V-dolly

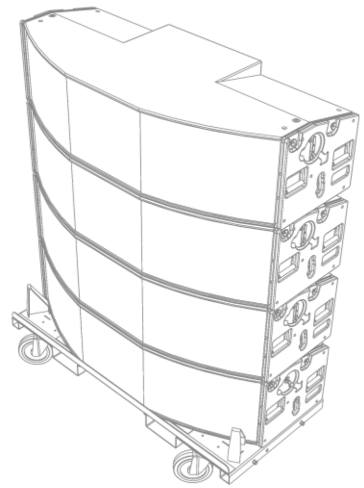
- Ensure that the dolly is level before attempting to place the bottom four Vero cabinets on it. Adjustable outriggers may be inserted into the corner points to level the dolly on uneven ground or to tilt a Vero stack – see ***Section 7.5*** next.
- The dolly's front guide chocks and rear recess locator allow you to place your Vero cabinets onto the dolly accurately
- Although the dolly is ruggedly built, be careful when placing the array on the dolly. Use single motor clicks for precise placement
- Ensure that the bottom four Vero cabinets are sitting flat on the dolly surface
- Unpin the top loudspeaker from the above speaker's Lambda arms
- Always ensure that Vero cabinets are transported in perfect show-ready condition with *all* their pins in place



4 x Vero cabinets on dolly – rear view



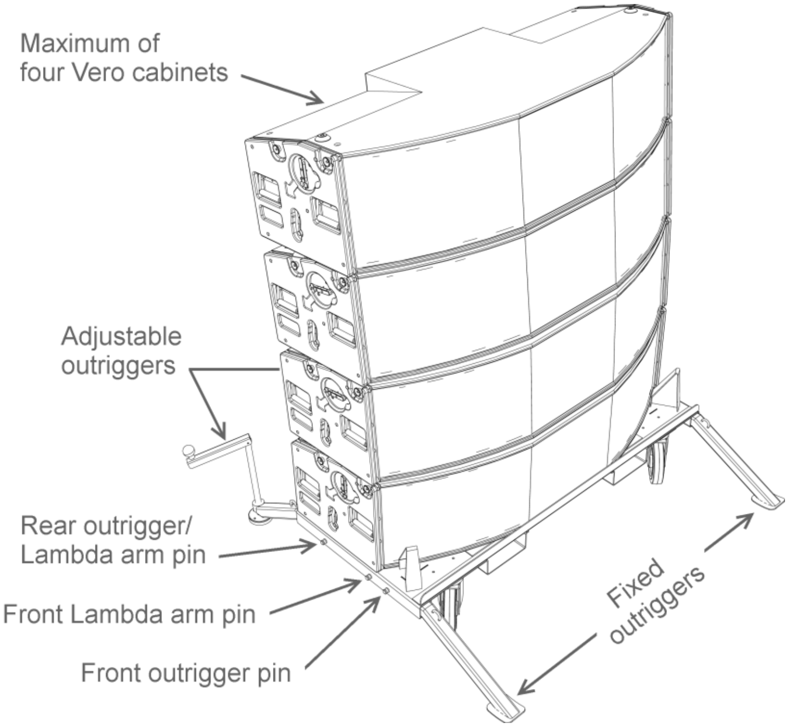
Rear cabinet recess locator
(Also see top left illustration)



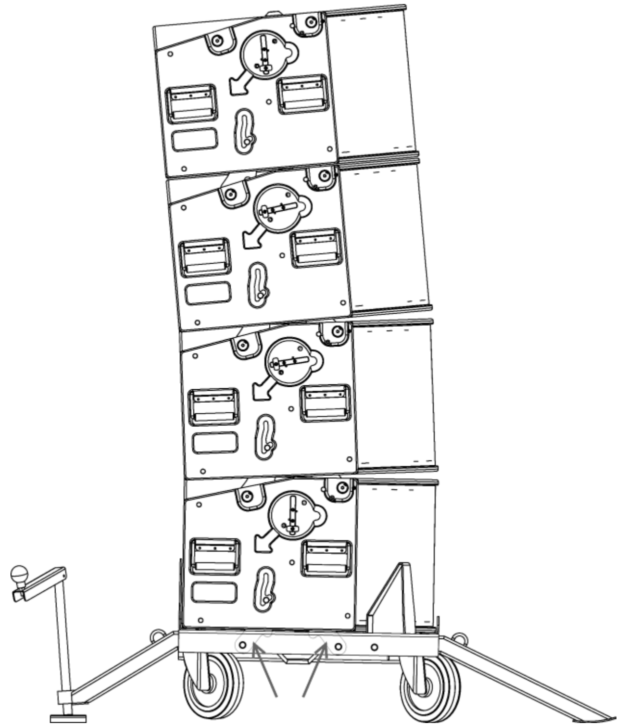
4 x Vero cabinets on dolly – front view

7.5 Dolly-mounted Vero stacks using outriggers

Vero stacks should be limited to four cabinets per stack.
Use spreader boards *(not shown)* under the outrigger feet on soft ground
Vero stacks should be tethered to prevent toppling if used on uneven surfaces – especially in gusty conditions.



Dolly-mounted Vero stack *(with adjustable outriggers at rear in this example)*



Bottom cabinet's Lambda arms *(arrowed – usually hidden, of course)* pinned at 0°

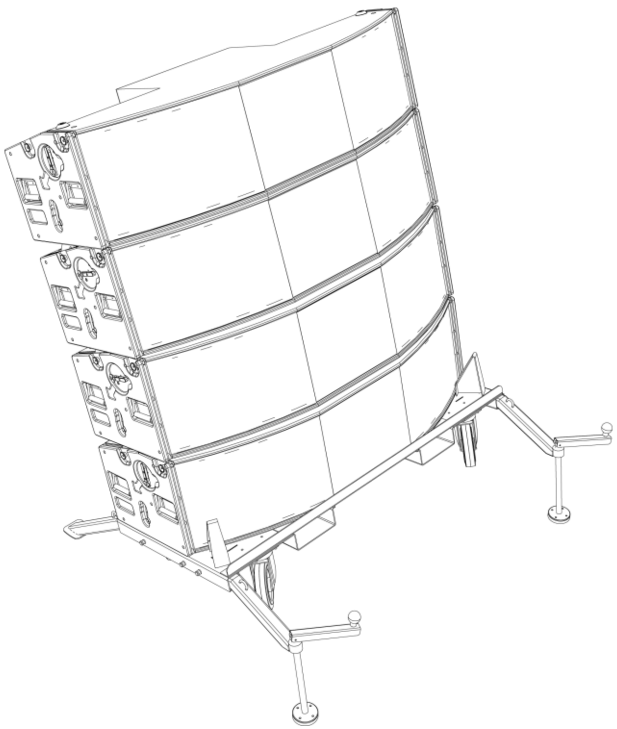
Fixed and adjustable outriggers may be plugged into the Vero dolly chassis ends *(as shown above)* to stabilise Vero stacks and to provide overall tilt adjustment.

Three pins per side *(arrowed in left illustration above)* are used to secure the outriggers and the bottom cabinet to the dolly:

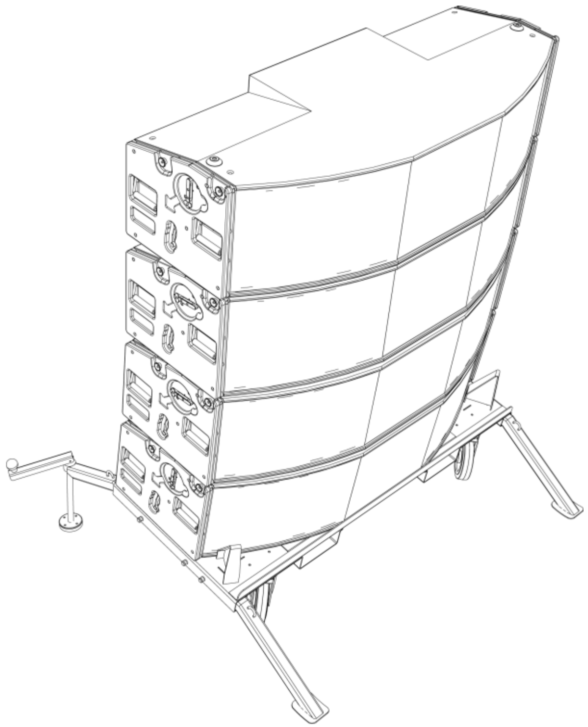
- The front pins secure the front left and right outriggers
- The middle pins secure the bottom Vero cabinet's front left and right Lambda arms *(set to 0°)*
- The rear pins serve a dual role: They secure the rear left and right outriggers and secure the bottom Vero cabinet's rear left and right Lambda arm.

Fixed and adjustable outrigger deployment

The choice of front or rear adjustable outrigger position depends on the type of tilt required. Use front deployment for up-tilts and rear deployment for down-tilts.



Dolly mounted Vero stack – front outrigger adjustment providing accurate up-tilt



Dolly mounted Vero stack – rear outrigger adjustment providing accurate down-tilt

The left-hand illustration shows adjustable outriggers deployed at the front. This configuration makes it easy to set accurate up-tilt angles. Up-tilted set-ups can be used for temporary distributed systems around sports stadiums or for ground-stacked bleacher fills.

In the right-hand illustration, the adjustable outriggers have been deployed at the rear to set accurate down-tilt angles – often required when a stack is placed on high stage wings, for example.

Note that normal array curvature design rules apply where larger audience areas are to be covered. As with flown systems, Vero stacks provide the smoothest audience coverage when smaller inter-cabinet angles are used at the top of the array - see the side view *(right-hand illustration on previous page)*.

Where cabinet 1 is at the top, use a smaller inter-cabinet angle between cabinets 1 & 2 *(set using the top cabinet's index handles)*, a slightly larger angle between cabinets 2 & 3 *(using cabinet 2's index handles)* and the same or slightly larger angle still between cabinets 3 & 4 *(using cabinet 3's index handles)*. Cabinet 4's index handles are usually set to 0° so that it sits firmly on the dolly when pinned in place.

8 In-concert limiter considerations

8.1 Vero limiter dos and don'ts

Vero series limiters are preset in the Dolby® Lake™ controller's ***Load Libraries*** and may be adjusted via the Dante network.

Please note that, for warranty purposes, and to avoid prematurely ageing your system:

- Vero preset limiter thresholds should only be adjusted by approved system technicians
- Vero preset limiter thresholds must never be increased
- Vero preset limiters may only be reduced to cater for clipped programme signals or to maintain array performance and spectral balance

Reducing limiter thresholds to cater for clipped programme signals

The ***initial*** load library limiter settings provide overdrive protection for your Vero series loudspeakers for normal, *unclipped* music and speech use. However, clipped signals can dump up to twice as much power into your system.

Although Vero series loudspeakers are active systems and less prone to damage than a full-range passive system would be, it is worth reducing the initial limiter thresholds by 3dB to prevent premature ageing of components when renting out your system to, perhaps, less professional users.

For more about clipping, see ***Appendix D*** towards the end of this user guide.

8.2 Final limiter adjustments for spectral balance

Another reason for reducing limiter thresholds is to maintain spectral balance when the system is being driven into limit.

Mid and high frequency drivers tend to have lighter, more delicate voice coils and smaller magnetic assemblies than lower frequency elements. This causes them to heat up much faster and, therefore, have lower AES power ratings.

However, these lower power ratings don't usually restrict the mid and high frequency output spl because:

1) Mid and high frequency horns can be designed for very high efficiency and…
2) Natural music and speech signals tend to have quite a "pink" rms spectrum - i.e. signal levels tend to reduce slightly with increasing frequency

Not all amplified material is "natural" - we're a creative species. Most club-goers love sound systems with lots of that classic Funktion One bass impact and this means that, as levels rise *(and despite the mids and highs having lower power ratings)*, the bass system is often the first to go into limit leaving the mids and highs to carry on increasing in level.
It is often, therefore, a good idea to trim the limiter thresholds of the mid and high sections to maintain full-limit spectral balance and to protect your audience from discomfort.

Balanced sound 'On the limit'

Funktion One's John Newsham explains;

"Here are some common scenarios:

a) You've got your mix set up and ''in the pocket'' then the singer gets a bit excited and shouts ''Hello (insert your town here)! Are we all having a good time??!'' His vocal compressor is set at a low ratio (or there isn't one) and you're not quick enough on the fader to catch the sudden burst of level. Happily, your system limiters are set a couple of dB before the clip point of the amps in accordance with the power handling of the drivers and the amps don't clip

b) A sudden burst of feedback from a vocal mic too near a side fill and the same thing happens

c) An inexperienced guest engineer loses control of the mix and ends up running too loud, the limiters start to flash so we ask him to back off a bit. He's surprised that it starts to sound louder when the masters are pulled down a little and the limiters stop pumping. (How many times have I seen this?).

In all these cases the limiters are set for system safety and nothing gets broken, all good but what does the system sound like and how loud is it when the limiters come into play?

Mid and HF horns are usually more efficient than low mid and bass drivers. Of course, they handle less power, but you can often get a situation where if everything is just set for safety the system is capable of being driven into a state of imbalance where it sounds bad. It can even be way too loud for the venue and the safety of the people in it.

Bass speakers these days can handle a massive amount of power (if the signal is clean) so large amps are used and limiters set to prevent clipping.

Mid and high limiters are set progressively tighter in respect to the drivers' lesser power handling.

It's still a good idea to use a large amp though. We are happy to allow transient spikes like snare hits or cymbal crashes, to get past the limiters because these are valid parts of the mix. The dynamics and excitement of the music are preserved, and these transients are not long enough to overheat the drivers - assuming they're not clipping, of course.

So, what happens as the system comes up to full power?

1) Often the bass will limit first, then as the system is driven harder the low-mid starts to limit, the high-mids are still going strong as are the highs

2) If the bass limits 4 dB before the mid-highs this is effectively the same as turning up the mid-highs by 4dB. The balance of the system is completely wrong, and all the hard work of tuning and sound check is wasted.

The thing to do here is to reduce the limiter threshold settings still further so that the mids and highs start to limit at around the same point as the bass. This way the frequency content of the music will be maintained when the system is being pushed into limit and the mids will stay at a level where they still relate to the low end.

The settings for this will be different for live and club systems. You will still, of course, want to leave a bit more headroom for the dynamics of live music and in a night club there may be noise restrictions, or the owner may have an idea of how loud he wants the system to go.

The principle is the same. In both cases run the system up to the required maximum level, reduce the limiters so that the limit lights are flashing then open a dB or so to account for when the room is warm and full of people, come back later when the venue is running and fine tune if needed. Above all, lock the system to safeguard against tampering!

Your system will now not only be safe against damage, but it will also be safe against being driven out of balance and sounding bad.”

9 Specifications

Vero V60 mid/high system

Horizontal dispersion:	60°		
Vertical splay angle range:	0° - 6° for contiguous coverage. Up to 12.5° available if required for coverage gaps		
Frequency response:	200Hz – 18kHz		
Driver complement:	Mid:	2 x 16Ω, 10” cone drivers in parallel	
	HF:	3 x 32Ω, 1.4” compression drivers in parallel	
Nominal impedance:	Mid:	Nominally 8Ω over its pass-band. See curves later	
	HF:	Nominally 10.67Ω over its pass-band. See curves later	
Power ratings:	Mid:	500W AES, 2kW peak	
	HF:	225W AES, 900W peak	
Voltage ratings:	Mid:	63.2vrms, 126.4vpk	
	HF:	69vrms, 138vpk	
Broadband sensitivity:	Mid:	108dB for 1W at 1m in free space	
	HF:	112dB for 1W at 1m in free space	
Calculated max spl:	Mid *(1m from source)*:	135dB spl rms, 141dB spl peak in free space	
See Appendix F	HF *(1m from source)*:	135dB spl rms, 141dB spl peak in free space	
Connectors:	2 x 8-pin Syntax *(pin-to-pin male-female pair)*, A - D link through, 2 x parallel mids on E+/F-, 3 x parallel HFs on G+/H-		
Enclosure material:	15mm thick birch ply enclosure		
Enclosure finish:	High durability Polyurea coating		
Rigging:	Integral Lambda® rigging system with colour-coded Index dial *(blue)*		
Working load limit:	2,700kg *(5,952lb)*		
Handling:	Generous flip handles		
Weight:	115kg *(253.5lb)*		
Dimensions:	1,679mm *(5’ 6 2/16”)* wide x 430mm *(1’ 4 15/16”)* high x 716mm *(2’ 4 3/16”)* deep		

Vero V90 mid/high system

Horizontal dispersion:	90°	
Vertical splay angle range:	0° - 12.5° for contiguous coverage	
Frequency response:	200Hz – 18kHz	
Driver complement:	Mid:	2 x 16Ω, 10" cone drivers in parallel
	HF:	2 x 32Ω, 1.4" compression drivers in parallel
Nominal impedance:	Mid:	Nominally 8Ω over its pass-band. See curves later
	HF:	Nominally 16Ω over its pass-band. See curves later
Power ratings:	Mid:	500W AES, 2kW peak
	HF:	150W AES, 600W peak
Voltage ratings:	Mid:	63.2vrms, 126.4vpk
	HF:	69vrms, 138vpk
Broadband sensitivity:	Mid:	108dB for 1W at 1m in free space
	HF:	110dB for 1W at 1m in free space
Calculated max spl:	Mid *(1m from source)*:	135dB spl rms, 141dB spl peak in free space
See Appendix F	HF *(1m from source)*:	132dB spl rms, 138dB spl peak in free space
Connectors:	2 x 8-pin Syntax *(pin-to-pin male-female pair)*, A - D link through, 2 x parallel mids on E+/F-, 3 x parallel HFs on G+/H-	
Enclosure material:	15mm thick birch ply enclosure	
Enclosure finish:	High durability Polyurea coating	
Rigging:	Integral Lambda® rigging system with colour-coded Index dial *(yellow)*	
Working load limit:	2,700kg *(5,952lb)*	
Handling:	Generous flip handles	
Weight:	107kg *(235.9lb)*	
Dimensions:	1,679mm *(5' 6 2/16")* wide x 430mm *(1' 4 15/16")* high x 716mm *(2' 4 3/16")* deep	

