

PROJECT FEATURES

DX 1616

MATRIX AUDIO PROCESSOR

PROJECT
DX 1616 GP 1 Rev1



TABLE OF CONTENTS

1. GENERAL INFORMATIONS	3
1.1 GETTING STARTED	3
2. PROJECT FEATURES	4
2.1 PROJECT STRUCTURE	6
2.2 INPUT SECTION	6
2.3 SIGNAL ROUTING	7
3. EXAMPLES	8
3.1 SETTING UP AN HDL 20-A SYSTEM	8
3.2 HDL 20-A ARRAY SETTING	9
3.2 DX 1616 I/O SET UP	10
3.3 DX 1616_GP_1_rev1 – PRESET 2 - GENERAL INFORMATIONS	10
Figure 1: Network View	3
Figure 2: Mapping Devices	3
Figure 3: Program architecture	4
Figure 4: Program modules architecture and connections	5
Figure 5: Input Module	6
Figure 6: Analog Input Module	6
Figure 7: Analog Output Module	7
Figure 8: Dante Output Module	8
Figure 9: HDL 20-A – Example	8
Figure 10: Venue and Angles data	9
Figure 11: Array and Pick-Up point data	9

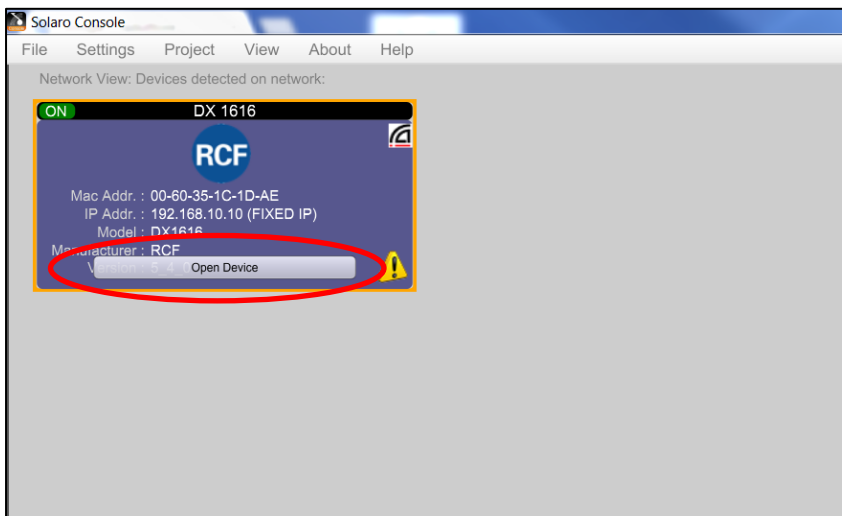
1. GENERAL INFORMATION

Before you start working with the Solaro Console software and your DX 1616 processor, visit RCF's website on www.rcf.it to check and download the DX 1616 project files (.pjson files) available.

IMPORTANT NOTE: This manual only describes the characteristics of the **DX_1616_GP_1 project files**. For more information about using the Solaro Console and Control software, installations, network configurations and customization of the Control interface, please see the respective manuals.

