Unified communications solution for SMB

iPECS

ACT-50

IP AUDIO CONFERENCE TERMINAL

A USER-FRIENDLY MULTI-INTERFACE VOICE CONFERENCING SYSTEM WITH RICH AND CLEAR SOUND.

LG-Ericsson's new iPECS Audio Conference Terminal delivers high performance audio conferencing with multiple sources. Connect to a LAN and ACT-50 is ready to bring your conference experience to a new level with advanced audio technology and high-fidelity sound.

Easy on talkers

No need to raise your voice to be heard. The top mounted array of 16 microphones reliably reproduces your voice from 2 to 20 feet away. Also, the high performance adaptive echo canceller prevents annoyances such as echoes, cut-outs, and drops in volume when participants speak at the same time, ensuring that your conference proceeds smoothly. For larger conference rooms, daisy chain ACT-50s for great performance even in difficult audio environments.

Easy on listeners

Audio is delivered by the 4 speaker array, reproducing audio with rich and clear sound.

A selection of terminals support many types of conferences

ACT-50 has three built-in interfaces. The LAN port provides high sound quality conferencing between conference phones. The LINE port (for analog phone lines) facilitates emergency calls with land line phones, cell phones, and analog phone terminals. And to connect to a PC as a microphone speaker for Web conferencing or soft phone use, you can use the AUDIO(IN/OUT) port. ACT-50 offers a multi-interface voice conference system that can be used in various ways according to the application. [Figure 1]

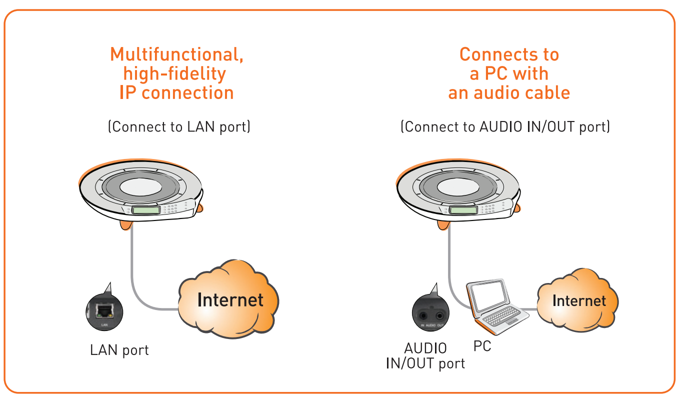

Figure 1

Easy to connect

ACT-50 quickly connects to an IP network to establish a VoIP call. In addition, a standard audio input can connect to external sources such as a PC softphone. Also, the high performance adaptive echo canceller facilitates conferencing in a variety of environments from large, open-space offices to small meeting rooms with high reverberation.

Integrated audio mixer enables conferencing with mixed line types connects to a PC with an audio cable

With its various interfaces, ACT-50 can open calls with IP, analog phone, and audio connections. Parties away from the office can join IP audio conferences from their cell phones, or join Web conferences from a telephone (voice only). [Figure 2]

Microphone mode

ACT-50 provides 3 microphone modes for various environments.
The default "Zone mode" is used for quiet environments and picks up audio from all directions. Spot mode is used when the number of talkers is limited to one or two, or when there is equipment that produces noise such as a projector nearby. ACT-50 fixes the audio pick up area to the front of the microphone by narrowing the directivity. Tracking mode automatically tracks and focuses on the audio of the talker. The audio can be picked up with narrow directivity so that other noises can be reduced. This mode is suitable to pick up the audio clearly in a noisy environment.