Vero V315 mid-bass system

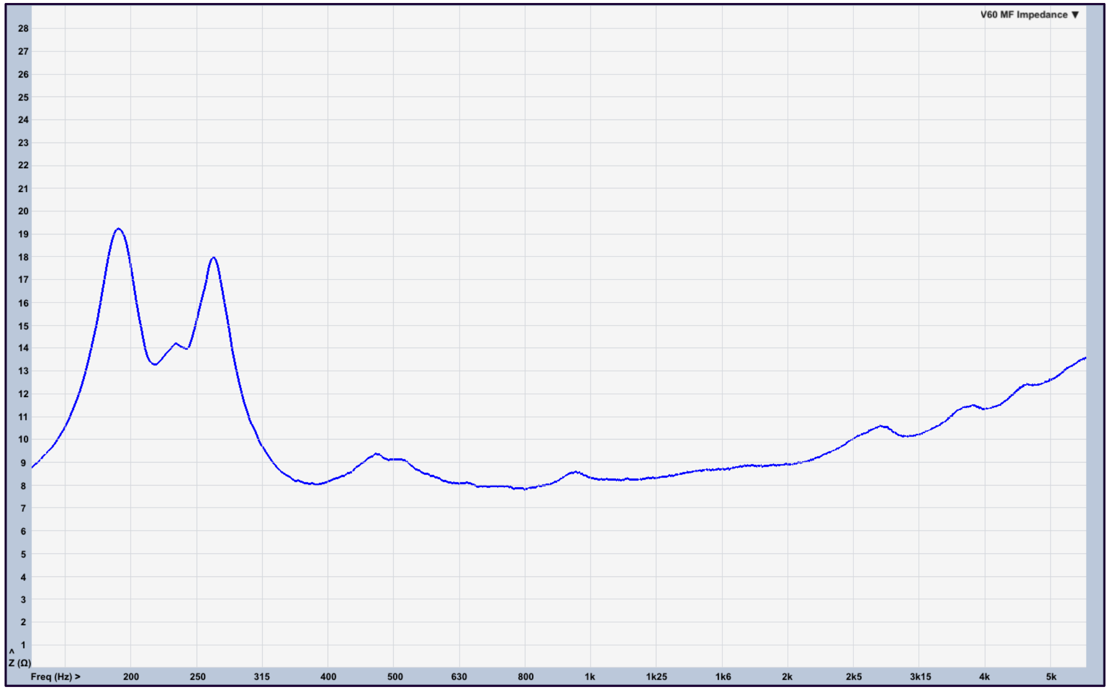
Horizontal dispersion:	70° at 200Hz
Vertical splay angle range:	0° - 5°
Frequency response:	50Hz – 250Hz
Driver complement:	3 x 8Ω, 15" cone driver in parallel
Nominal impedance:	Low mid: Nominally 2.67Ω over its pass-band. See curves later
Power ratings:	1.2kW AES, 4.8kW peak
Voltage ratings:	56.5vrms, 113vpk
Broadband sensitivity:	106dB for 1W at 1m in free space
Calculated max spl: ***See Appendix F***	136dB spl rms, 142dB spl peak in free space
Connectors:	Single SpeakOn NL4. *(3 x parallel drivers on 1+, 1-)*
Enclosure material:	15mm thick birch ply enclosure
Enclosure finish:	High durability Polyurea coating
Rigging:	Integral Lambda® rigging system with colour-coded Index dial *(red)*
Working load limit:	2,700kg *(5,952lb)*
Handling:	Generous flip handles
Weight:	116kg *(255.7lb)*
Dimensions:	1,679mm *(5' 6 2/16")* wide x 430mm *(1' 4 15/16")* high x 716mm *(2' 4 3/16")* deep

Vero V221 system

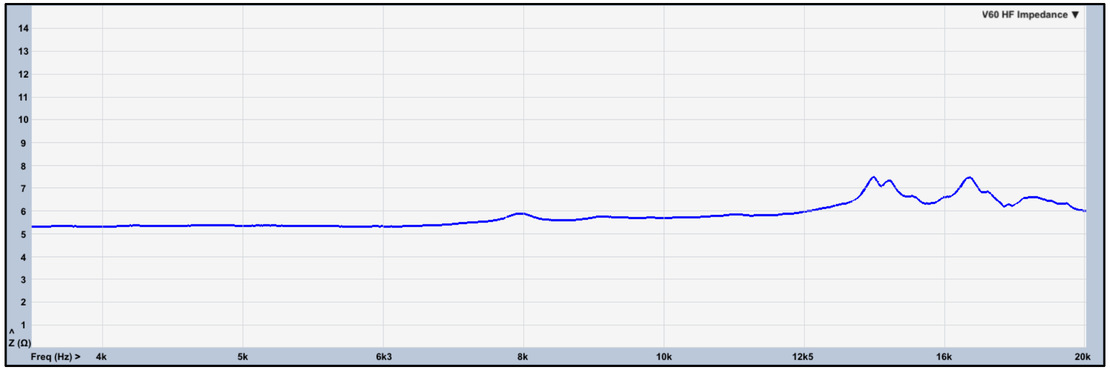
Frequency response:	40Hz – 150Hz
Driver complement:	2 x 8Ω 21" cone driver. 4Ω when connected in parallel
Power ratings:	1.5kW AES, 6kW peak
Voltage ratings:	77.4vrms, 154.8vpk

Sensitivity:	107dB for 1W at 1m in half space
Calculated max spl: ***See Appendix F***	139dB spl rms, 145dB spl peak in half space
Connectors:	2 x SpeakOn NL4 *(pin-to-pin male-female pair)*, one driver on 1+/1-, the other on 2+/2-
Enclosure material:	18mm thick birch ply enclosure
Enclosure finish:	High durability Polyurea coating
Weight:	130kg *(286.6lb)*
Dimensions:	1,688mm *(5’ 6 7/16”)* wide x 589mm *(1’ 11 3/16”)* high x 1002mm *(3’ 3 7/16”)* deep

