ARCHITECT'S AND ENGINEER'S SPECIFICATION

PRODUCTS SUPPORTED:
ClearOne CONVERGE® Pro 2

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CONVERGE PRO 2 128

Digital Signal Processor, Automatic Mixer with Acoustic Echo Cancellation

Architect's and Engineer's Specification

The Automatic Mixer / Digital Signal Processor (DSP) with Distributed Echo Cancellation shall provide twelve balanced mic/line inputs, each with full bandwidth (20 Hz - 22 kHz) AEC, and eight balanced mic/line outputs, using Euro block connectors. Inputs and outputs shall be analog, with internal 24-bit A/D & D/A converters operating at a sample rate of 48 kHz.

The Acoustic Echo Cancellation (AEC) feature on each mic/line input shall have a frequency response of 20 Hz to 22 kHz, tail time of 250 ms, convergence rate of 120 dB/sec, Noise Cancellation (NC) depth up to 25 dB and AEC + NC latency of 32 ms.

The automatic mixer shall include advanced functions to prevent false activation of microphones including: Microphone gating, PA Adaptive mode, First Mic Priority and Adaptive Ambient. USB and Ethernet communications shall be utilized for software control and configuration.

Software shall be provided for configuring and adjusting system parameters within each hardware unit and within a daisy-chained system. Available system components shall include: Automatic microphone mixer, matrix mixer with cross-point level control, acoustic echo cancellation, active noise cancellation, fader channels, feedback elimination, filters, crossovers, dynamics/gain, controls, routers, delays, meters, signal generators, and diagnostics. This software shall provide a matrix view and a signal flow view of the complete system with controls to select and filter the view. This software and command interface shall allow remote diagnostics and user control of input/output channel levels, recall of presets and activation of macros. Custom and third-party control systems shall interface to the unit through the RS232 port, GPIO control/status ports and/or Ethernet port.

The DSP unit shall have a built-in USB Type B port for connecting with a USB host like a PC to provide 2 input audio and 2 output audio channels, at a sample rate of 48 kHz and bit depth of 24. The USB port shall be compatible with USB 2.0 or higher.

This DSP unit shall work as a standalone unit and shall be capable of daisy-chaining with the same DSP unit type or other related DSP mixer type units to make one combined system. Multiple DSP units shall be linkable via a digital, proprietary audio and control bus with cable lengths between units up to 45 meters (150 feet) and up to 144 mic/line inputs in total. The digital audio and control bus shall allow sharing of digital audio and control signals across a multi-unit system. The DSP unit shall be linkable to other types of devices via a digital audio, control and power bus capable of linking with proprietary peripheral devices including the Beamforming Microphone Array 2, USB expander, GPIO expander and Wireless Receiver, with cable lengths between units up to 61 meters (200 feet) and up to three units of each peripheral device.

The DSP unit shall have LED indicators for Power, Status and Locate. It shall also have a capacitive touch button for locating the unit in the software.

Software shall operate on a PC running Windows® 7/10 and above.

Dialer Controller software shall provide mute and volume control functions and custom buttons for the user to program and this software shall operate on a PC, Mac computers, Android and iOS mobile devices.

The Automatic Mixer/Digital Signal Processor shall be compliant with EU 2014/30/EU EMC Directive, the 2011/65/EU RoHS Compliance Directive.

Warranty shall be 3 years with an option to extend the total warranty to 5 years.

The Automatic Mixer/Digital Signal Processor shall be CONVERGE Pro 2 128.

CONVERGE PRO 2 128D

Digital Signal Processor, Automatic Mixer with Acoustic Echo Cancellation and Built-in Dante Digital Audio Network Interface

Architect's and Engineer's Specification

The Automatic Mixer/Digital Signal Processor (DSP) with Distributed Echo Cancellation shall provide twelve balanced mic/line inputs, each with full bandwidth (20 Hz - 22 kHz) AEC, and eight balanced mic/line outputs, using Euro block connectors. Inputs and outputs shall be analog, with internal 24-bit A/D & D/A converters operating at a sample rate of 48 kHz. It shall have built-in Dante digital audio network interface with primary and secondary ports to support 16 input audio channels and 16 output audio channels.

The Acoustic Echo Cancellation (AEC) feature on each mic/line input shall have a frequency response of 20 Hz to 22 kHz, tail time of 250 ms, convergence rate of 120 dB/sec, Noise Cancellation (NC) depth up to 25 dB and AEC + NC latency of 32 ms.

The automatic mixer shall include advanced functions to prevent false activation of microphones including: Microphone gating, PA Adaptive mode, First Mic Priority and Adaptive Ambient. USB and Ethernet communications shall be utilized for software control and configuration.

Software shall be provided for configuring and adjusting system parameters within each hardware unit and within a daisy-chained system. Available system components shall include: Automatic microphone mixer, matrix mixer with cross-point level control, acoustic echo cancellation, active noise cancellation, fader channels, feedback elimination, filters, crossovers, dynamics/gain, controls, routers, delays, meters, signal generators, and diagnostics. This software shall provide a matrix view and a signal flow view of the complete system with controls to select and filter the view. This software and command interface shall allow remote diagnostics and user control of input/output channel levels, recall of presets and activation of macros. Custom and third-party control systems shall interface to the unit through the RS232 port, GPIO control/status ports and/or Ethernet port.

