EBS-FXO

4, 8 OR 12 ANALOG LINE EXTERNAL BOARD

www.khomp.com

Main Characteristics

- 4, 8 or 12 analog channels
- 1 HMP channel for each analog channel (when used witch CTI solutions)
- DSPs for processing of audio and signaling
- Web interface for control, visualization and download of logs
- Classification of call answering (Call Analyzer)

Typical Applications

- PBX
- IP PBX
- Gateway
- IVR
- DAC

Models

- EBS-FXO 40, with 4 FXO interfaces + 4 VoIP channels
- EBS-FXO 80, with 8 FXO interfaces + 8 VoIP channels
- EBS-FXO 120, with 12 FXO interfaces + 12 VoIP channels

Overview

The EBS-FXO was developed for CTI applications that require analog trunking. Added value platforms, Audible Response Unit (IVR), Telemarketing, Voice Mail, Conference, IP PBX, and Help desk, among other items, are examples of applications in which the EBS-FXO can be used.
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, among other items, in addition to echo canceling, are performed in the hardware without consuming the processing capacity of the host. This robust architecture allows for the use of the EBS-FXO in high density applications using servers with less density.

Exclusive Resources of the EBS-FXO:

- Network channels: 4, 8 or 12 analog channels
- 1 HMP channel for each analog channel (when used witch CTI solutions)
- Possibility for expansion of SIP channels with the acquisition of additional licenses.
- Network protocols: FXO
- PBX Protocols: Flash generation and detection
- Assistant for configuration of call progress.
- FSK detection for call identifier (Bina)
- Reports inversion of polarity
- Reports failure of physical call on the line

Physical characteristics:

- Connectors: RJ11
- Weight: from 2.08 to 2.21Kg
- Maximum power consumption: xxxx

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, etc.)
- 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

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

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- 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

EBS-FXO

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

Rear view of model with 8 FXO

Rear view of model with 4 FXO

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

Application Model