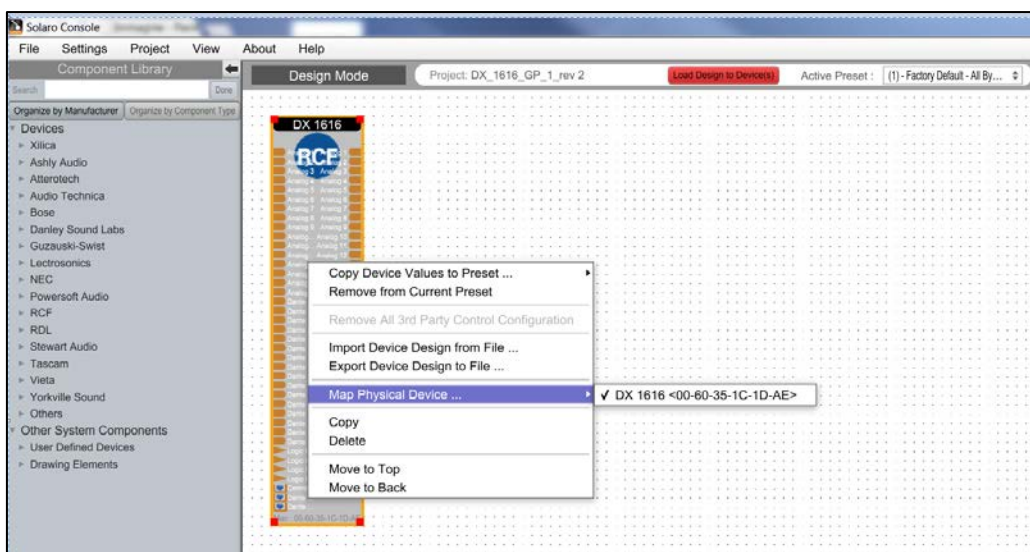
1.1 GETTING STARTED

After connecting the device to our Solaro Console software we'll access to the Network view as follow:



Clicking on “Open Device” button we access at all the modules and parameters of this project to manage our audio system. (See figure 2). Notice that in this way we can control the modules without the possibility to save or load any preset.

Figure 1: Network View



To do it we have to load the project.pjson file; **File>Open Project...** and then we have to “Map Physical Device”, in this case our DX 1616 processor. Now our project is matched to the device and we can access at the load and save

Figure 2: Mapping Devices

preset functions. See the “EXAMPLE” chapter.

2. PROJECT FEATURES

- 16 analog inputs
- 8 digital AES/EBU inputs
- 8 Network Dante inputs
- Pink Noise Generator
- Full processing
- 16 analog outputs
- 8 digital AES/EBU outputs
- 16 Network Dante Outputs

In the following figure we can see the architecture of the program:

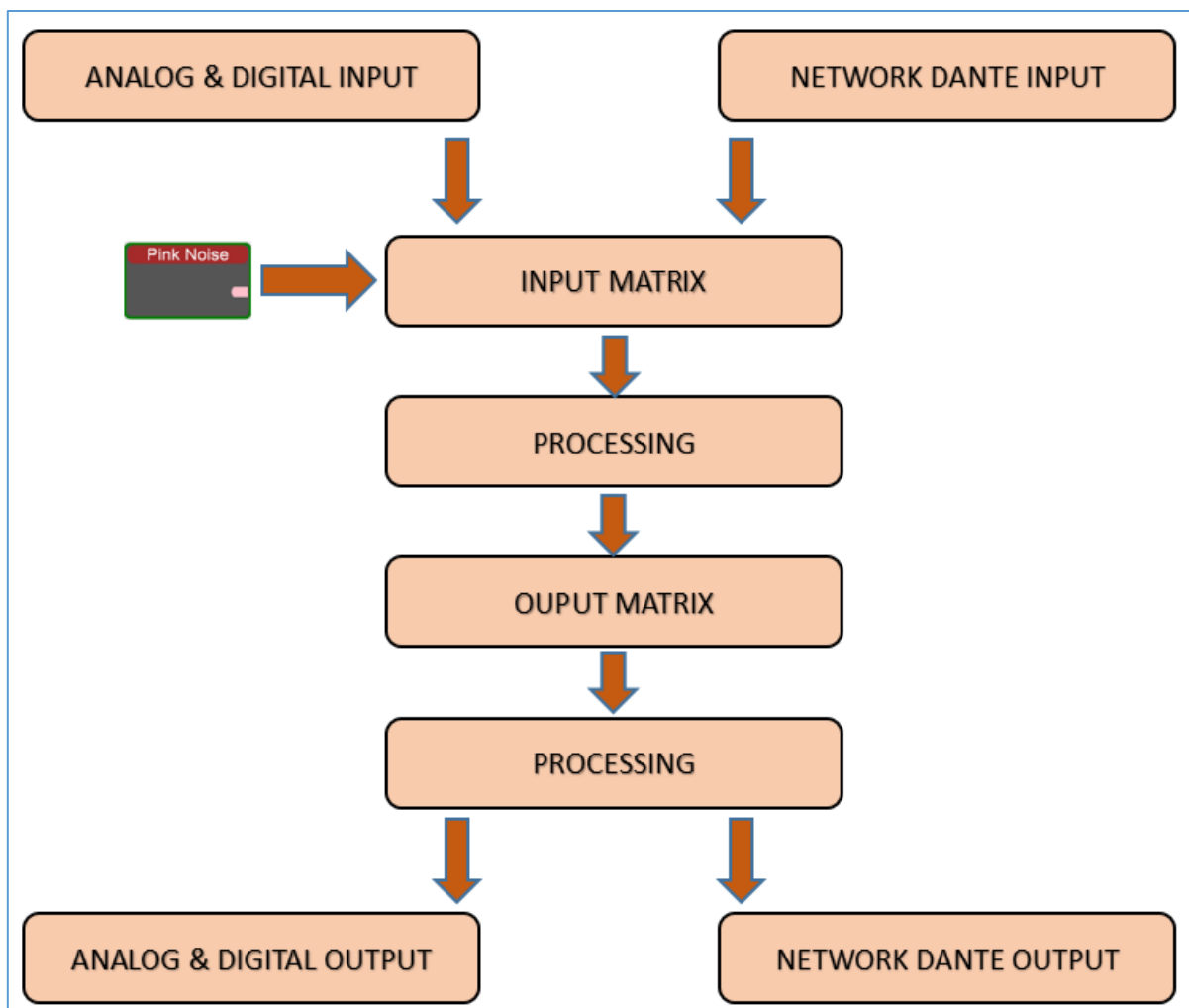


Figure 3: Program architecture

The project was developed using the first 6 channels (input and output) as 3 stereo channel pairs. In the following figure we can see the Solaro Console interface with all the modules of the project with the relative connections among them.

2.1 PROJECT STRUCTURE

Once the DSP Hardware Device is open, the design file can be accessed as shown in Figure 2.

The module control interface can be launched by clicking on individual modules and the user has the possibility to change the parameters.

Note: If necessary, it is important to select and set the gain structure in the Analog domain first. Digital Trim is POST (after) the Analog gain setting.

2.2 INPUT SECTION

By double clicking on the “INPUT” module, the user can access to the Digital Trim, Mute and Phase control, and also an RMS meter which is post Analog and Digital Gain/Trim.

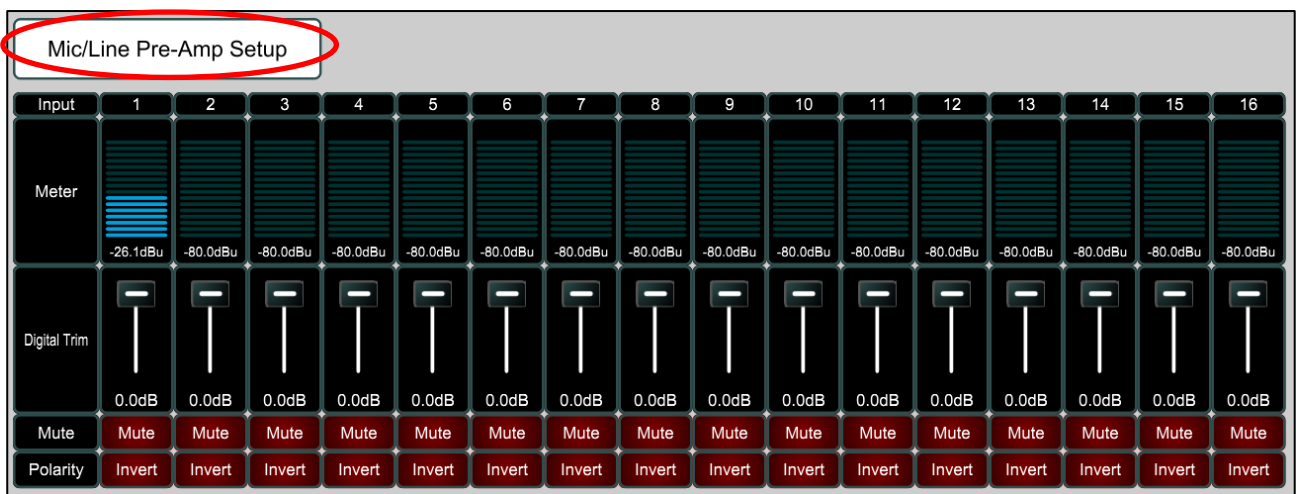


Figure 5: Input Module

Clicking on the <Mic/Line Pre-Amp Setup> we'll access to the analog input section as shown in figure 4.