The DSP unit shall have a built-in USB Type B port for connecting with a USB host like a PC to provide 2 input audio and 2 output audio channels, at a sample rate of 48 kHz and bit depth of 24. The USB port shall be compatible with USB 2.0 or higher.

This DSP unit shall work as a standalone unit and shall be capable of daisy-chaining with the same DSP unit type or other related DSP mixer type units to make one combined system. Multiple DSP units shall be linkable via a digital, proprietary audio and control bus with cable lengths between units up to 45 meters (150 feet) and up to 144 mic/line inputs in total. The digital audio and control bus shall allow sharing of digital audio and control signals across a multi-unit system. The DSP unit shall be linkable to other types of devices via a digital audio, control and power bus capable of linking with proprietary peripheral devices including the Beamforming Microphone Array 2, USB expander, GPIO expander and Wireless Receiver, with cable lengths between units up to 61 meters (200 feet) and up to three units of each peripheral device.

The DSP unit shall have LED indicators for Power, Status and Locate. It shall also have a capacitive touch button for locating the unit in the software.

Software shall operate on a PC running Windows® 7/10 and above.

Dialer Controller software shall provide mute and volume control functions and custom buttons for the user to program and this software shall operate on a PC, Mac computers, Android and iOS mobile devices.

The Automatic Mixer/Digital Signal Processor shall be compliant with EU 2014/30/EU EMC Directive, the 2011/65/EU RoHS Compliance Directive.

Warranty shall be 3 years with an option to extend the total warranty to 5 years.

The Automatic Mixer/Digital Signal Processor shall be CONVERGE Pro 2 128D.

CONVERGE PRO 2 128T

Digital Signal Processor, Automatic Mixer with Acoustic Echo Cancellation and Built-in Telephone Interface

Architect's and Engineer's Specification

The Automatic Mixer / Digital Signal Processor (DSP) with Distributed Echo Cancellation shall provide twelve balanced mic/line inputs, each with full bandwidth (20 Hz - 22 kHz) AEC, and eight balanced mic/line outputs, using Euro block connectors. Inputs and outputs shall be analog, with internal 24-bit A/D & D/A converters operating at a sample rate of 48 kHz. The automatic mixer shall provide a built-in telephone line interface.

The Acoustic Echo Cancellation (AEC) feature on each mic/line input shall have a frequency response of 20 Hz to 22 kHz, tail time of 250 ms, convergence rate of 120 dB/sec, Noise Cancellation (NC) depth up to 25 dB and AEC + NC latency of 32 ms.

The automatic mixer shall include advanced functions to prevent false activation of microphones including: Microphone gating, PA Adaptive mode, First Mic Priority and Adaptive Ambient. USB and Ethernet communications shall be utilized for software control and configuration.

Software shall be provided for configuring and adjusting system parameters within each hardware unit and within a daisy-chained system. Available system components shall include: Automatic microphone mixer, matrix mixer with cross-point level control, acoustic echo cancellation, active noise cancellation, fader channels, feedback elimination, filters, crossovers, dynamics/gain, controls, routers, delays, meters, signal generators, and diagnostics. This software shall provide a matrix view and a signal flow view of the complete system with controls to select and filter the view. This software and command interface shall allow remote diagnostics and user control of input/output channel levels, recall of presets and activation of macros. Custom and third-party control systems shall interface to the unit through the RS232 port, GPIO control/status ports and/or Ethernet port.

The telephone interface shall enable direct connection to standard analog telephone lines via an RJ11 connector. This DSP shall provide line-echo cancellation with 250 Hz to 3300 Hz frequency response, caller ID decoding, ring detection and validation, DTMF tone dialing, DTMF decoding, and call progress tone decoding.

The DSP unit shall have a built-in USB Type B port for connecting with a USB host like a PC to provide 2 input audio and 2 output audio channels, at a sample rate of 48 kHz and bit depth of 24. The USB port shall be compatible with USB 2.0 or higher.

This DSP unit shall work as a standalone unit and shall be capable of daisy-chaining with the same DSP unit type or other related DSP mixer type units to make one combined system. Multiple DSP units shall be linkable via a digital, proprietary audio and control bus with cable lengths between units up to 45 meters (150 feet) and up to 144 mic/line inputs in total. The digital audio and control bus shall allow sharing of digital audio and control signals across a multi-unit system. The DSP unit shall be linkable to other types of devices via a digital audio, control and power bus capable of linking with proprietary peripheral devices including the Beamforming Microphone Array 2, USB expander, GPIO expander and Wireless Receiver, with cable lengths between units up to 61 meters (200 feet) and up to three units of each peripheral device.

The DSP unit shall have LED indicators for Power, Status and Locate. It shall also have a capacitive touch button for locating the unit in the software.

Software shall operate on a PC running Windows® 7/10 and above.

