Presenting Vortex® Installed Voice Products

Next Generation Voice Conferencing

Application Note

Polycom Installed
Voice Business Group

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INTRODUCTION

Welcome to Polycom's Vortex® product line, designed specifically for the challenges of installed room conferencing applications. The Vortex line currently comprises five products plus a number of accessories, and integrates easily with Polycom voice and video products for a comprehensive conferencing experience that will please integrators and end users alike.

This paper will cover the key features and capabilities of the following Vortex products:

- Vortex EF2280, a 12 input/12 output, 8 channel acoustic echo and noise canceller with built-in automatic microphone mixers, parametric EQ's, AGC's and matrix mixer.

- Vortex EF2241, an 8 input/8 output, 4 channel acoustic echo and noise canceller with built-in automatic microphone mixers, DSP based telephone interface, power amplifier, matrix mixer and processing functions.

- Vortex EF2211, a 3 input/3 output, single channel acoustic echo and noise canceller with DSP based telephone interface and power amplifier and processing functions.

- Vortex EF2210, a 3 input/3 output, single channel acoustic echo and noise canceller with power amplifier and processing functions.

- Vortex EF2201, a DSP based telephone hybrid for connection to PSTN (analog) telephone lines.

Typical applications for Vortex products include video and audio teleconferencing rooms, distance learning classrooms, boardroom audio systems, courtroom audio systems, and any facility where conferencing is required or may be necessary in the future.

All Vortex products include our Conference Composer™ software for Windows®, featuring the Polycom *Instant*Designer™ Vortex setup wizard. We invite you to download the Conference Composer software for your evaluation. You may find it on our website at www.polycom.com or e-mail us and we will send you a CD-ROM.

For a demo of the Vortex EF2280 or any of our other Vortex products, call the Polycom Installed Voice Business Group anytime at: 1-800-932-2774.

Polycom, Inc.
Installed Voice Business Group
9040 Roswell Road, Suite 450
Atlanta, GA 30350
Phone: 770.350.4140
Fax: 770.350.4142
vortex@polycom.com
www.polycom.com

1 VORTEX EF2280/EF2241 FEATURES AND FUNCTIONS

The Vortex EF2280 is a 12 input/12 output, 8 channel acoustic echo and noise canceller with built-in automatic microphone mixers, parametric EQ's, AGC's and matrix mixer. The Vortex EF2241 is an 8 input/8 output, 4 channel acoustic echo and noise canceller with automatic mixing, parametric EQ's, AGC's, matrix mixer, telephone hybrid and power amp. These products' key features are:

- Multiple channels of mic/line acoustic echo cancellation (AEC), noise cancellation (NC), and automatic gain control (AGC).
- 48 VDC phantom power on microphone inputs, software selectable.
- 40 dB/second convergence rate on each AEC channel.
- 270 msec tail time.
- 20dB headroom on all channels.
- 12 line level outputs on Vortex EF2280; 8 line level outputs on Vortex EF2241 (0dBu nominal level), user adjustable.
- Polycom patented noise cancellation algorithms remove steady-state noise signals on microphone inputs (and on telephone line output in Vortex EF2241); user adjustable up to 15 dB of noise reduction.
- 5-band parametric filters for input and output equalization on each of the inputs and outputs.
- Digitally controlled analog trim pots on A/D inputs for precise input level adjustment.
- 24-bit codecs, 32-bit floating-point processing.
- 48kHz sample rate $\Rightarrow$ 22 kHz bandwidth on all channels.
- RS-232 for interface to control systems and Conference Composer software.
- 24 configurable logic input pins and 20 configurable logic output status pins.
- High speed, low-delay EF expansion bus links Vortex devices.
- 32 user presets, 16 factory presets, 256 macros (each up to 256 commands).
- Scalable automatic microphone mixer – supports up to 64 microphones or can be arbitrarily split into 2 automixers per Vortex.
- Fully controllable 25x18 cross point main matrix with arbitrary gain on cross-points (1dB increments on level).
- Configurable cross point mix-minus interface from the expansion bus to the main matrix: select only the channels from the bus that you want!
- Conference Composer software for easy configuration, project management, and project documentation.
- *Instant*Designer setup wizard reduces configuration time from hours to minutes – its capabilities will amaze you!
- Front panel LCD for controlling and configuring the product without a computer.
- Internal signal generator for noise masking applications and calibration.

2 VORTEX EF2211/EF2210 FEATURES AND FUNCTIONS

- Single channel of mic/line acoustic echo cancellation (AEC), noise cancellation (NC), and automatic gain control (AGC).
- 48 VDC phantom power on microphone input, software selectable.
- 2 line level inputs with adjustable input level.
- 40 dB/second convergence rate on each AEC channel.
- 270 msec tail time.
- 20dB headroom on all channels.
- 3 line level outputs (0dBu nominal level), user adjustable.
- Polycom patented noise cancellation algorithms remove steady-state noise signals on mic/line input (and on telephone line output in Vortex EF2211); user adjustable up to 15 dB of noise reduction.
- 5-band parametric filters for input and output equalization on each of the inputs and outputs.
- Digitally controlled analog trim pots on A/D inputs for precise input level adjustment.
- 24-bit codecs, 32-bit floating-point processing.
- 48kHz sample rate ⇒ 22 kHz bandwidth on all channels.
- RS-232 for interface to control systems and Conference Composer software.
- 24 configurable logic input pins and 20 configurable logic output status pins.
- High speed, low-delay EF expansion bus links up to 8 Vortex devices with 4 channels per device plus AEC reference.
- 32 user presets, 16 factory presets, 256 macros (each up to 256 commands).
- Conference Composer software for easy configuration, project management, and project documentation.
- *Instant*Designer setup wizard reduces configuration time from hours to minutes – its capabilities will amaze you!
- Front panel LCD for controlling and configuring the product without a computer.
- Internal signal generator for noise masking applications and calibration.

3 VORTEX EF2201 FEATURES AND FUNCTIONS

- Echo cancellation 40 dB, total 60 dB, convergence rate 30dB/second.
- Line echo cancellation span 32 msec.
- 20dB headroom.
- Telephony signals are communicated with a Vortex EF2280 over the digital P-bus channel of the EF bus, eliminating the need for external analog inputs or outputs to/from the Vortex EF2280.
- Polycom patented noise cancellation algorithms remove steady-state noise signals on telephone line output; user adjustable up to 15 dB.
- Audible ring indication, dialtone, and DTMF signals (user adjustable levels).
- User configurable entry and exit tones.
- Up to 256 speed dial memories.
- Country-specific telephone configuration settings.
- Selectable auto hang-up via loop drop and call progress detection.
- RS-232 for interface to control systems and Conference Composer software.
- 24 configurable logic input pins and 20 configurable logic output status pins.
- Parametric EQ for tailoring sound of telephone line to user requirements.
- Conference Composer software for easy configuration, project management, and project documentation.
- *Instant*Designer setup wizard reduces configuration time from hours to minutes – its capabilities will amaze you!
- Front panel LCD for controlling and configuring the product without a computer.

4 VORTEX PRODUCTS: WHAT'S NEW

The Vortex incorporates the latest advances in audio technology for use in audio conferencing applications and more. The highlights of what makes the Vortex unique are described here.

Instant*Designer *v2.0

This powerful setup wizard can help you configure the Vortex products in your installation, including loudspeaker zoning, in a matter of minutes. Not only will you save time, you'll be assured that matrix settings, gains and other tricky components of your setup are accurate. This reduces the chance of service calls later and improves your profitability on the job.

Improved latency specifications

Total latency in a Vortex unit (mic/line inputs to outputs, processing enabled) does not exceed 13 ms. This results in more natural sounding audio and is better for sound reinforcement applications.

Interoperability with other Polycom products

The Vortex EF2280 and Vortex EF2241 are designed to work easily with the SoundStation VTX 1000™, Polycom's unique wideband conference phone. This product gives users a familiar interface to the phone line and easy control over levels and other system adjustments. In addition, the Vortex EF2280/EF2241 work seamlessly with the VSX™ 8000 video conferencing system. The VSX remote can be used to control the Vortex products.

Special product bundles are available that package the Vortex EF2280 or Vortex EF2241 with these Polycom voice and video products. Ask your Polycom representative for more information.

Noise Cancellation

Polycom owns two patents on our proprietary noise cancellation technology that can eliminate background acoustic noise such as HVAC, computer noise, projector noise, or any noise with stationary statistics. The noise cancellation not only removes noise, but also helps the acoustic echo canceller converge faster in noisy rooms. Vortex noise cancellation is user-adjustable from 0 to 15 dB on all mic/line inputs.

Speech Driven Automatic Gain Control

Each of the microphone/line inputs on the Vortex includes an automatic gain control that has been specifically developed for use with speech and audio signals. This AGC is controlled by a neural network and uses advanced signal analysis to determine when to change the gain of the input signal. Unlike some other products, you won't experience ramping on background noise with the Vortex! This is a sophisticated tool that we recommend should be enabled at all times. Try it and we think you will agree.

5 SOME QUESTIONS AND ANSWERS

Acoustic Echo Cancellation (AEC) Questions

What makes the Vortex AEC better?

The Vortex AEC has been designed to get the best trade-off between full-duplex audio (quick interaction of both sides of a conversation and having both sides hear each other) and removal of echo when both sides of the conversation are moving about their rooms while both sides are speaking. This is a measure of the doubletalk performance.

Why does the Vortex sound better?

There are several reasons including the full bandwidth of the signal – 22 kHz of bandwidth all the time, the digital trim pots that adjust the analog gain before the A/D converter – this maximizes the signal to noise ratio and removing the high frequency hiss you can hear with other devices. The noise cancellation also helps to bring down the ambient room noise so that the echo canceller can do a better job of adapting to the room acoustics. The AGC also helps the AEC performance by bringing all the signals to the proper level.

What is room gain and why is it important?

Room gain is the difference in level (in dB) between the outgoing level from the room device to the incoming level to the room device. In Figure 1, you can see which two points are being used to measure this difference. Factors that affect room gain are the amount of amplification in the loudspeaker, the amount of gain applied to the microphone inputs, and how closely coupled the microphones and loudspeakers are.

Room gain is important because the acoustic echo canceller compares the signal that is sent to the amplifier/loudspeaker to the signal picked up by the microphones. If the signal picked up by the microphones is much larger than the signal sent to the amplifier/loudspeaker (which is the case if the room gain is positive), then the AEC has a tougher time removing the echo and may become less full duplex.

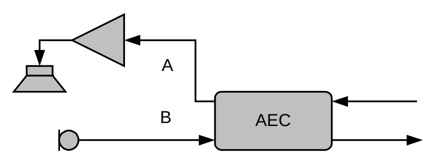


Figure 1. Room gain is measured from output A to input B as $20\log_{10}(B/A)$.

How do I know what my room gain is?

The Conference Composer software has a built in mode for determining room gain and can display the room gain just like signal meter information. Go to the room diagnostics page within Conference Composer and select the Levels button. On the room diagnostics page you will see the room gain for each microphone and you will also see the amount of gain (or attenuation) the AGC is applying to each inputs 1 to 8. The room gain measurement is most accurate when the local talkers are quiet and only the remote site is talking.

Is the Vortex easier to setup than other products?

Yes. Our *Instant*Designer wizard beats everything else in terms of speed and accuracy. Even if you opt not to use the *Instant*Designer, it's still easy to set up a Vortex product. Setting the Vortex input channel gains and the calibration process is easy because of the room diagnostics page within Conference Composer. You can set input gains more quickly and get immediate visual feedback as to whether your input gains are set too high or too low. If the AGC is continually pulling the signal up by 10dB (as indicated by the AGC level meter) then your input gains should be adjusted up by 10dB. If the AGC is continually pulling the input channel down by 8dB then your input gain should be adjusted down by 8dB. It is that easy.

What is convergence speed and why is it important?

Convergence speed is how fast the AEC can adapt and "lock" onto to the local echo and effectively begin removing it. You can think of convergence speed as the rate at which audio from the remote site played into the local room will be attenuated by the AEC before being sent back to the remote site. This is related to convergence time, which is the amount of time it takes the AEC to reduce the echo to some level. With most echo cancellers, before the AEC locks into the echo, there is some amount of clipping that happens to the out going audio to prevent an echo from being sent back to the remote side. This clipping makes it difficult to have a natural full-duplex conversation. Because participants in a room are constantly moving, or the microphones are being moved or muted, or the volume is being changed, the echo path is always changing. Faster convergence means the AEC can lock onto changes in the room faster and sound better faster.

Why does the Vortex's AEC converge faster than everyone else?

It is because we spent more than eight years in the development of our AEC algorithm, and we are continuing to work on it. We have learned a great deal over this time frame. By tightly integrating the AEC with the noise canceller we can make better decisions about when to adapt the AEC and when to not adapt. Adapting at the wrong time is worse than not adapting at all because it is possible to make the echo worse if the AEC trains on the wrong signals, i.e., if it trains on the local talker's audio, mistaking it for an echo from the remote side.

What is tail time and why is a longer tail time important?

Tail time is the maximum length of time that an echo may bounce around the local room and still have the AEC recognize it as an echo of the remote talker's speech and remove it. Echoes that are longer than the tail time (because of a large room, room with hard surfaces, etc.) will not get recognized by the AEC and will not be removed. The remote talkers will always hear a delayed version of their audio if the echo length of the local room is larger than the tail time of the local echo canceller. A longer tail time is important because it means the echo canceller can handle larger rooms or rooms with hard surfaces.

Noise Cancellation Questions

How does the Vortex's noise canceller work?

The Vortex's noise cancellation system works by looking at the background noise between spoken words and estimating the noise statistics. Once the system has determined the characteristics of the noise, it then removes the noise from the input signal. Any subsequent processing, such as in the automixer and matrixer, work with the noise-reduced version of the signal. The adaptation to the noise happens continuously, so as the background noise changes, such as HVAC is turned on and off, the system adapts nearly instantaneously to the new noise characteristics.

How is the Vortex's noise canceller different from other products that say they reduce noise?

Many products say they remove noise by filtering. When this happens, they will also remove any signal in the same frequency band as the noise. The Vortex noise canceller is different from other products because it uses an adaptive frequency selective algorithm to remove the noise only where there is noise and to leave the desired signals (speech, audio, etc.) alone.

What type of noise does it work on?

The Vortex noise canceller works best with steady state signals, i.e., signals that are random, but their randomness can be thought of as fixed. Examples of this are white or pink noise – the signal is random but the type of randomness is known, periodic signals such as tones – the signal is steady state, and crowd noise – a large collection of voices, when mixed together, sound like random background noise. Certainly combinations of these types of signals can also be effectively removed such as the noise from power drills – there is a combination of periodic signals and random noise. The types of noise that the Vortex noise canceller cannot remove include short impulsive noises, like a hammer blow, which are not present over a long enough time window to have the algorithm adapt and remove it.

Automatic Gain Control Questions

How does the AGC work?

The Vortex's AGC operates by looking at the input signal and making a determination as to whether the signal is a valid speech input or not. It does this by analyzing the signal looking for the speech-like components. Once it finds valid speech (and it is always looking for valid speech), it will estimate the nominal level of the incoming signal. The AGC will then determine what level to apply to the signal to get it to a 0dBu nominal level. If the signal is too loud then it will bring it down. If the signal is too quiet, then it will bring it up. If the signal is so quiet that the automixer would not gate the microphone on, then the AGC leaves the signal alone.

How is the Vortex's AGC different from other AGC's?

The Vortex's AGC algorithm uses a neural network to help make decisions about what is speech and what is not speech. A neural network is an artificial intelligence concept that is a form of nonlinear adaptive filter with a built-in state machine. The neural net helps make a much better judgment on whether the input data is speech-like every couple of hundred milliseconds. Other AGC algorithms may simply use the envelope of the signal to determine if the signal should be increased or decreased. This has the unintended consequence of training on background noise and making the signal too loud if there is prolonged silence.

How do we know when something is speech?

The Vortex AGC works in conjunction with the information available from the acoustic echo canceller, the noise canceller, and the automatic microphone mixer. The acoustic echo canceller determines which side of a conversation is talking: the remote side, the local side, both sides, or no one. This can help the AGC determine whether it should be adapting or not. The noise canceller, together with the neural network, determines whether a signal is speech or noise. If these systems determine that the input is noise, the AGC will not try to adjust the level of the signal. If the automixer does not gate on for a particular channel, then the AGC will leave that alone because it will treat the signal as a side conversation that does not need to be increased in level and heard by the remote participants.

When will the AGC not work?

The AGC will not change the level of the incoming signal if the signal was not strong enough to gate on a microphone or the automixer is configured so that the microphone cannot gate on – for example the priority of that microphone is too low and this microphone gating on would exceed the maximum number of open microphones that can be open for this mixer, or chairman mode is on and this microphone is not the chairman. If a conference participant mutes his/her microphone, the AGC will not turn up his/her speech in an adjacent microphone.

Can I control the parameters?

Conference Composer will allow the user to change the maximum amount of gain that can be applied to the input signal. Currently the factory defaults are to apply no more than 15dB of input gain and as much as 15dB of attenuation.

How fast does the Vortex's AGC work?

