**Architects and Engineering specification**

The PLENA matrix digital audio system.

The system is comprised of the following products;

 PLM-8M8 – DSP matrix mixer

 PLM- 4Pxxx- DSP amplifier

 PLM-8CS – Call station

 PLM- WCP

**PLM-8M8 DSP matrix mixer**

The DSP matrix mixer shall be able to control 8 independent zone outputs simultaneously and provide 4 mic/line inputs via combo XLR/TRS connectors, 3 Line level inputs via RCA/Cinch connections and a dedicated call station paging bus via RJ45. Audio output connection to amplifiers or powered speakers shall either be via Phoenix line output connectors or via a pair of Cat5 cables using the “Amplink” RJ45 connectors. Up to 8 call stations and up to 16 control panels can be connected to the DSP matrix mixer at any time using the RJ45 connectors provided.

The control software GUI shall provide a minimum 2 levels of user access with password protection for the administrator level. The first level of user control will be both on an iOS format and Windows PC and it shall provide the ability to control the input mix, select/toggle line input sources and adjust output level for up to 8 assignable independent zone outputs. This level of user control shall also be able to be controlled via 3rd party API control.

The second level of administrator access shall only be available via Windows PC. It shall be able to access all user level controls plus the following features;

* All mic/line inputs shall provide input preamp gain control of 51dB, -16dBPad, +48V phantom power 100Hz HPF
* All inputs shall provide DRC (Dynamic Range Compression)
* All inputs shall provide 5 Parametric EQ filters.
* All line inputs shall provide input trim level control +/- 12dB

The 8 outputs shall have:

* Output channel assignment
* Crossover
* 7 band parametric with Bell, All pass, Notch, 1st and 2nd order Shelf filters
* 120ms Delay
* DRC and Max output level adjustment.

Mixing of the outputs shall be provided by a separate user interface screen that shall be able to control all input and output levels. The GUI shall provide a min of 5 user configuration presets, and administrator and User level access.

Ducking shall be provided for paging inputs with 4 priority settings. Mic/Line inputs shall also be able to duck the BGM sources. The DSP matrix mixer will also provide an audio override input that shall override all inputs into the system in emergency situations.

**PLM-4P125 DSP Amplifier**

Amplifiers shall provide 4 independent channels of 125W for each channel. The amplifier shall have 4 ohm, 8ohm, 70V and 100V connectors for each output channel. All output impedances shall be rated at a minimum of 125Watts per output channel. Bridging and Input sensitivity between outputs will be available via dip switch settings on the rear of the unit.

The device shall provide 4 line inputs via combo XLR/TRS connectors, Phoenix connectors and input from a DSP matrix mixer via an RJ45 “Amplink” connector. The amplifier shall also provide a line level override input that is activated via logic switch.
Each of the 4 channels shall be equipped with:

* a 4:1 input mixer
* pink noise generator
* override input level control
* DRC (dynamic range compression)
* Crossovers to the 8th order
* 8 band parametric with Bell, All pass, Notch, 1st and 2nd order Shelf filters
* Dynamic bass enhancement on/off
* 120ms Delay
* Max output level adjustment
* 5 device presents ( store/ recall)
* Hardware and software password lock out.

Control of the DSP shall be via Ethernet with a windows GUI for administrator level control. The DSP should also support an iOS app and be compatible with 3rd party controls for basic user level controls.

The amplifier shall be able to run in a <6Wstandby mode with both NO or NC connections that enable the unit to be connected to motion detection systems with up to a 4 hour activation delay. The amplifier shall provide a 12V output suitable to power motion detectors.

**PLM-4P220 DSP Amplifier**

Amplifiers shall provide 4 independent channels of 220W for each channel. The amplifier shall have 4 ohm, 8ohm, 70V and 100V connectors for each output channel. All output impedances shall be rated at a minimum of 220Watts per output channel.

Bridging and Input sensitivity between -will be available via dip switch settings on the rear of the unit.

The device shall provide 4 line inputs via combo XLR/TRS connectors, Phoenix connectors and input from a DSP matrix mixer via a RJ45 connector. The amplifier shall also provide a line level override input that is activated via logic switch.
Each of the 4 channels shall be equipped with a 4:1 input mixer, pink noise generator, override input level control, DRC (dynamic range compression), Crossovers to the 8th order, 8 band parametric with Bell, All pass, Notch, 1st and 2nd order Shelf filters, Dynamic bass enhancement on/off, 120ms Delay, Max output level adjustment, 5 device presents ( store/ recall), Hardware and software password lock out. Control of the DSP shall be via Ethernet with a windows GUI for administrator level control. The DSP should also support an iOS app and be compatible with 3rd party controls for basic user level controls.

The amplifier shall be able to run in a <6Wstandby mode with both NO or NC connections that enable the unit to be connected to motion detection systems with up to a 4 hour activation delay. The amplifier shall provide a 12V output suitable to power motion detectors.

**PLM-8CS Call station**

The call station shall be able to be connected to the DSP matrix mixer via RS485 protocol over Cat5 cable. The call station shall be powered via the cat5 cable. Devices shall be able to be daisy chained in topology.

The call station shall have 8 capacitive touch areas that act as zone selection areas. Each of the zone selection areas can be programmed via the DSP matrix GUI to support multiple zones groups. Each call station can be individualized for priority, pre-announcement chimes (on/off), mic level, auto clearing of zone selections (on/off) and name. The Call station shall have a mechanical push to talk button, with 3 colour LED indications to give the status of the system/zone (Zone Busy, Chime indication, Mic open).

A max of 8 call station shall be able to connect into one DSP matrix system. Labels for the units shall be able to be printed out via the DSP matrix GUI.

**PLM-WCP Wall Control Panel**

The wall control panel shall be a surface mounted device with a bracket connection. The unit shall be dedicated to a particular zone via dip switch selection on the rear panel and it shall provide input source selection and volume control for that particular zone.

The device shall be able to be connected to the DSP matrix mixer using RS485 protocol over Cat5 cable. The device shall be powered via the cat5 cable from the DSP matrix mixer. Devices shall be able to be daisy chained in topology. The controller shall be able to be set up to control either; microphone inputs providing a mix, or provide switching of BGM sources via toggling.

Labels for the units shall be able to be printed out via the DSP matrix GUI.