Dialer software shall provide call control functions that include make and receive PSTN calls, puts calls on-hold, show call history, mute and volume control functions, custom buttons for the user to program and this software shall operate on a PC and Mac computers, Android and iOS mobile devices.

The Automatic Mixer/Digital Signal Processor shall be compliant with EU 2014/30/EU EMC Directive, the 2011/65/EU RoHS Compliance Directive.

Warranty shall be 3 years with an option to extend the total warranty to 5 years.

The Automatic Mixer/Digital Signal Processor shall be CONVERGE Pro 2 128T.

CONVERGE PRO 2 128TD

Digital Signal Processor, Automatic Mixer with Acoustic Echo Cancellation, Built-in Telephone Interface and Built-in Dante Digital Audio Network Interface

Architect's and Engineer's Specification

The Automatic Mixer / Digital Signal Processor (DSP) with Distributed Echo Cancellation shall provide twelve balanced mic/line inputs, each with full bandwidth (20 Hz - 22 kHz) AEC, and eight balanced mic/line outputs, using Euro block connectors. Inputs and outputs shall be analog, with internal 24-bit A/D & D/A converters operating at a sample rate of 48 kHz. The automatic mixer shall provide a built-in telephone line interface. It shall have built-in Dante digital audio network interface with primary and secondary ports to support 16 input audio channels and 16 output audio channels.

The Acoustic Echo Cancellation (AEC) feature on each mic/line input shall have a frequency response of 20 Hz to 22 kHz, tail time of 250 ms, convergence rate of 120 dB/sec, Noise Cancellation (NC) depth up to 25 dB and AEC + NC latency of 32 ms.

The automatic mixer shall include advanced functions to prevent false activation of microphones including: Microphone gating, PA Adaptive mode, First Mic Priority and Adaptive Ambient. USB and Ethernet communications shall be utilized for software control and configuration.

Software shall be provided for configuring and adjusting system parameters within each hardware unit and within a daisy-chained system. Available system components shall include: Automatic microphone mixer, matrix mixer with cross-point level control, acoustic echo cancellation, active noise cancellation, fader channels, feedback elimination, filters, crossovers, dynamics/gain, controls, routers, delays, meters, signal generators, and diagnostics. This software shall provide a matrix view and a signal flow view of the complete system with controls to select and filter the view. This software and command interface shall allow remote diagnostics and user control of input/output channel levels, recall of presets and activation of macros. Custom and third-party control systems shall interface to the unit through the RS232 port, GPIO control/status ports and/or Ethernet port.

The telephone interface shall enable direct connection to standard analog telephone lines via an RJ11 connector. This DSP shall provide line-echo cancellation with 250 Hz to 3300 Hz frequency response, caller ID decoding, ring detection and validation, DTMF tone dialing, DTMF decoding, and call progress tone decoding.

The DSP unit shall have a built-in USB Type B port for connecting with a USB host like a PC to provide 2 input audio and 2 output audio channels, at a sample rate of 48 kHz and bit depth of 24. The USB port shall be compatible with USB 2.0 or higher.

This DSP unit shall work as a standalone unit and shall be capable of daisy-chaining with the same DSP unit type or other related DSP mixer type units to make one combined system. Multiple DSP units shall be linkable via a digital, proprietary audio and control bus with cable lengths between units up to 45 meters (150 feet) and up to 144 mic/line inputs in total. The digital audio and control bus shall allow sharing of digital audio and control signals across a multi-unit system. The DSP unit shall be linkable to other types of devices via a digital audio, control and power bus capable of linking with proprietary peripheral devices including the Beamforming Microphone Array 2, USB expander, GPIO expander and Wireless Receiver, with cable lengths between units up to 61 meters (200 feet) and up to three units of each peripheral device.

The DSP unit shall have LED indicators for Power, Status and Locate. It shall also have a capacitive touch button for locating the unit in the software.

Software shall operate on a PC running Windows® 7/10 and above.

Dialer software shall provide call control functions that include make and receive PSTN calls, puts calls on-hold, show call history, mute and volume control functions, custom buttons for the user to program and this software shall operate on a PC and Mac computers, Android and iOS mobile devices.

The Automatic Mixer/Digital Signal Processor shall be compliant with EU 2014/30/EU EMC Directive, the 2011/65/EU RoHS Compliance Directive.

Warranty shall be 3 years with an option to extend the total warranty to 5 years.

The Automatic Mixer/Digital Signal Processor shall be CONVERGE Pro 2 128TD.

CONVERGE PRO 2 128V

Digital Signal Processor, Automatic Mixer with Acoustic Echo Cancellation, Built-in VoIP Interface

Architect's and Engineer's Specification

The Automatic Mixer / Digital Signal Processor (DSP) with Distributed Echo Cancellation shall provide twelve balanced mic/line inputs, each with full bandwidth (20 Hz - 22 kHz) AEC, and eight balanced mic/line outputs, using Euro block connectors. Inputs and outputs shall be analog, with internal 24-bit A/D & D/A converters operating at a sample rate of 48 kHz. The automatic mixer shall provide a built-in VoIP (Voice over Internet Protocol) interface which shall provide direct connection to SIP based phone systems.

