# EBS-E1 VoIP

VOIP CHANNELS EXTERNAL BOARD WITH 1 TO 4 E1/T1 LINKS


## Main Characteristics

- 30 channels via SIP protocol for each E1/T1 link
- DSPs for processing of audio and signaling
- Web interface for control, visualization and download of logs
- Classification of call answering (HMP Analytics)

## Typical Applications

- PBX
- IP PBX
- Gateway
- IVR
- DAC

## Models

- EBS-E1 3300, with 1 E1/T1 + 30 VoIP channels
- EBS-E1 6600, with 2 E1/T1 + 60 VoIP channels
- EBS-E1 9900, with 3 E1/T1 + 90 VoIP channels
- EBS-E1 13200, with 4 E1/T1 + 120 VoIP channels

## Overview

The EBS-E1 family with SIP channels is designed to add high availability to the robust CTI solutions with an IP network. Maintaining the standard and quality of the Khomp family of boards, the EBS-E1 VoIP performs all of the audio and signaling processing within its own hardware. Its voice resources, which include detection of voice mail, detection and suppression of DTMF and AGC, call progress, reproduction and recording of audio messages, detection of fax signals, conference, among other items, are performed in the module's DSP, in addition to echo canceling, which is critical in IP telephony solutions. This robust architecture allows for the use of EBS for interconnecting up to 1000 TDM channels (34 E1/T1) with VoIP channels in a single server *.

Audible Response Unit (IVR), Telemarketing, Help Desk, Customer Service, Voice Mail, Call Center, Conference, IP PBX, among other features, are examples of applications in which the EBS-E1 VoIP can be used.

** Consult with the Khomp technical team to find out about servers certified with this density*

## **Exclusive Resources** of the EBS-E1 VoIP:

- Network channels: 1 to 4 E1/T1 links
- VoIP channels: 30 channels via SIP protocol for each E1/T1 link
- Network protocols: ISDN and R2 Digital (with 30 MFC exchangers per E1/T1 link)
- PBX Protocols: EL7, Line Side, LC and QSIG (SSCT and CT)
- Optional EBS-E1 Bypass contingency board for each 2 E1/T1, for handling failures.
- Reports call monitoring events in R2D and RDSI signaling to the application, including number of A and B

Physical characteristics:

- Connectors: 75 Ohms BNC or RJ45 connectors
- Weight: from 2.60 to 2.78Kg

# Resources available on the entire EBS family of products

## Voice processing

High capacity resources:

- All voice resources available simultaneously on all channels
- DSPs for processing of audio and signaling

Detection and generation of tones (DSP)

- MFC exchange (R2 signaling)
- Detection and generation of DTMF digits, fax tones, 425Hz (dial tone) and TDD messages *(Telecommunications Device for the Deaf)*
- Detection of intercept tones (voice mail, collect calls)
- Generation of programmable tones (beep)
- Detection of silence and presence of audio before and after answering
- Detection of fax signal and voice mail with standard signaling: 600Hz/450ms – 1000Hz/450ms or 300Hz/250ms
- Detection of programmable frequencies (for example: portability tone, non-standard voice mail, etc)

Audio enhancement features

- DTMF suppression
- Manual and automatic volume control (AGC)
- *Carrier grade* echo canceling in hardware
  - Up to 64ms (512 TAPS) simultaneously on all channels, independent of other resources
  - Convergence and automatic delay adjustment during the entire call
  - Compatible with ITU-T G.165 and G.168 norms (2000 and 2002)

Call signaling and handing

- Detection of collect calls through recognition of tones, signaling or double answering

## Features programmable via API K3L

Switching of channels:

- Conference calls with up to 5 participants between any channels
- Full commutation between all channels and between modules

Recording and reproduction of voice messages

- Full-duplex mono or stereo recording
- Codecs available for recording and reproduction: G.711 (A-law and μ-law), GSM and ADPCM, PCM8, PCM16 and AMR.
- Reproduction of messages (play) in the PCM8, PCM11, PCM16, A-law and μ-law, GSM and DVI4 (ADPCM) formats

## VoIP channel features

- All voice resources available for network and VoIP channels
- VoIP calls use the host Ethernet port (fast or giga ethernet)
- Codecs available for VoIP: G.711 (A-law and μ-law), ADPCM, GSM, iLBC

## OAMPT

- Automated installer for updating and implementing new systems
- Web system for configuration, monitoring and diagnostics
- Native integration with SNMP
- Signaling analyzer
- Remote monitoring in real time (via web)
- Web interface for control, visualization and download of logs

- Call progress for generation of call control events in FXO interfaces and PBX protocols
- Classification of call answering *(Call Analyzer)*

## High availability

- 2 Ethernet ports for server connection (network redundancy)
- Server redundancy (supports virtual IP)

## Physical Characteristics

- Standard 1U Module and 1/2 19" rack
- Measurements in mm: 44.5 (height) x 220.5 (width) x 280 (length)
- Power source: Full Range (100~240Vac - 50/60 Hz)

## Guarantees and Certifications

- Factory warranty 3 years
- The entire EBS line is Anatel certified
- ISO 9001:2008 Industry certified

# Additional product images

Rear view of model with 4 E1/T1 links (BNC).

Rear view of model with 3 E1/T1 links (BNC).

Rear view of model with 2 E1/T1 links (BNC).

Rear view of model with 1 E1/T1 link (BNC).

Rear view of model with 4 E1/T1 links (RJ45).

# EBS-E1 VoIP

Example of 7 EBS modules arranged in a standard 19'' rack

## Application Model