The AGC is able to bring signals up in level at a rate of up to 5dB per second and down in level at a rate of up to 25dB per second. This rate is adjustable if you want to limit the speed of the AGC from 1dB per second up to 5dB per second.

Should I leave the AGC on all the time?

Yes. We have done extensive testing of the AGC and it can dramatically improve the quality of an audio conference. We recognize that your experience from other AGC's may not be that favorable, but we definitely suggest you leave the Vortex's AGC on all the time because it is very robust. When *Instant*Designer creates the design, it will enable the input AGC will configure it with a +/- 3 dB gain range by default.

Automixer Questions

What style is the Vortex's automixer?

The automixer (or more accurately, both automixers) in the Vortex use a gating style of automixing. The automixers make a determination of which signals should be gated on and this decision gets translated into a decision of what gain to apply to the input signal as it gets mixed with other signals to create an output signal in the matrix. The automixer takes in up to 8 microphone/line inputs and produces 8 line output signals and gain information to be applied to each of these 8 gated outputs as they are used in the matrix. The matrix does the actual mixing of the signals, but the automixer determines the gains for the signals. If you select the gated version of an input at a cross point in the matrix then you will get the signal scaled by the gain that the automixer has determined should be applied to that signal.

What parameters can you control?

There are a number of parameters to control and they are categorized as global parameters, mixer parameters (remember there are two automixers), and channel parameters. The global parameters are hold time, decay time. The mixer parameters are maximum number of open microphones, last mic mode, chairman microphone mode, and the amount of off attenuation. The channel parameters are which mixer the input is assigned to, the gate ratio, adaptive or fixed threshold, manual threshold, and the gate priority. Details of these options are given in Section 9: Automatic Microphone Mixing.

What is different about it?

The noise canceller helps remove noise so the gating decisions can be made more intelligently. The speech detector in the noise canceller also helps to determine whether the channel should gate on.

What about NOM attenuation, can I use it over 8 units?

NOM stands for the number of open microphones and NOM attenuation means the attenuation applied to a sum of microphone signals to compensate for having some number of open microphones. With the Vortex, NOM attenuation can be applied to each output signal that is made up of sums of the input microphones. For instance if an output contains microphones 1-8 which have been automixed and 3 of these microphones are currently open, then an attenuation of $10\log_{10}(3) = 4.77$ d**B** can automatically be applied to the sum of these microphones before they are summed with the other signals that make up that output.

Now, if you have multiple Vortexes, information about the number of open microphones can be sent to other devices by using the W, X, Y or Z busses. This means that an output signal made up of microphone signals from linked devices will have the appropriate NOM attenuation applied to the output signal: $10\log_{10}(NOM)$ where NOM is the number of open microphones from all the linked devices being summed together to form that output. So yes, NOM attenuation can be applied across all 64 available microphone channels.

Matrix Questions

Why does a 12-input/12-output device have a 27x18 matrix?

There are 12 physical analog inputs into the Vortex EF2280 and there are an additional 14 digital inputs that can come from the high speed digital EF Bus. Finally the signal generator (white noise, pink noise, or sinusoids) can be mixed into any output. This accounts for the 27 inputs to the matrix. On the output side there are 12 physical analog line level output signals, 4 digital bus signals (W, X, Y, and Z), and 2 echo cancellation reference signals that are used internally for the acoustic echo cancellers. This accounts for the 18 outputs from the matrix.

Why cross point gains?

We wanted the Vortex to be as flexible as possible and felt that a simple router instead of a matrix was too limiting. Cross point gains allow you to map any input to any output with arbitrary gain. An input can be used in many different outputs with different gains each time the signal is used.

What can I do with them?

With arbitrary cross point gains you can do much more interesting conferencing with sound reinforcement designs. Also, as we'll see with the sub-matrix, the cross points in the sub-matrix allow you to very easily share resources such as codecs and other signals across all linked Vortexes.

Why is that different?

Previous products gave you no choice in how inputs were mapped to outputs. You took the full strength audio or possibly mapped full strength inputs into a sub bus and then you could use the sub bus. That technique forced you to think differently – why not just map the inputs to the outputs with whatever gain you want? This is what the Vortex allows you to do.

Can I use all inputs and outputs?

Yes. All microphone/line inputs, the A, B, C, and D inputs, the gated microphone 1-8 signals, the 14 bus signals and the signal generator can be used as inputs. Any of these signals can be mapped to any output with an arbitrary cross point gain. The exception to this is that the expansion bus inputs cannot be mapped to the same expansion bus output, i.e., the X bus input cannot be mapped to the X bus output.

How do I change a matrix cross point?

The easiest way is to use the Conference Composer software and left click on the matrix cell you would like to change. To do this manually with an RS-232 command, you would simply use the MGAIN command with the row and column of the cross point you would like to change and then specify the value. Assuming a Vortex EF2280 has a device ID of 0, the following command sets the cross point of input 5 to output A to –6dB: F00MGAIN5,A,-6. That's all there is to setting cross point gain values. You can increment (or decrement) the cross point gain with the MGAIN command too. F00MGAIN5,A,>3 increments the cross point gain by 3dB. (<3 decrements the gain by 3dB).

Processing Questions

What processing is available?

The processing in the Vortex includes the NOM attenuation on all 12 outputs, the acoustic echo cancellation, the noise cancellation, automatic gain control, 5 parametric filters for each of the inputs and outpus, and adjustable delay on the output channels. The parametric filters include highpass, lowpass, notch/boost, high shelf, low shelf filters, and Linkwitz-Riley lowpass and highpass.

How do I access it?

The processing is most easily accessed via RS-232 commands. Conference Composer simplifies the process of enabling the processing and configuring the necessary parameters. The user can also design the filters by setting the appropriate frequencies, bandwidths, and attenuation/gain values.

Where can I apply it?

The input equalization can be used to compensate for varying characteristics of microphones or input room conditions. The AGC can help compensate for loud or soft talkers. The noise cancellation can help remove background noise.

What is a parametric EQ? Why would I use it? How do I use it?

Parametric equalization is a technique for applying filters where you need them with arbitrary gains and center frequencies. This differs from graphic equalizers where the filters have fixed bandwidth and center frequencies. You might use this to compensate for different microphone frequency characteristics, to compensate for some background noise such as a 60Hz hum, or to remove some of the room effects.

When would I use a high/low pass filter? How does it work?

You might use a highpass filter to remove some low frequencies such as some boominess in a room. A lowpass filter could be used to remove high frequency background noise (as well as any desired signal in the same frequency band). When using *Instant*Designer, a 80 Hz highpass filter will be enabled on each microphone input signal to reduce the room boominess.

Bussing Questions

The EF Bus is used to send high-speed data and control between Vortexes devices.

How does the bussing work?

Within the EF bus there are five busses: the P, W, X, Y, and Z busses. Each of these busses contains up to 8 signals (one from each linked Vortex). Each Vortex can place an output signal on each of the P, W, X, Y, and Z busses. The P bus output is only found on the telephony devices and is typically used to transmit the phone signal ('P' for phone) between devices. Since there can be 8 Vortexes linked, this means there are a total of 32 signals on the EF bus at any given time and each linked Vortex can get any of these signals off the bus. With the addition of the P bus on the telephony products, up to 8 EF2280's and 8 EF2201's can be linked together for a total of 16 devices. The W, X, Y, and Z busses include NOM information and are most often used to link microphones between devices, but can also be used to share line level sources between devices.

What is the sub-matrix? And how many are there?

The sub-matrix is a smaller matrix that feeds the main matrix. The inputs to the sub-matrix are the bus signals from all the other devices that are linked. In the Vortex there are five sub-matrices, one for each of the P, W, X, Y, and Z busses.

Each sub-matrix has 7 inputs (the maximum number of devices that can be linked to any given device). Since a device's bus cannot be linked to itself, there are only 7 inputs, not 8 to each sub-matrix. In the Vortex, each sub-matrix for the W, X, Y, and Z busses has 3 outputs and the sub-matrix for the P bus has 2 outputs, which become inputs to the main matrix.

Any of the three outputs from each sub-matrix can have arbitrary cross points applied to them so you can weight the signals from other devices to enable arbitrary zoning or mute some of them to extract just the signals from the bus that you want.

If you are more comfortable using the bus as a simple mix-minus of all the other bus signals on the particular bus (W, X, Y, or Z) then any of the three outputs from each sub-matrix (one for W, X, Y, and Z) can be configured as a mix-minus of the other linked device's entire W, X, Y, and Z expansion bus signals, respectively.

Why is it different from the main matrix?

The five sub-matrices make the main matrix more manageable. Essentially the bus signals from other devices are mixed down to 12 main matrix inputs in the sub-matrix. Each of the 14 outputs from the sub-matrix (3 for each of the W, X, Y, Z busses and 2 for the P bus) can have arbitrary cross point gains or mutes before combining the channels for presentation to the main matrix.

How many channels can I use?

Each Vortex can map its inputs to the P, W, X, Y, and Z expansion busses just like it would create any other matrix output signal. These five outputs are available to all the other linked devices as inputs to the sub-matrix.

How do I access them?

If you want to use the linked signals individually or with different gains, you should access the signals from the sub-matrix. If just want a mix-minus of all the other devices, you can configure any of the sub-matrix outputs to be the sum of all the other devices excluding the local device.

Can I share codecs and program audio sources easily through the busses?

Yes. Any input to a Vortex can be shared with other linked Vortexes by putting the signal on the expansion bus and then any other Vortex can get that signal off the bus by muting all the other sub-matrix cross points except the one that it wants.

How far can the bussing be run?

Because the bus is very high speed, it is required to keep the length of each cable segment to less than 14 ft (4 m).

RS-232 Control Questions

Will RS-232 work with AMX and Crestron?

Yes. Absolutely. Polycom has provided all the programming information to AMX & Crestron and macros are available.

What can I control via RS-232?

Absolutely everything. Every feature in the Vortex has an RS-232 command and documentation as to how to control the feature. The Vortex has a very readable and friendly command set. Section 12 lists the Vortex RS-232 commands.

How does the Vortex work across multiple Vortex devices?

A linked network of Vortexes uses a single serial port on the host PC or controller. The remaining Vortexes receive their commands via the EF Bus. This way only one Vortex needs to be connected to the control system and the remaining Vortexes are linked in a daisy chain fashion. However, if you would like to connect multiple controllers to multiple linked Vortexes, you can do that too. Each Vortex can be connected to its own controller (although in most applications you would only need one controller to control all the linked devices). See the application note Interfacing to Multiple Vortex Devices via Multiple RS-232 ports for more information.

Does the Vortex give mixer gating information to the controller? How?

Yes. With a single RS-232 command (GATEEN1 for GATE ENable On or CGATEEN1 for Camera GATE Enable On), the Vortex will send event-driven gating status commands to the main controller. As the gating status of microphones change, the Vortex will automatically send out a status message. Another command can be used to turn off this flow of information. Gating information can also be mapped to the logic outputs, but it is usually simpler to have the Vortex continuously sending the data via RS-232 to the control system (no polling is necessary). The Camera gating feature allows you to setup an amount of time (up to 5 seconds) that the microphone must be gated on before a camera gating indicator will be sent. This information can be used for camera positioning.

What about IP addressing?

Polycom has an IP accessory, the Vortex Ethernet Interface, that can be used to configure the Vortex devices remotely.

Logic Control Questions

How do you address logic pins? How many controls can you have?

The product has 24 logic input pins and 20 logic output status pins. The logic pins are most easily accessible through the Conference Composer software. Logic input pins can be programmed to execute a preset, a macro, or an individual command on either closure, opening, or held open/closed.

What about volume control?

Volume control can be done either by mapping a single level increment command to the logic pin, or multiple levels (like a stereo volume) can be changed by creating a macro and assigning the macro to a logic input pin. The product supports output gain control, input gain control, and matrix cross point control with simple RS-232 commands.

Is it re-configurable?

Absolutely. You can change pin assignments and functions very easily.

What are macros? Why would I need them?

Macros are a collection of up to 256 Vortex commands that can be executed with a single RS-232 command or contact closure. You might use macros to ramp stereo volume, customize a preset, do sound masking, send commands to a video codec for camera positioning, or anything else you can think of.

Can I combine multiple logic pins into a group to handle room combining applications?

Yes. The product supports 4 groups of pins where each group can have up to 5 logic pins assigned (32 combinations). The groups can be used in room combining applications to select different presets or macros to execute based on the combinations of input pins.

Can I get gating information from groups of mics on a single logic output pin?

Yes. You can assign the gating status of a single input or groups of inputs to a single logic pin. When assigning a group of inputs (such as microphones 1, 2, and 6), you can choose whether you want the logic output status to be TRUE if ALL microphones are gating (1 and 2 and 6 in this example), or if ANY of the microphones are gating (1 or 2 or 6 in this example), or if ALL microphones are NOT gating or if ANY of the microphones are NOT gating. The syntax for the command to assign status information to output pins is very easy to understand.

How can I configure the Vortex to automatically "duck" outputs based on microphone gating activity?

You can easily setup a macro to duck particular outputs based on gating activity of a collection of microphones. By connecting a logic output pin (configured to show gating information for the particular microphones of interest) to a logic input (tied to the ducking macro), you can duck outputs every time the collection of microphones gates on and "un-duck" when the inputs gate off.

Preset Questions

How many presets are in the Vortex?

There are 16 factory presets and 32 user presets for a total of 48 presets. The presets store all the information in the device including input gains, matrix cross points. Also, each device stores a complete set of text labels of inputs, outputs, macros, presets, and logic pins.

How do the factory presets work? How are they set up?

The factory presets are stored in an area within the Vortex where they cannot be overwritten by a user. They are provided to make it easy to get started using the Vortex. The factory presets are common setups that may be sufficient for many different applications. The factory presets can be used as the basis for user presets, and then modified and saved.

Does the Vortex come up as a "blank slate"?

The Vortex as shipped from the factory comes up in the default configuration of preset 0 and the power on preset is set to 0. The Vortex can be configured to power up with any of the factory or user presets and will not come up blank unless you save a blank preset and have it power up to that preset.

How do I select which preset the device powers up into?

On the presets page within Conference Composer you can select which preset to use when powering up. This can also be set from the LCD and with RS-232 commands. The default from the factory is to power up into factory preset 0.

What if I make changes? Will the preset still be there?

Presets are stored in non-volatile memory. Once loaded into the Vortex, changes can be made to the settings of the Vortex. However, unless the changes are saved as a preset (either a new preset or overwriting the existing preset), the old preset values will be used the next time the device is powered up or reset. The changes are not saved to the non-volatile memory unless you tell the settings to be saved.

Can I password protect presets?

Yes. There is an optional write-protect password (in addition to the front panel lock out password) that can be used to prevent users from overwriting presets within the Vortex. When the Vortex is write-protected, the user can change which preset is running in the device, but cannot save a new preset and cannot change the preset selected upon power-up.

Software Questions

What type of computer do I need?

The minimum computer configuration for running Conference Composer is a 500 MHz Pentium computer with 128MB of memory and 50MB of free disk space. The recommended system is a 700MHz or better Pentium with 256MB of memory and 50MB of available disk space. Windows 2000 or Windows XP is recommended.

Can I work offline?

Yes. With Conference Composer you can work offline, design the system, and save files to disk. You can easily restore files and upload information into a product when you decide to go online.

What do I need do do to set up my Vortex unit for operation?

In many cases, you don't need to do anything – a Vortex preset will work fine for your application. If you are dealing with multiple microphones, different types of microphones, multiple loudspeakers and other equipment, we recommend that you use the *Instant*Designer software wizard. When you first open Conference Composer, you will be given the choice to create a project with *Instant*Designer. This easy-to-use tool takes you through a set of pull-down menus in which you select the number and types of microphones, amplifiers, and other equipment; *Instant*Designer then automatically generates a Conference Composer system file. It will determine your matrix settings, gain structures, etc. If you need to further adjust settings, our Conference Composer software will allow you to change anything in your system configuration.

What parameters can I have access to in Conference Composer?

Conference Composer makes every parameter in the Vortex accessible. Conference Composer is organized into tabs where each tab shows the controls for that particular function. There are tabs for the device inputs, the device outputs, the automixer, matrixer, the logic inputs and outputs, presets, macros, room diagnostics, system, and the global settings of the device.

What about the labels? How do they work?

Text labels can be entered for all the input and output signals, the logic pin names, and macro and preset names. Each of these labels are stored within the Vortex and can be up to 16 characters in length.

What are the different levels on the left side of the Conference Composer software?

Conference Composer organizes your projects into a hierarchy of Workspace, Project, Device Chain, and Device. Starting from the bottom of the list, a device is simply that – some piece of equipment that will be used in the installation. A Device Chain is a collection of devices that are linked together through the EF Bus (that's where the name device *chain* comes from). Typically these represent all the devices that are tied to a particular serial port on your computer/controller. The project represents a collection of device chains while the workspace allows you to store a collection of projects. You can keep multiple independent projects in your workspace. You can think of the workspace as your hard drive, the projects as directories, the device chains as files and the devices as items in the files.

Will Conference Composer automatically get information from a Vortex?