The Acoustic Echo Cancellation (AEC) feature on each mic/line input shall have a frequency response of 20 Hz to 22 kHz, tail time of 250 ms, convergence rate of 120 dB/sec, Noise Cancellation (NC) depth up to 25 dB and AEC + NC latency of 32 ms.

The automatic mixer shall include advanced functions to prevent false activation of microphones including: Microphone gating, PA Adaptive mode, First Mic Priority and Adaptive Ambient. USB and Ethernet communications shall be utilized for software control and configuration.

Software shall be provided for configuring and adjusting system parameters within each hardware unit and within a daisy-chained system. Available system components shall include: Automatic microphone mixer, matrix mixer with cross-point level control, acoustic echo cancellation, active noise cancellation, fader channels, feedback elimination, filters, crossovers, dynamics/gain, controls, routers, delays, meters, signal generators, and diagnostics. This software shall provide a matrix view and a signal flow view of the complete system with controls to select and filter the view. This software and command interface shall allow remote diagnostics and user control of input/output channel levels, recall of presets and activation of macros. Custom and third-party control systems shall interface to the unit through the RS232 port, GPIO control/status ports and/or Ethernet port.

The VoIP interface shall provide support for SIP v2 and G.711uLaw, G.711ALaw, G.722 wideband audio Codec support. The port type shall be RJ45 port. This shall support two phone lines by default with an option to upgrade to get more lines.

The DSP unit shall have a built-in USB Type B port for connecting with a USB host like a PC to provide 2 input audio and 2 output audio channels, at a sample rate of 48 kHz and bit depth of 24. The USB port shall be compatible with USB 2.0 or higher.

This DSP unit shall work as a standalone unit and shall be capable of daisy-chaining with the same DSP unit type or other related DSP mixer type units to make one combined system. Multiple DSP units shall be linkable via a digital, proprietary audio and control bus with cable lengths between units up to 45 meters (150 feet) and up to 144 mic/line inputs in total. The digital audio and control bus shall allow sharing of digital audio and control signals across a multi-unit system. The DSP unit shall be linkable to other types of devices via a digital audio, control and power bus capable of linking with proprietary peripheral devices including the Beamforming Microphone Array 2, USB expander, GPIO expander and Wireless Receiver, with cable lengths between units up to 61 meters (200 feet) and up to three units of each peripheral device.

The DSP unit shall have LED indicators for Power, Status and Locate. It shall also have a capacitive touch button for locating the unit in the software.

Software shall operate on a PC running Windows® 7/10 and above.

Dialer software shall provide call control functions that include make and receive VoIP calls, multi-party conference calls, put calls on-hold, show call history, mute and volume control functions, custom buttons for the user to program and this software shall operate on a PC and Mac computers, Android and iOS mobile devices.

The Automatic Mixer/Digital Signal Processor shall be compliant with EU 2014/30/EU EMC Directive, the 2011/65/EU RoHS Compliance Directive.

Warranty shall be 3 years with an option to extend the total warranty to 5 years.

The Automatic Mixer/Digital Signal Processor shall be CONVERGE Pro 2 128V.

CONVERGE PRO 2 128VD

Digital Signal Processor, Automatic Mixer with Acoustic Echo Cancellation, Built-in VoIP Interface and Built-in Dante Digital Audio Network Interface

Architect's and Engineer's Specification

The Automatic Mixer / Digital Signal Processor (DSP) with Distributed Echo Cancellation shall provide twelve balanced mic/line inputs, each with full bandwidth (20 Hz - 22 kHz) AEC, and eight balanced mic/line outputs, using Euro block connectors. Inputs and outputs shall be analog, with internal 24-bit A/D & D/A converters operating at a sample rate of 48 kHz. The automatic mixer shall provide a built-in VoIP (Voice over Internet Protocol) interface which shall provide direct connection to SIP based phone systems. It shall have built-in Dante digital audio network interface with primary and secondary ports to support 16 input audio channels and 16 output audio channels.

The Acoustic Echo Cancellation (AEC) feature on each mic/line input shall have a frequency response of 20 Hz to 22 kHz, tail time of 250 ms, convergence rate of 120 dB/sec, Noise Cancellation (NC) depth up to 25 dB and AEC + NC latency of 32 ms.

The automatic mixer shall include advanced functions to prevent false activation of microphones including: Microphone gating, PA Adaptive mode, First Mic Priority and Adaptive Ambient. USB and Ethernet communications shall be utilized for software control and configuration.

Software shall be provided for configuring and adjusting system parameters within each hardware unit and within a daisy-chained system. Available system components shall include: Automatic microphone mixer, matrix mixer with cross-point level control, acoustic echo cancellation, active noise cancellation, fader channels, feedback elimination, filters, crossovers, dynamics/gain, controls, routers, delays, meters, signal generators, and diagnostics. This software shall provide a matrix view and a signal flow view of the complete system with controls to select and filter the view. This software and command interface shall allow remote diagnostics and user control of input/output channel levels, recall of presets and activation of macros. Custom and third-party control systems shall interface to the unit through the RS232 port, GPIO control/status ports and/or Ethernet port.