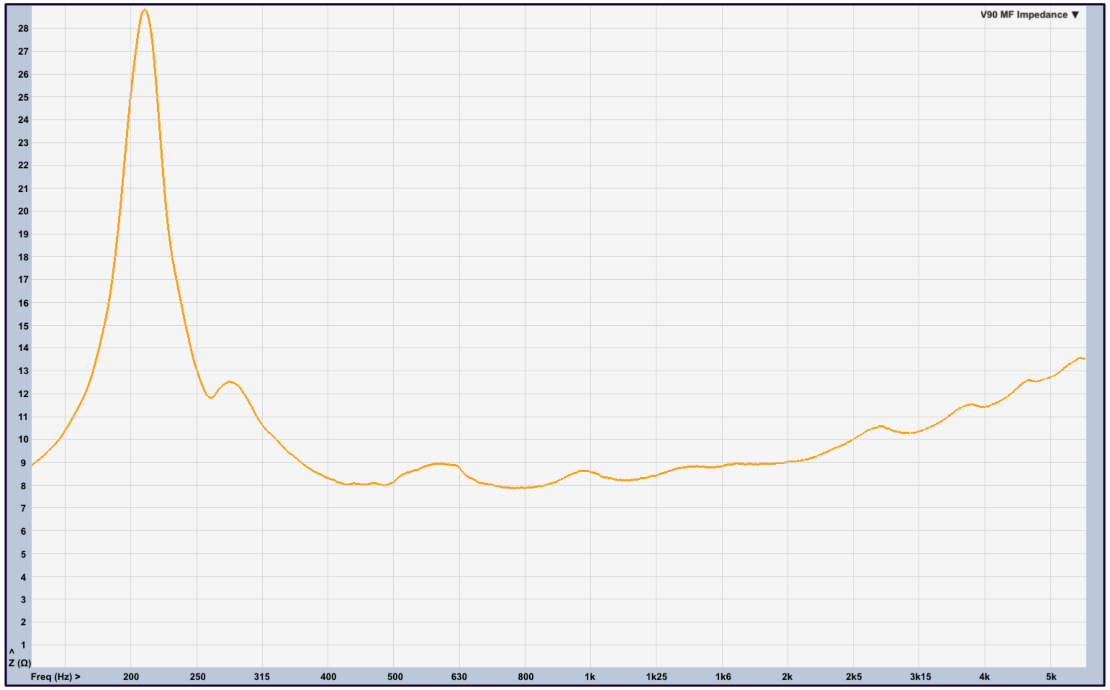
9.1 V60, V90 & V315 impedance curves



V60 mid frequency impedance

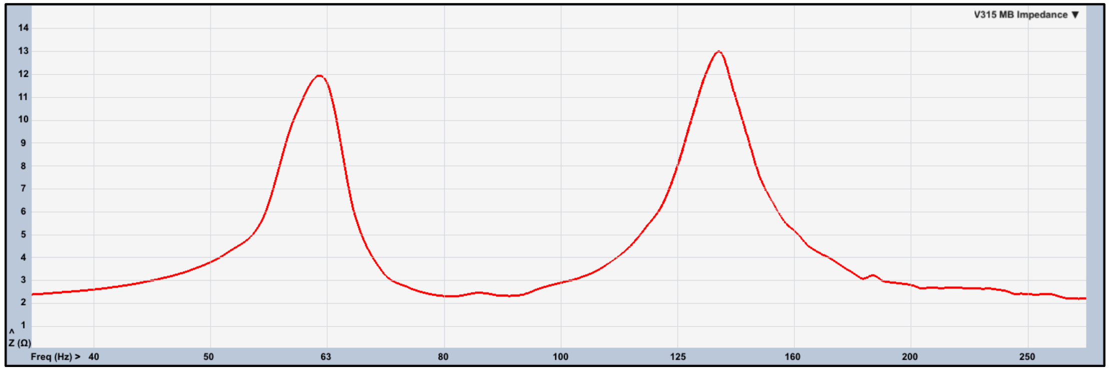


V60 high frequency impedance



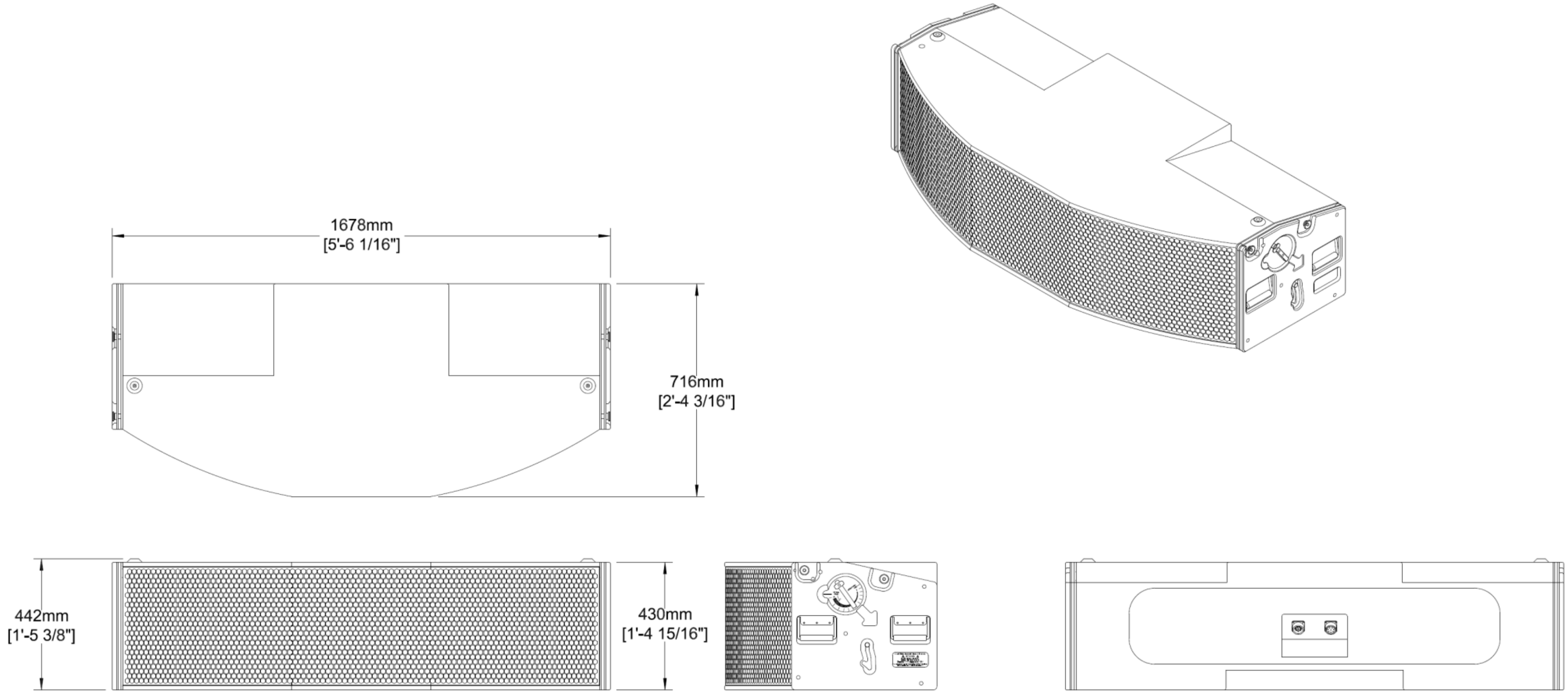
V90 mid frequency impedance

V90 high frequency impedance

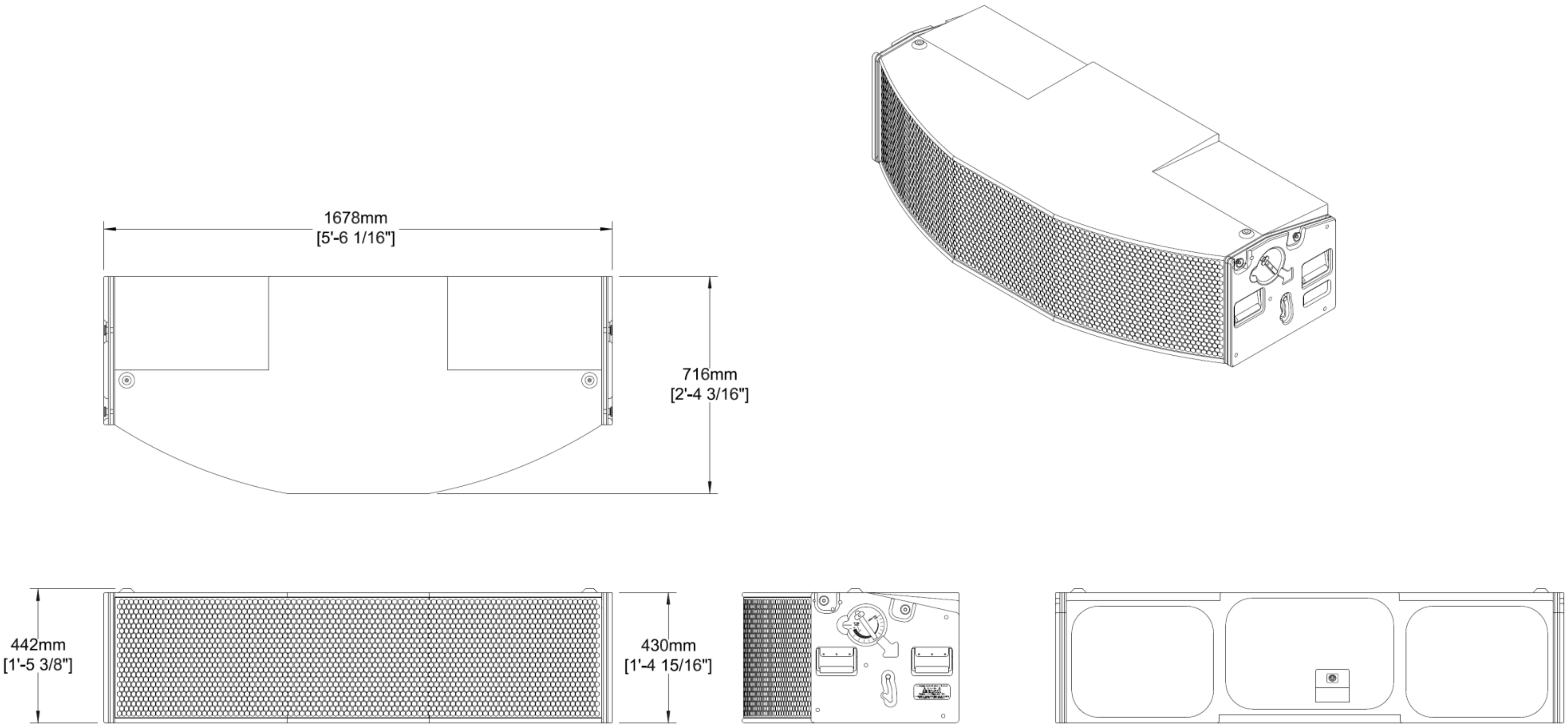


V315 mid-bass impedance

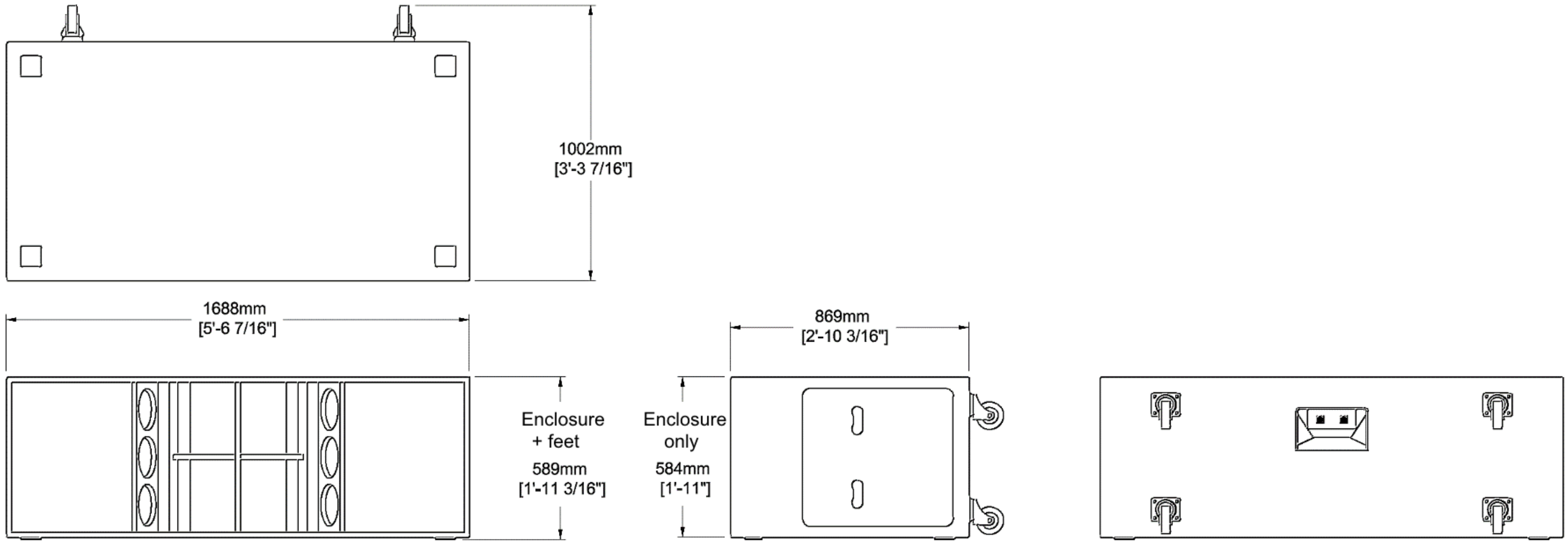
9.2 Cabinet dimensions



V60, V90 dimensions in millimetres *(and feet, inches & 1/16th inches)*



V315 dimensions in millimetres *(and feet, inches & 1/16th inches)*

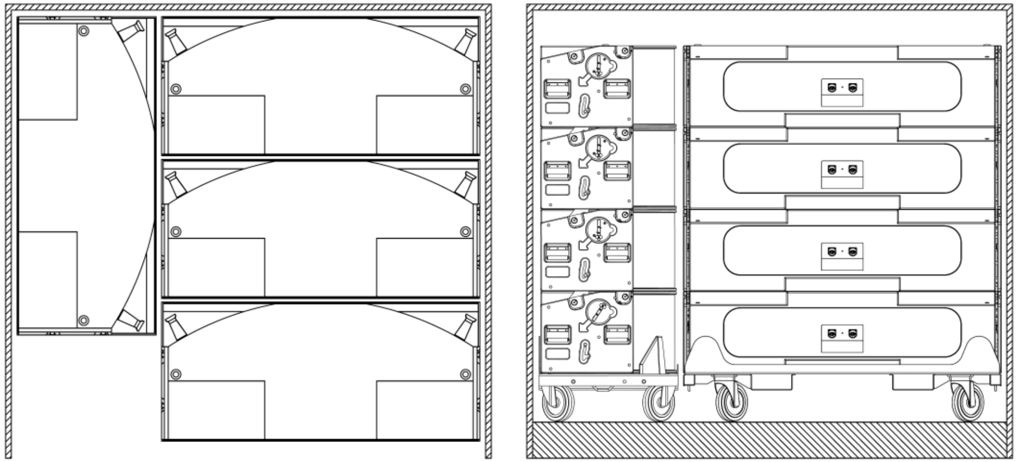


V221 dimensions in millimetres *(and feet, inches & 1/16th inches)*

Further Vero-compatible bass systems

For further Vero-compatible bass systems and to keep up-to-date with application examples, hints and tips, see ***www.vero-system.com/system/bass_range/***.

9.3 Truck pack - and 4-cabinet dolly, amp rack & rigging trunk dimensions



Vero system truck pack

2.44m (8ft) wide truck pack

⇦ *See illustration*

When trucking Vero systems...

i) Always transport Vero cabinets on their dollies

ii) Before loading each dolly, ensure that:
   a) The lower cabinet's index handle is set to ***Park***
   b) The upper three cabinets' index handles are set to ***0° Dolly TRANSPORT***

iii) Load each dolly with the grilles innermost for protection

iv) Turn the dolly wheels inwards (as illustrated left) for maximum pack density and stability

v) Place load restraints (e.g. shoring bars) or ratchet straps across the rear of the load for stability.
(To avoid damage, don't shore or strap across the cabinet grilles)

4 x Vero dolly dimensions

See illustration ⇨

Amplifier rack dimensions *(per 8 x V60/V90 + 4 x V315 + 6 x V221)*
(not shown)

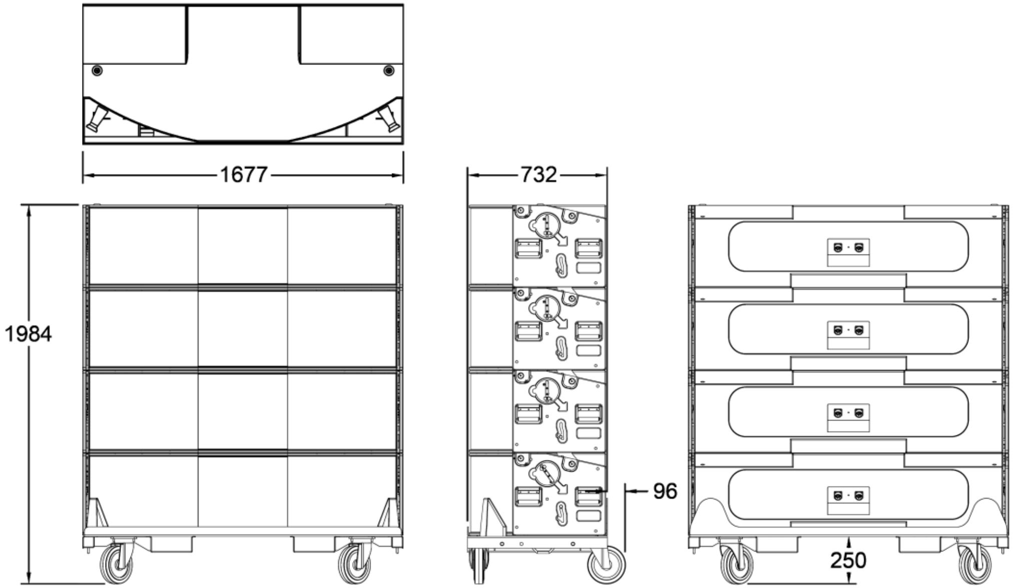
543mm *(w)* x 646mm *(h)* x 639mm *(d)*
(includes 128mm high dolly-wheels)

Houses 3 x amplifiers, mains distribution panel and 1U space for network switch

Rigging trunk *dimensions (per array)*
(not shown)

1735mm *(w)* x 637mm *(h)* x 570mm *(d)*
(includes lid and 128mm high dolly-wheels)

Houses 1 x FlyGrid, lever-hoist and accessories.



4 x Vero dolly dimensions (in mm)

9.4 4-cabinet dolly cover and lid

A 4-cabinet dolly cover is available to protect your Vero system from the rigours of regular trucking.

Vero 4-cabinet dolly cover

Lid

A sturdy lid under the cover provides a solid extra surface for efficient truck packing.

Easy rear access

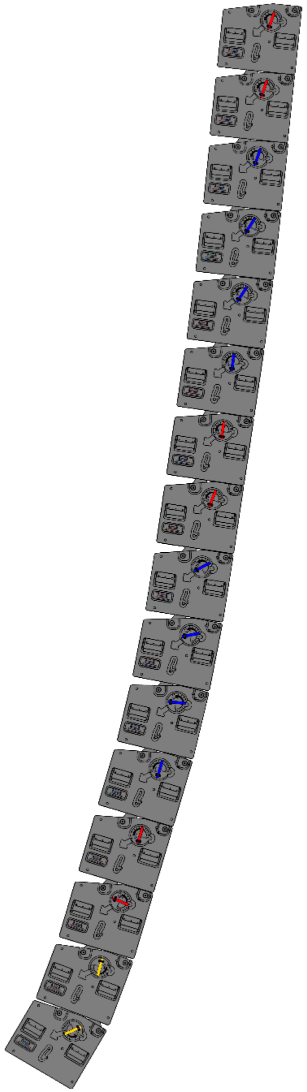
Easy weather-protected access is available to the rear connectors and the lower three sets of side handles.

Inspection windows allow Index settings to be checked.

Inspection window

Weather-protected handle access

Appendix A - Geometric Energy Summation (GES)



Vero's Geometric Energy Sum *(GES)*

The easiest way to describe Funktion One's Geometric Energy Sum characteristic is to run through a large-scale Vero array design…

8 x V60 + 2 x V90 + 6 x V315 array for a flat large-scale festival field

Here is a 16-cabinet very high-power Vero array designed to cover a large-scale outdoor festival audience.
It comprises – from top to bottom:

- 2 x V315 mid-bass loudspeakers*
- 4 x V60 mid/high loudspeakers – arrayed at low vertical splay angles** for a high GES to cover the furthest audience areas
- 2 more V315 mid-bass loudspeakers*
- 4 x V60 mid/high loudspeakers – arrayed at medium vertical splay angles* for a medium GES to cover mid audience areas (often including *the mix position)*
- 2 more V315 mid-bass loudspeakers*
- 2 x V90 mid/high loudspeakers – arrayed at a wide vertical splay angle for front audience coverage

* Final V315 mid-bass positions will vary depending on the vertical array curvature *(which determines the vertical coverage profile)* and the required mid-bass coverage control. For instance, V315 vertical spacing may be chosen to deliberately create a mid-bass null beneath the array. Especially if the system is being used for orchestral amplification and there are low string sections near the stage wings. More about this later.

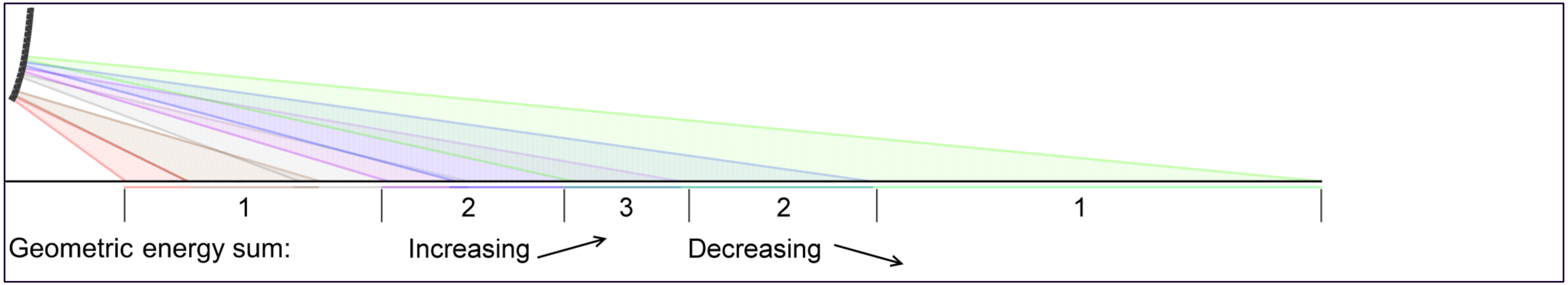
** For a flat audience area, the vertical inter-cabinet splay angles increase sequentially from top to bottom to provide a smooth transition from a higher GES at the top *(for distant audience coverage)*, through a medium GES for the middle sections *(for mix/ mid audience coverage)*, to a low or zero GES section at the bottom *(for front audience coverage)*.

Each loudspeaker in the Vero range exhibits a radial vertical coverage characteristic for excellent pattern control whilst minimising the multitude of HF lobes that severely reduce the bandwidth and definition of a typical "line array".

Although each Vero element's radial attenuation follows the inverse square law for sound intensity vs distance (so sound pressure level decreases linearly at 6dB per doubling of distance), the extra summation of more cabinets vs distance partially offsets the normal radial attenuation. See the following illustrations to see how…

Lower 2 x V90 wide-splayed

Little or no mid/high coverage overlap occurs here so the listener benefits from just one V90 at a time *(upper V90 coverage is brown, lower V90 coverage is red)*.

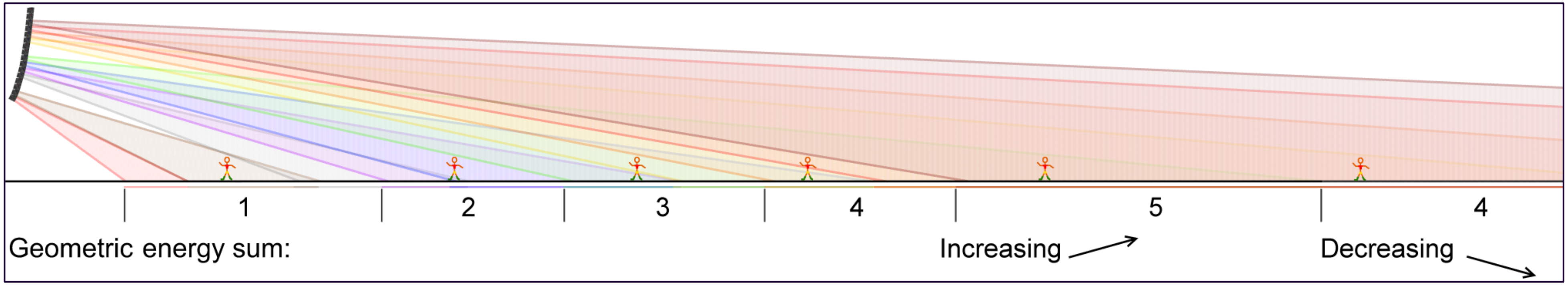


Lower 2 x V90 plus an additional middle 4 x V60 *(V315 mid-bass radiation left off for clarity)*

Mid/high geometric energy summation *(GES)* increases, initially, with distance. From single-cabinet coverage *(simply the red or the brown V90 coverage)*, to 2 x V60 *(the grey + the violet V60 then the violet + the blue V60)*, to 3 x V60 *(the violet + the blue + the green V60)*. See illustrations above…

This increasing GES reduces the normal *(6dB/distance doubling)* radial attenuation, maintaining smooth coverage with distance until all of the V60s are contributing. This maximum GES distance is usually referred to as the transition distance. See next page…

GES progressively reduces beyond that transition distance – back down to the sum of 2 cabinets *(the blue + the green V60)* and then to single-cabinet coverage *(the green V60)* where the inverse sound pressure vs distance law takes over again. See next page…



Lower 2 x V90 plus the middle 4 x V60 plus an additional 4 x V60 at the top *(V315 mid-bass radiation left off for clarity)*

Now let's have a look at a full system from a listener's point of view. ⇨ 

As listeners walk away from the tilted array, they'll pass through the single-cabinet V90 area *(1)*, through the 2-cabinet V60 area *(2)*, onto the 3-cabinet V60 area *(3)*, and through the 4-cabinet V60 area *(4 - increasing)* where GES is gradually increasing to partially counteract radial attenuation.

They'll eventually reach the 5-cabinet maximum GES area *(5)*. Beyond this transition distance, there are no further V60s to compensate for the inverse sound pressure vs distance.

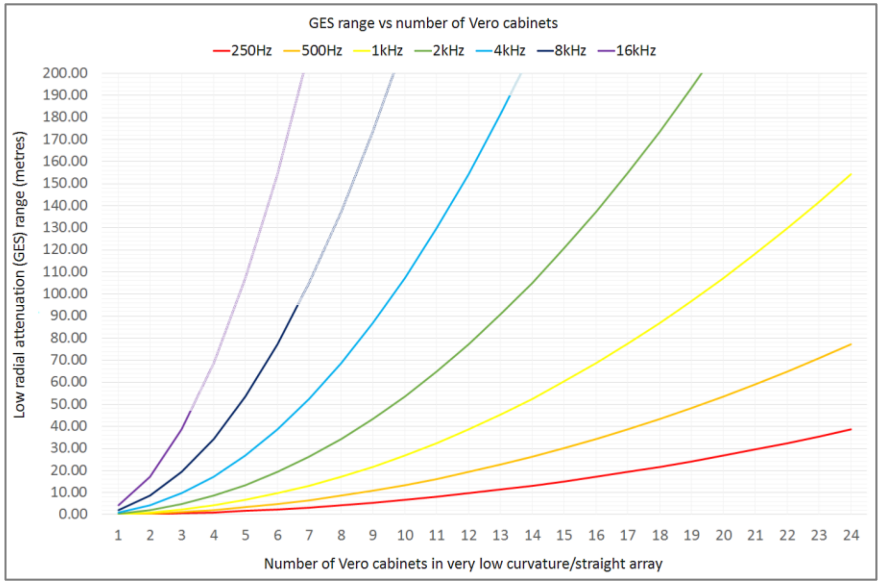
For a tilted array, as in the example shown, there is likely to be a rapid reduction in level and bandwidth as they go past the back of the intended audience area *(4 - decreasing)*.