You can choose to synchronize to a connected Vortex or collection of Vortex devices. This will cause Conference Composer to go out and have all the products send their information to the PC. As you change values on the PC, the corresponding changes take place in real-time on the connected devices.

Can I print my saved information?

Yes. There is a way to create a report from the configurations on your devices. Conference Composer makes it easy to print reports.

Can I copy information in the software? To another Vortex?

Yes. You can copy information from one device to another device in the Conference Composer Navigator panel by simply clicking and dragging the device into a second device or dragging any of the settings such as matrix settings or macro settings from one device to another.

General Questions

Can the Vortex be used with ceiling microphones?

Yes. The combination of the Vortex's noise cancellation (to reduce the amount of noise picked up by the microphones), the acoustic echo cancellation (the ability to handle positive room gain), and the AGC (to increase the signal level) allows you to use ceiling microphones and loudspeakers and have your audio conferencing system work well. While we can't break the laws of physics, we can at least make ceiling microphones work the way you would like them too.

How do I keep someone from messing up the system?

There are two levels of passwords with the Vortex. The first is front panel lockout. If you enable the front panel lockout then your users will be able to view the settings but not change any values from the front panel. The second level of security is the write-protect password (also referred to as the non-volatile memory lock). If the Vortex is write-protected, users will be able to change settings and change which preset is currently loaded, but they will not be able to overwrite presets, or macros, and will not be able to change which preset the Vortex powers up into.

Can multiple Vortex products be linked?

Yes, you can link up to eight Vortex devices in any combination with one exception. If you are using multiple Vortex EF2201 telephone hybrids, up to eight of them can be linked to up to eight Vortex EF2280 units. The reason is that the Vortex EF2280 and Vortex EF2201 can share the same device ID number.

6 SYSTEM BLOCK DIAGRAMS

While the Vortex performs a large variety of complex tasks, the control and implementation of these tasks is very simple to the user. The simplicity comes from the Conference Composer Windows™ software and the elegant design of the Vortex hardware. The power of the digital signal processors has allowed our engineers to design an echo canceller that has full bandwidth – not limited to only 7kHz or even 15kHz. With the Vortex, conferencing can now sound as good as the face-to-face experience. By designing the system this way, all the program audio and other sound reinforcement audio will have 22kHz audio bandwidth. This opens up the possibility of full bandwidth audio conferencing over broadband connections.

In its simplest form, the Vortex is an automatic microphone mixer with full cross point matrix. All the microphone parameters can be customized and any input can be mapped to any output with an arbitrary gain applied to the cross point. Adjustments in levels, matrix cross points, and other functions can be made in one of three ways: RS-232 control via the Conference Composer software or a dedicated control system, the logic input pins, and the front panel buttons and LCD.

In reality, the product is much more than an automatic microphone mixer, it includes automatic gain control for automatically leveling the microphone input signals, noise cancellation for reducing the amount of ambient noise picked up by the microphones, acoustic echo cancellation for removing the echo of the remote talkers voice from being sent back to them, and many other features that are detailed in the block diagram in Figure 2 and Figure 4.

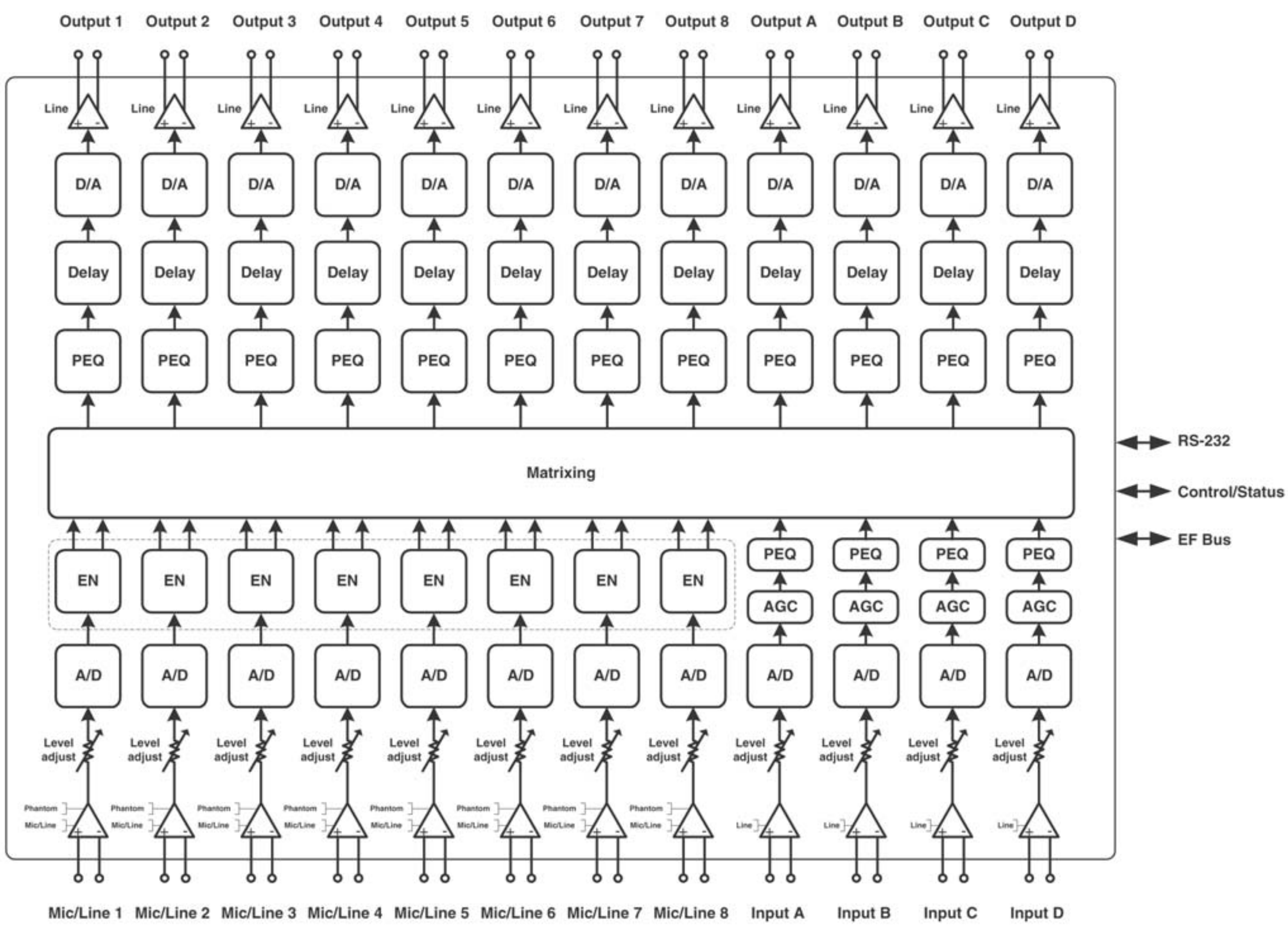


*Figure 2. Block diagram of the **Vortex**.*

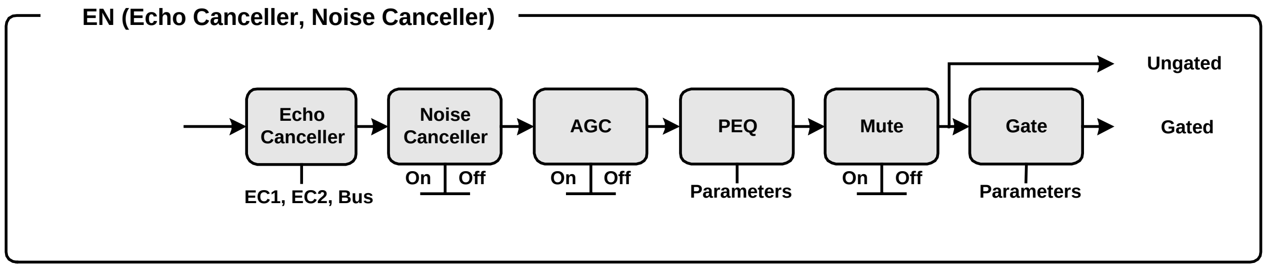
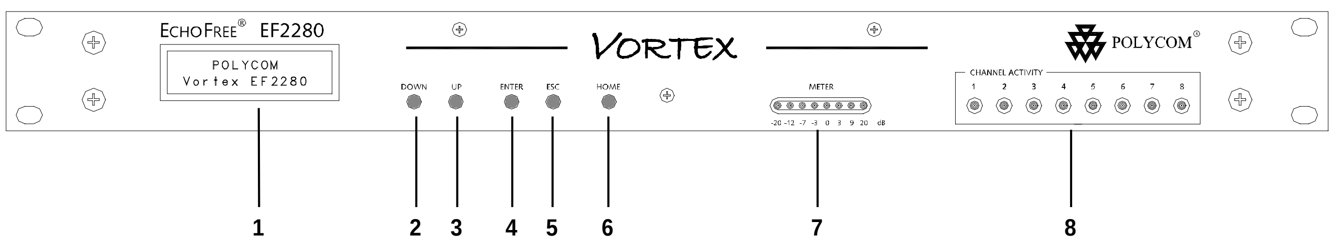


Figure 4. The channel processors for each of the 1-8 mic/line inputs.

Each of the 8 mic/line inputs has the EN processing block that is shown in Figure 3. The balanced input signal passes through a digitally controlled, analog gain stage, is converted to a digital signal and then processed by the acoustic echo canceller, the noise canceller, the automatic gain control, the parametric equalizer, and then the automatic mixer. The details of the processing are described in the next section.

Vortex ***Front and Rear Panels***



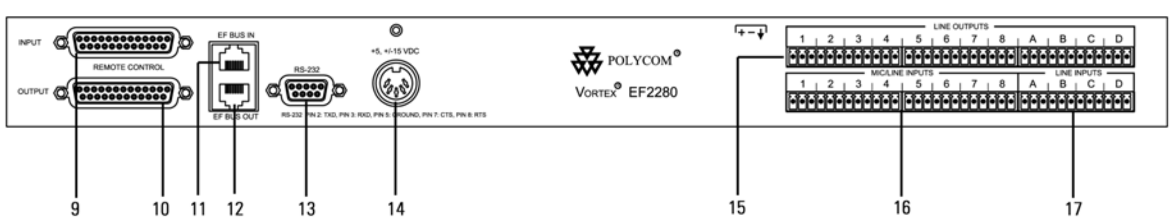


Figure 6. The product front and rear panels.

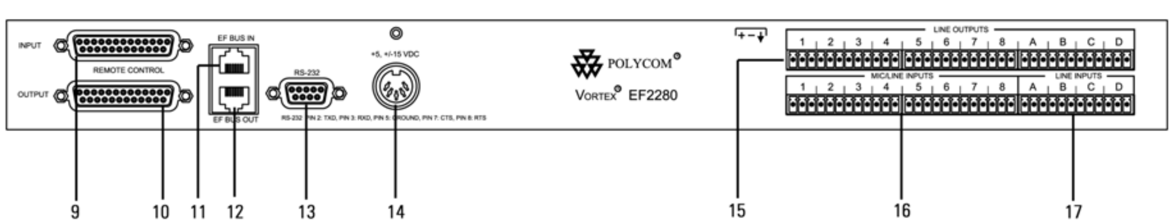


Figure 6 shows the front and rear panels of the Vortex along with numeric pointers for understanding what the different connectors are. Rack ears are built into the Vortex enclosure design.

The features of the Vortex are described below:

1. LCD Display. Displays menu instructions for configuration and operation of the Vortex.

2. Down Button. Scrolls backward through menu items at a particular level or decreases the value of a parameter.
3. Up Button. Scrolls forward through menu items at a particular level or increases the value of a parameter.
4. Enter. Enters the menu and allows you to select and change parameter values.
5. Esc. Returns to the next highest level of menus.
6. Home. Returns to the top of the menu structure.
7. Level Indicator. Indicates the level of the selected channel or parameter.
8. Channel Activity LED's. Indicates gating activity of the 8 mic/line channel inputs.
9. Input Logic Port. Parallel logic input.
10. Output Logic Port. Parallel logic output.
11. EF Bus In. Connects to EF Bus Out of another Vortex.
12. EF Bus Out. Connects to the EF Bus In of another Vortex.
13. RS-232 Serial Port. Connect this to an optional RS-232 remote control device, such as a touch panel or personal computer COM port.
14. Power Supply Input. Connects to the external power supply provided with the Vortex.
15. Line Outputs. Outputs 1 – 8 and A - D at line level.
16. Mic/Line Inputs. Connects to microphone at either mic or line level, with or without phantom power..
17. Line Outputs. Outputs 1-8 at line level.

7 VORTEX EF2280 INPUTS AND OUTPUTS

The Vortex EF2280 has 12 analog inputs: 8 mic/line inputs and 4 line inputs, and 12 line-level analog outputs as shown in Figure 2. All inputs and outputs are actively balanced.

Input level control (before the A/D converter) is executed in the analog domain using digitally controlled analog trim pots. This approach was taken to maximize the signal to noise ratio of the system – the A/D always gets the proper level to create the best sounding digital signal that is possible. The microphone switch provides 33 dB of gain and the software controlled analog level adjust can provide up to an additional 30 dB of gain on channels 1-8.

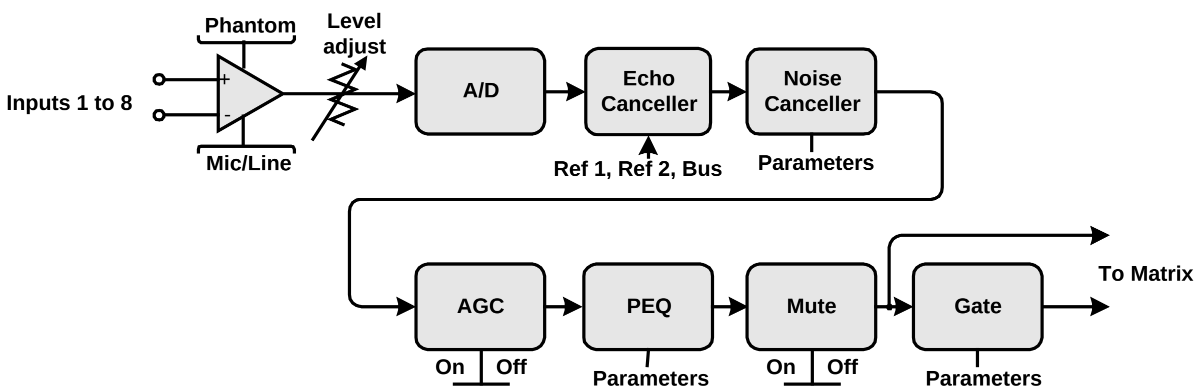


Figure 8. The input signal chain for inputs 1 to 8.

Mic/Line Inputs 1-8

Balanced audio appears at the rear panel mini Phoenix™ connector (3.5mm spacing). Microphone or line level is selected and phantom power can be enabled or disabled. As shown in Figure 8 the level adjustment to match the nominal level of the microphones is performed in the analog domain – this ensures that the signal to noise ratio of the signal is preserved. Once the proper gain is set for the input channel (this step is done in calibration), the signal is ready to be converted to the digital domain through the A/D converter.

Once the input signal is converted to a 24-bit digital signal, it is converted to a 32-bit floating-point number and presented to the processing blocks. When viewing the input signal meter on the Vortex front panel or through the Conference Composer software, the input meter shows the signal strength just after the input channel has been converted to a digital signal (right after the A/D converter). The Vortex uses floating-point arithmetic to maintain the highest signal to noise ratio possible through the processing.

The first processing block is the acoustic echo canceller (AEC) which, if enabled, will eliminate the acoustical coupling of the remote talkers' speech from being coupled back into the microphone and sent back to the remote talkers. The AEC can use one of three reference signals: Ref 1, Ref 2, or the EF Bus external reference. Ref 1 and Ref 2 can be built from any of the input signal by using the main matrix. The AEC reference should contain the appropriate mix of remote talker signals (such as the codec audio and phone add audio) with the local program audio. Local sound reinforcement should not be added into the reference. Integrated within the AEC is the noise cancellation processing which will analyze the signal and remove steady state random noise from the signal. The user can select various levels of noise cancellation from 0 to 15 dB with 10dB the default value.

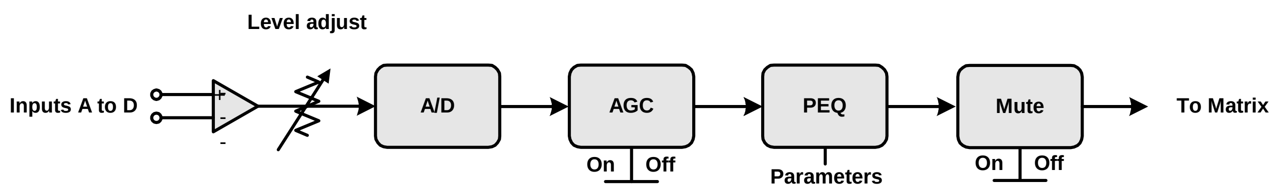


Figure 9. The input signal chain for inputs A-D.