The VoIP interface shall provide support for SIP v2 and G.711uLaw, G.711ALaw, G.722 wideband audio Codec support. The port type shall be RJ45 port. This shall support two phone lines by default with an option to upgrade to get more lines.

The DSP unit shall have a built-in USB Type B port for connecting with a USB host like a PC to provide 2 input audio and 2 output audio channels, at a sample rate of 48 kHz and bit depth of 24. The USB port shall be compatible with USB 2.0 or higher.

This DSP unit shall work as a standalone unit and shall be capable of daisy-chaining with the same DSP unit type or other related DSP mixer type units to make one combined system. Multiple DSP units shall be linkable via a digital, proprietary audio and control bus with cable lengths between units up to 45 meters (150 feet) and up to 144 mic/line inputs in total. The digital audio and control bus shall allow sharing of digital audio and control signals across a multi-unit system. The DSP unit shall be linkable to other types of devices via a digital audio, control and power bus capable of linking with proprietary peripheral devices including the Beamforming Microphone Array 2, USB expander, GPIO expander and Wireless Receiver, with cable lengths between units up to 61 meters (200 feet) and up to three units of each peripheral device.

The DSP unit shall have LED indicators for Power, Status and Locate. It shall also have a capacitive touch button for locating the unit in the software.

Software shall operate on a PC running Windows® 7/10 and above.

Dialer software shall provide call control functions that include make and receive VoIP calls, multi-party conference calls, put calls on-hold, show call history, mute and volume control functions, custom buttons for the user to program and this software shall operate on a PC and Mac computers, Android and iOS mobile devices.

The Automatic Mixer/Digital Signal Processor shall be compliant with EU 2014/30/EU EMC Directive, the 2011/65/EU RoHS Compliance Directive.

Warranty shall be 3 years with an option to extend the total warranty to 5 years.

The Automatic Mixer/Digital Signal Processor shall be CONVERGE Pro 2 128VD.

CONVERGE PRO 2 48T

Digital Signal Processor, Automatic Mixer with Acoustic Echo Cancellation and Built-in Telephone Interface

Architect's and Engineer's Specification

The Automatic Mixer / Digital Signal Processor (DSP) with Distributed Echo Cancellation shall provide four balanced mic/line inputs, each with full bandwidth (20 Hz - 22 kHz) AEC, and eight balanced mic/line outputs, using Euro block connectors. Inputs and outputs shall be analog, with internal 24-bit A/D & D/A converters operating at a sample rate of 48 kHz. The automatic mixer shall provide a built-in telephone line interface.

The Acoustic Echo Cancellation (AEC) feature on each mic/line input shall have a frequency response of 20 Hz to 22 kHz, tail time of 250 ms, convergence rate of 120 dB/sec, Noise Cancellation (NC) depth up to 25 dB and AEC + NC latency of 32 ms.

The automatic mixer shall include advanced functions to prevent false activation of microphones including: Microphone gating, PA Adaptive mode, First Mic Priority and Adaptive Ambient. USB and Ethernet communications shall be utilized for software control and configuration.

Software shall be provided for configuring and adjusting system parameters within each hardware unit and within a daisy-chained system. Available system components shall include: Automatic microphone mixer, matrix mixer with cross-point level control, acoustic echo cancellation, active noise cancellation, fader channels, feedback elimination, filters, crossovers, dynamics/gain, controls, routers, delays, meters, signal generators, and diagnostics. This software shall provide a matrix view and a signal flow view of the complete system with controls to select and filter the view. This software and command interface shall allow remote diagnostics and user control of input/output channel levels, recall of presets and activation of macros. Custom and third-party control systems shall interface to the unit through the RS232 port, GPIO control/status ports and/or Ethernet port.

The telephone interface shall enable direct connection to standard analog telephone lines via an RJ11 connector. This DSP shall provide line-echo cancellation with 250 Hz to 3300 Hz frequency response, caller ID decoding, ring detection and validation, DTMF tone dialing, DTMF decoding, and call progress tone decoding.

The DSP unit shall have a built-in USB Type B port for connecting with a USB host like a PC to provide 2 input audio and 2 output audio channels, at a sample rate of 48 kHz and bit depth of 24. The USB port shall be compatible with USB 2.0 or higher.

The DSP unit shall have 2 channels x 10 Watts speaker outputs, with 8 Ohms load, with dual binding posts.

This DSP unit shall work as a standalone unit and shall be capable of daisy-chaining with the same DSP unit type or other related DSP mixer type units to make one combined system. Multiple DSP units shall be linkable via a digital, proprietary audio and control bus with cable lengths between units up to 45 meters (150 feet) and up to 144 mic/line inputs in total. The digital audio and control bus shall allow sharing of digital audio and control signals across a multi-unit system. The DSP unit shall be linkable to other types of devices via a digital audio, control and power bus capable of linking with proprietary peripheral devices including the Beamforming Microphone Array 2, USB expander, GPIO expander and Wireless Receiver, with cable lengths between units up to 61 meters (200 feet) and up to three units of each peripheral device.