Understanding the basics will help with your large-scale array designs

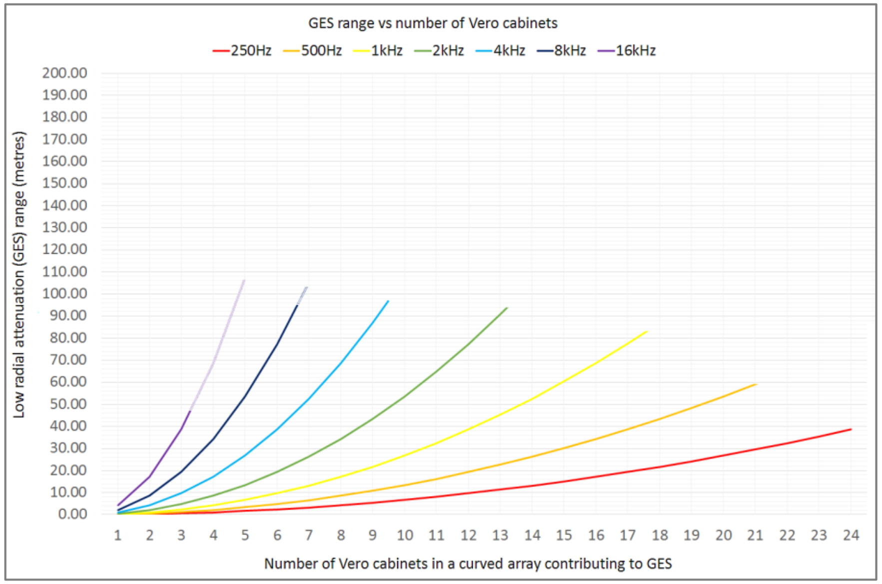
Although you don't need a physics degree to design excellent arrays using Funktion One's Projection software, a basic understanding of vertical array behaviour will help you make good initial decisions - about the length of your array for a given audience distance, for instance.

Theoretical transition distances vs number of Vero cabinets

As would be expected, the longer and straighter the array, the longer the low attenuation *(GES)* range – i.e. the greater the transition distance before the attenuation reverts to the normal 6dB/doubling of distance.



Theoretical transition distances vs array length for low curvature/straight array



Transition distances reduced by array curvature

The plots show how the transition distances vary with array length and frequency – from 250Hz *(red)* to 16kHz *(purple)*.

The vertical scale is the transition distance in metres and the horizontal scale is the number of V60 cabinets in the array.

The lighter sections on some of the 16kHz, 8kHz and 4kHz traces indicate that, in practice, excess air attenuation could limit mid-high and HF propagation.

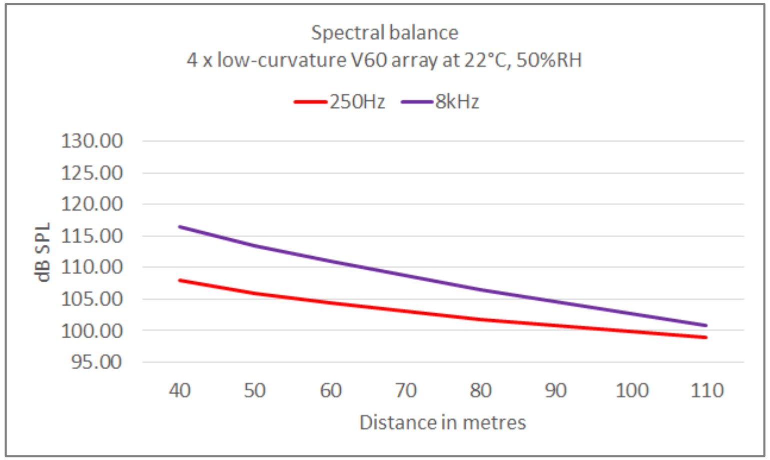
The left-hand plot shows the transition distances for a very low curvature/straight array whilst the right-hand plot shows the shorter transition distances at mid-high and HF expected from a slightly curved array. Transition distance are more sensitive to array curvature at shorter wavelengths where curvature reduces the number of cabinets contributing to the sound level at specific listening distances.

Spectral balance vs array length when using low-curvature arrays for large scale set-ups

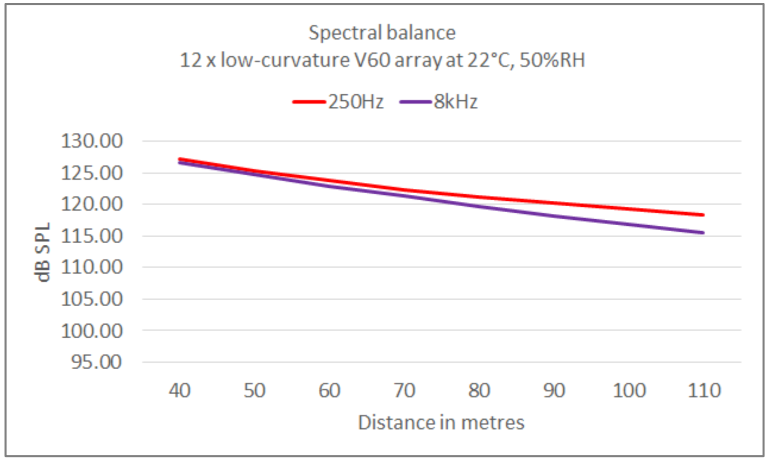
Let's have a look at some more theoretical plots to illustrate some basic large-scale array design dos and don'ts. We'll base our plots on the need for vocal intelligibility and warmth, so we'll concentrate on a V60 range from 250Hz to 8kHz.

As you can see from the transition distance plots above, the transition distance *(or low attenuation (GES) range)* of a very low curvature or straight array is proportional to frequency and to the square of the array length.

Although a short array will provide adequate mid-high and HF propagation, the array will need to be much longer for good low frequency propagation due to the longer low frequency wavelengths. That doesn't mean that mid-high and HF propagation is easy. These upper frequency ranges are not only affected by array curvature, but also by air absorption in low relative humidity conditions *(see later)*.



Short array with poor spectral balance *(weak LF)*



Longer array with better spectral balance for large scale coverage

The plots show how spectral balance varies with array length and listening distance *(shown for typical Northern European temperature and relative humidity)*.

The vertical scale is sound level in dB SPL and the horizontal scale is the distance from the array. The red trace is 250Hz and the purple trace is 8kHz.

The short straight array's spectral balance *(above left)* is poor – and inconsistent with audience distance. Curving the array to reduce the HF level would simply make the whole thing less powerful. And the curvature would cause a lack of HF at the back of the audience.

The more suitably sized array *(above right)* provides more vocal warmth and "authority" with better spectral balance over the full audience distance. The improved spectral balance also responds better to any HF compensation that may be required once curvature has been considered – see later…

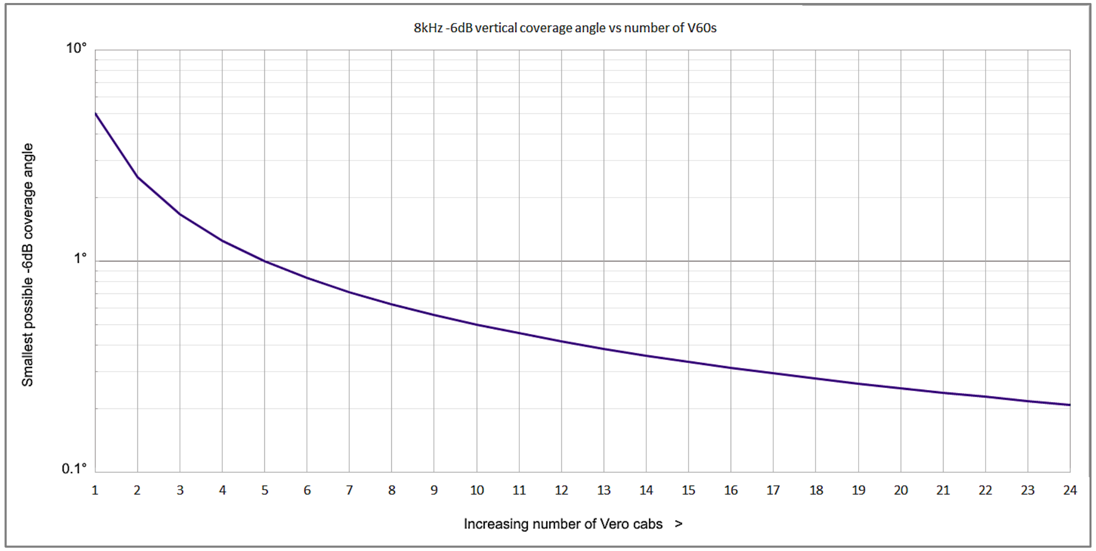
A good rule of thumb for clarity and impact is to start your designs with an array length of around 6-8% of the distance to be covered.

Curvature considerations

Unfortunately, long, low-curvature or straight arrays are very directional and don't provide the vertical coverage required to cover real audiences - especially at mid and high frequencies where the vertical beam width can become impractically narrow.

The trick is to curve the array so that your individual cabinets' on-axis rays are regularly spaced through the audience – as in earlier examples. The resultant array shape will tend to have small inter-cabinet angles at the top, medium inter-cabinet angles for mid-field/mix position and larger inter-cabinet angles at the bottom - tending towards individual-cabinet point source coverage at the front.

So, how many cabinets can we use in the upper, low-curvature/straight part of the array?



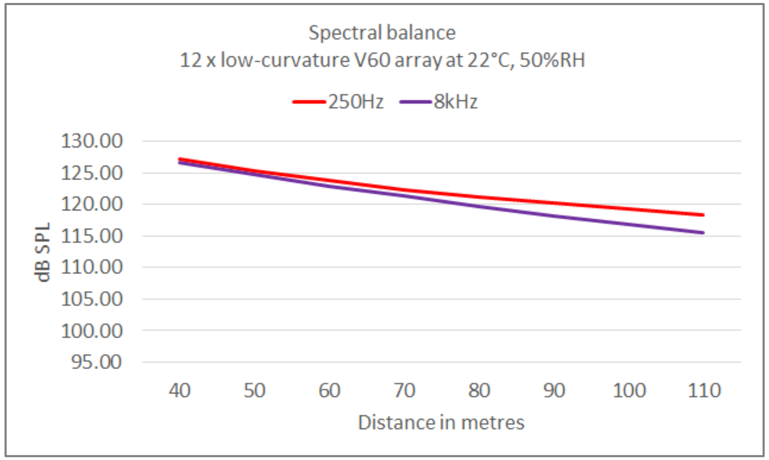
8kHz *(-6dB to -6dB)* vertical beam-width vs number of V60 cabinets

The plot shows -6dB to -6dB vertical beam-width *(vertical scale degrees)* vs the number of V60 cabinets in the straight section of the array *(horizontal scale)*.

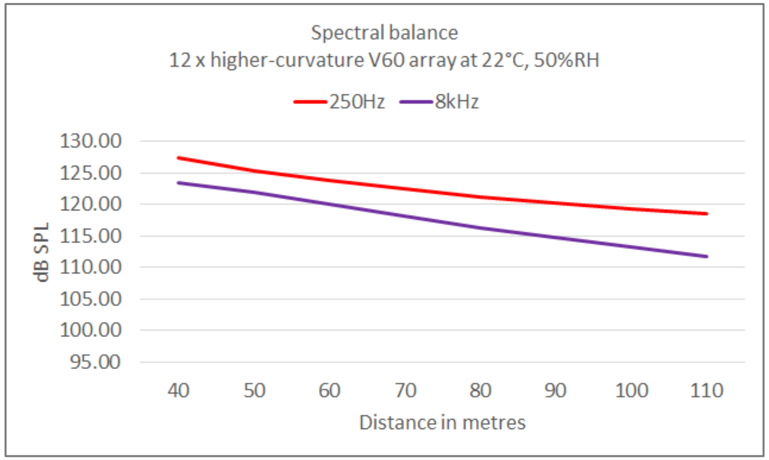
To cover a large-scale audience area, the array will tend to be progressively curved *(although the sequence may vary around V315 zones)* for equal ray spacing. The smallest inter-cabinet splay angle is 0.5° so the tightest multiple-cabinet beam-width allowable is ±0.5° = 1°.

The 1° 8kHz beam-width line is crossed at 5 cabinets for the V60 (see plot above). This suggests that the maximum number of V60s in the upper, straighter part the array should be limited to five cabinets or less to ensure contiguous geometric energy summation with the lower, progressively curved sections.

Curving an array reduces the number of cabinets contributing to mid-high and HF summation at distant listener positions and can reintroduce an LF-HF imbalance – by weakening mid-high/HF coverage in this case.

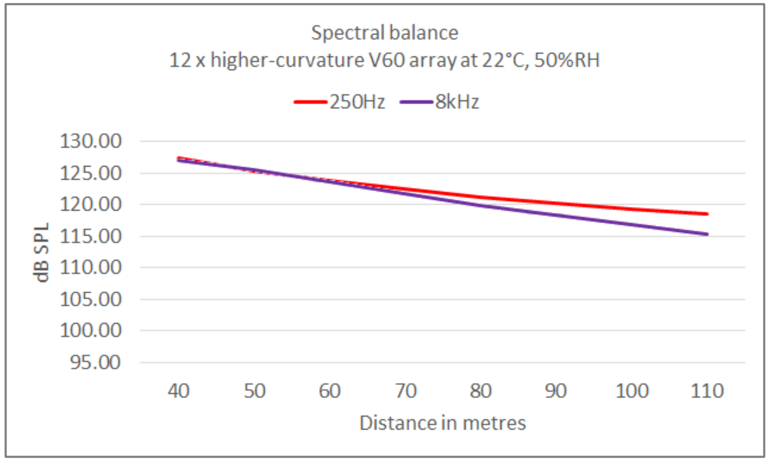


Good spectral coverage from a low-curvature array



HF reduction caused by extra curvature

Assuming the array has been chosen for initial spectral balance, a little overall HF compensation can be applied *(about +5dB, in this case)*.



Partial HF compensation using simple overall HF shelving

Excess attenuation at high frequencies

In practice, of course, high frequency propagation will also be limited by air absorption. This is often referred to as excess attenuation as it is in addition to the normal radial attenuation that we're all familiar with.

> ***A Scientific explanation for the curious…***
> *Excess attenuation is caused by a process of energy transfer due to the excitation and relaxation of air molecules in a sound field. When an oxygen or nitrogen molecule's relaxation frequency coincides with the sound frequency propagating through it, it converts some of the sound energy into heat and returns some, after a sequence of molecular collisions, as phase-shifted pressure - hence the attenuation.*
>
> *Maximum losses occur* ***per wavelength*** *at these relaxation frequencies. As there are more wavelengths over a given distance at higher frequencies, air absorption rises with increasing frequency.*

Sound usually propagates radially from a source, so its broadband attenuation is quantified as a number of dBs per doubling of distance. Excess HF attenuation, however, is quantified in ***dB per metre*** *(due to its per-wavelength characteristic)*. So, it is more significant when sound is propagated over large distances.

In addition to being frequency dependent, excess attenuation is heavily influenced by relative humidity. Relative humidity is a measure of how saturated the air is with water vapour – which, in turn, is affected by temperature. It all gets very complicated, so we'd suggest using a calculator if you wish to work out the figures.

> See ***www.brusi.com/downloads.shtml#PAC*** for a useful ***PAcalculate*** App for iOS, Android, Windows Phone & Blackberry. The App is free of charge.
>
> **Note that Funktion One cannot take responsibility for 3rd party software.**

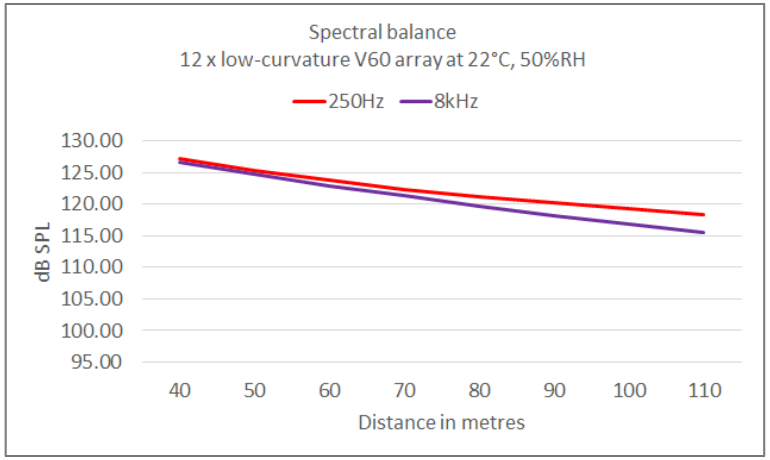
At low relative humidity figures, air attenuation down the length of a football pitch can be as high as 28dB at 20°C.