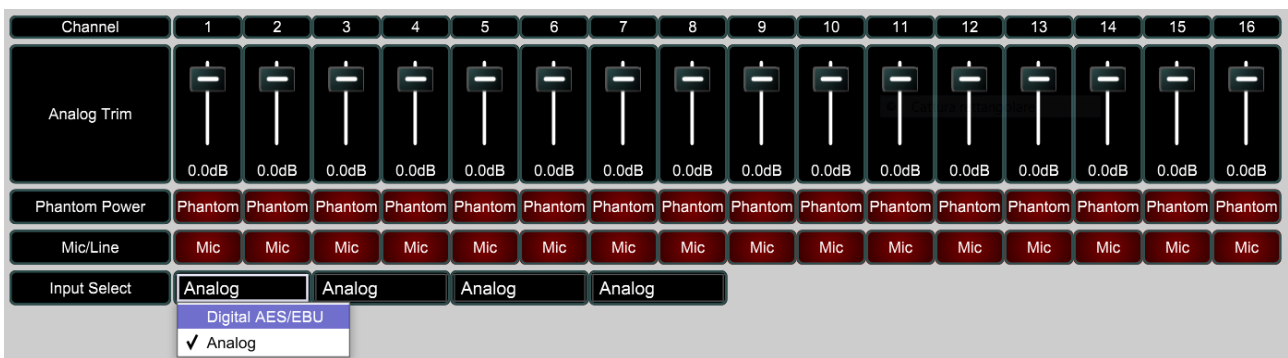


Figure 6: Analog Input Module

In the first 8 channels we can select the Analog or Digital AES/EBU Input Mode. We can also select the Mic Input Mode plus, if necessary, Phantom (48V) power.

In this section we have also a Pink Noise Generator connected to the MATRIX INPUT module for measurement and testing operations.

2.3 SIGNAL ROUTING

The first **INPUT MATRIX** follows the input section. As mentioned before, the first 6 channels are 3 stereo channel pairs and the following 10 channels are individual mono. In this module we can make, for instance, mono sum of the first channels for the management of the subwoofers, etc.

After we have the input processing section that consist in:

- High Pass Filter 6-48 dB/oct
- Low Pass filter 6-48 dB/oct
- 5 Band Parametric EQ
- Delay from 1 to 500 ms
- Compressor

The following module is the **CORE MATRIX** that allows sending the signal to the output section. Before going to the outputs the signal can be processed using the following modules:

- Low Shelv Filter
- High Shelv Filter
- 5 Band Parametric EQ
- Delay from 1 to 100 ms
- All Pass Filter
- Limiter

The 16 analog output channels are in parallel with the Dante Network output channels. The output modules (Analog and Network), allow to change level, mute, polarity and also to see the output level using the RMS meter. The following figure shows the output modules:

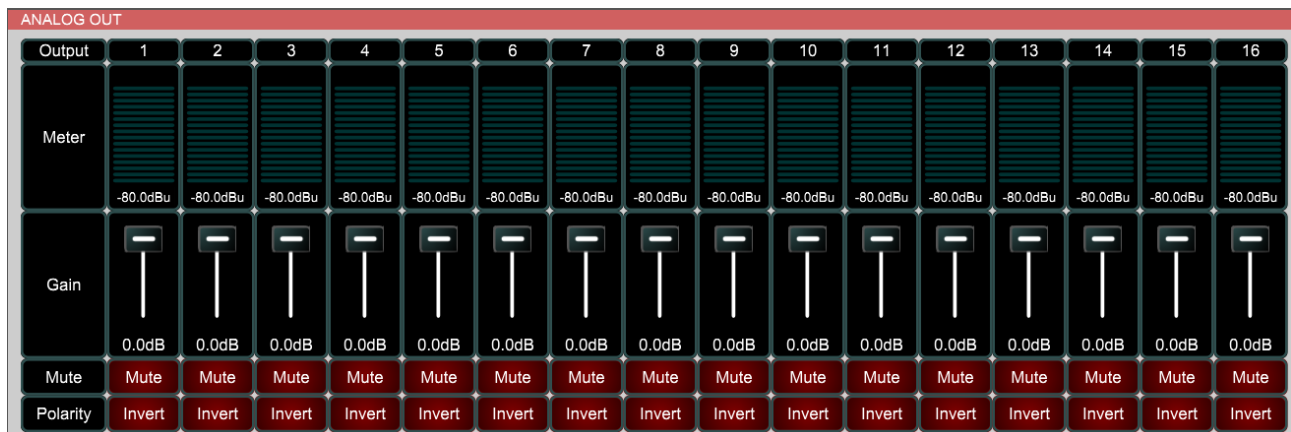


Figure 7: Analog Output Module

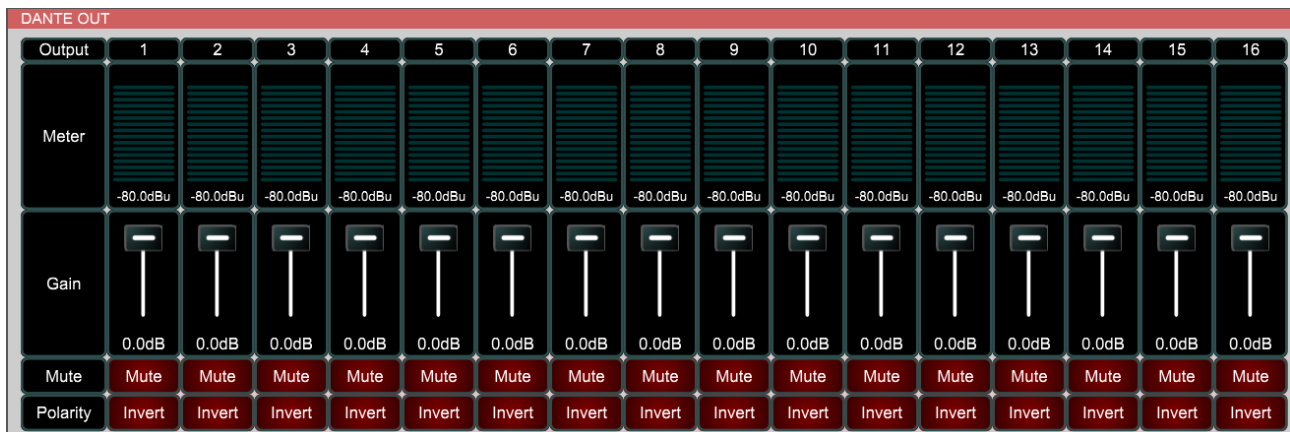


Figure 8: Dante Output Module

3. EXAMPLES

The first preset available in the preset list is the “Factory Default” where all the modules are bypassed and the routing of the processor is Input 1-16 to Output 1-16 so by selecting this preset we can return in a factory default situation.

3.1 SETTING UP AN HDL 20-A SYSTEM

The second preset is an example of a set up related to 6+6 HDL 20-A Line array system coupled with a straight line (sub array) of 5 subs 8006-AS plus 2 HDL 20-A used as front-fill.

