Polycom® Vortex® Products

Frequently Asked Questions

What is the Vortex line?

Polycom Vortex products provide advanced acoustic echo and noise cancellation, automatic microphone mixing, room zoning, and other functions for installed conferencing and collaboration applications.

Designed specifically for system integration, the Vortex line installs easily with a wide range of video codecs, room audio components, and controllers such as AMX® and Crestron®.

What products are available in the Vortex line?

Vortex EF2280 - provides 8 mic/line level inputs, 4 line level inputs, 12 line level outputs

Vortex EF2241 - provides 4 mic/line level inputs, 4 line level inputs, 8 line level outputs, a 10 Watt power amplifier, and a DSP-based telephone hybrid for phone adds

Vortex EF2211 - provides 1 mic/line level input, 2 line level inputs, 3 line level outputs, a 10 Watt power amplifier, and a DSP-based telephone hybrid

Vortex EF2210 - provides 1 mic/line level input, 2 line level inputs, and 3 line level outputs

Vortex EF2201 - a DSP based telephone hybrid for phone adds, designed to work in conjunction wih other Vortex products

What do the Vortex model numbers mean?

The Vortex products are numbered EF22XY where:

EF = Echo Free (a Polycom trademark)

22 = 22kHz wideband audio

X = the number of microphone inputs

Y = the number of phone adds (analog telephone lines)

For example, the Vortex EF2241 provides 22kHz wideband audio, permits direct connection of 4 microphones, and has an integrated phone add.

When should I use a Vortex product vs. tabletop?

A Vortex solution should be selected when you need to have multiple microphones around the room, including wireless and ceiling microphones, or distributed loudspeakers to provide full room coverage.

In larger rooms, 20'x20' and above, or when there are many participants, additional microphones and loudspeakers may be required to allow all participants to hear and be heard clearly. Also, if multiple telephone lines are required to allow multiple simultaneous callers, the Vortex products should be used as they allow you to have up to 8 simultaneous telephone callers. If there are requirements for wireless microphones, podium microphones, or multiple video codecs, a Vortex should be used.

When using video conferencing, the Vortex will improve the audio quality of your installations and can be used with nearly all Polycom video codecs.

If the highest audio quality is a requirement, the Vortex will give you the highest quality, the most flexibility, and the expandability to support your future audio and conferencing needs.

How do I know which Vortex product to specify for my application?

If you are designing a room with several microphones, plan on using the Vortex EF2241 or Vortex EF2280. These products provide individual channel acoustic echo and noise cancellation, giving you the best possible voice quality for multi-mic applications.

If you have a mixer already and would like to add teleconferencing, the Vortex EF2211 (with built-in telephony interface) or Vortex EF2210 will do the job nicely.

To add PSTN (analog) telephone calls to a voice or video conference, you can use the Vortex EF2241, Vortex EF2211, or the Vortex EF2201. Vortex products can also interface with the Polycom SoundStation VTX 1000™, which gives users a familiar telephone controller and can provide wideband (7kHz) telephone audio when connected to another SoundStation VTX 1000 on the far end.

How many Vortex products can be linked together?

Up to 8 units* can be linked. This means you can use up to 64 microphones in a system using 8 Vortex EF2280 units. NOM (number of open microphones) information can be specified across all linked Vortex units.

*The Vortex EF2280 and Vortex EF2201 phone add can share the same device ID, allowing you to have up to 8 Vortex EF2280 units AND 8 Vortex EF2201 units, if desired.

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What other Polycom products can be interfaced to Vortex products?

The Polycom SoundStation VTX 1000 wideband conference phone can be used to add wideband telephony audio to a Vortex. The SoundStation VTX 1000 user interface can be used to dial phone calls, mute the audio, and change the volume of the incoming wideband telephone audio.

In addition, the new Polycom VSX™ 8000 video conferencing codec interfaces easily to the Vortex, with its infra-red remote control capable of executing commands within the Vortex for volume control and muting Vortex microphones.

All Polycom video codecs, as well as those made by other companies, interface easily with Vortex products, and Vortex can also be easily controlled with AMX or Crestron control systems.

