Polycom SoundStructure™: Architect's and Engineer's Specifications

SoundStructure™ C16, C12, and C8 Multi-channel Acoustic Echo / Noise Canceller with Automatic Mic / Matrix Mixer

The digital audio device shall be available in hardware configurations of 16 inputs by 16 outputs, 12 inputs by 12 outputs, or 8 inputs by 8 outputs. All the inputs and outputs are balanced analog signals accessible via terminal strip connectors with 24-bit analog to digital and digital to analog converters operating at a sample rate of 48 kHz. Each input shall have software selectable 48V phantom power capable of sourcing 7.5mA of current and have an analog input gain adjustment applied to the analog signal that is software programmable from -20dB to +64dB in 0.5 dB increments.

Each input of the device shall provide 22kHz bandwidth stereo acoustic echo cancellation processing, up to 20 dB of ambient noise cancellation, automatic gain control, ten bands of parametric equalization, ten bands of adaptive feedback reduction, dynamics processing including compression, limiting, expansion, digital gain with a range of +20 to -100dB, and up to one second of audio delay.

Each output of the device shall have up to ten bands of parametric equalization or up to thirty-one bands of graphic equalization, dynamics processing including expansion, compression, and limiting, and up to one second of audio delay.

The device shall support up to thirty-two AEC reference signals per device with each audio input accepting up to two AEC references. A total of 16, 12, or 8 microphone / line inputs shall be provided and all inputs shall have equivalent input processing and support microphone or line level signals. The matrix mixer will scale seamlessly from 8x8 to 128x128 with up to eight devices linked together. All inputs from the linked devices shall be available to all outputs. Matrix crosspoints shall be gain adjustable in 0.1 dB increments over a range of +20 dB to 100 dB. The input section of the matrix shall include up to 128 microphone or line inputs, up to 128 submixes, and up to 8 signal generators. The output section of the matrix shall include up to 128 line level signals and 128 submixes.

The digital audio device shall operate with 22kHz audio bandwidth (0Hz – 22kHz) and 48 kHz sample rate and be capable using two simultaneous references for stereo operation on every input. AEC convergence shall be no slower than 40 dB / second. The AEC shall have a minimum echo cancellation span of 200 ms per channel. The AEC shall be capable of operation at room gain levels of more than +15 dB. In addition, the unit shall provide user selectable amounts of ambient noise cancellation on each microphone input from 0 to 20dB. The noise cancellation shall effectively cancel steady-state ambient noise at all frequencies without causing any perceptible degradation of human voice or other transient sounds.

The device shall have Ethernet and RS-232 ports where both ports can be used for simultaneous complete control of the device. A programmable logic interface shall be

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provided, with 22 logic inputs, 22 logic outputs, and 2 analog gain controls. The device shall connect digitally to the Polycom HDX video codec systems via a single cable connection that provides bi-directional stereo digital audio between the HDX system and the digital audio device.

The digital audio device can be linked with up to seven additional devices to act as one large system, with control of open microphones across all microphone channels in linked systems.

The device shall be powered by an internal, UL approved, power supply. The digital audio device shall be no larger than one rack unit in size and shall accept one plug-in card. The device shall comply with the ITU G.167 Recommendation for AEC, FCC part 15, CE, and RoHS requirements.

The Polycom SoundStructure C16, C12, or C8 is specified.

SoundStructure™ SR12 Multi-channel Noise Canceller with Automatic Mic / Matrix Mixer

The digital audio device shall be available in hardware configuration of 12 inputs by 12 outputs. All the inputs and outputs are balanced analog signals accessible via terminal strip connectors with 24-bit analog to digital and digital to analog converters operating at a sample rate of 48 kHz. Each input shall have software selectable 48V phantom power capable of sourcing 7.5mA of current and have an analog input gain adjustment applied to the analog signal that is software programmable from -20dB to +64dB in 0.5 dB increments.

A total of 12 microphone / line inputs shall be provided and all inputs will have equivalent input processing and support microphone inputs. Each input of the device shall have 22kHz bandwidth with up to 20 dB of ambient noise cancellation, automatic gain control, ten bands of parametric equalization, ten bands of adaptive feedback reduction, dynamics processing including compression, limiting, expansion, digital gain with a range of +20 to -100dB, and up to one second of audio delay.

The digital audio device shall provide user selectable amounts of ambient noise cancellation on each microphone input from 0 to 20dB. The noise cancellation shall effectively cancel steady-state ambient noise at all frequencies without causing any perceptible degradation of human voice or other transient sounds.

Each output of the device shall have up to ten bands of parametric equalization or up to thirty-one bands of graphic equalization, dynamics processing including expansion, compression, and limiting, and up to one second of audio delay.

The matrix mixer will scale seamlessly from 12x12 to 96x96 with up to eight similar devices linked together. All inputs from the device will be available to all outputs. The

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device shall also link seamlessly with SoundStructure C-series devices. Matrix crosspoints shall be gain adjustable in 0.1 dB increments over a range of +20 dB to 100 dB. The input section of the matrix shall include up to 96 microphone or line inputs, up to 96 submixes, and up to 8 signal generators. The output section of the matrix shall include up to 96 line level signals and 96 submixes.

The device shall have Ethernet and RS-232 ports where both ports can be used for simultaneous complete control of the device. A programmable logic interface shall be provided, with 22 logic inputs, 22 logic outputs, and 2 analog gain controls. The digital audio device can be linked with up to seven additional devices to act as one large system, with control of open microphones across all microphone channels in linked systems.

The device shall be powered by an internal, UL approved, power supply. The digital audio device shall be no larger than one rack unit in size and shall accept one plug-in card. The device shall comply with FCC part 15, CE, and RoHS requirements.
The Polycom SoundStructure SR12 is specified.

SoundStructure TEL1 Telephony interface

The unit shall be a line echo canceller for interface to the public switch telephone network. An integral AGC (Automatic Gain Control), dynamics processing, and noise cancellation shall be provided on the telephony input. In addition, a ten-band parametric EQ shall be provided on the telephony input and output channels.

The telephony interface shall be through a field-installable plug-in card for direct connection to a standard (dial-up) phone line using RJ-11 connectors for both line and telephone set. Tip and ring from the phone line shall be present at the line connector when the hybrid is off. The telephony interface shall have a 250-3.6 kHz +/-3dB frequency response with 32ms echo cancellation span, 0 to 20 dB noise cancellation, and 30dB/second LEC convergence. The device shall have a built-in DTMF dialer, call progress tone detector, user-set "Entry" and "Exit" tones, "Privacy" mode to alternate between telephone hand-set and conference modes, and international telephone line support.

The Polycom SoundStructure TEL1 is specified.

SoundStructure TEL2 Telephony interface

The unit shall be a two-line line echo canceller for interface to two simultaneous public switch telephone connections. An integral AGC (Automatic Gain Control), dynamics processing, noise cancellation shall be provided on both telephony inputs. In addition, a ten-band parametric equalizer shall be provided on both the telephony input and output channels.

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The telephony interface shall be through a field-installable plug-in card for direct connection to two standard (dial-up) phone line using two RJ-11 connectors for both telephone line inputs. Both telephone line interfaces may be used independently or together. Each telephony interface shall have a 250-3.6 kHz +/-3dB frequency response with 32ms echo cancellation span, 0 to 20 dB noise cancellation, and 30dB/second LEC convergence. The device shall have a built-in DTMF dialer, call progress tone detector, user-set "Entry" and "Exit" tones, and international telephone line support on both telephony inputs.

The Polycom SoundStructure TEL2 is specified.

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