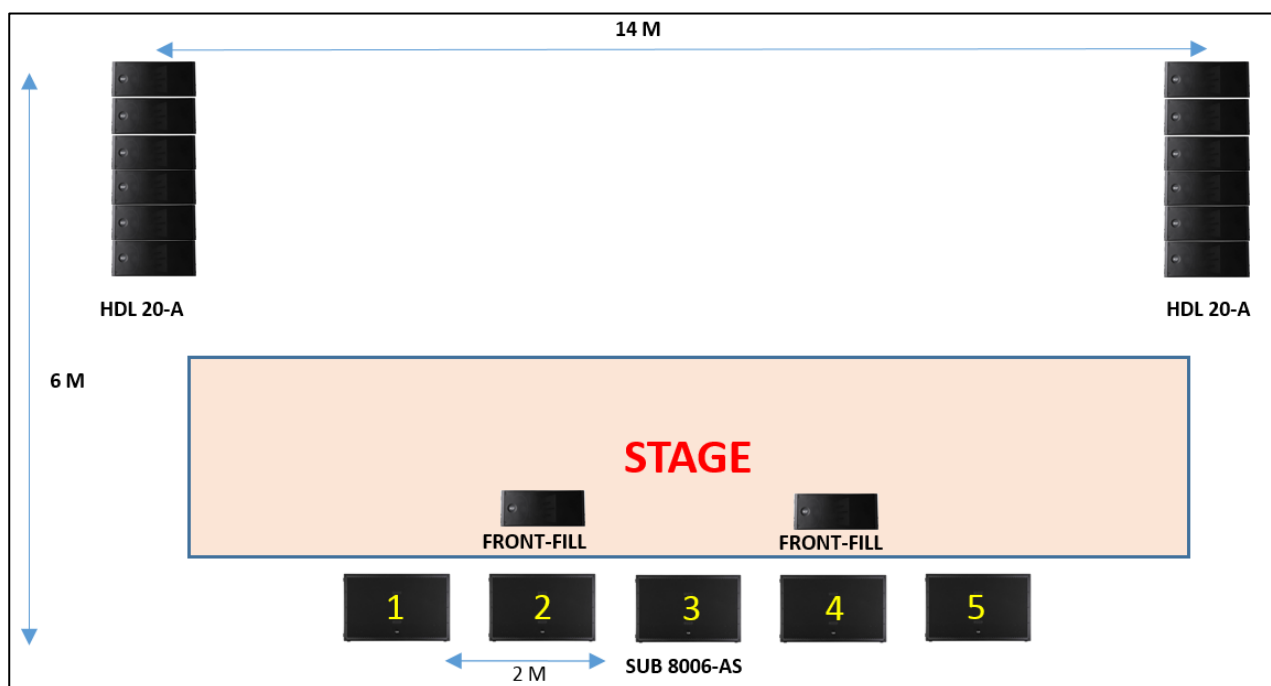


Figure 9: HDL 20-A – Example

3.2 HDL 20-A ARRAY SETTING

The project of this chapter regards a venue of depth of 45 meter; the coverage of the system will start at 7 to 45 meter. Follow the indications in figure 9 and 10 to set up the array.