After the AEC and Noise cancellation, the signal passes through a neural network controlled automatic gain control system. The role of the AGC is to increase the level of a low signal and decrease the level of a strong signal. By using the information about the signal from the AEC and NC components, the AGC can very effectively adapt the signal level to match the target level in the device. The neural network enables the AGC to only adapt when it should adapt, e.g., when there is a valid speech signal present. Because the AEC knows what the AGC is doing to the signal, the AEC operates very effectively with the gain adjustments of the AGC.

Following the AGC, the user has control of 5 bands of filters to equalize the input signal. The types of filters include Parametric, High Pass, Low Pass, High Shelf, Low Shelf, and Linkwitz-Riley Low Pass and High Pass. The user can cascade up to five of these filters (any combination) to perform input equalization. The parameters that control the filters are the cut-off frequencies, center frequencies, and the amount of gain or attenuation specified. These filters can be used to eliminate unwanted parts of the signal such as removing the boominess or a room (using a high pass filter). When different microphones are used in an installation, the equalizer can make all the microphones have the same frequency response so they sound the same.

The final step in the signal path before the matrix is the gating automatic microphone mixer. The automixer makes decisions about which channels should be gated on and which ones should be gated off (i.e., off attenuation applied). If a particular microphone is muted, the muting will be applied after gating decisions have been made – this way muting a particular microphone does not simply cause the next closest microphone to gate on. After automixing, both the gated and non-gated signals are available for use in the full cross point matrix. See Section 9: Automatic Microphone Mixing for more information about the automatic microphone mixers.

The system has been designed to have 20dB of headroom. This means that the input levels should not exceed +20dBu to avoid clipping. This results in a dynamic range of over 90dB and a passband from 20Hz to 22kHz. (Trivia buffs: That's where the 22 in EF2280 comes from!)

Line Inputs A-D

As with the microphone/line level inputs, balanced audio appears at the rear panel mini Phoenix™ connector (3.5mm spacing). As shown in Figure 9 the signal level is adjusted to match the nominal level of the line level input in the analog domain – this ensures that the signal to noise ratio of the signal is preserved. Once the proper gain is set for the input channel, the signal is ready to be converted to the digital domain through the A/D converter.

Once the input signal is converted through the 24-bit A/D converter, it is converted to a 32-bit floating-point number where it can optionally be muted and then sent to the matrix. The Vortex uses floating-point arithmetic to maintain the highest signal to noise ratio as possible through the processing.

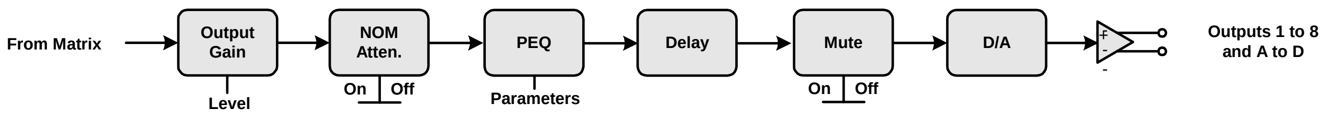


Figure 10. The signal chain feeding the output.

Line Outputs 1-8, A-D

All 12 outputs on the EF2280 (or 8 outputs on the EF2241) operate in an identical fashion. The line level output signals appear at the rear panel mini Phoenix™ connector (3.5mm spacing). As shown in Figure 10 the output processing consists of an arbitrary user gain or attenuation, NOM attenuation, followed by the output filtering, signal delay, and signal mute and then conversion from a digital signal to a balanced analog audio signal. When viewing the output signal meter on the Vortex front panel or via the Conference Composer software, the output meter shows the signal level just prior to conversion to an analog signal at the D/A converter.

The user can apply up to 20 dB of gain and as much as 100 dB of attenuation (the range is +20 to -100). This allows the user to interface to external equipment that may not support a 0dBu nominal level.

The NOM attenuation, if enabled, will compensate for the number of open microphones (NOM) that are assigned to that particular output by applying an attenuation of $10\log_{10}(NOM)$. Please note that NOM attenuation is computed on a per output basis. This keeps the overall gain of the signal constant and helps eliminate acoustic feedback in sound reinforcement applications. A unique feature of the Vortex is that the NOM attenuation is computed for each of the 12 outputs and can be different for each output depending on what microphones are mapped to that particular output.

The outputs of the Vortex are line level with a nominal level of 0dBu with a maximum value of +20dBu. Output gain or attenuation adjustment is possible on a per channel basis.

8 APPLICATIONS

Vortex products may be used in a large variety of applications including voice and video conferencing, courtrooms, distance learning, telemedicine and more. They have even been used in unique television broadcast applications including town hall meetings and sports team draft coverage.

Several application notes and other resources are available discussing the various Vortex applications, product setup, matrix settings, and system configuration. Check out these documents (all available via the Polycom Resource Center on the internet):

Vortex – VSX™ 8000 Integration
Interfacing to the SoundStation VTX 1000™
InstantDesigner Release 2.0
Interfacing to Video Codecs
Vortex User Guides for Vortex EF2280, EF2241, EF2211, EF2210, and EF2201

9 AUTOMATIC MICROPHONE MIXING

The Vortex EF2280 and Vortex EF2241 have two independent automatic microphone mixers that can have their own settings for off attenuation, last microphone mode, chairman microphone settings, and NOM limit. There are a number of parameters to control and they are categorized as global parameters (the entire Vortex), per mixer parameters (remember there are two automixers), and channel parameters (channels 1-8). The global parameters are hold time and decay time. The mixer parameters are maximum number of open microphones (NOM limit), last mic mode, chairman microphone mode, and the amount of off attenuation. The channel parameters are which mixer the input is assigned to, the gate ratio, adaptive or fixed threshold, manual threshold, and the gate priority. Details of these options are given in below.

Remember, if a microphone is not assigned to one of the two automixers in the Vortex, then the microphone does not affect gating decisions.

Figure 12 shows the different parameters for how the Vortex automixers function. These parameters are described below.

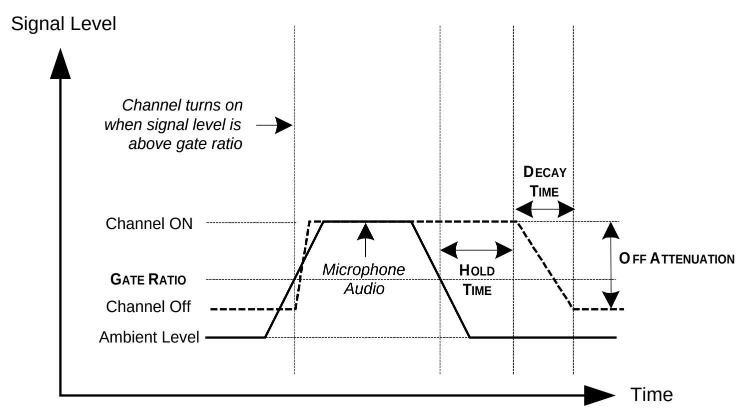


Figure 12. The gating automixer parameters and operation.

Gate Ratio

This allows you to determine when to gate a microphone on based on an estimate of the background noise level. The default value is to gate a channel on if it is more than 15 dB louder than the background noise level. Values range from 0 to 100 dB.

Off Attenuation

Off Attenuation affects how much the signal level is reduced when the microphone is gated off. The values range from 0 to 100 dB by 1 dB increments. The default value is 15 dB of attenuation (or a gain of –15dB).

Hold Time

This is the amount of time the microphone stays on after the energy in the channel drops below the gating threshold. The default value is 500 msec and the range is 1 to 5000 msec. Microphone channels gating On and Off too frequently during short pauses in speech might be the result of setting the Hold Time too low while too many microphones gating on at the same time may be the result of Hold Time values that are too high.

Decay Time

Decay time is the amount of time the microphone audio takes to ramp down to the Off Attenuation level after Hold Time. Decay Time values range from 0 to 5000 msec. The default value is 1000 msec.

Microphone activation

There are two modes of microphone activation for an input that is assigned to one of the two automixers: auto gate, or forced-on/forced-off. In auto gate mode, a microphone will gate on or off depending on the signal activity and the settings for that input channel and for that mixer. This is the normal operating mode. If a channel is set to forced-on, the channel will operate as if it is always gated on. This will affect the count of the number of open microphones which will affect the NOM attenuation that is applied to any output signal that uses that input channel in its mix. If a microphone is forced-off, then the signal will never be gated on and will always be attenuated by the amount of off attenuation.

Chairman Mode (On or Off). Any of the inputs to the automixers may be selected as a Chairman microphone. When the Chairman microphone gates on, all the other microphones assigned to that mixer will be gated off. This allows the chairman to take complete control of the meeting.

NOM Limit

The NOM limit feature allows the user to set the maximum number of microphones that may be automatically gated on for each mixer (remember there are two independent automixers built into the Vortex). The actual maximum number of microphones that will be on at any given time depends on the signal activity and whether any channels are forced-on. Forcing a microphone on will override the NOM limit. For example, if NOM limit is set to 2 and there are two microphones that are presently gated on, if the user forces a third microphone on, that third microphone will be forced on making the actual number of microphones gated on 3. When the NOM attenuation is applied to an output that includes these three microphones in its mix, an attenuation of $10\log_{10}(3) = 4.77$ dB will be applied to the output signal.

Priority

There are four levels of gating priority associated with each microphone where priority 1 is the highest priority and priority 4 is the lowest priority. This is a softer version of the Chairman Mode where higher priority microphones are allowed to displace lower priority microphones. Here's how it works: the microphones with the higher priority are more important to the automixer when it makes its gating decisions. For instance assume the NOM Limit is 2 (only 2 microphones are allowed to automatically gate on at any given time) and a priority 1 microphone is gated and a priority 3 microphone is gated. If another priority 1 or priority 2 microphone is loud enough to gate the mixer then that microphone will be gated on and at the same time the priority 3 microphone will be gated off because it is the lowest priority. In this example, because NOM Limit is set to 2, only 2 microphones can be automatically gated on at one time.

In another example, if NOM Limit is 4 and two microphones with priority 1 are gating and two microphones with priority 2 are gating, then a priority 3 or 4 microphone will have to wait until one of the priority 1 and 2 gated microphones gates off. Once that happens the lower priority microphone can be gated (if there is enough signal activity to gate the microphone) and once gated it will remain gated until a higher priority microphone gates on and takes priority – forcing the lower priority microphone to gate off.

Last Mic On Mode

Each automixer can be configured with Last Mic On mode in one of three ways. Last mic mode can be disabled, can be set to a particular channel, or can be set to the last microphone that was gated on.

10 AUDIO MATRIXING

One of the most important features of the Vortex is the full bandwidth full cross point matrix. Not just a router, the Vortex's full matrix allows audio from any input to be changed in gain and mapped to any output. No longer do you have to be constrained with using a sub-bus to change the gain of groups of input signals. The Vortex now makes it possible to have arbitrary gains on every cross point - any combination of inputs to outputs is now possible. Any input signal can be mapped to an output by applying a cross point (in dB) and unmuting the cross point. For instance to map input 1 to output A with 6 dB of attenuation, you would put a –6 in the Row 1, Column A cell. To map inputs 1-8 to output A with 6 dB of attenuation you would put a –6 in Rows 1-8, Column A cells. It is that easy! The cross points can be changed via RS-232, the LCD front panel, or through presets and macros activated by contact closures.

To explore the matrix further, let's start with reviewing the inputs and outputs.

Inputs

The inputs to the matrix are 8 Microphone/Line inputs (1-8), 4 line inputs (A-D), a signal generator (white noise, pink noise, and sinusoids), and the 14 outputs of the EF Bus mix-minus sub-matrix (P_M0, P_M1, W_M0, W_M1, W_M2, X_M0, X_M1, X_M2, Y_M0, Y_M1, Y_M2, Z_M0, Z_M1, Z_M2) for a total of 27 inputs. Don't worry, the EF Bus cross point mix-minus sub-matrix is described in Section 17: Linking Multiple Devices.

Mic/Line Inputs

The microphone/line inputs (1-8) to the matrix are the outputs of the processing block shown in Figure 11 -- the signal is echo cancelled, noise cancelled, automatic gain controlled, filtered, and has passed through the automixer. Both the gated and ungated signals can be used in the matrix. For example input 1 could be sent directly to output 1 as an ungated signal (for example a direct output used in a court system) and it can be used as part of output A as a gated signal (to be sent to a codec).

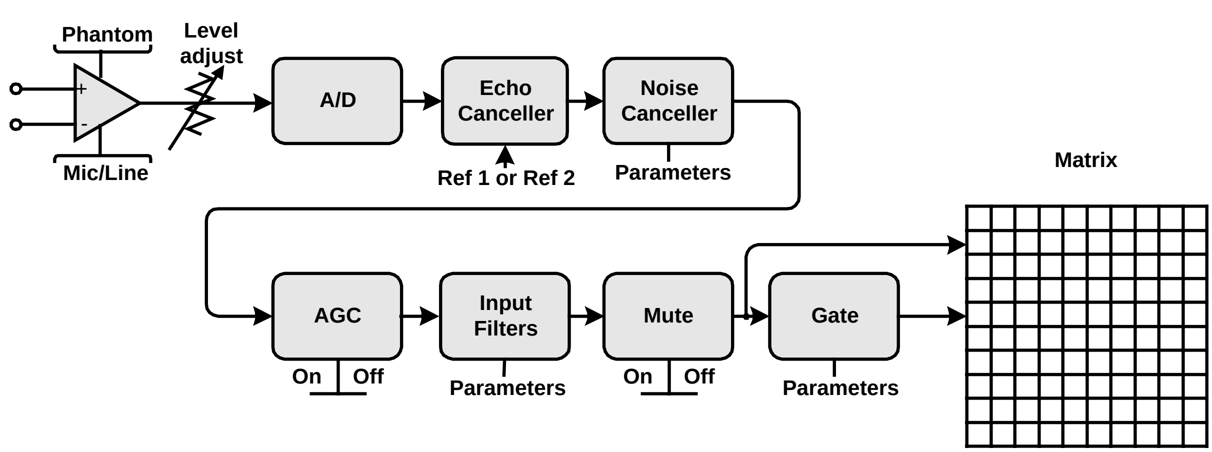


Figure 13. Input 1-8 signal path to the matrix.

Line Inputs

The line inputs to the matrix are typically inputs from video codec signals or program audio devices such as VCRs or DVDs. In a typical application these signals must be heard in the local room and can be mapped through the matrix to outputs.

Outputs

The outputs of the matrix are 12 line outputs (1-8, A-D), the two echo canceller references, and the four EF Bus signals W, X, Y, and Z for a total of 18 outputs. The echo canceller references are not physical outputs of the device, but are used internally. You do not need to use an output for the echo canceller reference. This is all shown in Figure 14.

Outputs 1-8 (Vortex EF2280)

Outputs 1-8 can contains any combinations of inputs. The default mappings in most of the 16 factory presets configure outputs 1-8 as direct outputs of inputs 1-8.

Outputs A-D

The A-D outputs are typically connected to amplifiers, video codecs, and recorders. Typically this audio is comprised of the local microphones, the codec, and the phone add signals.

EF Bus

This is a high-speed digital bus that appears at every Vortex networked together. This bus allows signals placed on the bus from one device to appear at all the other networked devices, except the sending device. The Vortex EF2280 supports placing 4 signals on this EF bus: the W, X, Y, and Z expansion signals. All four bus channels, W, X, Y, and Z, can be used in the same way since they all carry information about the number of open microphones.

P-bus.

This bus is the default bus for sending the telephone signals to other devices. Only Vortex devices that have an integrated telephony interface can put signals onto the P bus, but all devices can take signals from the P bus.

W-bus.

This bus is the default bus for sending the microphone signals to other devices. In addition to the signal data this bus sends information about how many open microphones there are in the sum of the microphones that are mapped to the W bus output.

X-bus.

This bus is the same as the W-bus and is used for sharing microphones with other devices. This bus sends information about how many open microphones there are in the sum of the microphones that are mapped to the X bus output.

Y-bus.

This bus can be used for sending any signal between different devices. This bus also can be used for sharing microphone information. Typically applications may be for sending stereo program audio between devices (one channel on the Y-bus and one channel on the Z-bus).

Z-bus.

This bus is the same as the Y bus and can be used for sending any signal between different devices. This bus also can be used for sharing microphone information. Typically applications may be for sending stereo program audio between devices (one channel on the Y-bus and one channel on the Z-bus).

EF Bus EC Reference

This bus allows one device to make its echo canceller reference available for other devices that are linked to it. For instance the user can put one of the devices' echo canceller reference (either Ref 1 or Ref 2) onto the EC Reference bus signal. Other linked devices can choose to take their echo canceller reference from the EC Reference bus, or can build their own reference (their local Ref 1 or Ref 2) signals.