Here are some more mid & HF attenuation figures…

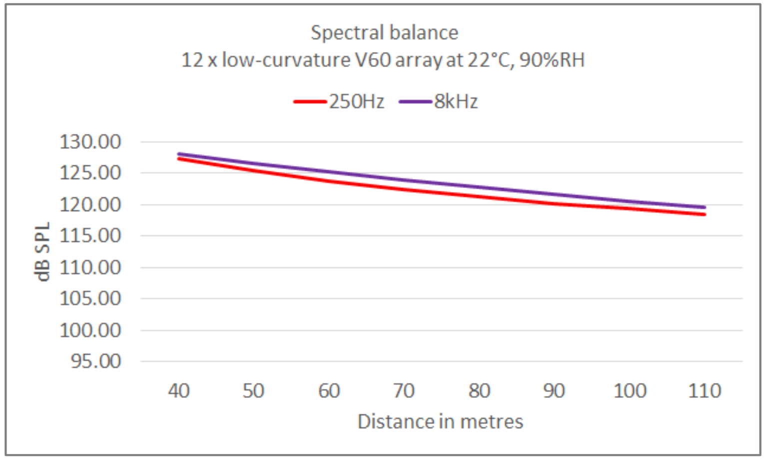
Relative humidity *(How saturated the air is with water vapour)*	**Frequency**				
	2kHz	**4kHz**	**6kHz**	**8kHz**	**10kHz**
	Sound attenuation over 100m				
20% *(Air feels dry over a wide range of temperatures. Lips tend to feel dry)*	2.2dB	7.5dB	14.5dB	21.7dB	28.4dB
50% *(Air feels comfortable over a wide range of temperatures)*	1.0dB	3.0dB	3.0dB	10.5dB	15.9dB
80% *(Weather feels humid when warm and damp and murky when cold)*	0.9dB	2.1dB	2.1dB	6.9dB	10.5dB

HF attenuation over 100m vs relative humidity at 20°C due to air absorption *(as per ISO 9613-2)*

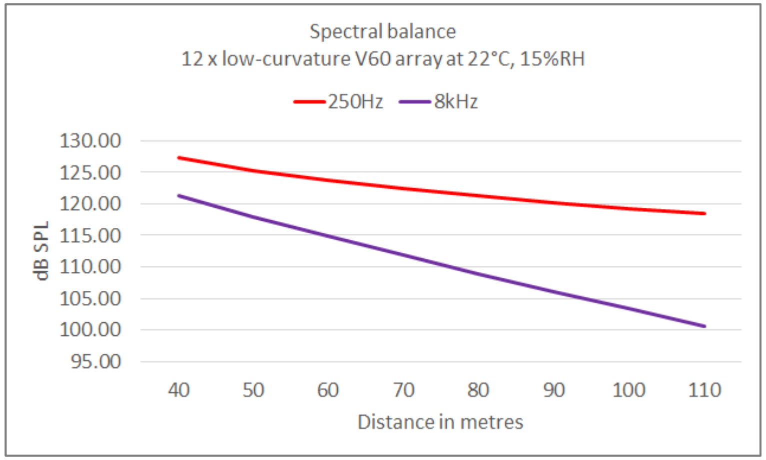
Effect of humidity on spectral balance



Normal spectral coverage at 22°C, 50% relative humidity



Increased HF at 22°C with very high *(90%)* relative humidity



Heavily attenuated HF at 22°C with very low *(15%)* relative humidity

Note that very low relative humidity conditions *(right hand plot)* can lead to heavily attenuated HF at the back of the audience area.

Relative humidity figures as low as 20% are not unusual on dry summer days – even in Northern Europe.

So, should we use HF compensation?

If you're going to be using a system in very low humidity conditions - in the Middle East or Arizona, for example – and it is not possible to fly a very large array, it's usually better to specify delay systems beyond about 70 metres *(if the budget will allow)* rather than use excessive HF compensation.

Before going to site, make sure that your array design is adequate for the job. It may be better to play safe and "over design" your system for a heavy rap, metal or dance festival over 140m in a very dry climate. A very large-scale array – up to 24 cabinets – will not only provide more power but will also allow you to keep inter-cabinet angles small to maximise geometric energy summation at mid and high frequencies.

And, if you're already on site and struggling to get HF projection, re-check your array design *(via your Projection software)* – you may need smaller inter-cabinet angles at the top of the array. Also check that the array is aimed correctly. For instance, has that last-minute tie-back caused the array to aim too low?

It's best to conduct on-site listening tests remembering that most listeners will expect the sound to be duller at a distance.

Experienced system technicians only

If you are an experienced system technician working in exceptionally dry conditions and have confirmed that there's nothing wrong with your array design or its deployment, you may decide that some HF lift would be beneficial. Use a minimal amount *(to maintain system headroom)* and apply it incrementally - from none at the bottom of the array, through just a few dBs for the mid-throw section to more HF boost at the top. Keep the dB changes between adjacent sections as small as possible – say, a maximum of 4dB* per section. *(*Don't worry. You won't hear up to a 4dB step as overlapping cabinet outputs will blend together).*

- Don't try to achieve for a flat response "from DC -to- light". An exceptionally flat rear audience response may look good on an analyser but:
  - a) You'll eat up lots of HF headroom and…
  - b) The system will sound unnatural to an audience that far from the stage
- Apply any HF compensation pre-crossover *(using each amplifier's Overlay settings – mentioned in **here in Section 6.2**)* to avoid asymmetrical crossover effects and subsequent crossover frequency side-lobes
- Use shelving filters and reduce the knee frequency as the boost is increased for the upper sections to avoid running out of HF headroom *(see below)*. ⇨

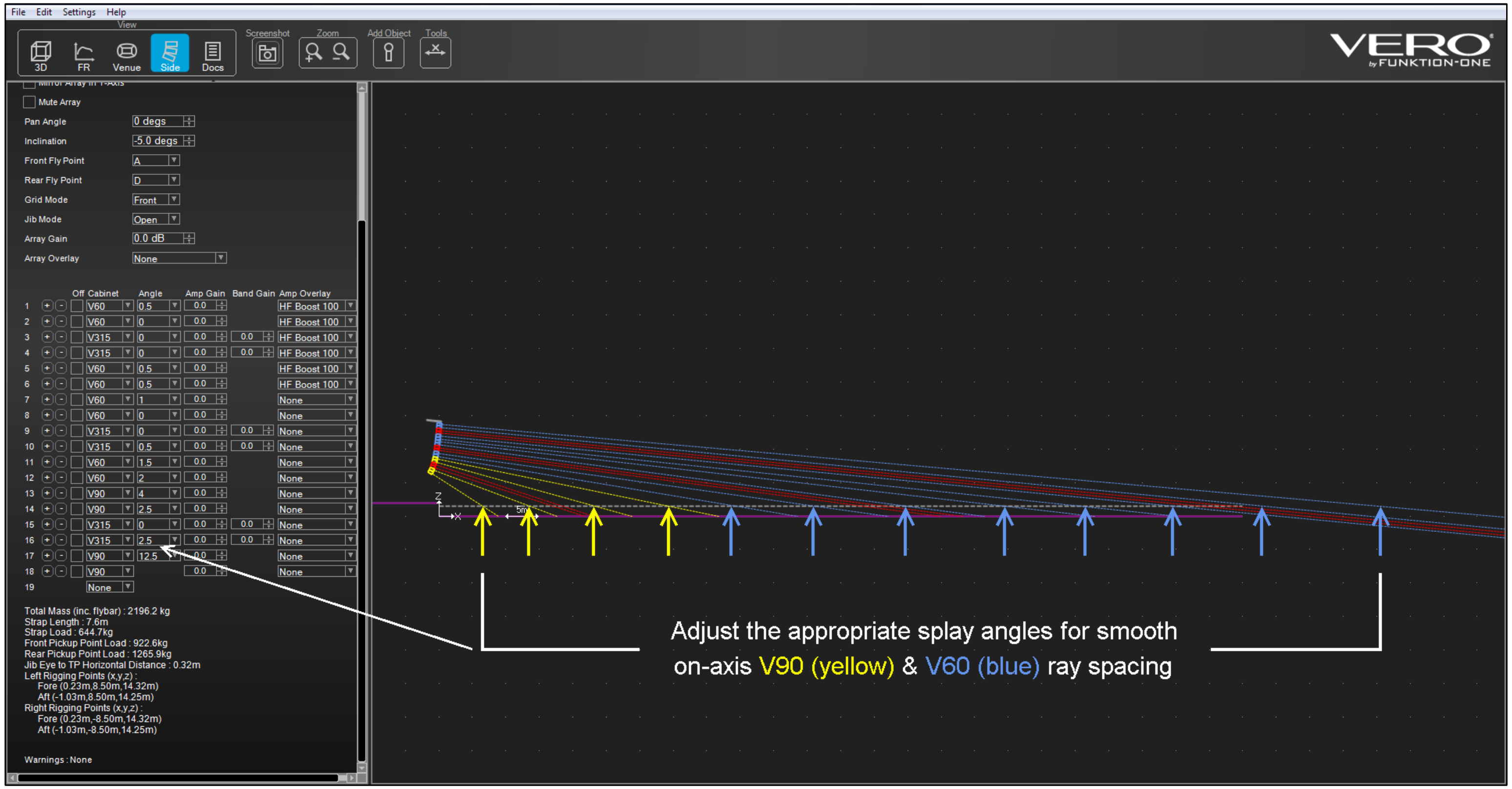
Large scale *(24-cabinet)* HF compensation example – for dry air conditions (<40%RH)

With a large-scale array *(24 cabinets – numbered from the top)* covering up to 120m in dry air, you could try the following 1st order HF shelving overlays:

Upper 4 x V60:	+6dB to +8dB, levelling off at 8kHz
Upper middle 4 x V60:	+4dB to +6dB, levelling off at 10kHz
Lower middle 4 x V60:	2dB to +4dB, levelling off at 14kHz
Lower 4 x V90:	Overlay = None
All 8 x V315:	Follow whatever the V60s are doing on the same amplifier - as all four amplifier channels are ganged. *(See **here in Section 6.2**)*

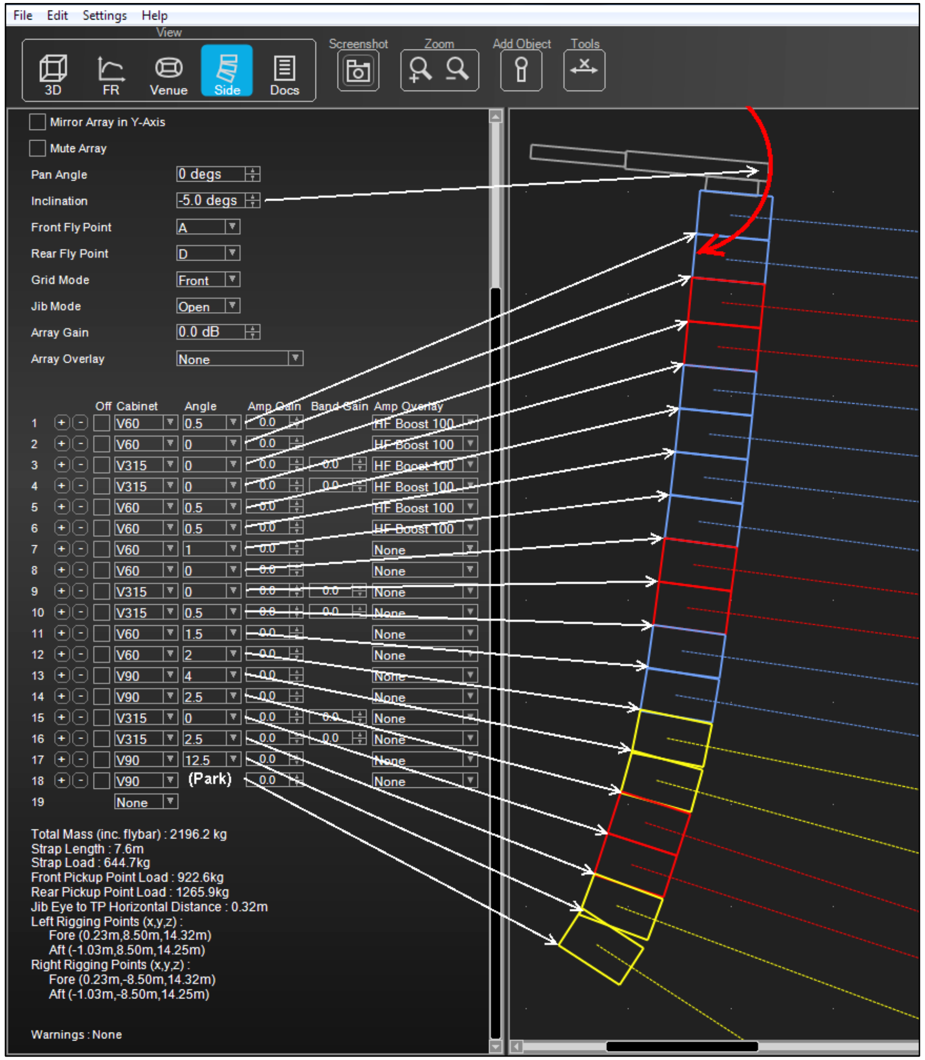
Very large-scale array design using your Funktion One Projection software

The Projection-based design process for large scale arrays follows the same procedure as before. Here's an 18-Vero *(10 x V60, 2 x V90, 6 x V315)* example:



- Inter-cabinet splay angles are set, initially, for equally spaced ray positions at the audience plane. Again, fine adjustments may be required once the coverage predictions have been checked.
- Rigging properties are updated as you design the array – see earlier examples of rigging warnings…

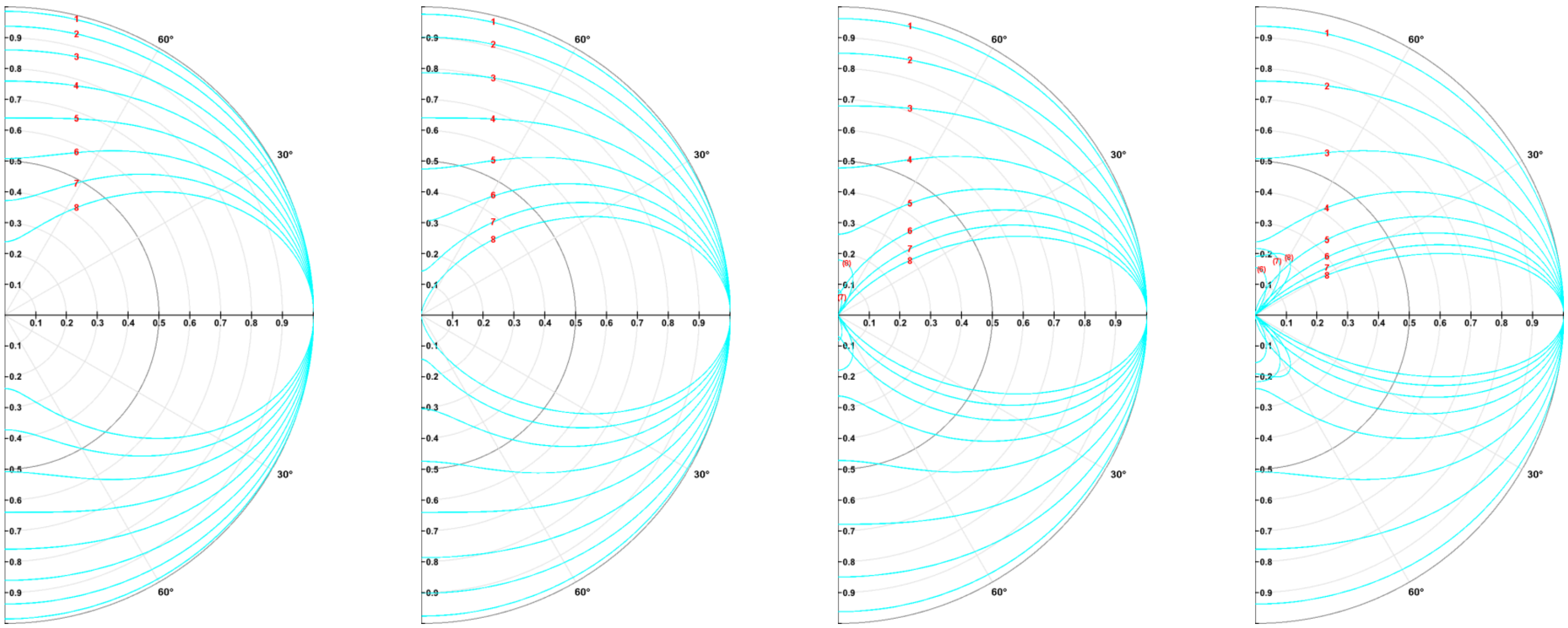
And, as before:



- The ***Angle*** settings refer to the Lambda mechanisms at the bottom of each cabinet and set the splay angle between each cabinet and the one below
- The ***Inclination*** angle sets the overall array up-tilt *(+ve)* or down-tilt *(-ve)* by tilting the grid. The top cabinet is always parallel to the grid

Appendix B

Far-field mid-bass directivity responses for multiple Vero-sized radiators



80Hz **100Hz** **125Hz** **160Hz**

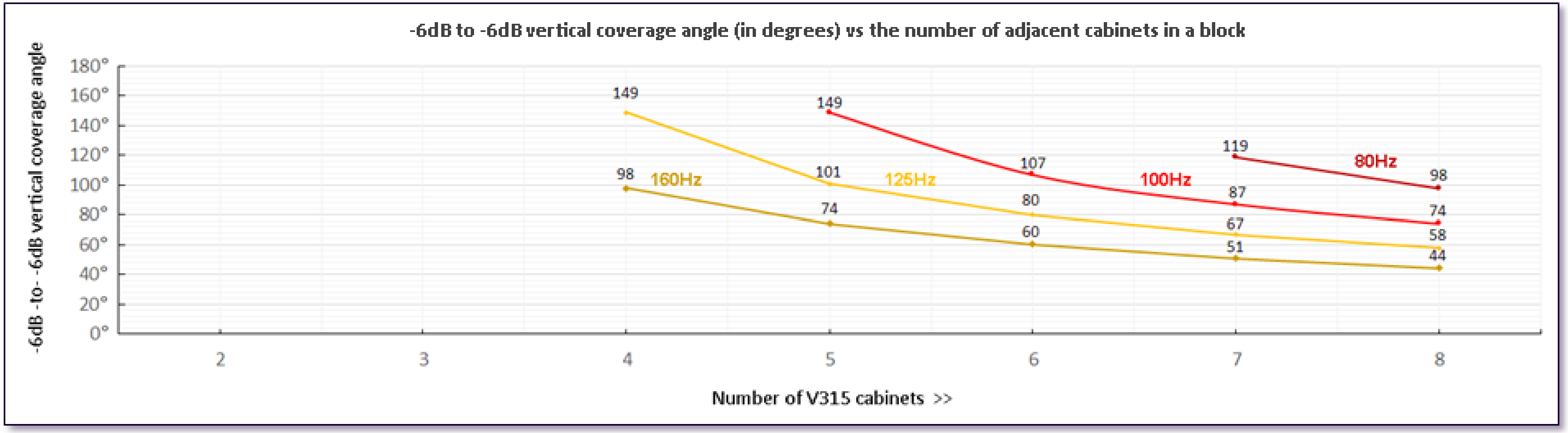
Vertical far-field directivity response vs number of adjacent or combined V315s

The above plots' pressure level coordinates (wrt normalised on-axis maxima) are linear so the 0.5 radial line = the -6dB point.
Angular coordinates are shown in degrees wrt on-axis (0°) maxima.