The DSP unit shall have LED indicators for Power, Status and Locate. It shall also have a capacitive touch button for locating the unit in the software.

Software shall operate on a PC running Windows® 7/10 and above.

Dialer software shall provide call control functions that include make and receive PSTN calls, puts calls on-hold, show call history, mute and volume control functions, custom buttons for the user to program and this software shall operate on a PC and Mac computers, Android and iOS mobile devices.

The Automatic Mixer/Digital Signal Processor shall be compliant with EU 2014/30/EU EMC Directive, the 2011/65/EU RoHS Compliance Directive.

Warranty shall be 3 years with an option to extend the total warranty to 5 years.

The Automatic Mixer/Digital Signal Processor shall be CONVERGE Pro 2 48T.

CONVERGE PRO 2 48V

Digital Signal Processor, Automatic Mixer with Acoustic Echo Cancellation, Built-in VoIP Interface

Architect's and Engineer's Specification

The Automatic Mixer / Digital Signal Processor (DSP) with Distributed Echo Cancellation shall provide four balanced mic/line inputs, each with full bandwidth (20 Hz - 22 kHz) AEC, and eight balanced mic/line outputs, using Euro block connectors. Inputs and outputs shall be analog, with internal 24-bit A/D & D/A converters operating at a sample rate of 48 kHz. The automatic mixer shall provide a built-in VoIP (Voice over Internet Protocol) interface which shall provide direct connection to SIP based phone systems.

The Acoustic Echo Cancellation (AEC) feature on each mic/line input shall have a frequency response of 20 Hz to 22 kHz, tail time of 250 ms, convergence rate of 120 dB/sec, Noise Cancellation (NC) depth up to 25 dB and AEC + NC latency of 32 ms.

The automatic mixer shall include advanced functions to prevent false activation of microphones including: Microphone gating, PA Adaptive mode, First Mic Priority and Adaptive Ambient. USB and Ethernet communications shall be utilized for software control and configuration.

Software shall be provided for configuring and adjusting system parameters within each hardware unit and within a daisy-chained system. Available system components shall include: Automatic microphone mixer, matrix mixer with cross-point level control, acoustic echo cancellation, active noise cancellation, fader channels, feedback elimination, filters, crossovers, dynamics/gain, controls, routers, delays, meters, signal generators, and diagnostics. This software shall provide a matrix view and a signal flow view of the complete system with controls to select and filter the view. This software and command interface shall allow remote diagnostics and user control of input/output channel levels, recall of presets and activation of macros. Custom and third-party control systems shall interface to the unit through the RS232 port, GPIO control/status ports and/or Ethernet port.

The VoIP interface shall provide support for SIP v2 and G.711uLaw, G.711ALaw, G.722 wideband audio Codec support. The port type shall be RJ45 port. This shall support two phone lines by default with an option to upgrade to get more lines.

The DSP unit shall have a built-in USB Type B port for connecting with a USB host like a PC to provide 2 input audio and 2 output audio channels, at a sample rate of 48 kHz and bit depth of 24. The USB port shall be compatible with USB 2.0 or higher.

The DSP unit shall have 2 channels x 10 Watts speaker outputs, with 8 Ohms load, with dual binding posts.

This DSP unit shall work as a standalone unit and shall be capable of daisy-chaining with the same DSP unit type or other related DSP mixer type units to make one combined system. Multiple DSP units shall be linkable via a digital, proprietary audio and control bus with cable lengths between units up to 45 meters (150 feet) and up to 144 mic/line inputs in total. The digital audio and control bus shall allow sharing of digital audio and control signals across a multi-unit system. The DSP unit shall be linkable to other types of devices via a digital audio, control and power bus capable of linking with proprietary peripheral devices including the Beamforming Microphone Array 2, USB expander, GPIO expander and Wireless Receiver, with cable lengths between units up to 61 meters (200 feet) and up to three units of each peripheral device.

The DSP unit shall have LED indicators for Power, Status and Locate. It shall also have a capacitive touch button for locating the unit in the software.

Software shall operate on a PC running Windows® 7/10 and above.

Dialer software shall provide call control functions that include make and receive VoIP calls, multi-party conference calls, put calls on-hold, show call history, mute and volume control functions, custom buttons for the user to program and this software shall operate on a PC and Mac computers, Android and iOS mobile devices.

The Automatic Mixer / Digital Signal Processor shall be compliant with EU 2014/30/EU EMC Directive, the 2011/65/EU RoHS Compliance Directive.

Warranty shall be 3 years with an option to extend the total warranty to 5 years.

The Automatic Mixer / Digital Signal Processor shall be CONVERGE Pro 2 48V.

CONVERGE PRO 2 120

Digital Signal Processor, Automatic Mixer with Acoustic Echo Cancellation, with only Inputs

Architect's and Engineer's Specification

The Automatic Mixer / Digital Signal Processor (DSP) with Distributed Echo Cancellation shall provide twelve balanced mic/line inputs, each with full bandwidth (20 Hz - 22 kHz) AEC, using plug-in Euro block connectors to increase the total number of mic/line inputs in a DSP system. Inputs shall be analog, with internal 24-bit A/D & D/A converters operating at a sample rate of 48 kHz.