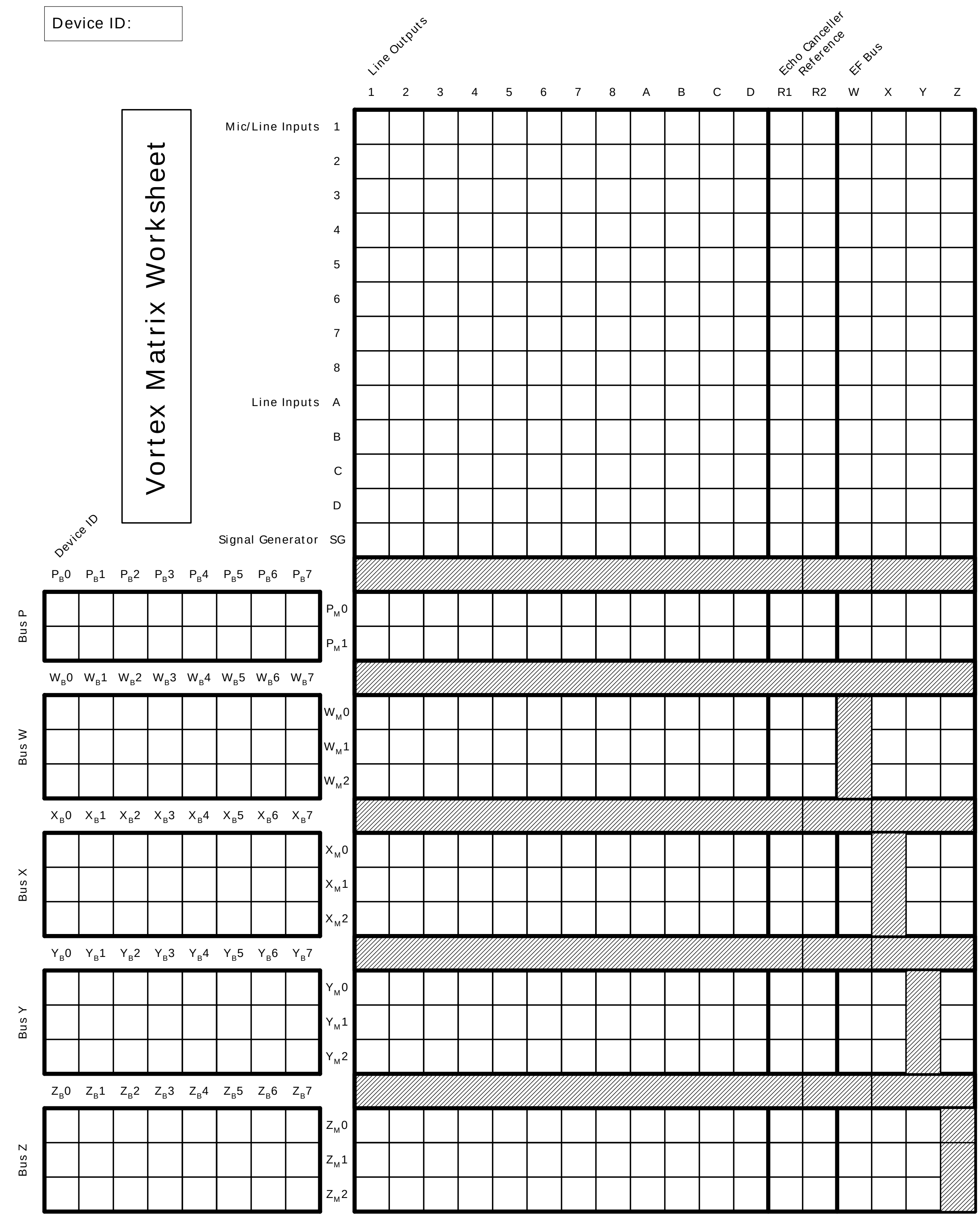


Figure 14. The matrix worksheet including the full matrix and the EF Bus sub matrix. You can use this matrix to design your input and output mappings. Inputs to the main matrix are on the left and outputs are on the top. Inputs to the sub-matrices (PB0, …, PB7, WB0, …, WB7, …) are from the Vortex with device ID 0, …, 7, respectively.

11 PRESETS AND MACROS

The Vortex has built-in non-volatile storage that can store 16 factory default presets, 32 user presets, and 256 user macros. A preset is a complete setting for the Vortex and stores over 2,000 user configurable parameters such as mic/line gains, filter parameters, matrix cross points, etc. A macro is a much smaller collection of settings (up to 256 settings) and can be used to make changes to the Vortex settings or even send commands out the RS232 port to control other devices. Macros can be thought of as mini-presets, i.e., they configure some settings – what the user specifies to configure - but not everything. A macro can change specific cross points, input gains, output gains, muting, etc. For example the user can have a preset and then have multiple macros that can be executed based on contact closures to reconfigure gains, assignments of inputs to automixers, etc. All macros and presets can have a 16-character text label to make it easy to remember what they do.

The factory default presets (presets 0 to 15) represent common configurations for the Vortex and are present on any Vortex you use. The user presets (presets 16-47) can be used to store any collection of settings that the designer requires for a given installation.

The user can write-protect the Vortex with a password so that only authorized users can save a preset and change which preset the device powers up into.

All the analog inputs and outputs and the logic control inputs and logic status outputs of the Vortex can have text labels that may be up to 16 characters in length. These labels are stored within the device just like the presets and macros and can be downloaded into a PC along with the other settings.

Macros are created within Conference Composer on the Macro screen by either typing in commands or selecting commands from a pull down menu.

12 SERIAL REMOTE CONTROL

The Vortex has been designed to be configured with a PC via the RS-232 port and the included Conference Composer software and then operated with the RS-232 connected to a remote control system. As with previous EF products, multiple devices can be linked together and controlled via a single RS-232 connection. All functions of the Vortex can be controlled via RS-232. It is also possible to download new firmware into the Vortex via RS-232.

The Conference Composer software makes it extremely easy to configure features and makes it possible to save configurations to disk and to restore configurations.

Please note that while an external control system is not required (because many of the functions of the product are accessible via the contact closures by assigning functions or macros to pins) it is often desirable to have an external controller particularly in an installation with other equipment.

All commands to the product are executed in real-time, e.g., device settings are updated immediately. A document describing the entire command set of the product and programming tips is available in the Conference Composer software help file This document is also available on the CD that is shipped with the Vortex.

Each command name can be from 1 to 7 characters long followed by the specific arguments. Command names will be specific to device types as some products have an integrated telephony interface and others don't. There are some commands, such as PING, that are common among the various command sets. The Vortex sends status messages via RS-232 and EF Bus any time one of its internal parameters changes. This means that the host program does not need to continually poll the Vortex in order to detect status changes. Status messages are in the same format as the commands used to set the corresponding parameter.

A powerful feature of the command set is the ability to use wildcards which allows you to affect multiple channels with a single command or to query the status of multiple channels and return an array status message. Another powerful feature is the absolute and relative parameter settings. Now you can set a gain value absolutely, or change it relative to its current value. A list of the commands is included in the programmers reference guide.

13 CONFERENCE COMPOSER SOFTWARE

Conference Composer is the design and configuration software that is supplied with the Vortex. Its purpose is to make it easy and intuitive to configure the Vortex, share configuration files, create presets, create macros, and assign events to logic input and output pins. Conference Composer makes using the Vortex simple and intuitive. Conference Composer operates under Windows™ NT, 2000, and XP.

A powerful addition to Conference Composer is our *Instant*Designer setup wizard. For more information on the Polycom *Instant*Designer, see the next section or check out our application note on *Instant*Designer.

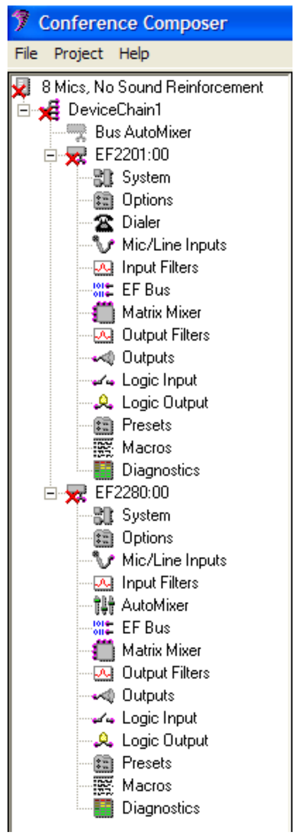


Figure 16. Project with an EF2201 at device ID 0 and an EF2280 at device ID 0.

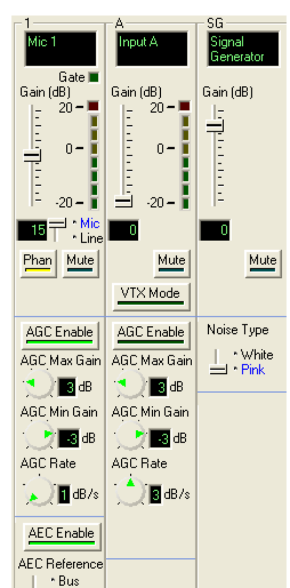
Conference Composer may either be run online (connected via RS-232) with a Vortex or off-line (not connected to a Vortex) depending on how you would like to use it. If you work off-line it is easy to save the settings to a file (with CCP file extension), share the files with others, and then upload the settings into a Vortex when you go online. The online or off-line operation of Conference Composer is the same, although if you are operating off-line you will need to add devices to the project before you can start working with the devices.

Conference Composer organizes your projects into a hierarchy of Project, Device Chain, and Device. Starting from the bottom of the list, a device is simply that – some piece of equipment that will be used in the installation. A Device Chain is a collection of devices that are linked together through the EF Bus (that's where the name device *chain* comes from). Typically these represent all the devices that are tied to a particular serial port on your computer or controller. The project represents a collection of device chains. Multiple projects may be open at the same time.

Each device in Conference Composer is represented by a list of sub-entries or tabs for that device. For the Vortex, those fields are System, Mic/Line Inputs, etc. as shown in Figure 16. Clicking on either a tab or the field name will bring you to a page with specific controls for that set of functions.

Figure 17 shows some of the controls that are available to configure mic/line inputs 1-8, line inputs A-D, and the signal generator.

There are similar controls for the parametric EQ capabilities, the automixer, the matrixer, the outputs, etc. See the Conference Composer documentation for more information about using Conference Composer.



Vortex Matrix

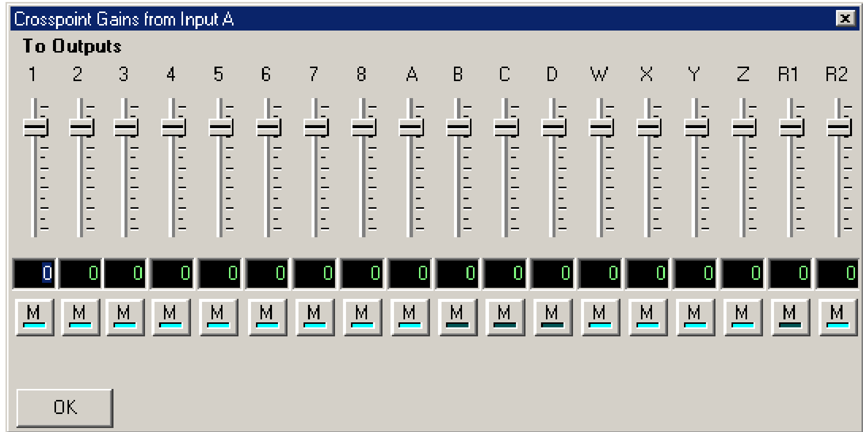
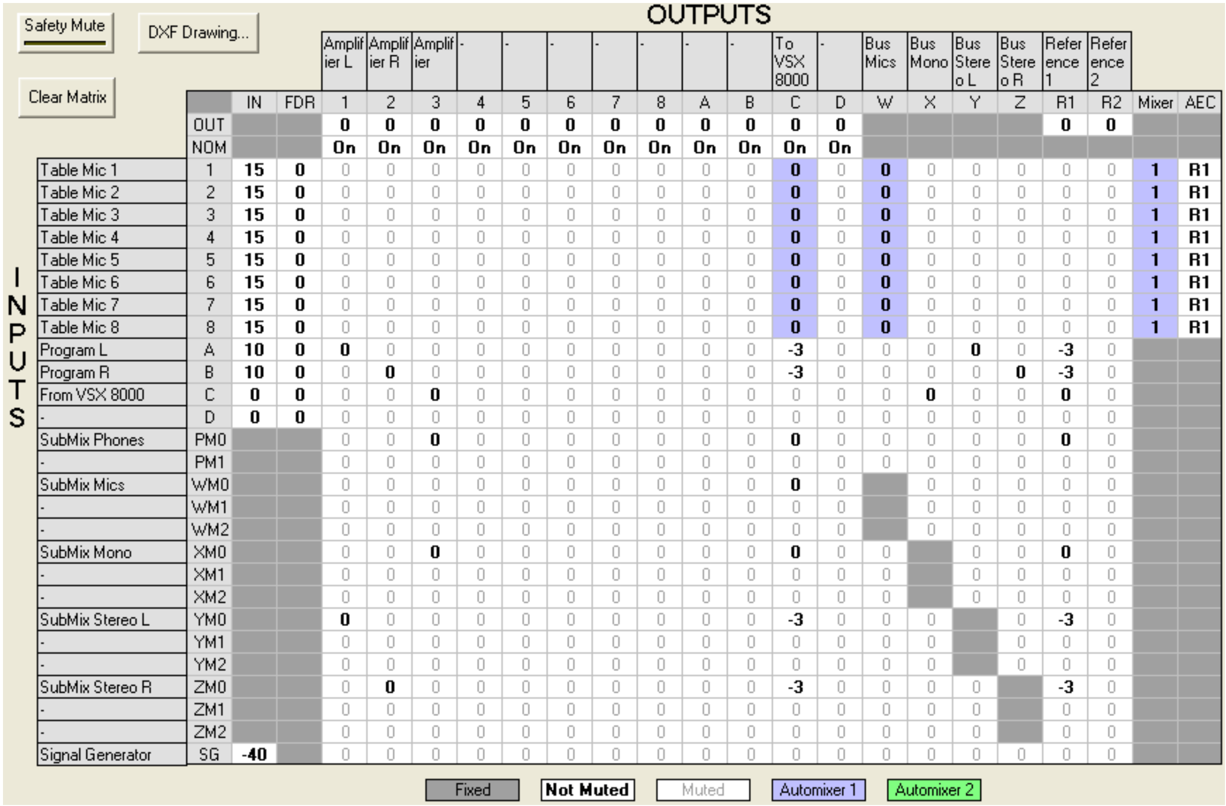


Figure 13. Cross point slider view of how input A is used in all the output channels.

The cross point matrix in Figure 20 allows the user to map any input to any output with an arbitrary gain applied. To set the gain of a cross point single-click on the cell. To mute the cross point simply right-click on it. To set the gain of a collection of cross points click and drag to select a rectangular area. A slider control with a text box will appear to allow you to change the gain, select mute/unmute, or, when working with microphones, allow you to choose the gated version of the input. The gain of the gated signals should be set to 0dB since the matrix will sum the automixed signals and use the appropriate weightings of the signals based on the information from the automixer.

To change the gain of an entire row or column you may simply click on the column or row header to see a collection of sliders as shown in Figure 13. To change gating status, Control-click (hold the control button and left click). To change the automixer assignment, click on a Mixer column entry.

The R1 and R2 outputs correspond to the echo canceller references Ref 1 and Ref 2. While these outputs are not physical outputs on the product, they allow you to build the echo canceller reference without stealing physical outputs from the device. In the example in Figure 20, we used stereo program audio and consequently had a local left output (output 1) and a local right output (output 2) in the room. The echo canceller reference R1 included both the codec audio (input C), the phone add audio (input PM0) and a mono sum of the left and right program audio. The program audio is attenuated by 3dB before adding it into the R1 output.



		IN	FDR	Amplifier L 1	Amplifier R 2	Amplifier 3	- 4	- 5	- 6	- 7	- 8	- A	- B	To VSX 8000 C	- D	Bus Mics W	Bus Mono X	Bus Stereo L Y	Bus Stereo R Z	Reference 1 R1	Reference 2 R2	Mixer	AEC
	OUT			0	0	0	0	0	0	0	0	0	0	0	0					0	0		
	NOM			On	On	On	On	On	On	On	On	On	On	On	On								
Table Mic 1	1	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 2	2	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 3	3	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 4	4	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 5	5	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 6	6	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 7	7	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 8	8	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Program L	A	10	0	0	0	0	0	0	0	0	0	0	0	-3	0	0	0	0	0	-3	0		
Program R	B	10	0	0	0	0	0	0	0	0	0	0	0	-3	0	0	0	0	0	-3	0		
From VSX 8000	C	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		
-	D	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		
SubMix Phones	PM0			0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		
-	PM1			0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		
SubMix Mics	WM0			0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0	0		
-	WM1			0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0	0		
-	WM2			0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0	0		
SubMix Mono	XM0			0	0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0		
-	XM1			0	0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0		
-	XM2			0	0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0		
SubMix Stereo L	YM0			0	0	0	0	0	0	0	0	0	0	-3	0	0	0		0	-3	0		
-	YM1			0	0	0	0	0	0	0	0	0	0	0	0	0	0		0	0	0		
-	YM2			0	0	0	0	0	0	0	0	0	0	0	0	0	0		0	0	0		
SubMix Stereo R	ZM0			0	0	0	0	0	0	0	0	0	0	-3	0	0	0	0		-3	0		
-	ZM1			0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		0	0		
-	ZM2			0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		0	0		
Signal Generator	SG	-40		0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		

Figure 20. The full cross point matrix for a system using a phone add, a codec, and stereo program audio. All cross point gains are in dB. Grayed cells are muted.