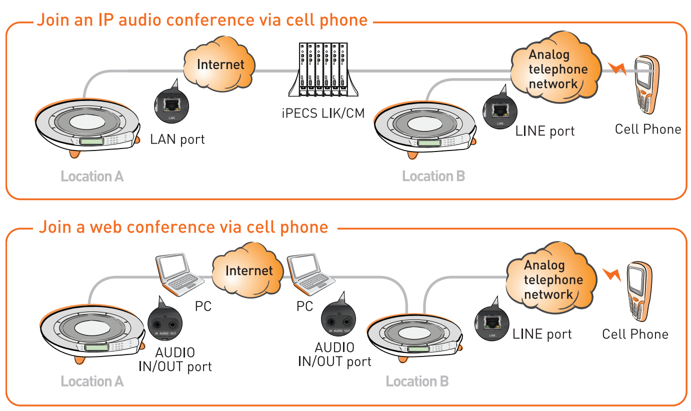

Figure 2

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1. LINE input
2. LAN port
3. DC IN 12V
4. AUDIO IN
5. AUDIO OUT

- DHCP and static IP support
- Time synchronization using the SNTP server (Time zone setup, SNTP interval setup)
- Standard SIP compliance
- Port number for SIP : 5060 (UDP)
- Port number for RTP/RTCP : 57000 to 57010 (UDP)
- Echo length control (Room size = Large/Medium/Small)
- Silence suppression
- Natural Voice Enhancer feature (high frequency boost)
- MIC/audio-in/audio-out gain control
- Ring tone volume control
- Speaker volume control
- Arrayed microphones (directivity controlled)
- Zone audio pickup function
- Spot audio pickup function
- Call history (50 calls)
- Address book (16 SIP addresses, 50 PSTN addresses)
- LCD back-light/contrast control
- Manual F/W update
- Web GUI
- Password protection
- Syslog support
- PSTN: DTMF and pulsed line support
- PSTN: hook time control
- External audio support
- Daisy chain expansion

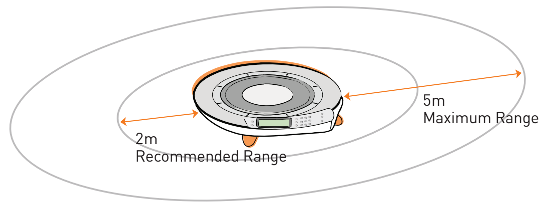
FUNCTIONS

Microphone **16** | Speaker **4** | Max. volume **85dB** | Pick up reg* Rec.2m Max.5m

- *Varies depending on communication methods and environment

Adaptive echo canceller | Noise reduction | SIP compliant | Zone Pick up | Spot Pick up | Microphone Auto tracking | Supports RoHS

PICKUP RANGE IMAGE



General	External interfaces	Ethernet(10/100Base-TX), Analog phone/modular jack, Stereo analog audio in/out x 1 each (mini-jacks), AC adapter connector (DC-12V IN): for the provided AC power adapter
	Power consumption	Approx. 8W
	Radio interference standard	FCC part 15 (US), EN 55022 (EU)
	Operating environment	Operating temperature:0-40℃ (32-104°F), Operating humidity:20%-85% (no condensation)
	Size	283.4(W) x 51.5 (H) x 297.5 (D)mm
	Weight	Approx. 1.4kg (excl.power adapter)
	Power supply	100 to 240 V AC (50/60 Hz)
	Accessories	AC adapter, power cable, LAN cable, phone cable, owner's manual
	Other	Logging facility (SYSLOG), Firmware revision update (HTTP, TFTP) Settings (Web menu and front panel key operations)
Audio	Arrayed microphones	Zone audio pickup function, spot audio pickup function, microphone auto tracking function
	Arrayed speakers	Output level: 85dB
	Frequency range	300-7000Hz(wide band)
	Signal processing	Adaptive echo canceller, noise reduction, microphone/speaker array control
Communications	Supported audio codec	G.722 G.711 A-Law / μ-Law G.729
	Other functions	Compliance with DHCP, and SIP: internal clock time synchronization with SNTP server

	Connecting locations	Mesh connection (max. 4 locations), cascade connection (max. 8 locations)			
Communications	Audio Codec Required Transmission Bandwidth	G.722	G.711 (μ-Law/A-Law)		G.729a
		160kbit/s	90kbit/s		24kbit/s

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