Are Vortex products difficult to install and set up?

Not at all! All connections use industry-standard connectors (Phoenix, DB9, etc.), and our *Instant*Designer™ setup wizard makes it possible to configure even the most complex systems in a short time.

How does the Polycom *Instant*Designer wizard work?

A recent addition to our Conference Composer™ design software for Windows®, the patent pending Polycom *Instant*Designer guides the system designer through the steps required to create high performance Vortex audio conferencing and sound reinforcement solutions.

The A/V integrator or consultant simply chooses the necessary inputs, outputs (including the new Polycom VSX 7000 or VSX 8000 video codecs) and optionally the sound reinforcement zones needed for the desired system, and Polycom *Instant*Designer automatically selects the appropriate Vortex devices necessary to implement the system, maps the inputs and outputs to the devices, and creates the Conference Composer design files required to implement the design. It is easy to upload these design files to the Vortex devices to complete the configuration. All of this can be done within a matter of minutes.

The Polycom *Instant*Designer handles all the details of creating zone to zone gains for sound reinforcement, configuring the acoustic echo cancellers, interfacing to common audio and video equipment, bussing between devices, configuring presets, creating volume control macros, and preconfiguring logic ports for push to talk microphones.

For more information, see the *Instant*Designer application note (it may be downloaded in PDF format via the Polycom Resource Center on the internet).

How do I get a copy of the *Instant*Designer software?

The *Instant*Designer software is included in the Conference Composer software and can be downloaded from the Polycom Resource Center at http://extranet.polycom.com.

Is a remote control available for Vortex products?

Yes, the optional Vortex EF-IR11 infrared controller provides direct control of the Vortex EF2280, EF2201, EF2211 and EF2241. It can be used to adjust volume, mute and unmute audio, activate a phone line, dial out or answer calls, or execute any of 35 macros that are user-programmed into the Vortex units. And, as noted earlier, Vortex products are easily controlled with external room controllers such as Crestron or AMX.

What makes the Vortex Acoustic Echo Cancellation (AEC) better than competing units?

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 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. Vortex products also have the fastest convergence speed of any acoustic echo canceller on the market.

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 Polycom's Vortex sound better than other units?

There are several reasons including the full bandwidth of the signal (22kHz), the digital trim pots that adjust the analog gain before the A/D converter (this maximizes the signal to noise ratio), and our patented ambient noise cancellation. The Vortex's neural network-based AGC also helps the AEC performance by bringing all signals to the proper level. Vortex units also provide better room gain performance than competing products.

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. 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), many AECs have a tougher time removing the echo and may become less full duplex.

For more information on room gain, see the Polycom application note on the Vortex EF2280.

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

Many products remove noise by filtering. When this happens, they will also remove any signal in the same frequency band as the noise. The patented Polycom 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.

The Polycom noise canceller works best with steady state signals, i.e., signals that are random, but their randomness can be thought of as fixed. Example 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. 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 Polycom 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.

How does the Polycom AGC (Automatic Gain Control) work, and how is it different from competing products?

Polycom'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. The Polycom algorithm looks at the input signal and makes a determination as to whether the signal is a valid speech input or not. Once it finds valid speech (and it is always looking for valid speech), it will estimate the nominal level of the incoming signal. It will then determine what level to apply to the signal to get it to a 0dBu nominal level. 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.

What type of automixers do the Vortex EF2280 and Vortex EF2241 use?

The automixer -- or more accurately, both automixers, since you can split the mixer in each product-- in the Vortex EF2280 and Vortex EF2241 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 in the Vortex EF2280 (4 in the Vortex EF2241) and produces the same number of line output signals and gain information to be applied to each of these gated outputs as they are used in the matrix. 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 types of filters and other parameters do the Vortex products provide?

User-programmable delays on all outputs, 5-band parametric EQ on all inputs and outputs, 256 macros, 16 factory presets, 32 user-configurable presets, system diagnostics and other tools are provided to ensure the best performance for all room conditions.

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