14 POLYCOM *INSTANT*DESIGNER V 2.0

Following is a brief description of our *Instant*Designer setup wizard for the Vortex products. For more comprehensive information, see the application note on *Instant*Designer.

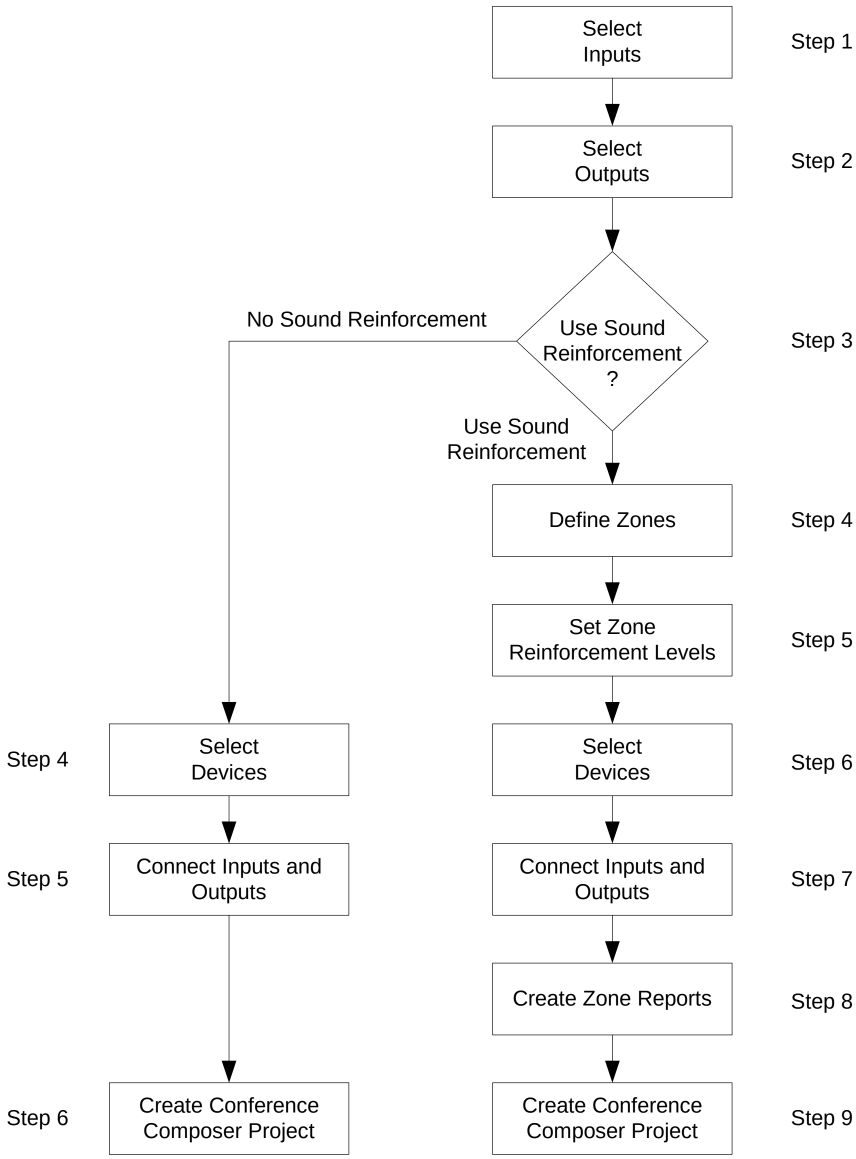
Polycom **Error! Bookmark not defined.** release 2.0 encapsulates the design expertise of the Vortex engineering team and the experience of thousands of successful audio and video conferencing installations. We have learned both the easy way and the hard way what works and how to best use the Vortex products. By making Polycom **Error! Bookmark not defined.** available to you, the A/V specialist, we want to make your job easier and allow you to spend more time doing the things that can help your design and integration firm be more profitable. To give you an idea of how much the design process can be accelerated, below is a partial list of what Polycom **Error! Bookmark not defined.** will do for you. With just a few mouse clicks, Polycom **Error! Bookmark not defined.** will automatically:

- Select the Vortex equipment necessary to implement the system
- Map the input and output equipment to Vortex inputs and outputs (e.g., lectern microphone is input 1 on Vortex EF2280:00)
- Set the input and output gains in the Vortex required to interface to the selected input and output equipment (e.g., most video codecs typically require 10dB of input gain, and 5dB of output attenuation)
- Configure the automatic microphone mixer for linked operation for larger systems
- Create all the matrix crosspoints necessary to use the resulting system (e.g., local audio is sent to the remote site and remote audio is played back into the room)
- Create loudspeaker zones for sound reinforcement and allow you to specify the microphones that are part of that zone and reinforced into other zones
- Make it easy to modify the zone to zone crosspoint gains needed for sound reinforcement across zones
- Configure all the bussing across linked devices (e.g., local microphones are automatically sent to all devices), including sound reinforcement zone gains
- Configure the acoustic echo cancellation (e.g., the echo canceller reference is configured and all linked devices are configured to use the bus reference)
- Customize the text labels for all inputs and outputs including the bussing signals
- Save the settings to Preset 16 and set the power-on preset on all devices
- Create macros for muting, unmuting, and volume control
- Create logic assignments making it easy to connect push-to-talk microphones or other hardwired control systems
- Create a project summary of the wiring connections and necessary equipment
- Create formatted PDF reports of your system configurations
- Create a zone to zone sound reinforcement summary report
- Create a signal routing report so you can see how the signals are routed and bussed to create the appropriate sound reinforcement zones
- Create DXF drawings of your designs
- Upload the Conference Composer project to your Vortex devices

The list keeps growing and growing. Imagine what you can do with Vortex products now – perform more conference designs with your current team, create new conferencing applications, build more expertise within your team, etc. If you haven't tried the Vortex products – you can't afford not to.

What's new in *Instant*Designer Version 2.0:

The main difference is its ability to help you set up microphone/loudspeaker zones. From a high level perspective, there are two ways you can use *Instant*Designer, depending on whether sound reinforcement is required.



Once the project has been created, it is a simple matter to go into Conference Composer and make any desired final changes to the system. You can practice (or just play) with the *Instant*Designer wizard by downloading Conference Composer from the Polycom Resource Center. Please refer to the *InstantDesigner Release 2.0 Application Note* for complete instructions on using this powerful tool.

15 FRONT PANEL CONTROL

Using a PC with the Conference Composer / *Instant*Designer software is the easiest way to configure the Vortex. However, since the user may not always have a PC available, it is possible to control many of the functions directly from the front panel interface.

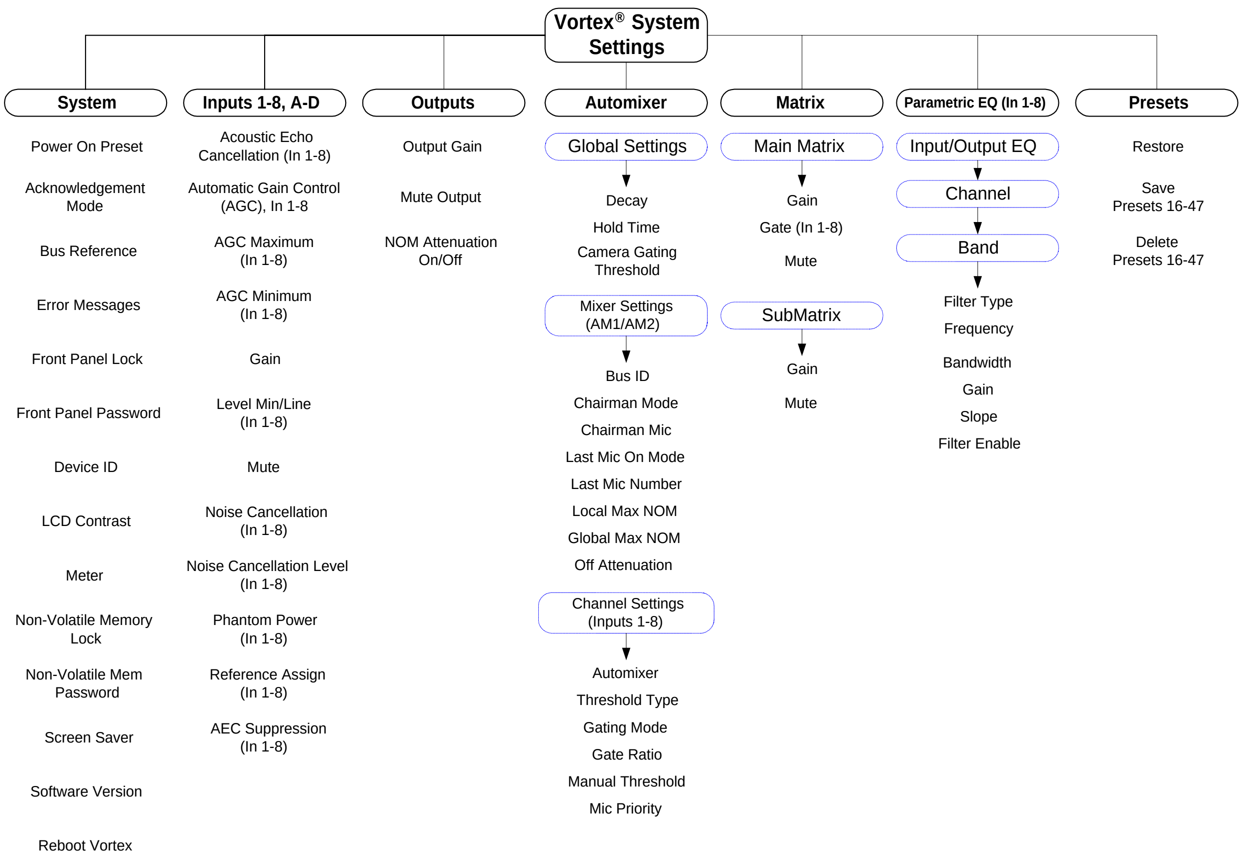


Figure 16. The LCD menu structure of the Vortex.

The Vortex has five menu buttons on the front panel for navigation in the menu tree. Press the HOME button from anywhere in the menu tree to return to the top of the menu. The ENTER button enters the menu and the ESC button returns to the next highest level of menus. To scroll back through menu items at a particular level, use the DOWN button. To scroll forward through menu items at a particular level, use the UP button. To adjust a parameter, first locate the parameter by scrolling to the appropriate menu (with combinations of the UP/DOWN and ENTER buttons). The display will show the parameter field and the parameter value. To change the parameter, the parameter must be flashing. To make the parameter flash (assuming the front panel is not locked) press ENTER. Once the parameter is flashing, use the UP and DOWN buttons to adjust the parameter value. The parameter is instantly updated while it is being adjusted - you should hear changes as the parameter is changing.

As you update settings from the front panel, RS-232 control strings are also sent via the RS-232 port so your remote control device is instantaneously updated as well. Press ENTER to select and store the parameter value or press ESC to cancel the selected value and return to the old value. Pressing HOME has the effect of pressing ESC then HOME, so the selected value will be cancelled and the menu will return to the top of the menu tree. Parameters that toggle or select among a list of options will wrap around when you reach the end, but parameters that adjust numeric values will not wrap around once the maximum or minimum value is reached. While adjusting a parameter, the UP/DOWN button must be held down briefly before the repeat rate increases.

16 Remote Control: Logic Input and Output Pins

The product has 24 logic input pins and 20 logic status output pins. Each of the logic input pins can be assigned to trigger an event based on contact closure, contact open, or contact held closed or open. In addition, the user can invert the polarity of a pin (what was executed on closure would execute based on opening if the polarity was changed after the events were assigned). The logic pins are unassigned by default.

Events can be preset changes, macros to execute, or single RS-232 commands. Input pin 25 is a ground pin and can be used as a common ground for contact closures. A simple example of a macro being assigned to a logic pin would be to raise or lower a stereo volume control based on a single logic input. The macro would raise or lower the volume of the left and right program audio each time it executed.

Up to four groups of logic pins may be created where each group can have up to 5 logic pins assigned to the group. These groups of logic pins can be used to trigger events based on the binary sequence the logic contacts form. For instance if logic input pins 1, 2, and 3 are assigned to a group, then there are 8 possible combinations of settings that can be generated by these 3 logic pins (3 bits). These assignments can be related to contact closures for room combining or any other connections the designer desires.

The logic output pins allow you to assign status signals such as gating or mute status to output pins. Each output pin can provide up to 20mA of current to drive LED's or other electronics.

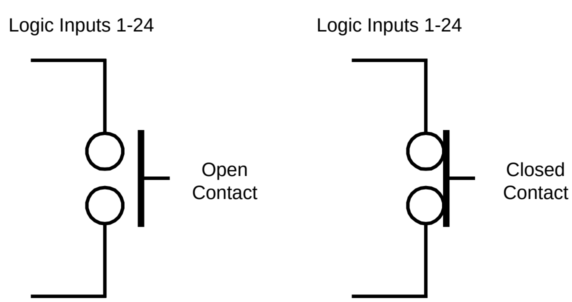


Figure 17. An example of an open contact (left) and closed contact (right).

17 LINKING MULTIPLE DEVICES

One of the most important features of the Vortex is its ability to be linked with other Vortex's when your design requires more than 8 microphones. The Vortex allows the user to link up to 8 devices and have it operate as a single 64-channel automixer with 32 line level inputs, sharing signals across the EF bus.

The EF bus is a high-speed network protocol that allows up to 8 Vortex units (any combination) to be linked together – in addition, up to 8 EF2280's and 8 EF2201's can be linked together for a total of 16 devices. The EF bus is used for both command and control information between devices and for high speed data linking. Each Vortex can place 4 or 5 output signals on the EF Bus – the P, W, X, Y, and Z signals – and use 14 mixes of the up to 40 input signals from the bus in its local matrix. In addition one device can place its echo canceller reference on the bus so the linked devices can use it. Only the telephony products have the ability to put signals on the P bus.

Since each device (up to 8) can place 4 or 5 channels of audio on the bus, there are a total of 40 (5 for each device times 8 devices) full-bandwidth signals that are accessible by all the devices. Each linked Vortex can access the bus and choose which signals to read from the bus and use with its local matrix.

To understand the linking, each of these busses will be introduced and then described from both an output view and an input view.

P bus An audio bus that is used to share telephony signals between devices. Only Vortex devices with telephony options (EF2241, EF2211, and EF2201) can put signals on this bus. All Vortex devices can read data from the P bus.

W bus An audio bus that includes signal information and information about the number of open microphones for each device that puts microphone information into this bus.

X bus An audio bus that includes signal information and information about the number of open microphones for each device that puts microphone information into this bus.

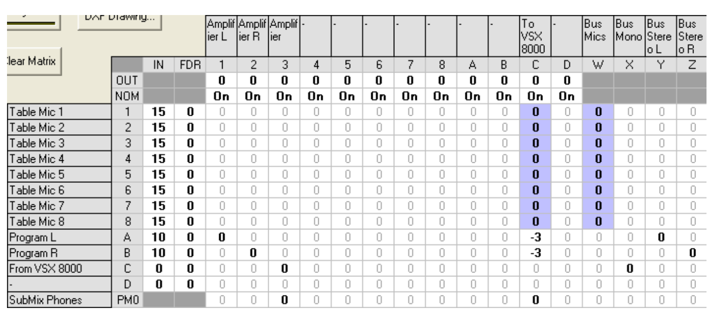
Y bus An audio bus that includes signal information and information about the number of open microphones for each device that puts microphone information into this bus.

Z bus An audio bus that includes signal information and information about the number of open microphones for each device that puts microphone information into this bus.

EC Ref An audio bus that allows one device to put its echo cancellation reference on the bus to be used by other devices.

Expansion Outputs Example

Let's assume the Vortex EF2280 we are using is device id 0 (the device id's can range from 0 to 7). As described above, Vortex EF2280:00 has an output signal that it can place on each of the four busses: W, X, Y, and Z. Each of these four outputs is created through the matrix by mapping input signals to the output signals through the matrix mixer. Figure 18 shows a portion of an EF2280:00 matrix and shows outputs 1-8, A-D and the expansion busses W, X, Y, and Z. In this example output W is a gated sum of inputs 1-8 (the blue background signifies the microphone signals are "gated" through the automatic microphone mixer), output X is the codec audio from input C, output Y is the



				Amplifier L	Amplifier R	Amplifier	-	-	-	-	-	-	-	To VSX 8000	-	Bus Mics	Bus Mono	Bus Stereo L	Bus Stereo R
		IN	FDR	1	2	3	4	5	6	7	8	A	B	C	D	W	X	Y	Z
	OUT			0	0	0	0	0	0	0	0	0	0	0	0				
	NOM			On	On	On	On	On	On	On	On	On	On	On	On				
Table Mic 1	1	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
Table Mic 2	2	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
Table Mic 3	3	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
Table Mic 4	4	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
Table Mic 5	5	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
Table Mic 6	6	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
Table Mic 7	7	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
Table Mic 8	8	15	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
Program L	A	10	0	0	0	0	0	0	0	0	0	0	0	-3	0	0	0	0	0
Program R	B	10	0	0	0	0	0	0	0	0	0	0	0	-3	0	0	0	0	0
From VSX 8000	C	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
-	D	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0
SubMix Phones	PM0			0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0

Figure 18. A screen shot of the Conference Composer software showing the mapping input signals to the expansion busses.