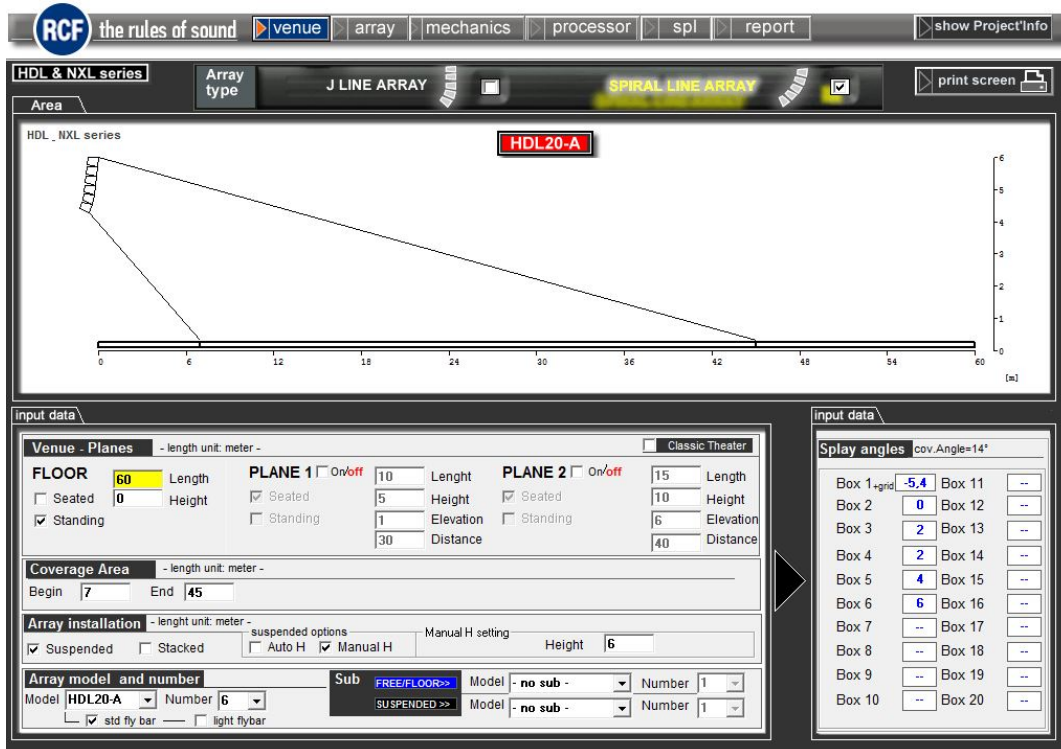


Figure 10: Venue and Angles data

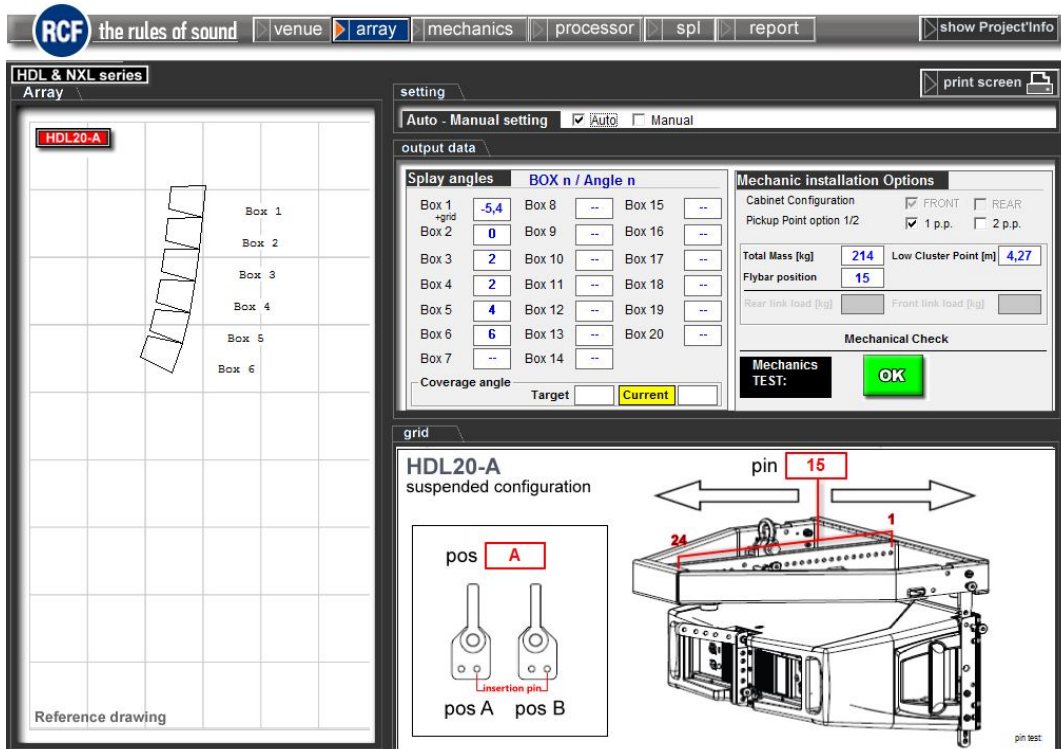


Figure 11: Array and Pick-Up point data

3.2 DX 1616 I/O SET UP

DX 1616 Input:

- Input 1 - from FOH Mixer Left
- Input 2 - from FOH Mixer Right
- Input 7 - from FOH Mixer Mono (for subs)

DX 1616 Output:

- Out 1 - to Cluster Left
- Out 2 - to Cluster Right
- Out 7 - to Sub 1
- Out 8 - to Sub 2
- Out 9 - to Sub 3
- Out 10 - to Sub 4
- Out 11 - to Sub 5
- Out 12 - to Front-Fill Left
- Out 13 - to Front-Fill Right

Note: If the FOH mixer or the FOH engineer works in a set-up without mono signal, we have to manage the input matrix to build the mono using Left and Right signal.

3.3 DX 1616_GP_1_rev1 – PRESET 2 - GENERAL INFORMATION

The cross over frequency of the system is centred 75 HZ and the sub array is already delayed by 4 ms. Of course users must check the alignment of the sub array with a proper measurement system. 4 ms is the value that we used to align the system in our RCF demo area. (Probably it will work in the mayor part of situations).

After this delay, the sub array is electronically curved (using the delay modules after the Core Matrix), to achieve 120 ° of horizontal low frequency coverage (from 30 to 75 Hz).

The clusters are slightly equalized to work properly in the moment of the simulation (humidity, temperature, etc.). The users have to check if this configuration can match their needs. Our suggestion is to start with the module bypassed and then trying to turn off the bypass, listening if it work properly or no.

The Front-Fill are delayed by 4 ms and there is a low shelv filter centred at 300 Hz that decreases the low frequency (that comes from the main system).

We remark that this configuration is developed in order to give to the users the possibility to see some functions of the DX 1616. Try to navigate into the modules to see the setting.

Of course if users want use this preset in a real venue they have to check all the phase alignment with a proper measure. (And using our ears).

RCF Engineering Support Group is at your disposal for any information and clarification you might require techsupport.pro@rcf.it.



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