Multiple V315 outputs will tend to combine smoothly when tight-packed or spaced less than 1/3rd wavelength apart.

- V315s will combine with adjacent V60s or V90s and follow the array curvature at the crossover frequency *(hence no coverage plot at 200Hz)*
- Adjacent V315s will combine over their full operating range

Vertical coverage vs number of adjacent or combined Vero-sized cabinets



80Hz, 100Hz, 125Hz and 160Hz plots showing the -6dB to -6dB vertical coverage angle *(in degrees)* vs the number of adjacent cabinets in a block

Figures are not shown for some lower numbers of V315 cabinets where coverage would exceed 180°. In practice, of course, the baffling effects of adjacent V60 or V90 cabinets will limit the V315 coverage to 180° *(plus any curvature)*. See the next page for the full, theoretical polar data for 1 – 8 cabinets at 80Hz, 100Hz, 125Hz and 160Hz.

Appendix C - Achieving under-array bass attenuation

Vero's Geometric Energy Summation is based on sound level summation at the listener position. Other manufacturers insist that their vertical arrays must be mechanically contiguous, robbing the user of flexibility. As long as the individual V60 and V90s in the mid-high sections of a Vero array are aimed correctly for listener-position summation, those sections may be opened up to allow V315 mid-bass deployment to be optimised for best low frequency gain-before-feedback without resorting to excessive, impact-robbing equalisation.

V315s exhibit excellent pattern control when deployed in a Vero array but, where extra bass attenuation is required beneath the array – perhaps, for instance, LF feedback has caused problems with low string sections on previous occasions - V315 placement may be chosen to create specific frequency nulls under the array.

Off-axis response nulls are related to odd numbers of half-wavelength spacing between cabinet or block centres. Calculating them can get quite complicated and the results can be a little unintuitive, especially where multiple and asymmetrical V315 block sizes and spacing are used. So, we've predicted V315 null frequencies under the array using Funktion One's ***Projection*** software.

Under-array attenuation *(wrt approx. 30m on-axis levels)* is indicated below: ✓ = -6dB, ✓✓ = -12dB, ✓✓✓ = lower than -12dB.

Array size	**Array layout** *(top -to- bottom)*	**Projection-predicted V315 null frequency ranges under the array** ***to nearest 1/6 octave*** *(and likely feedback notes fixed - based on A_4 = 440Hz)*										
No of cabs	**V60/V90-V315**	**50Hz** G_1-$G^\#_1$/A^b_1	**56Hz** A_1-$A^\#_1$/B^b_1	**63Hz** B_1-C_2	**71Hz** $C^\#_2$/D^b_2-D_2	**80Hz** $D^\#_2$/E^b_2-E_2	**90Hz** F_2-$F^\#_2$/G^b_2	**100Hz** G_2-$G^\#_2$/A^b_2	**112Hz** A_2-$A^\#_2$/B^b_2	**125Hz** B_2-C_3	**140Hz** $C^\#_3$/D^b_3-D_3	**160Hz** $D^\#_3$/E^b_3-E_3
6	**1-1-2-1-1**						✓	✓	✓✓	✓✓✓	✓✓	✓
9	**1-2-1-2-1-2**			✓	✓✓	✓✓✓	✓✓	✓✓	✓✓	✓	✓✓	✓✓✓
10*	**2-1-1-2-1-1-2**						✓	✓✓	✓✓	✓✓✓	✓✓	✓✓
12*	**3-1-1-2-1-1-3**						✓	✓✓	✓✓	✓✓✓	✓✓	✓✓
12	**3-2-2-2-3**				✓	✓	✓✓	✓✓✓	✓✓	✓		
12	**2-2-4-2-2**	✓	✓✓✓	✓✓	✓✓	✓						
15	**2-2-4-2-2-1-2**	✓✓✓	✓✓					✓				
18	**2-2-4-2-4-2-2**	✓✓	✓		✓	✓✓✓	✓	✓			✓	✓
18	**2-4-8-2-2**						✓	✓✓✓	✓		✓	✓
18	**2-2-4-2-2-1-2-1-2**	✓✓✓	✓✓	✓	✓	✓✓	✓	✓				✓
24	**4-4-8-4-4**						✓	✓✓	✓		✓	✓✓✓

Projection-predicted bass null frequency ranges vs V315 placement within the array

*(*Preferred 10 and 12-box layouts for best mid-bass pattern control. The 12-box example is shown on right ⬀)*

3-1-1-2-1-1-3
12-box layout

Appendix D - Clipping

Caution

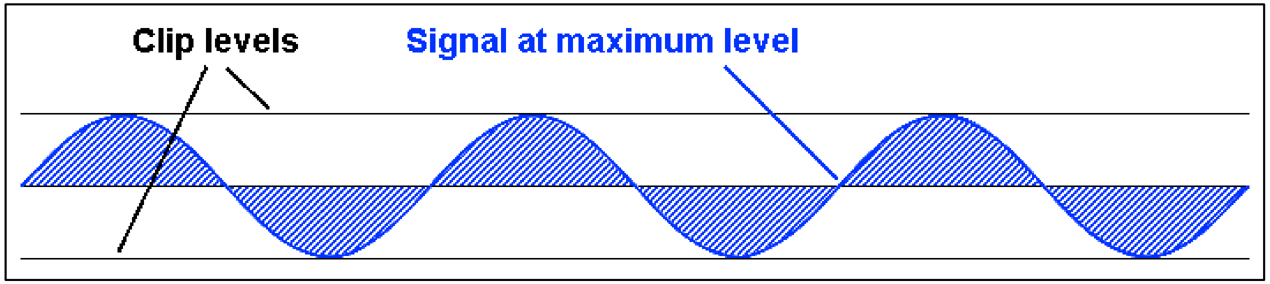
Prolonged heavy clipping can subject loudspeaker voice coils and passive crossover components to double their rated power dissipation leading to overheating, premature ageing or early failure. The high frequency components are particularly vulnerable in systems with passive crossovers.

In extreme cases this could pose a fire hazard.

What is clipping and why is it a problem?

When an audio signal is amplified beyond the maximum voltage or full scale digital capabilities of the equipment in use, the peaks of the waveform can get flattened. This is referred to as clipping and is heard as distortion.

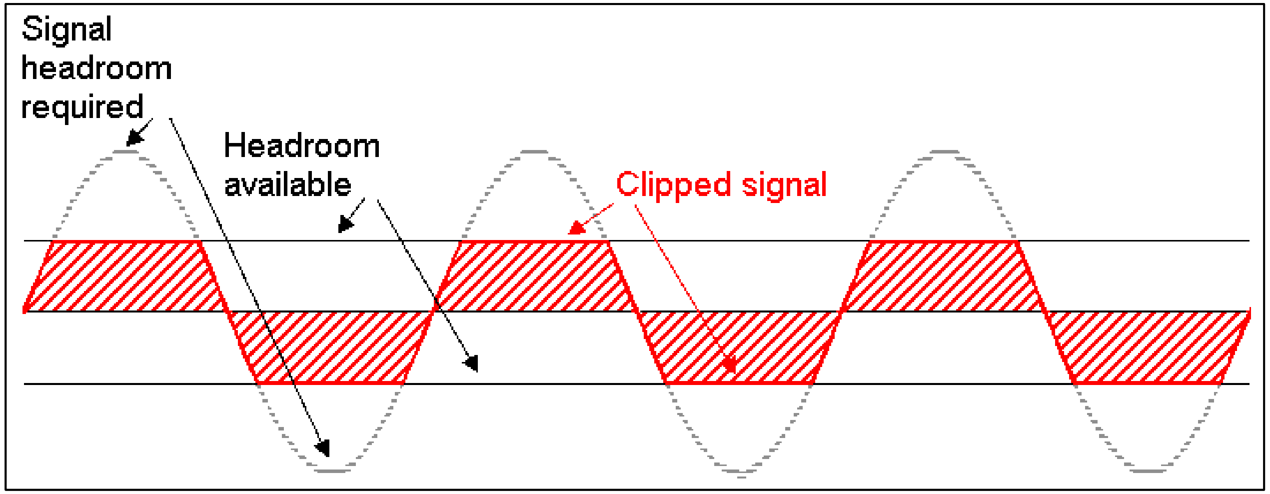
Here's a normal signal at maximum level – just below clipping and still within the available "headroom".
(The clip level sets the maximum available headroom)



Pure tone *(single frequency)* signal just below clip level
(Vertical = voltage, horizontal = time)

The blue positive and negative shaded areas represent the power the signal would cause to be dissipated in a load – usually a loudspeaker voice coil.

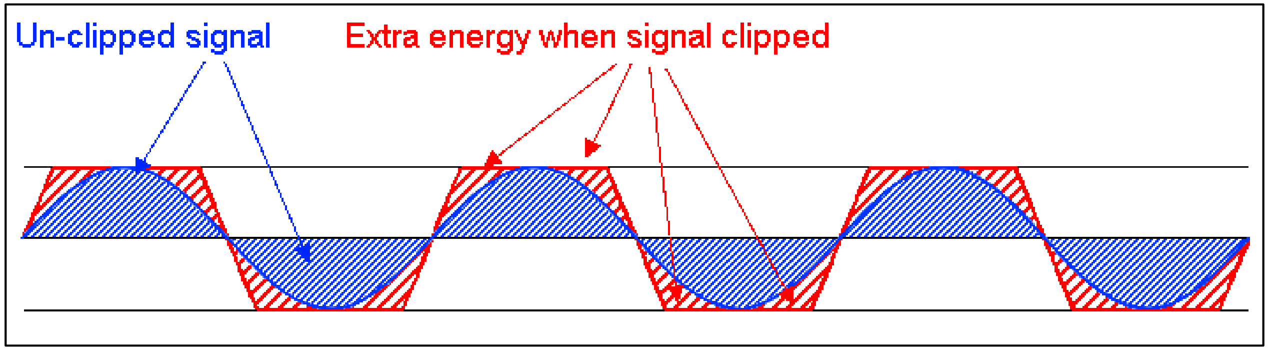
Here's the same signal amplified beyond the available "headroom" – i.e. beyond the clip level.



Signal amplified beyond clip level

When a signal is clipped, its waveform squares off.

The signal stays at the maximum positive and negative values for more of each cycle in cross-hatched regions. This is easier to see if we compare the non-clipped ***(blue)*** waveform with the clipped ***(red)*** waveform. The clipped signal's extra content can be seen as the extra red shaded areas.

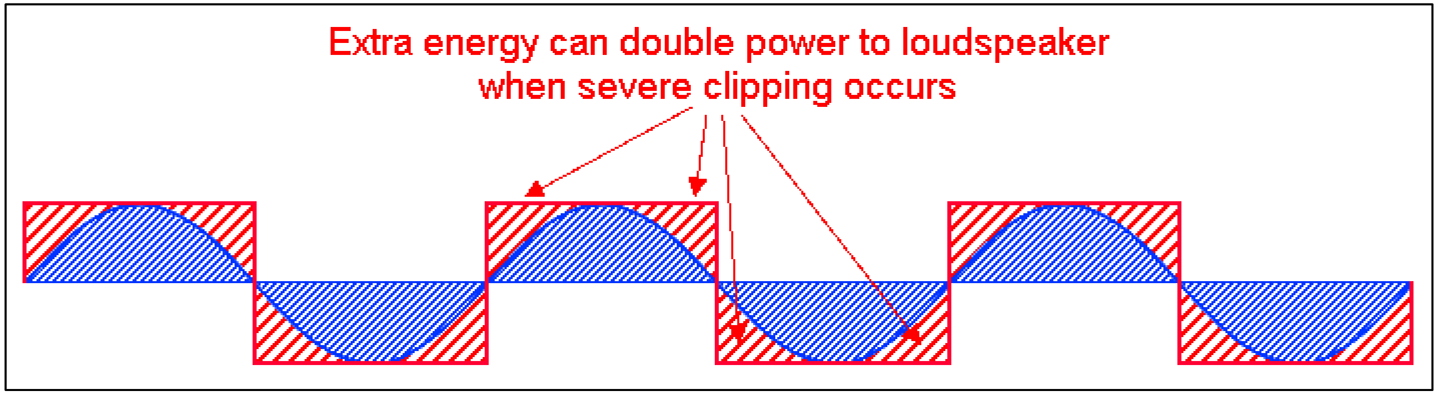


Extra energy of a clipped signal

In extreme cases, the waveform is so heavily clipped that it can resemble a square wave. This increases the rms voltage delivered to the loudspeaker to √2 (approximately 1.4142) times what would be expected.

The power delivered to the loudspeaker is dependent on the rms voltage^{2} ***(squared)***.

√2 x √2 = 2 **so the power delivered to your loudspeakers is doubled!**



Note that, in all cases, the two waveforms have similar peak voltage levels, so this extra energy doesn't always show up on digital audio systems where meters are usually calibrated with respect to full scale digital level - usually written as dBFS.

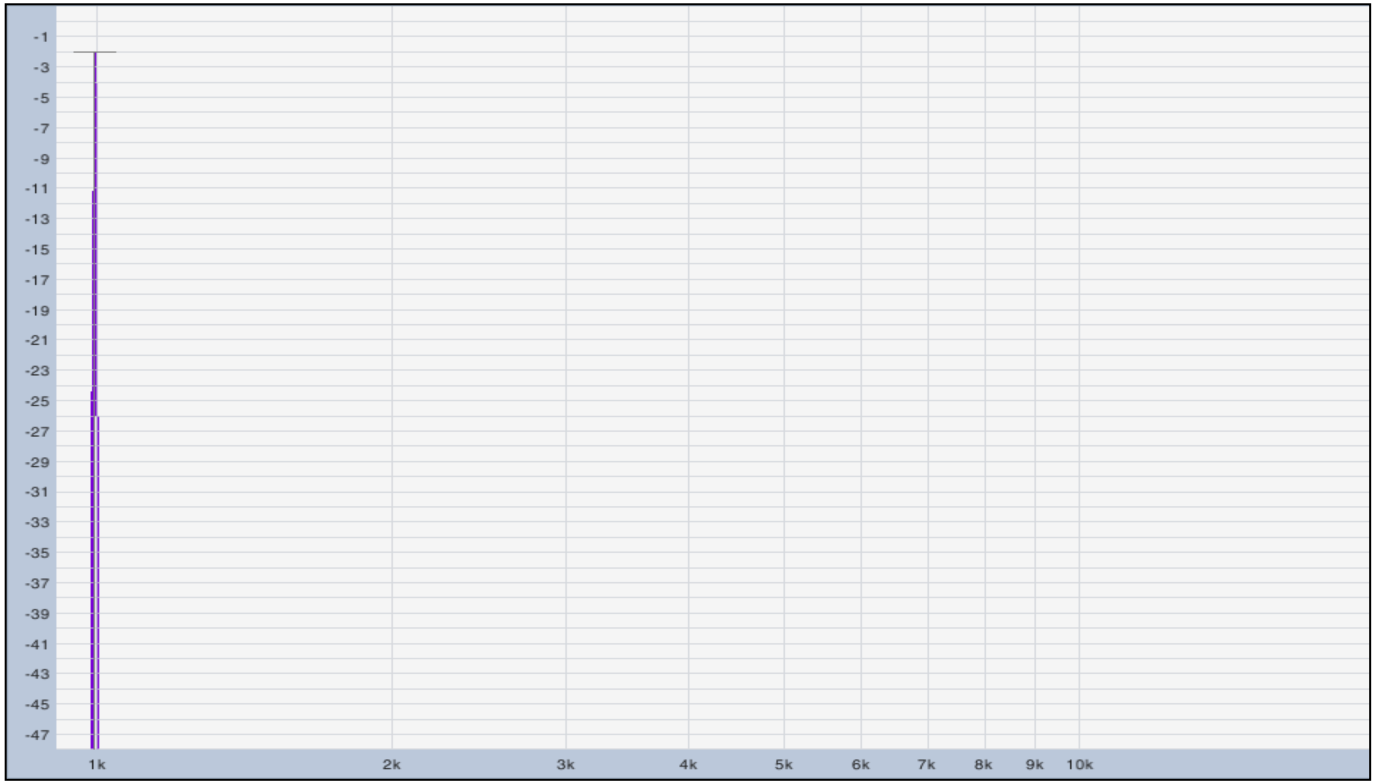
Digital level meters don't usually take the rms value of the waveform into account, so the extra power capability of squarer waveforms is often missed. Hence the warning at the beginning of this Appendix.

To avoid overheating loudspeaker voice coils – particularly HF voice coils in passive loudspeaker systems – try not to clip signals repeatedly or for long periods.

Clipping and its effect on frequency content

All waveforms are made up of combinations of fundamental frequencies and their harmonics *(also known as overtones)* in a variety of amplitude and phase relationships. Although harmonics are part of everyday speech and music, unwanted or unexpected harmonics - added by clipping, for instance – may be heard as distortion.