Program L signal (the left channel of the program audio), and output Z is the Program R signal (the right channel of the program audio).

That's all there is to do to put signals on the expansion bus. The W, X, Y, and Z signals from EF2280:00 can now be used by other linked Vortex devices.

Expansion Inputs Example

Now, let's look at it how another Vortex device can read these bus signals and use them locally. Let's assume that Vortex EF2280:04 wants to use the microphone signals that Vortex EF2280:00 has put on the W bus. How does it do that? When a Vortex device wants to use signals that are on the bus, it can create sums of the bus signals that can be used in its main matrix. The bus signals from other Vortex devices are summed together (with arbitrary gains) to create the combination signals WM0, WM1, WM2, XM0, … as shown in Figure 19. These combination signals can then be used in the main matrix and used as any other input to the matrix as shown in Figure 20. In Figure 19, WM0 is a user defined summation of the W signals that have been placed on the W bus by the other Vortex devices and can contain any combination of W signals (with the exception of a Vortex's own W bus signal) that are on the bus. In this figure we can also see that WM1 and WM2 only contain muted inputs from the bus (the light gray crosspoints indicate the signals are muted). By having Vortex EF2280:00 output the sum of its microphones to the W bus (this is done in the main matrix for the EF2280:00), Vortex EF2280:04 can pull the EF2280:00 W signal off the bus and create a mix on outputs WM0, WM1, or WM2 – three different combinations of signals from the W bus can be created. To do this, the user would go to the EF Bus page on Vortex EF2280:04 and unmute the WB0 signal (the W signal from Vortex EF2280:00) in the row that is summed together to form WM0 as shown in Figure 19. Just as the WM0 signal is a combination of signals from the W bus, the EF2280:04 bus combination signal XM0 is a user defined combination of all the X bus signals that the other devices (except Vortex EF2280:04 in this case) put onto the expansion bus and so forth for the other bus signals YM0, …, YM2, ZM0, …, and ZM2. The next section provides an example of using these signals.

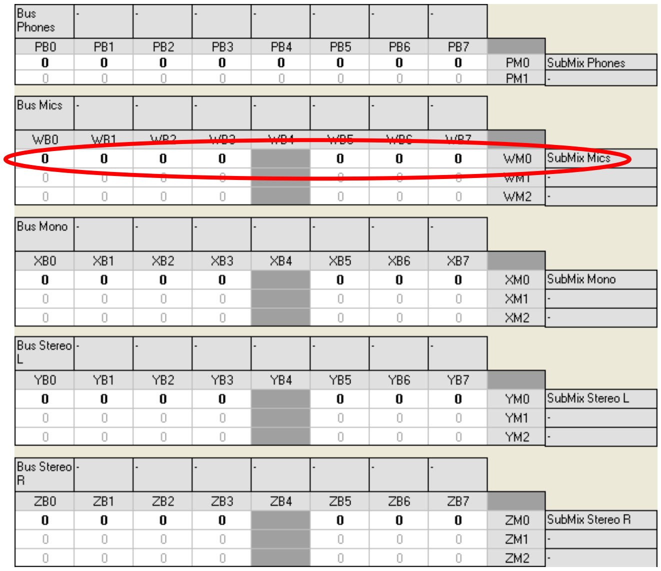


Figure 19. A screen shot of the bus mixer for Vortex EF2280:04 showing the sub matrix associated with the EF Bus. Notice that column 4 is grayed out – EF2280:04 cannot use as input signals the output signals that it put on the bus. The highlighted row shows the sum of all the W signals from other linked Vortex devices being summed together to create the signal WM0.

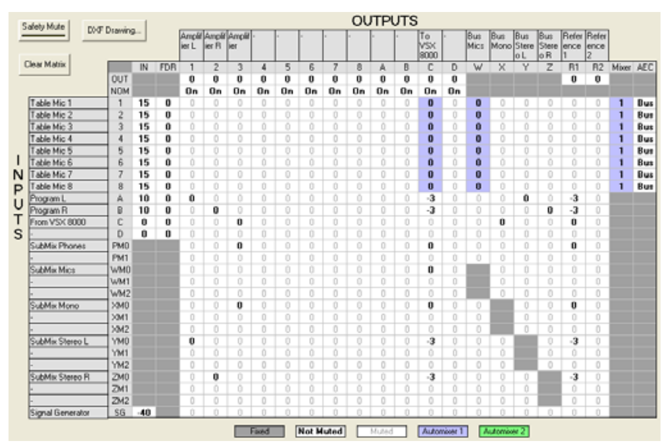


Figure 20. A screen shot of the matrix for EF2280:04 showing that EF2280:04 is using the bus input signals with the PM0 bus combination is sent to output 3, the WM0 bus combination is sent to output C, the Y bus combination is to outputs 1 and C (with 3dB of attenuation), and the Z bus combination is going to outputs 2 and C (with 3 dB of attenuation).

When Mix-Minus is not enough

The example in the previous section showed how the expansion signals (P, W, X, Y, and Z) from multiple Vortex devices could be used to send signals from one device to another.. What if you don't want to use a mix of all the W signals from the linked devices, but only want the W signal from device 2? What if you just want the Y signal from one of the devices, such as from device 0, but not from device 7? The answer is in how the sub-matrix is used.

Figure 19 shows the 8 signals that are part of the P, W, X, Y, and Z busses. Each of these 8 signals from the 8 devices with device id's 0, 1, … 7 are denoted WB0, WB1,…, WB7 for the W bus, XB0, XB1,…, XB7 for the X bus and so forth. The last digit on these signals represents the device id of the Vortex that originally put the signal on the bus. So WB0, XB0, YB0, and ZB0 are the bus output signals from Vortex EF2280:00. Each mini-matrix for W, X, Y, and Z has three outputs which go to the main matrix: WM0, WM1, WM2, XM0, XM1, XM2, YM1, YM2, ZM1, and ZM2. The mini-matrix for the P bus has only two outputs PM0 and PM1.

You can arbitrarily mute or change the gain on the bus signals (PB0, PB1, …, PB7, WB0, WB1…, ZB7) as you create the combination signals PM0, PM1, WM0, WM1, …, ZM2 The signals PM0, PM1, WM0, WM1, …, ZM2 become inputs to the main matrix and can be used like any other input and output to the main matrix. The actual bus signals PB0, … are not affected by using different crosspoint gains, only their use in creating the combination signal PM0, … is affected by setting a crosspoint gain or muting the particular bus input signal.

Figure 21 shows an example of how the different cross point gains on the bus signals can be useful. Assume we have four zones that we would like to do conferencing and sound reinforcement with 4 microphones per zone as shown in the table diagram in Figure 21. The sound reinforcement levels that we would like in adjacent levels are shown in Figure 23. We can easily accomplish this goal by having both Vortex EF2280's put four microphones on the W bus and four microphones on the X bus. Each of these busses contains NOM information, so the appropriate overall gain and automixer performance will be maintained. By entering this design into *Instant*Designer release 2.0, we can automatically create the appropriate bussing, matrix configuration, and wiring solution as will be described below. The automatically created main matrix is shown in Figure 25.

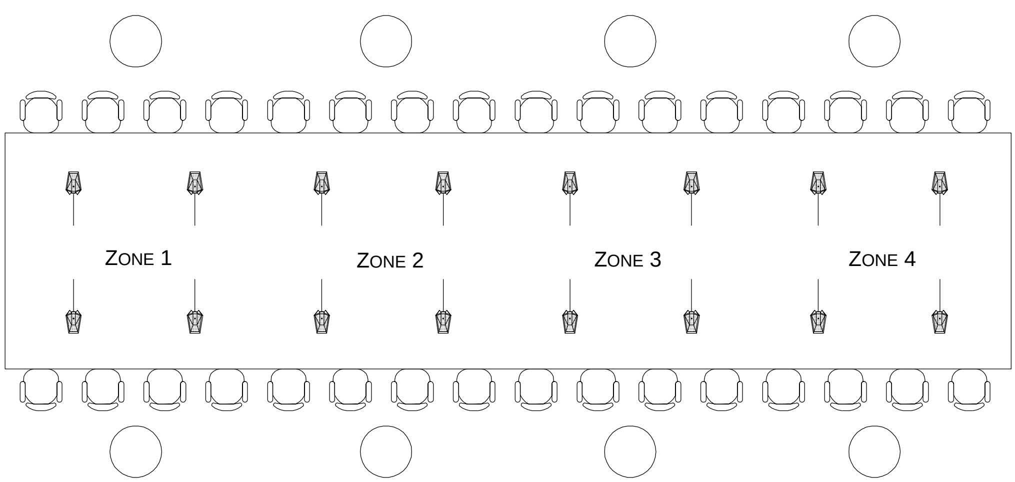


Figure 21. A conference room table with 32 participants, four speaker zones (two loudspeakers in each zone), and 16 microphones (4 microphones in each zone).

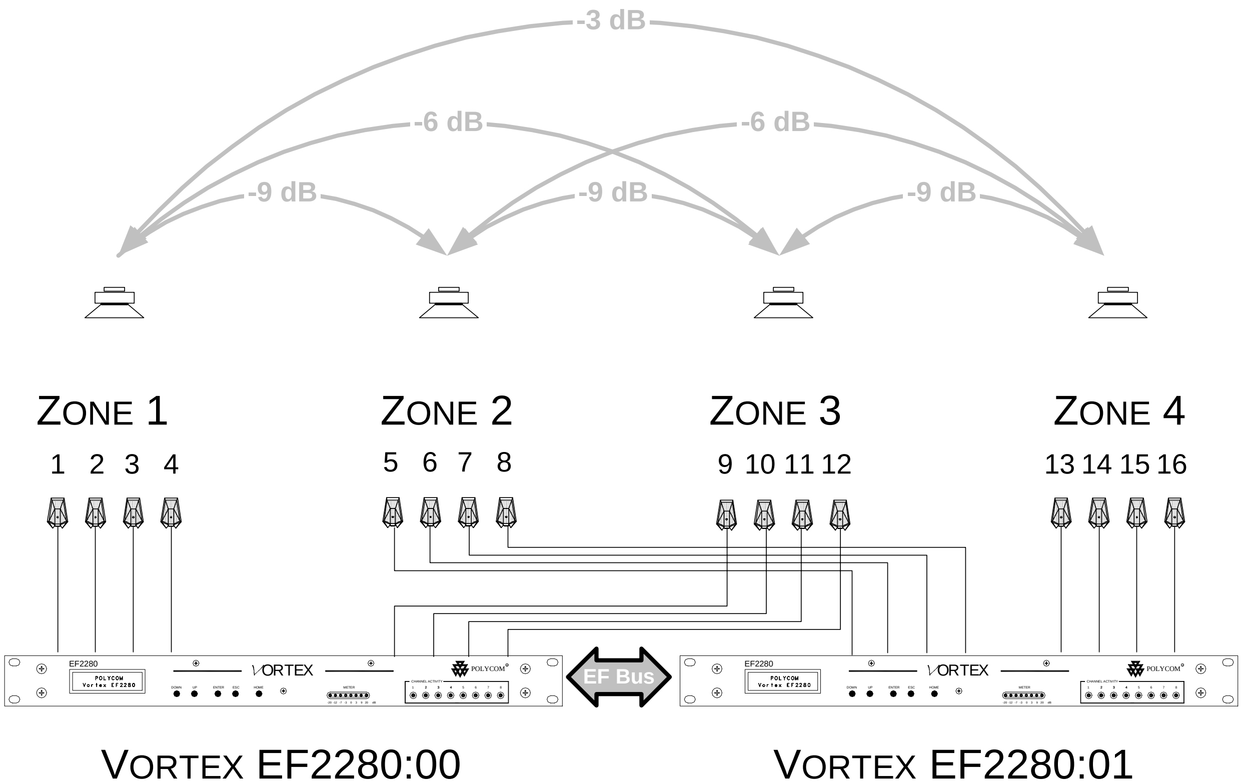


Figure 23. A four zoned conferencing system with sound reinforcement. The left Vortex EF2280 has device ID 0 as indicated by the :00 and right Vortex EF2280 has device ID 1 as indicated by the :01.

Safety Mute | DXF Drawing... | Clear Matrix

OUTPUTS

				Amplifier 1	Amplifier 3	-	-	-	-	-	-	To VSX 8000	-	-	-	Zone 1	Zone 3	Bus Mono /L	Bus Mono /R	Reference 1	Reference 2		
		IN	FDR	1	2	3	4	5	6	7	8	A	B	C	D	W	X	Y	Z	R1	R2	Mixer	AEC
	OUT			0	0	0	0	0	0	0	0	0	0	0	0					0	0		
	NOM			On	On	On	On	On	On	On	On	On	On	On	On								
Table Mic 1	1	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 2	2	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 3	3	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 4	4	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 9	5	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 10	6	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 11	7	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 12	8	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1

INP

Safety Mute | DXF Drawing... | Clear Matrix

OUTPUTS

				Amplifier 2	Amplifier 4	-	-	-	-	-	-	-	-	-	-	Zone 2	Zone 4	Bus Mono /L	Bus Mono /R	Reference 1	Reference 2		
		IN	FDR	1	2	3	4	5	6	7	8	A	B	C	D	W	X	Y	Z	R1	R2	Mixer	AEC
	OUT			0	0	0	0	0	0	0	0	0	0	0	0					0	0		
	NOM			On	On	On	On	On	On	On	On	On	On	On	On								
Table Mic 5	1	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus
Table Mic 6	2	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus
Table Mic 7	3	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus
Table Mic 8	4	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus
Table Mic 13	5	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus
Table Mic 14	6	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus
Table Mic 15	7	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus
Table Mic 16	8	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus

INP

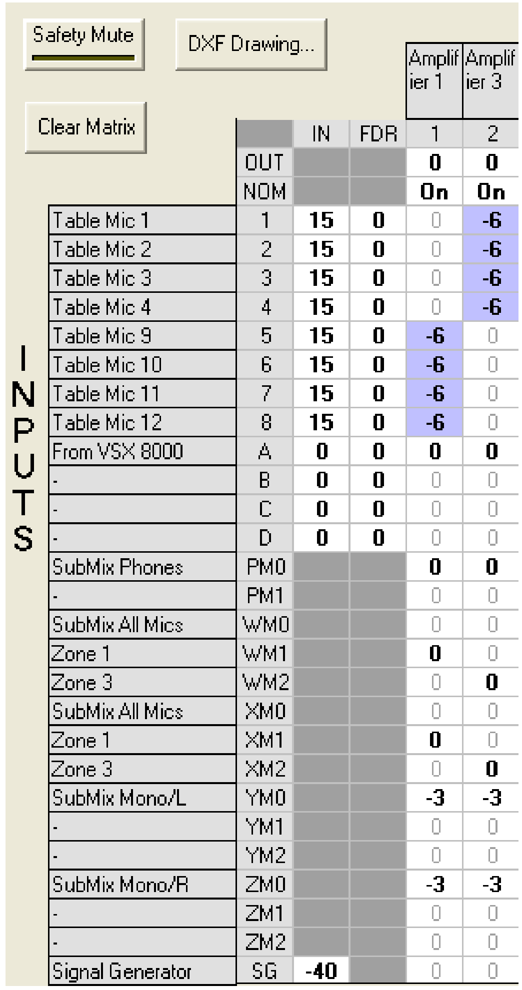
Figure 25. A cut-away of the main matrix for EF2280:00 (top) and EF2280:01 (bottom) showing how each device will map its zone's microphones to the W and X bus, respectively. EF2280:00 is creating the echo canceller reference, R1, and using that locally and exporting it to EF2280:01. EF2280:01 is using the bus reference as shown in the figure.

Now that each Vortex has put its microphones on the W and X bus respectively, let's look how we can accomplish the voice lift application on each Vortex. For this example, we will assume that two of the zone outputs are coming from each Vortex.

Vortex EF2280:00 Configuration

We specified in Figure 23 that in Zone 1 we want the Zone 2 microphones reinforced at a level of -9dB, the Zone 3 microphones reinforced at a level of –6dB and the Zone 4 microphones reinforced at a level of -3dB. InstantDesigner defined the first four microphones on EF2280:00 as being in Zone 1, the next four microphones as being in Zone 3 and the first four microphones on EF2280:01 as being in Zone 2, and the remaining 4 microphones as being in Zone 4. To create the appropriate sound voice lift attenuations, we can either apply attenuations on the incoming signal in the sub-matrix, or we can set the sub-matrix cross point gain to 0dB and apply the appropriate attenuation in the main matrix. For this example, we will apply the attenuation in the sub-matrix to help illustrate how to use the sub-matrix. In a larger system with more than two Vortexes, we would have to use the sub-matrix to get the appropriate signals and levels. As shown in Figure 28, the microphones from Zone 2 (on EF2280:01) are available on Vortex EF2280:00 as input WB1 and the microphones from Zone 4 are available on EF2280:00 on the input signal XB1 (the "1" on the bus signal XB1 name indicates the signal is sent from Vortex EF2280:01 which has device ID 1).

