Khomp EBS-E1 SPX 900, With 3 E1s Gateway

Product Name: Khomp EBS-E1 SPX 900, With 3 E1s Gateway

Manufacturer: -

Model Number: EBS-E1-SPX-900

The Khomp EBS-E1 SPX 900, With 3 E1s Gateway was developed for use in applications with PABX open source software.Features and Benefits

• 3 E1 digital interfaces
• Compatible with Asterisk and FreeSWITCH
• 2 Ethernet ports for connecting to server
• Network protocols implemented by the board: RDSI, SS7 (ISUP) and R2 (with up to 120 MFC signal changers). You can set a different protocol on each link
• PABX protocols implemented by the board: EL7, Line Side, LC and QSIG (SSCT and CT)

Critical services related to PABX digital intersection, such as signals (R2D or ISDN) interchanging with public central, DTMF dialing detection, tone generating (DTMF, MFC and 425Hz) and echo blockage are run by DSPs on the module. EBS has high resource availability for handling these services in real time and in all channels simultaneously, without using the processing capability of the host.The Khomp EBS-E1 SPX 900, With 3 E1s Gateway was also developed by Khomp to meet a demand of work expansion, with more channels or different interfaces, without the need to stop the operation for opening the server or worry about the connection bus, ensuring continuity and reliability of the solution.Together with EBS, Khomp provides drivers for integration with main open source PABX apps, such as Asterisk and FreeSWITCH
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• It generates DTMF and 425Hz signals
• DTMF dialling detection, even on message playing
• Detection of fax signal and inbox. Standard signalling (600Hz/450ms – 1000Hz/450ms)
• Silence and audio presence detection before and after answering
• DTMF dialling suppression in set calls
• Information about signals and channel status is reported through AMI interface
• Answering detection available via dial plan and AMI interface
• Signalling specific commands available via AGI and AMI interfaces
• Balance for calls between channels with one or more output routes
• It can control and generate call control tones
• Selective Drop Collect Call, based on SoftPBX dial plan
• Full commutation between all channels and between modules
• Echo blocking on all channels simultaneously, regardless of using other resources
• Echo blocking of up to 64ms (512 TAPS) per channel, included
• Echo blocking compatible with ITU-T G165 and G.168 (2000 e 2002) standards, with automatic delay control and convergence during the call
• Echo blockage is effective and carrier grade, ensuring a clear communication with excellent audio quality
• Available with 75 Ohms BNC or 120 Ohms
• RJ45 connectors
• Standard 1U module and 1/2 rack 19"
• Power Supply AC 110/220V or DC 48V
• Dimensions in mm: 44.5 (height) x 220.5 (width) x 280 (length)

Please Enquire