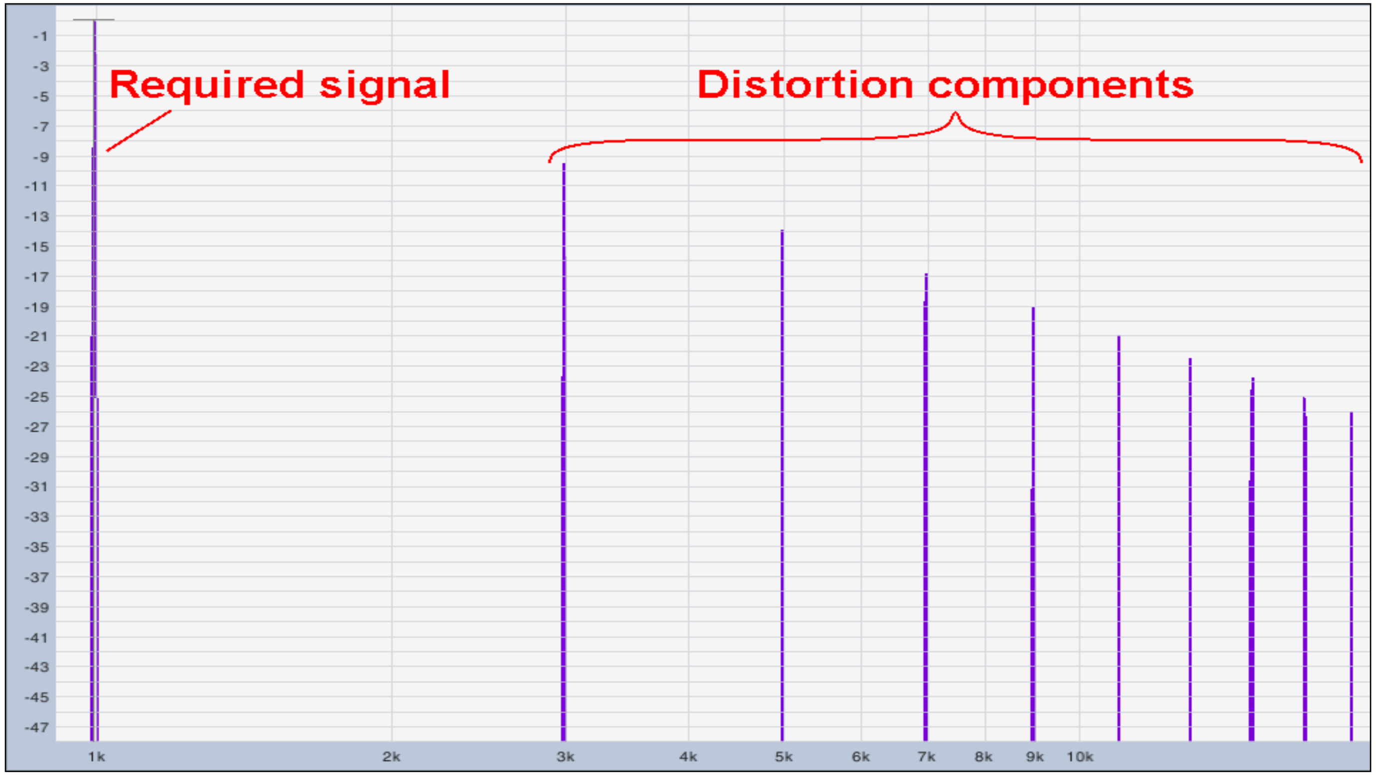
Spectrum analysers can be used to show this. In the screen shot below, an unclipped pure tone (approximately 1kHz) can be seen as a single frequency line.



1kHz pure tone just below clip level
(Vertical = level in decibels, horizontal = frequency)

The spectral plot above is that of an undistorted pure tone. Note the single, pure, frequency point.

Now let's see what happens when you clip a signal…
The heavily clipped signal on the next page shows clipping distortion as a series of odd harmonics at 3kHz, 5kHz, 7kHz etc., whose levels decrease with frequency at approximately 6dB per octave. Where clipping occurs before the system crossover, clipping harmonics can crossover into the HF driver and the HF driver can be called upon to dissipate far more power than normal.



1kHz heavily clipped signal

The example shows a single frequency fundamental with odd order harmonics caused by severe symmetrical clipping.

Musical signals, and any associated clipping distortion, will be a lot denser. Asymmetrical clipping *(from studio or backline valve amplifiers, for instance)* would also include even harmonics. The presence of even harmonics often makes subtly overloaded signals sound richer, often making guitar amplifier distortion acceptable as part of the overall sound.

As mentioned earlier, long-term unwanted clipping can be avoided if you choose mixers, effects units, controllers and amplifiers with adequate headroom and set your limiter thresholds to below clipping levels. You can also avoid clipping your signals at source by setting your console and effects system "gain structure" carefully. *See **Appendix E.***

Appendix E – Gain structure

What is gain structure all about and why is it important?

When an audio signal passes through an analogue or digital audio system, it is important to maintain the optimum signal operating level through the various sections of the signal path. Microphone levels must be boosted to make them compatible with line level and analogue-to-digital converter stages and mix stages must allow enough headroom for multi-channel summation.

The various gain settings throughout a piece of audio equipment are collectively known the system's gain structure.

Signal level too low

If the signal level is too low it will be wallowing around in the analogue noise floor or losing resolution and becoming fuzzy, distorted and noisy in the digital domain.

Avoiding digital garbage

If you have to go digital for your multi-channel live sound operations, use 24-bit systems or higher - if they are available - so that you can maintain enough headroom to mix without sinking into the digital mush. And, of course, if you like to use lots of effects, try to use studio-quality effects package that runs its internal processing at 32-bit floating-point or better. This reduces the build-up of mush you get every time your signal goes through a different processor.

Signal level too high

If your operating levels are too high, there won't be enough headroom to allow for performance peaks and multi-channel mixing. You may end up with distortion caused by clipping.

If you're using all-analogue equipment, the odd peak clip may go unnoticed as long as it's not sustained long enough to damage your HF drivers. This is because analogue clipping components are usually harmonically related to the original signal.

Avoiding digital mush

Unfortunately, digital overload isn't as simple. You get clipping when the digital stage runs out of digits – usually referred to as 0dBFS – and the higher order harmonics generated can then alias with clock frequencies creating all sorts of strange sum and difference frequencies. These aliasing products can be interesting if you normally work with robots with ring modulators for voices. These digital clipping artefacts turn cymbal "tings" into "shhhhs", make vocals sibilant and causing listener fatigue; the kind where you know something's wrong but can't quite put your finger on it.

To make matters worse, any over-sampling or delay-based processes built into effects units, can stretch these nasty artefacts in the time domain and make them far more audible than their relatively low levels would suggest. Also remember that aliasing will be compounded as the overloaded signal passes through successive processes.

Signal level just right

If your operating signal levels are just right, you won't run out of headroom, clip or run out of digits – even on musical peaks. And you'll still run well above the analogue noise floor and stay clear of the digital mush.

So, what is "just right"?

It all depends on the type of audio material and your application.

Live sound mixing

If you're going to be mixing audio from a variety of live musical instruments or you're going to be creating some broadband effects over a wide dynamic range, you'll need to allow plenty of operating headroom.

DJ mixing with pre-recorded material

If you are only ever going to work with pre-recorded, auto-levelled material, you'll probably get away with just enough headroom to mix tracks and add the odd effect – but read the whole section anyway, you may find it instructive.

Live sound operating levels

It is important to have enough headroom to cater for the transients and level changes encountered when mixing live material. You also need to allow for the natural signal level build-up through the console, once you start tweaking eqs, adding effects and sub-grouping multiple channels. Sub-group levels tend to rise by 3-to-6dB per doubling of channels routed to them.

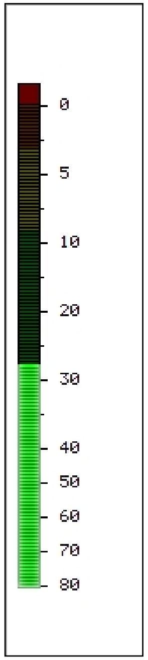
Assuming a professional quality analogue or 24-bit digital console, run each channel's pre-fader level and each group's mix level at around ***0dBU/+4dBu (analogue)*** or ***-18dBFS (digital)*** if at all possible.

Modern 24-bit systems with good studio quality converters usually work well at -18dBFS but some MI products may use lower grade converters which will need to be run at higher operating levels to avoid sounding mushy. If you suspect this is the case with your system, you'll need to run PFL and group levels around -12dBFS or even higher and keep an eye open for signal clipping during the live performance.

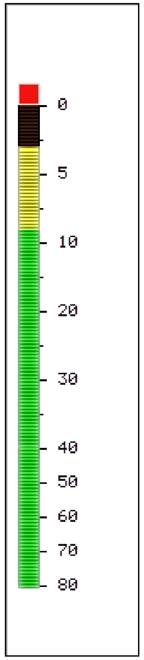
Understanding level meters

This section spends a lot of time discussing meters because, without an understanding of their typical calibration levels, it's difficult to get your gain structure and operating levels right.

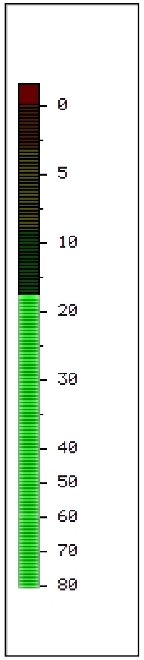
Professional live sound meters calibrated with respect to "Full Scale" (dBFS)



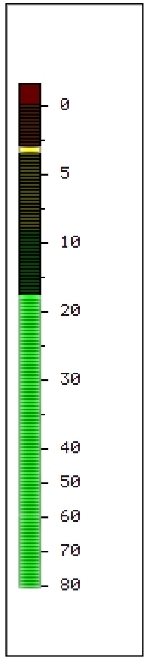
Too low *(-28dBFS)*
Wasting dynamic range



Too high *(-3dBFS)*
Peaks could be clipped



Just right *(-18dBFS)*
Noise-free without clipping



Peak hold option
May not be true peak

The illustrations above show levels on a typical bar meter with a dedicated peak LED at the top. Some analogue peak LEDs don't respond to very fast transients and some digital ones don't respond until the processor has seen several maximum bits in a row. Peak-hold indicators often hold maximum rms – not true waveform peaks – so they are not always a good indicator of potential clipping.

If your bar meter is way above the -18dB mark for live sound, you're probably clipping peaks and transients irrespective of what the peak LED or peak-hold indicator is telling you.

Q & A

Q)* *Surely 18dB below clipping (-18dBFS) won't drive my loudspeaker system hard enough?

A) Remember that 18dB below clipping represents about +4dBu/0VU output level on most pro-audio consoles.
Effects, eq., mixing and natural performance dynamics will require the 18dB of headroom through your channel, effects and subgroup sections.

Lower-cost digital console calibration and limited converter quality may push you towards an operating level only 12-to-9dB below clip level.

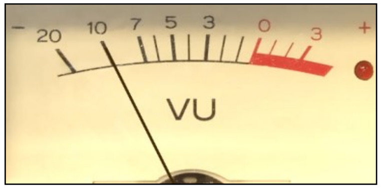
Once you've done all this without crashing the mix, you can then use the **master faders** to push the required level to your main system amplifier racks where your controller limiters will take care of the odd rogue peak.

Standard EBU alignment level for digital audio is also 18dB below full scale *(maximum digital signal level)* - usually written as -18dBFS - so sticking to the 18dB rule will also make you compatible with broadcasters when working at festivals and VIP DJ events.

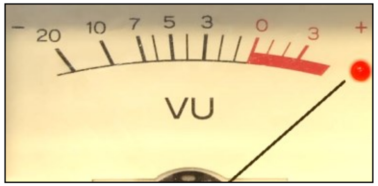
Professional live sound analogue meters calibrated with respect to +4dBu

Most pro-audio analogue equipment can handle internal and output signal levels of approximately 10vrms before clipping. 10vrms is approximately +22dBu so 18dB below that clip levels should equate to approximately +4dBu.

Traditionally, most pro-audio VU-meters (and VU-scaled bar meters) were calibrated so that 0VU corresponded to the industry standard +4dBu to give you 18dB of headroom.



Too low *(-10VU = 28dB below clip)*
Wasting dynamic range



Too high
Peaks could be heavily clipped

Just right *(0VU = 18dB below clip)*
Noise-free without clipping

More Qs & As

Q) What's a dBu?

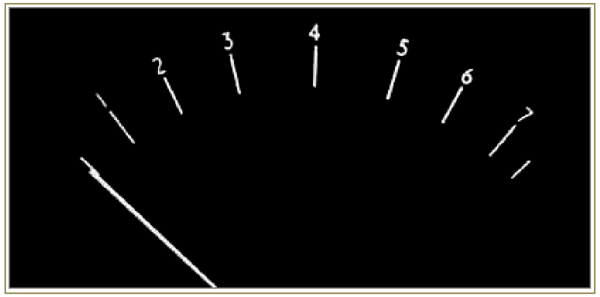
A) dBu *(sometimes written as dBv)* refers to so many dBs above or below 0.775vrms. You've probably seen the 0.775vrms (0dBu) standard used for power amplifier sensitivity.

Q) Why the strange voltage reference?

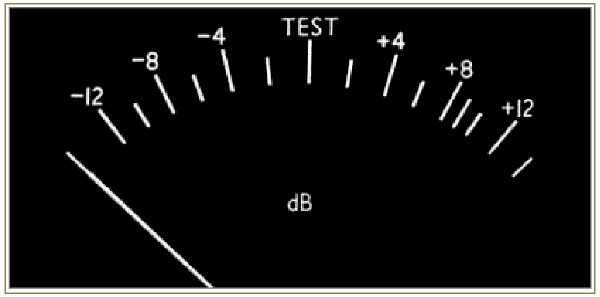
A) The original standard was devised for telephone circuitry and early audio. It was originally called dBm and was defined as 1mW into 600 ohms. Work out $0.775^2/600$ and you'll get a very close to one milliwatt.

Professional analogue and digital peak programme meters (PPMs)

Peak programme meters don't actually measure short signal bursts or transients. They are designed for estimating programme levels.



UK "BBC" PPM scale – in 4dB steps
"4" is usually set for -18dBFS



EBU PPM scale – also in 4dB steps
"Test" is usually set for -18dBFS

There are almost as many PPM scales as there are broadcasting authorities – including a "DIN" standard - where "0" is only -9dBFS!

PPMs move faster in response to tone-bursts than most rms meters. And they fall back more slowly – making them easier to read. But they still underestimate very short bursts and often ignore transients altogether so the 18dB below clip rule must still apply.

K-meters

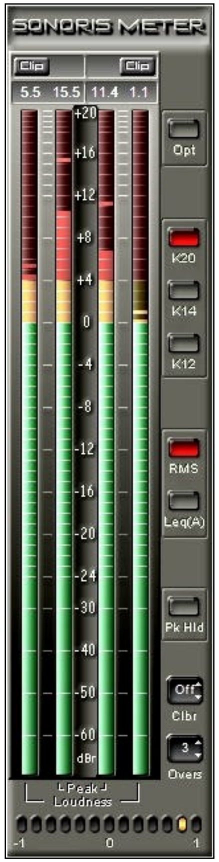
K-meters usually measure rms levels with a fixed 600ms integration and fall-back time over a very wide level range.

Some versions include peak facilities. K-meters usually have switchable scaling: K-20 indicates 0dB at -20dBFS, K-14 has 0dB at -14dBFS and K-12 has 0dB at -12dBFS.

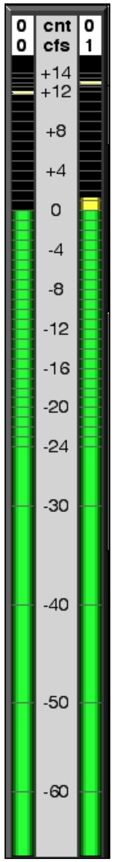
As most K-meters are software-based so it is possible to select K-20, K-14 etc via software. Live sound users should use the K-20 standard wherever possible.

K-meters were designed for recording applications in an attempt to get some consistency in control room monitoring levels. For recording purposes, 0dB is set for 83dBspl but this isn't particularly relevant for live sound use.

A K-meter's main advantage in live sound applications is its accurate indication of levels with reference to full scale digital signal levels.



Sonoris Meter
(Set to K-20)



Spectrafoo meter
(Set to K-14)

Gain structure basics for live sound console users

1: Ensure that all inserts, gates, compressors or plug-ins are bypassed before initially setting your gain structure! See ***Outboards effects set-up*** later.

2: Adjust your input channel mic/line gains for PFL *(pre-fade listen)* levels around +4dBu/0VU *(analogue)*, -18dBFS *(digital)* or the top green indicator *(DJ)*

You may need to trim input gains if the input levels change dramatically during the set. Remember that musicians tend to play louder with an excitable audience in place so allow at least 6dB of spare gain control

If your signals are too hot, even with the gain control below the 9 o'clock position, use a less sensitive input, perhaps line instead of mic, or the console pad switch

If your mic signals are too weak, even with the gain control above the 3 o'clock position, recheck your patch and make sure any input pad or insert points are switched out. It's also worth checking your mic specifications –in case the mic needs phantom power

3: Once you have a healthy PFL level, adjust the relevant subgroup or fader for AFL *(after-fade listen)* levels around +4dBu/0VU *(analogue)*, or -18dBFS *(digital)* with the channel routed and faded up to the nominal 0dB fader mark

Note that, for professional analogue consoles, this assumes that the nominal 0dB fader mark is 10dB below full fader level and that there is gain make-up between the sub-group fader output and the AFL point. Non-professional consoles may not have this gain make-up, so you may need to run at lower AFL levels *(around -6dBu)* to maintain pre-subgroup fader headroom

4: Channel faders will eventually be set to the levels required for artistic balance. Large numbers of channels routed to the same subgroup or output will raise overall mix bus levels and you may have to drop channel fader levels by approximately 3dB every time you double the number of channels routed to the same mix bus

5: Adjust your master/output faders for the required sound level

6: Readjust channel gain controls for suitable PFL levels if input levels change significantly, or if major channel effects or eq changes have been made

Note that short-term transients and peaks in live music - plus mix summing - can easily use up your headroom. Don't be tempted to drive channel rms levels much beyond +4dBu/0VU/-18dBFS if you want to minimise peak clipping and signal degradation further down the signal path.

Again, remember that you can always push the system harder using the master fader, once you've developed a clean mix.

Outboard effects set-up

Assuming you have set the relevant console channel with everything bypassed, switch in your inserts. If the outboard gear isn't too noisy or mushy, use the 18dB-below-clip rule.