Based on the sound reinforcement zone definitions in Figure 23, we will attenuate the Zone 2 microphones going to Zone 1 by 6dB before sending to the Zone 1 output on EF2280:00 and attenuate the Zone 4 microphones by 3dB before sending to the Zone 1 output on EF2280:00. This attenuation is performed by setting the cross point gain of WB1 and WM1 to –6dB, and by setting the cross point gain of XB1 and XM1 to –3dB. In the main matrix for Vortex EF2280:00, the WM1 and XM1 signals will be summed together to form the local Zone 1 output as shown in Figure 23.



		IN	FDR	Amplifier 1 1	Amplifier 3 2
	OUT			0	0
	NOM			On	On
Table Mic 1	1	15	0	0	-6
Table Mic 2	2	15	0	0	-6
Table Mic 3	3	15	0	0	-6
Table Mic 4	4	15	0	0	-6
Table Mic 9	5	15	0	-6	0
Table Mic 10	6	15	0	-6	0
Table Mic 11	7	15	0	-6	0
Table Mic 12	8	15	0	-6	0
From VSX 8000	A	0	0	0	0
-	B	0	0	0	0
-	C	0	0	0	0
-	D	0	0	0	0
SubMix Phones	PM0			0	0
-	PM1			0	0
SubMix All Mics	WM0			0	0
Zone 1	WM1			0	0
Zone 3	WM2			0	0
SubMix All Mics	XM0			0	0
Zone 1	XM1			0	0
Zone 3	XM2			0	0
SubMix Mono/L	YM0			-3	-3
-	YM1			0	0
-	YM2			0	0
SubMix Mono/R	ZM0			-3	-3
-	ZM1			0	0
-	ZM2			0	0
Signal Generator	SG	-40		0	0

Figure 23. The full matrix creating zone outputs 1 and 3 on EF2280:00.

Similarly we will attenuate the Zone 4 microphones 9dB for output to Zone 3. This attenuation is performed by setting the cross point gain of XB1 to sub-matrix output XM2 to –9dB. In the main matrix for Vortex EF2280:00 we will sum WM2 and XM2 to form the local Zone 3 output.

The full main matrix for Vortex EF2280:00 is shown in Figure 26**Error! Reference source not found.**. In this figure, the local microphones (Zone 1 and Zone 2 microphone inputs 1-8), the Zone 3 microphones (WM0), and the Zone 4 microphones (XM0) are summed and sent to the codec on output A and are placed on the bus to be sent to the Phone interface (an EF2201 in this example) which also take the four zones of microphone audio off the bus and create a sum of these signals before sending it to the telephone line. The output for Zone 1 (output 1 on EF2280:00) consists of the codec, the phone add (from PM0), the microphones from Zone 2 attenuated by 9dB (WM1 from Figure 28), the microphones from Zone 3 attenuated by 6dB (attenuated in the main matrix), and the microphones from Zone 4 attenuated by 3dB (attenuated in the sub-matrix XM1 as shown in Figure 28). The output in Zone 3 (Vortex EF2280:00 Output 2) consists of the codec audio, the phone add audio, the microphones from Zone 1 attenuated by 6dB (attenuated in the main matrix), the microphones from Zone 2 attenuated by 9dB (attenuated in the sub-matrix WM2 in Figure 28), the microphones from Zone 4 attenuated by 9dB (attenuated in the sub-matrix XM2 in Figure 28). NOM attenuation is enabled for outputs 1, 2, and output A so the appropriate output attenuation (based on the number of open microphones from all the Zones) will be applied.

Bus Phones	-	-	-	-	-	-	-		
PB0	PB1	PB2	PB3	PB4	PB5	PB6	PB7		
0	0	0	0	0	0	0	0	PM0	SubMix Phones
0	0	0	0	0	0	0	0	PM1	-

Zone 1	Zone 2	-	-	-	-	-	-		
WB0	WB1	WB2	WB3	WB4	WB5	WB6	WB7		
	0	0	0	0	0	0	0	WM0	SubMix All Mics
	-9	0	0	0	0	0	0	WM1	Zone 1
	-9	0	0	0	0	0	0	WM2	Zone 3

Zone 3	Zone 4	-	-	-	-	-	-		
XB0	XB1	XB2	XB3	XB4	XB5	XB6	XB7		
	0	0	0	0	0	0	0	XM0	SubMix All Mics
	-3	0	0	0	0	0	0	XM1	Zone 1
	-9	0	0	0	0	0	0	XM2	Zone 3

Bus Mono/L	Bus Mono/L	-	-	-	-	-	-		
YB0	YB1	YB2	YB3	YB4	YB5	YB6	YB7		
	0	0	0	0	0	0	0	YM0	SubMix Mono/L
	0	0	0	0	0	0	0	YM1	-
	0	0	0	0	0	0	0	YM2	-

Bus Mono/R	Bus Mono/R	-	-	-	-	-	-		
ZB0	ZB1	ZB2	ZB3	ZB4	ZB5	ZB6	ZB7		
	0	0	0	0	0	0	0	ZM0	SubMix Mono/R
	0	0	0	0	0	0	0	ZM1	-
	0	0	0	0	0	0	0	ZM2	-

Figure 28. The sub-matrix for Vortex EF2280:00 which will feed the main matrix to create the local outputs for Vortex 0: Zone 1 output and Zone 3. Combination signals WM1 and XM1 contain the microphones from Zone 2 and 4 that need to be sent to the Zone 1 amplifier and combination signals WM2 and XM2 contain the mixes of microphones that need to be sent to the Zone 3 amplifier.

Safety Mute DXF Drawing... Clear Matrix

OUTPUTS

INPUTS

				Amplifier 1	Amplifier 3	-	-	-	-	-	-	To VSX 8000	-	-	-	Zone 1	Zone 3	Bus Mono/L	Bus Mono/R	Reference 1	Reference 2		
		IN	FDR	1	2	3	4	5	6	7	8	A	B	C	D	W	X	Y	Z	R1	R2	Mixer	AEC
	OUT			0	0	0	0	0	0	0	0	0	0	0	0					0	0		
	NOM			On	On	On	On	On	On	On	On	On	On	On	On								
Table Mic 1	1	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 2	2	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 3	3	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 4	4	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 9	5	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 10	6	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 11	7	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
Table Mic 12	8	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	R1
From VSX 8000	A	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	-3	-3	0	0		
-	B	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		
-	C	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		
-	D	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		
SubMix Phones	PM0			0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		
-	PM1			0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		
SubMix All Mics	WM0			0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0	0		
Zone 1	WM1			0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0	0		
Zone 3	WM2			0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0	0		
SubMix All Mics	XM0			0	0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0		
Zone 1	XM1			0	0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0		
Zone 3	XM2			0	0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0		
SubMix Mono/L	YM0			-3	-3	0	0	0	0	0	0	-3	0	0	0	0	0		0	-3	0		
-	YM1			0	0	0	0	0	0	0	0	0	0	0	0	0	0		0	0	0		
-	YM2			0	0	0	0	0	0	0	0	0	0	0	0	0	0		0	0	0		
SubMix Mono/R	ZM0			-3	-3	0	0	0	0	0	0	-3	0	0	0	0	0	0		-3	0		
-	ZM1			0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		0	0		
-	ZM2			0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		0	0		
Signal Generator	SG	-40		0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		

Figure 26. The main matrix for Vortex EF2280:00.

Vortex EF2280:01 Configuration

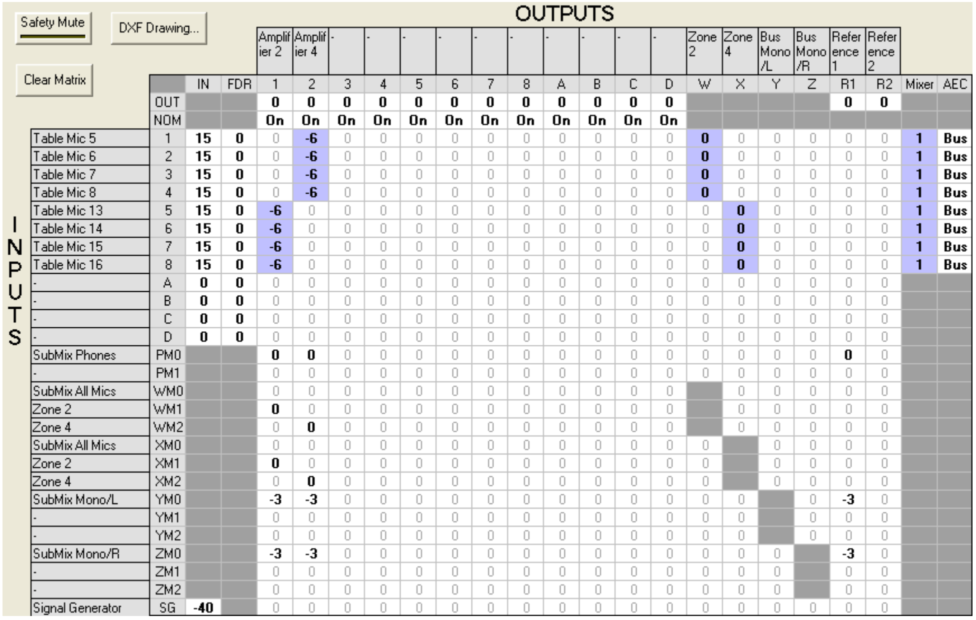
For Vortex EF2280:01, we will apply the same reasoning to create the zone 2 and zone 4 outputs. The EF bus sub matrix for EF2280:01 is shown in Figure 28.

POLYCOM

Bus Phones	-	-	-	-	-	-	-		
PB0	PB1	PB2	PB3	PB4	PB5	PB6	PB7		
0	0	0	0	0	0	0	0	PM0	SubMix Phones
0	0	0	0	0	0	0	0	PM1	-
Zone 1	Zone 2	-	-	-	-	-	-		
WB0	WB1	WB2	WB3	WB4	WB5	WB6	WB7		
0		0	0	0	0	0	0	WM0	SubMix All Mics
-9		0	0	0	0	0	0	WM1	Zone 2
-3		0	0	0	0	0	0	WM2	Zone 4
Zone 3	Zone 4	-	-	-	-	-	-		
XB0	XB1	XB2	XB3	XB4	XB5	XB6	XB7		
0		0	0	0	0	0	0	XM0	SubMix All Mics
-9		0	0	0	0	0	0	XM1	Zone 2
-9		0	0	0	0	0	0	XM2	Zone 4
Bus Mono/L	Bus Mono/L	-	-	-	-	-	-		
YB0	YB1	YB2	YB3	YB4	YB5	YB6	YB7		
0		0	0	0	0	0	0	YM0	SubMix Mono/L
0		0	0	0	0	0	0	YM1	-
0		0	0	0	0	0	0	YM2	-
Bus Mono/R	Bus Mono/R	-	-	-	-	-	-		
ZB0	ZB1	ZB2	ZB3	ZB4	ZB5	ZB6	ZB7		
0		0	0	0	0	0	0	ZM0	SubMix Mono/R
0		0	0	0	0	0	0	ZM1	-
0		0	0	0	0	0	0	ZM2	-

Figure 28. A screen capture of the sub-matrix for Vortex EF2280:01 which will feed the main matrix to create the local outputs for Vortex 1: Zone 2 output and Zone 4. Combination signals WM0 and XM0 contain the microphones from Zone 1 and 3 that need to be sent to the Zone 2 amplifier and combination signals WM2 and XM2 contain the mixes of microphones that need to be sent to the Zone 4 amplifier.

The main matrix for Vortex EF2280:01 is shown in Figure 30. In this figure, we see that the only local outputs are to the Zone 2 and Zone 4 amplifiers. The EF bus is used to send the microphone audio to the telephony interface and to send these same microphone zones to EF2280:00 to be played out into Zones 1 and 3. The output for Zone 2 (output 1 on EF2280:01) consists of the codec (from YM0 and ZM0), the phone add (from PM0), the microphones from Zone 1 attenuated by 9dB (signal WM1 from Figure 28), the microphones from Zone 4 attenuated by 6dB (attenuated in the main matrix), and the microphones from Zone 3 attenuated by 9dB (attenuated in the sub-matrix XM1 as shown in Figure 28). The output to Zone 4 (Vortex EF2280:01 Output 2) consists of the codec, the phone add, the microphones from Zone 1 attenuated by 3dB (signal WM2 in Figure 28), the microphones from Zone 2 attenuated by 6dB (attenuated in the main matrix), the microphones from Zone 3 attenuated by 9dB (attenuated in the sub-matrix XM2 in Figure 28). NOM attenuation is enabled for outputs 1, 2 so the appropriate output attenuation (based on the number of open microphones from all the Zones) will be applied.



				Amplifier 2	Amplifier 4	-	-	-	-	-	-	-	-	-	-	Zone 2	Zone 4	Bus Mono/L	Bus Mono/R	Reference 1	Reference 2		
		IN	FDR	1	2	3	4	5	6	7	8	A	B	C	D	W	X	Y	Z	R1	R2	Mixer	AEC
	OUT			0	0	0	0	0	0	0	0	0	0	0	0					0	0		
	NOM			On	On	On	On	On	On	On	On	On	On	On	On								
Table Mic 5	1	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus
Table Mic 6	2	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus
Table Mic 7	3	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus
Table Mic 8	4	15	0	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus
Table Mic 13	5	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus
Table Mic 14	6	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus
Table Mic 15	7	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus
Table Mic 16	8	15	0	-6	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	1	Bus
-	A	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		
-	B	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		
-	C	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		
-	D	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		
SubMix Phones	PM0			0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		
-	PM1			0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		
SubMix All Mics	WM0			0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0	0		
Zone 2	WM1			0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0	0		
Zone 4	WM2			0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0	0		
SubMix All Mics	XM0			0	0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0		
Zone 2	XM1			0	0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0		
Zone 4	XM2			0	0	0	0	0	0	0	0	0	0	0	0	0		0	0	0	0		
SubMix Mono/L	YM0			-3	-3	0	0	0	0	0	0	0	0	0	0	0	0		0	-3	0		
-	YM1			0	0	0	0	0	0	0	0	0	0	0	0	0	0		0	0	0		
-	YM2			0	0	0	0	0	0	0	0	0	0	0	0	0	0		0	0	0		
SubMix Mono/R	ZM0			-3	-3	0	0	0	0	0	0	0	0	0	0	0	0	0		-3	0		
-	ZM1			0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		0	0		
-	ZM2			0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		0	0		
Signal Generator	SG	-40		0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0	0		

Figure 30. The main matrix for Vortex EF2280:01.

*Instant*Designer will created all the bussing matrices and the complete system configuration that was described in this example. By simply entering in the appropriate attenuations for using the microphones in the different loudspeaker zones, the overall configuration was automatically created.

POLYCOM®

18 VORTEX EF2280 SPECIFICATIONS

MECHANICAL SPECIFICATIONS

Dimensions:	19" (483 mm) W x 9.6" (244 mm) L x 1.75" (45 mm) H (full rack unit)
Weight:	4 lb. (1.8 kg) dry
	5.5 lb. (2.5 kg) shipping
Connectors:	Audio: Mini (3.5mm) quick connect terminal blocks
	RS-232: DB9F
	EF Bus In/Out: RJ45
	Control/Status: DB25F

ELECTRICAL SPECIFICATIONS

Power:	110 - 240 VAC; 47-63 Hz
Power Consumption:	30 W

AUDIO I/O

Microphone input gain:	0 to 30 dB Mic/Line inputs in 1 dB steps, software adjustable
Mic/Line switch gain:	33 dB
Phantom Power:	48 V DC, software selectable
Line input gain:	0 to 20 dB in 1 dB steps, software adjustable
Maximum input amplitude:	+19 dBu, 1% THD+N
Nominal input level:	0 dBu (0.775 V rms)
Equivalent input noise:	<-124 dBu, 20–20,000 Hz
Input Impedance	10 kOhms
Output Gain:	-100 to 20 dB in 1 dB steps, software adjustable
Maximum output amplitude:	+23 dBu, 1% THD+N
Nominal output level:	0 dBu (0.775 V rms)
Output Impedance	50 Ohms (drives 600 Ohms)

PERFORMANCE SPECIFICATIONS

Frequency Response	20-20,000 Hz, +0.2/- 0.3 dB
Idle channel noise:	<-100 dB FS "A" weighted, 20-20,000 Hz
Dynamic range:	>100 dB FS "A" weighted, 20-20,000 Hz, 0dB gain
Linearity:	0 dB FS to -110 dB FS +/- 1dB
THD+N:	<-90 dB FS
Acoustic echo cancellation span:	270 ms
Total Cancellation:	> 65 dB
Convergence Rate:	40 dB/second
Noise Cancellation:	0 to 15 dB (continuous), software selectable
Operating Temperature:	0° - 40° C
Control Inputs:	Contact closure
Status Outputs:	5V, 20 mA each

COMPLIANCE	The Vortex® EF2280 complies with the ITU G.167 Recommendation for AEC, FCC part 15, and CE requirements.

Specifications on other Vortex products can be found on the rear page of the product data sheets.