The Acoustic Echo Cancellation (AEC) feature on each mic/line input shall have a frequency response of 20 Hz to 22 kHz, tail time of 250 ms, convergence rate of 120 dB/sec, Noise Cancellation (NC) depth up to 25 dB and AEC + NC latency of 32 ms.

The automatic mixer shall include advanced functions to prevent false activation of microphones including: Microphone gating, PA Adaptive mode, First Mic Priority and Adaptive Ambient. USB and Ethernet communications shall be utilized for software control and configuration.

Software shall be provided for configuring and adjusting system parameters within each hardware unit and within a daisy-chained system. Available system components shall include: Automatic microphone mixer, matrix mixer with cross-point level control, acoustic echo cancellation, active noise cancellation, fader channels, feedback elimination, filters, crossovers, dynamics/gain, controls, routers, delays, meters, signal generators, and diagnostics. This software shall provide a matrix view and a signal flow view of the complete system with controls to select and filter the view. This software and command interface shall allow remote diagnostics and user control of input channel levels, recall of presets and activation of macros. Custom and third-party control systems shall interface to the unit through the RS232 port, GPIO control/status ports and/or Ethernet port.

The DSP unit shall have a built-in USB Type B port for connecting with a USB host like a PC to provide 2 input audio and 2 output audio channels, at a sample rate of 48 kHz and bit depth of 24. The USB port shall be compatible with USB 2.0 or higher.

This DSP unit shall be capable of daisy-chaining with other related DSP mixer type units to make one combined system. Multiple DSP units shall be linkable via a digital, proprietary audio and control bus with cable lengths between units up to 45 meters (150 feet) and up to 144 mic/line inputs in total. The digital audio and control bus shall allow sharing of digital audio and control signals across a multi-unit system. The DSP unit shall be linkable to other types of devices via a digital audio, control and power bus capable of linking with proprietary peripheral devices including the Beamforming Microphone Array 2, USB expander, GPIO expander and Wireless Receiver, with cable lengths between units up to 61 meters (200 feet) and up to three units of each peripheral device.

The DSP unit shall have LED indicators for Power, Status and Locate. It shall also have a capacitive touch button for locating the unit in the software.

Software shall operate on a PC running Windows® 7/10 and above.

Dialer Controller software shall provide mute and volume control functions and custom buttons for the user to program and this software shall operate on a PC, Mac computers, Android and iOS mobile devices.

The Automatic Mixer/Digital Signal Processor shall be compliant with EU 2014/30/EU EMC Directive, the 2011/65/EU RoHS Compliance Directive.

Warranty shall be 3 years with an option to extend the total warranty to 5 years.

The Automatic Mixer/Digital Signal Processor shall be CONVERGE Pro 2 120.

CONVERGE PRO 2 012

Digital Signal Processor, Automatic Mixer with only Outputs

Architect's and Engineer's Specification

The Automatic Mixer / Digital Signal Processor (DSP) shall provide twelve balanced mic/line outputs, using plug-in Euro block connectors to increase the total number of outputs in a DSP system.

This software shall provide a matrix view and a signal flow view of the complete system with controls to select and filter the view. This software and command interface shall allow remote diagnostics and user control of output channel levels, recall of presets and activation of macros. Custom and third-party control systems shall interface to the unit through the RS232 port, GPIO control/status ports and/or Ethernet port.

The DSP unit shall have a built-in USB Type B port for connecting with a USB host like a PC to provide 2 input audio and 2 output audio channels, at a sample rate of 48 kHz and bit depth of 24. The USB port shall be compatible with USB 2.0 or higher.

This DSP unit shall be capable of daisy-chaining with other related DSP mixer type units to make one combined system. Multiple DSP units shall be linkable via a digital, proprietary audio and control bus with cable lengths between units up to 45 meters (150 feet) and up to 144 mic/line inputs in total. The digital audio and control bus shall allow sharing of digital audio and control signals across a multi-unit system.

The DSP unit shall have LED indicators for Power, Status and Locate. It shall also have a capacitive touch button for locating the unit in the software.

Software shall operate on a PC running Windows® 7/10 and above.

Dialer Controller software shall provide volume control functions and custom buttons for the user to program and this software shall operate on a PC, Mac computers, Android and iOS mobile devices.

The Automatic Mixer/Digital Signal Processor shall be compliant with EU 2014/30/EU EMC Directive, the 2011/65/EU RoHS Compliance Directive.

Warranty shall be 3 years with an option to extend the total warranty to 5 years.

The Automatic Mixer/Digital Signal Processor shall be CONVERGE Pro 2 012.

CONVERGE PRO 2 128SR

Digital Signal Processor, Automatic Mixer for Sound Reinforcement

Architect's and Engineer's Specification

The Automatic Mixer / Digital Signal Processor (DSP) shall provide twelve balanced mic/line inputs, and eight balanced mic/line outputs, using Euro block connectors. Inputs and outputs shall be analog, with internal 24-bit A/D & D/A converters operating at a sample rate of 48 kHz.