Outboard gain structure

1: With your outboard equipment's input gain set to a nominal level (about 1 o'clock if it's an analogue knob), set the relevant insert send control on your console so that your outboard gear's general level is indicating around 18dB below its own maximum (usually -18dBFS on digital gear).

2: With your outboard equipment's output control set to nominal, set the relevant insert return control on your console for PFLs around +4dBu/0VU (analogue) or -18dBFS (digital).

3: Listen to the required effect – on headphones first and then on the PA.

If the outboard gear is too noisy or has poor quality converters, you may have to compromise on headroom by increasing the insert send level by a few dB and decreasing the insert return level by the same amount.

If the outboard gear is starting to sound distorted at your likely maximum effect, you may need to bring its internal signal level down by reducing your console insert send level and increasing the insert return sensitivity by the same amount.

Appendix F – Maximum spl

A note about "maximum spl" specifications

Funktion One publishes a calculated "maximum spl figure" for guidance only. Potential purchasers should be aware, however, that neither calculated nor measured maximum spl figures are a reliable figure of merit or comparison as there are no hard and fast rules when it comes to assessing maximum sound pressure level.

Manufacturers tend to pick 'n' mix the parameters that make their product look best. Some calculate maximum spl, but ignore factors that might detract from that all-important magic number. Others measure responses but choose to ignore vital musical parameters such as power-bandwidth and distortion. Some only quote peak spl using ridiculously short test durations that bear little relationship to musical performance criteria.

And don't be fooled into thinking that manufacturers with impressive "educational" programmes publish more relevant figures. Sadly, the small print would suggest otherwise.

In short, it's a marketing jungle out there and the buyer should beware.

Introduction

Maximum spl figures are often thought of as a key parameter when assessing the suitability of a loudspeaker system. They are the starting point for sound designers' spl-versus-distance predictions and are regularly relied upon for loudspeaker system comparisons.

But are manufacturers' maximum spl figures reliable – or, indeed, comparable?

Manufacturers' small print suggests that a wide variety of calculation and measurement methods are in use. This makes it impossible to compare different manufacturers' data directly.

Most manufacturers quote ***calculated*** maximum spl figures based on their product's sensitivity and their driver supplier's power ratings. Theoretical calculations are easier to do than actual measurements and most manufacturers are honest enough to admit this, stating a lack of suitably isolated locations or facilities.

One major American manufacturer even implies that microphones aren't available to measure beyond 140dBspl. They've obviously never heard of ***Brüel & Kjær.***

Manufacturers who calculate maximum spl figures don't usually indicate the applicable bandwidth or the likely distortion levels. See the ***Calculation*** notes on the next page.

Manufacturers who measure maximum spl figures quote a variety of test signals, endurance time and boundary conditions – or none at all. As ***each*** measurement criterion can affect the resultant maximum spl figure by between 3 and 6dB, it is virtually impossible to compare different manufacturers' spl specifications once all possible criteria are considered. See the ***Measurement*** notes further down.

Calculation

You'd be forgiven for thinking that maximum spl ought to be easy to calculate. After all, we know how to measure loudspeaker sensitivity. We simply apply one watt of signal and see what spl we measure at one metre on axis. And loudspeakers' maximum power ratings have been standardised since the Audio Engineering Society *(AES)* published the original recommendations, AES2-1984, several decades ago. This was revised in 2003 and is still regularly used.

All we have to do is work out the AES power rating in dB, with reference to one watt, and then add that figure to the sensitivity figure, surely?

Working out the AES power rating in dB, with reference to one watt, and then adding that figure to the sensitivity figure is, indeed, a common way of calculating maximum spl so most manufacturers have little choice but to follow suit. This industry tradition appears to allow potential purchasers to compare different products from different manufacturers. But, unfortunately, the figures can cause unfair and misleading comparisons making the whole exercise pointless.

Here are some points to consider:

Sensitivity

1. Loudspeakers rarely have ruler flat spl versus frequency characteristics. Some manufacturers take advantage of this and quote the loudspeaker's sensitivity for a single octave centred on the highest peak, rather than quote an average sensitivity for the loudspeakers' full frequency range. This means that a loudspeaker with a nasty mid frequency resonance in its response could look better in terms of both sensitivity and maximum spl.

2. Quoted frequency ranges are not always the ± 3dB you might expect. They're often defined as the upper and lower frequencies where the loudspeaker's spl drops 10dB lower than the average level of the most sensitive octave *(see below).* So, instead of ±3dB, they're +0dB, -10dB with respect to an octave averaged response peak. This means that a system's sensitivity may be 10dB lower at the upper and lower ends of its frequency range.

3. Loudspeakers don't usually have flat impedance versus frequency characteristics either. Typical direct-radiator loudspeaker impedances tend to peak at the main system resonance, drop back a little, and then rise at high frequencies due to the voice coil inductance. They are also designed to be ***voltage*** driven rather than power driven.

   And most power amplifiers are designed to deliver a flat voltage-versus-frequency characteristic. So, it would be more realistic to quote a loudspeaker's sensitivity in terms of dBspl *(on axis at one metre)* versus drive voltage.

   Some manufacturers already do this, working out the nominal voltage required to deliver one watt at their quoted nominal impedance. But sensitivity is still rarely quoted versus frequency so points 1 and 2 still prevail.

AES power rating

1. A loudspeaker driver's AES power rating is its long-term *(typically, two hours)* ***free air*** power rating. Many loudspeaker manufacturers simply reiterate their driver supplier's figures which may not allow for the effects of voice coil heating *(compression)* under real-world loading conditions. This is particularly relevant for inefficient, heavily processed loudspeakers whose manufacturers like to get into the race for ever more unrealistic power handling claims.

Funktion One tries to avoid this kind of power war by concentrating on turning the electrical signal into sound through highly efficient loudspeaker system designs.

2. AES2-1984 *(r2003)* states “The rated power of the device shall be that power the device can withstand for two hours without ***permanent*** change in acoustical, mechanical, or electrical characteristics, greater than 10%”. This means that a loudspeaker’s AES power rating only refers to its resistance to permanent change *(or failure)*, not to its linear operating range. It is quite permissible for a loudspeaker to generate excruciating levels of distortion or to suffer from several dB of output compression as long as a ***permanent*** change doesn’t take place.

   AES2-1984 *(r2003)* recommendations mention distortion measurements being made at 10% of the AES power rating. This is perfectly reasonable for highly efficient loudspeakers, such as Funktion One systems, as most musical levels will be sitting well below the loudspeaker’s AES rating. But inefficient, power hungry systems are likely to be running much closer to their AES ratings producing unacceptable levels of distortion and mush.

   Current measurements of peak displacement limit maximum driver excursion *(Xmax)* to 10% deviation from linear displacement but manufacturers can get around this by choosing their test bandwidths carefully. The recommendations allow manufacturers to choose between quoting input current distortion or percentage deviation of displacement. The two types of distortion are not the same *(and don’t necessarily increase linearly with excursion)* so manufacturers could simply choose whichever result looks better in the calculations. Unfortunately, distortion is rarely quoted so all this work could be wasted anyway.

   There is now a proposal to measure Xmax at 10% total harmonic pressure distortion or at 10% 2nd or 3rd order modulation distortion using a two-tone signal where the upper frequency component is 8.5 x the frequency and at a 12dB lower level than the lower frequency component.

3. AES2-1984 *(r2003)* recommends that the test signal is band-limited pink noise and states that “The manufacturer shall state the upper and lower cut-off frequencies *(-3dB)* of the noise signal”. Most loudspeaker system manufacturers simply say “band-limited pink noise” without quoting upper and lower cut-off frequencies so it’s impossible to know if low frequency excursion is going to be a limiting factor in practice.

4. AES2-1984 *(r2003)* also recommends that the test noise has a 2:1 peak-to-rms voltage ratio – i.e. a 4:1 peak-to-average power ratio. This leads some manufacturers to quote peak power ratings of four times the long-term AES rating. This is quite permissible as long as the peak power figure is only used to supplement the normal AES power rating – and, of course, the peak power rating doesn’t cause over-excursion at very low frequencies. Again, some indication of upper and lower frequency cut-offs would be helpful.

5. Some manufacturers also add a nominal 6dB spl to their calculated maximum spl figure to allow for “half space” or “2π” *(hemispherical)* floor or wall loading. This can make sense at low frequencies, where small loudspeakers exhibit omnidirectional pressure characteristics. It also makes sense for subwoofers whose response extends low enough for the listener to be regarded as existing in half-space. It doesn’t make sense, however, for directional horn sections whose coverage doesn’t wrap around to the boundary. Again, an indication of bandwidth would help here.

To summarise calculated maximum spl problems:

- The quoted sensitivity – which is, after all, the basis of most calculated maximum spl figures - may be optimistic by up to 10dB at the upper and lower ends of the frequency range
- Likely distortion figures are rarely mentioned
- Most manufacturers' calculations ignore long-term voice coil heating and the resultant compression.
- Manufacturers who state ***peak*** spl may have added 6dB to figures calculated using their driver suppliers AES power ratings
- Manufacturers who state "half space" or "2π" conditions may have added a further 6dB to their figures

Measurement

As mentioned earlier, manufacturers quote a variety of test signals, measurement parameters, endurance times and boundary conditions:

1. As far as test signals are concerned, some manufacturers quote "IEC 60268 noise" that has bandwidth and peak-to-rms characteristics similar to the AES recommendations for power ratings *(see earlier).*

   Some simply state "band-limited noise" without specifying upper and lower frequency limits – whilst others don't specify their test signal at all.

   The choice of test signal makes quite a difference. For instance, the rms level difference between sine sweeps and AES or IEC noise can be 3dB. And regular *(non-AES or IEC)* pink noise differences can be considerably greater.

   The combined choice of measurement parameter and test signal can make quite a difference as well. A peak spl reading will look a lot more impressive than an rms reading – especially with regular pink noise instead of AES or IEC noise. The pink noise's peak measurement will be at a maximum but its unmentioned rms value – and, therefore, its tendency to heat the voice coil and cause output compression – is likely to be considerably lower than the AES or IEC recommendations.

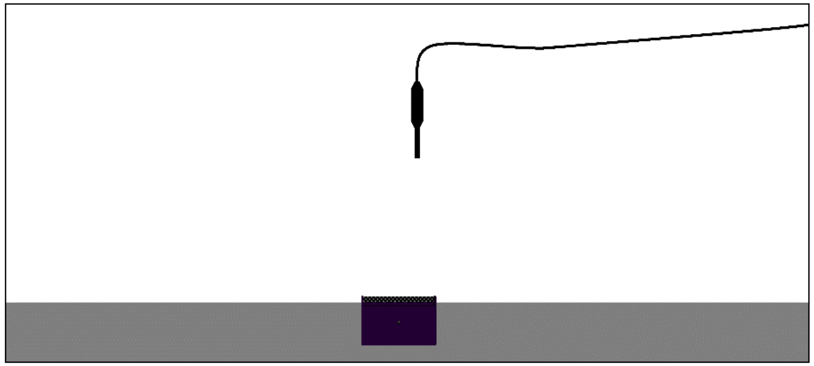
2. Also, the careful choice of spot frequencies or noise bands can exaggerate the maximum spl figures to unrepresentative levels.

3. Most manufacturers agree that loudspeaker systems need to be run for at least two hours for their magnetic assemblies and chassis to reach maximum operating temperature. However, very few manufacturers state measurement duration. One manufacturer quotes maximum spl figures based on peak readings of pink noise and further reading reveals that their product's "peak power handling capacity" is quoted for 10ms. That's only one cycle at 100Hz! Try explaining that to a bass or keyboard player who likes to play sustained notes!

4. Manufacturers who mention boundary conditions tend to state, "half space" or "2π" *(hemispherical)* conditions. Again, this can make sense for small loudspeakers at low frequencies. And, of course, for subwoofers whose response extends low enough for the listener to be regarded as existing in half-space.

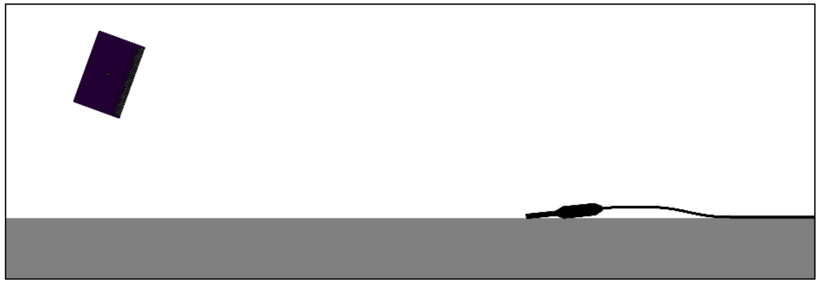
5. Note that there are two types of “half-space” set-up used to minimise the effects of delayed ground reflections where an anechoic chamber isn’t available:

   i) The most common set-up is where the loudspeaker is mounted in half space – typically in a pit facing upwards so that its baffle *(not its grille)* is flush with the ground. The measurement microphone is placed above the loudspeaker – usually on the acoustical crossover axis.



   This half space loudspeaker set-up will show a level enhancement of up to 6dB at very low frequencies where the loudspeaker response is almost omnidirectional. Results correlate nicely with the real-world listening experience.

   ii) The second set-up is where the loudspeaker is placed on or near the ground and its acoustical crossover axis is tilted towards a “boundary” microphone. The microphone is a small, but accurate, omnidirectional measurement microphone placed snugly against the ground. This half space microphone set-up will show a level enhancement of up to 6dB over the full frequency range of the loudspeaker being tested, as long as the surface is hard and smooth.



   The method can be problematic, though, depending on the loudspeaker’s off-axis polar response and ground irregularities at high frequencies.

   The boundary method also tends to enhance the mids and highs when compared with real-world listening conditions - where most listeners' ears tend to be several wavelengths above the floor at mid and high frequencies. The pit method *(i)* is preferred.

6. There are still some manufacturers who state, “open space” or “4π” *(spherical)* conditions. It’s a difficult condition to meet in practice as you need a measurement chamber that is anechoic down to the lowest frequency to be measured. Unless, of course, you hoist the loudspeaker and measurement microphone high above the ground.

To summarise measured maximum spl problems:

- There is usually no way of knowing what test bandwidth a manufacturer has used as this is rarely stated. Maximum spl measured at a response peak is useful for alarm sounders but meaningless if you’re interested in broadband performance
- Different manufacturers use different test signals, and these can have significant effects on, for instance, peak measurements, as accompanying rms levels will be different. Remember that pre-recorded contemporary music can be more compressed than classical recordings – so peak-to-rms factors may be relevant to your installation
- Some manufacturers don’t allow adequate warm-up time or measurement duration so maximum spl figures don’t include long-term compression Maximum peak or burst spl figures are useful – but they should only be used to supplement the normal AES power rating and not to replace it
- Manufacturers rarely define the test conditions adequately. Not all “half space” set-ups are equal. Half space loudspeaker measurements can make perfect sense, but half space microphone measurements may yield mid-high figures that are up to 6dB higher than is achievable at ear-height
- Again, distortion figures are rarely mentioned. Efficient designs tend to generate less distortion than inefficient ones for the same spl. Beware products with impressive maximum spl and AES power rating figures but no mention of distortion. Power hungry and loudspeakers doesn’t necessarily equate with musicality and projection

Powered loudspeakers

Maximum spl figures are often quoted for powered loudspeakers but very few manufacturers quote figures for continuous power as per the drivers’ AES power ratings.

Before miniature power modules were available, amplifier recommendations were based on the driver’s AES power rating. This was tested using noise with a 6dB peak-to-rms voltage ratio. This implied that a good power amplifier could supply the driver’s AES power continuously and that it would be able to supply four times that power in bursts.

Amplifiers built into loudspeakers tend to be quite small for aesthetic and weight reasons and are often based on designs with large enough voltage swings for instantaneous peaks, but inadequate power supply capacity for sustained performance. It is not unusual for powered loudspeaker manufacturers and power module OEM suppliers to quote maximum power for just a few hundred milliseconds *(often less)* and to quote peak or burst power for only a few tens of milliseconds. So, that impressive maximum spl figure wouldn’t be relevant for sustained bass or keyboard notes.

Such systems tend to sound impressive on opening percussion runs with single or sparse instrumentation, but they rapidly degenerate into a mush once the full band and vocals strike up.

Limiters

Limiters are usually employed to protect the drivers from accidental overload – especially in powered loudspeakers and professional touring systems. But they can also be used to provide a more acceptable spectral balance when the system is being driven hard.

Go to ***www.funktion-one.com/settings/*** for further information.

Most maximum spl figures are calculated from driver sensitivities and AES power ratings so limiters don't really feature in the arithmetic. It would be helpful if manufacturers stated the decrease in maximum spl expected with their recommended limiter settings dialled in. This, of course, should be with AES noise that is band-limited to the pass-band of the relevant loudspeaker or section.

Note that these figures will depend on limiter attack and decay characteristics, and not simply on limiter thresholds.

Again, when measured figures are quoted, it would be helpful if manufacturers quoted the maximum continuous spl and maximum peak or burst spl using the recommended or in-built power amplifier and using the recommended or pre-programmed limiter settings.

Conclusion

Neither calculated nor measured maximum spl figures are a reliable figure of merit or comparison as there are no hard and fast rules when it comes to assessing a loudspeaker's maximum sound pressure level.

Current methods are not transparent enough for potential purchasers to assess different products from a variety of manufacturers and make sensible comparisons.

Many years ago, loudspeaker driver manufacturers embraced AES2-1984 to provide consistency between driver manufacturers' power ratings. The professional audio industry has enjoyed an improvement in driver quality and reliability since then because the competitive focus shifted perceptibly from "smoke and mirror" sales techniques to genuine technical improvement.

We now need a similar industry shift towards an agreed standard for testing and reporting complete loudspeaker system specifications. These specifications should include distortion versus frequency – and versus operating levels. The current single-figure maximum spl specification owes more to the alarm industry that to a serious professional sound industry.