Software shall be provided for configuring and adjusting system parameters within each hardware unit and within a daisy-chained system. Available system components shall include: Automatic microphone mixer, matrix mixer with cross-point level control, fader channels, feedback elimination, filters, crossovers, dynamics/gain, controls, routers, delays, meters, signal generators, and diagnostics. This software shall provide a matrix view and a signal flow view of the complete system with controls to select and filter the view. This software and command interface shall allow remote diagnostics and user control of input/output channel levels, recall of presets and activation of macros. Custom and third-party control systems shall interface to the unit through the RS232 port, GPIO control/status ports and/or Ethernet port.

The DSP unit shall have a built-in USB Type B port for connecting with a USB host like a PC to provide 2 input audio and 2 output audio channels, at a sample rate of 48 kHz and bit depth of 24. The USB port shall be compatible with USB 2.0 or higher.

This DSP unit shall work as a standalone unit and shall be capable of daisy-chaining with the same DSP unit type or other related DSP mixer type units to make one combined system. Multiple DSP units shall be linkable via a digital, proprietary audio and control bus with cable lengths between units up to 45 meters (150 feet) and up to 144 mic/line inputs in total. The digital audio and control bus shall allow sharing of digital audio and control signals across a multi-unit system. The DSP unit shall be linkable to other types of devices via a digital audio, control and power bus capable of linking with proprietary peripheral devices – USB expander, GPIO expander and Wireless Receiver, with cable lengths between units up to 61 meters (200 feet) and up to three units of each peripheral device.

The DSP unit shall have LED indicators for Power, Status and Locate. It shall also have a capacitive touch button for locating the unit in the software.

Software shall operate on a PC running Windows® 7/10 and above.

Dialer Controller software shall provide mute and volume control functions and custom buttons for the user to program and this software shall operate on a PC, Mac computers, Android and iOS mobile devices.

The Automatic Mixer/Digital Signal Processor shall be compliant with EU 2014/30/EU EMC Directive, the 2011/65/EU RoHS Compliance Directive.

Warranty shall be 3 years with an option to extend the total warranty to 5 years.

The Automatic Mixer/Digital Signal Processor shall be CONVERGE Pro 2 128SR.

CONVERGE PRO 2 128SRD

Digital Signal Processor, Automatic Mixer for Sound Reinforcement with Built-in Dante Digital Audio Network Interface

Architect's and Engineer's Specification

The Automatic Mixer / Digital Signal Processor (DSP) shall provide twelve balanced mic/line inputs, and eight balanced mic/line outputs, using Euro block connectors. Inputs and outputs shall be analog, with internal 24-bit A/D & D/A converters operating at a sample rate of 48 kHz. It shall have built-in Dante digital audio network interface with primary and secondary ports to support 16 input audio channels and 16 output audio channels.

Software shall be provided for configuring and adjusting system parameters within each hardware unit and within a daisy-chained system. Available system components shall include: Automatic microphone mixer, matrix mixer with cross-point level control, fader channels, feedback elimination, filters, crossovers, dynamics/gain, controls, routers, delays, meters, signal generators, and diagnostics. This software shall provide a matrix view and a signal flow view of the complete system with controls to select and filter the view. This software and command interface shall allow remote diagnostics and user control of input/output channel levels, recall of presets and activation of macros. Custom and third-party control systems shall interface to the unit through the RS232 port, GPIO control/status ports and/or Ethernet port.

The DSP unit shall have a built-in USB Type B port for connecting with a USB host like a PC to provide 2 input audio and 2 output audio channels, at a sample rate of 48 kHz and bit depth of 24. The USB port shall be compatible with USB 2.0 or higher.

This DSP unit shall work as a standalone unit and shall be capable of daisy-chaining with the same DSP unit type or other related DSP mixer type units to make one combined system. Multiple DSP units shall be linkable via a digital, proprietary audio and control bus with cable lengths between units up to 45 meters (150 feet) and up to 144 mic/line inputs in total. The digital audio and control bus shall allow sharing of digital audio and control signals across a multi-unit system. The DSP unit shall be linkable to other types of devices via a digital audio, control and power bus capable of linking with proprietary peripheral devices – USB expander, GPIO expander and Wireless Receiver, with cable lengths between units up to 61 meters (200 feet) and up to three units of each peripheral device.

The DSP unit shall have LED indicators for Power, Status and Locate. It shall also have a capacitive touch button for locating the unit in the software.

Software shall operate on a PC running Windows® 7/10 and above.

Dialer Controller software shall provide mute and volume control functions and custom buttons for the user to program and this software shall operate on a PC, Mac computers, Android and iOS mobile devices.

The Automatic Mixer/Digital Signal Processor shall be compliant with EU 2014/30/EU EMC Directive, the 2011/65/EU RoHS Compliance Directive.

Warranty shall be 3 years with an option to extend the total warranty to 5 years.

The Automatic Mixer/Digital Signal Processor shall be CONVERGE Pro 2 128SRD.

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