JUSTEC SOLUTIONS®

JIP4002

CONVERGING PSTN AND VOIP SERVICES INTO ONE NETWORK

JIP4002 is a two-port FXS + FXO gateway and support H.323 v2/v3/v4 or SIP RFC 3261protocol. It supports an innovative intelligent call routing function that transparently routes calls to destination either through PSTN or internet. JIP4002 provides Voice over IP and FAX over IP services for ITSP / ISP (Internet Telephony Services Provider) and Office / SOHO IP-PBX application.

BENEFITS

- Easy access to IP from phone set or PBX
- Cost Saving - Telephone call from VPN or public Internet
- Follows the existing telephone call dial plan
- Easy interface to ADSL/Cable Modem or Leased line equipment

SPECIFICATIONS

MODEL	JIP4002
Protocol	• ITU-T H.323 v2/v3/v4 compliance SIP RFC3261
Supplementary service	• H.450 (Call Hold, Call Transfer, Call forward)
FAX support	• Automatically FAX detection • Group 3 Fax relay at 2.4 - 14.4 kbps • Support T.38 protocol • Support T.38 ECM: Error correction during the high speed mode • Support T.38 FAX Redundancy Depth • Support Abstract Syntax Notation 1 (ASN.1) • G.711 FAX Mode
Codec	• G.711A/μ-law, G.723.1, G.729A, G.729
Network Interface	• Two 10/100Base-T Ethernet RJ-45 port
FXS Interface	• One Telephone (FXS) RJ-11 port for JIP4001 • Two Telephone (FXS) RJ-11 ports for JIP4002
FXO Interface	• One Analog PSTN (FXO) RJ-11 port for JIP4001 • Two Analog PSTN (FXO) RJ-11 ports for JIP4002
FXS features	• Support auto-attendant (Tone or voice greeting) • PSTN polarity reversal detection • Provide 2nd dial tone to PSTN • Disconnect tone detection
FXO features	• Support auto-attendant (Tone or voice greeting) • PSTN polarity reversal detection • Provide 2nd dial tone to PSTN • Disconnect tone detection
Voice Quality	• VAD (Voice Activity Detection) • CNG (Comfort Noise Generation) • AEC (Acoustic Echo Cancellation) -- G.168/165 • Dynamic Jitter Buffer • Completed voice band signaling support • Provide In-band or Out-band DTMF generation/ detectio
QoS	• QOS by setting DSCP (Differentiated Service Code Point) parameters of VoIP packet
Console	• 1 D-SUB 9 pin RS-232 port
Tone	• Provide call progress tone
TCP/IP	• Support Fixed IP and DHCP
ADSL Environment	• PPPoE

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SPECIFICATIONS

MODEL	JIP4002
Firewall Environment	• Behind NAT Router or IP sharing device
Configuration & management	• Console port, TELNET and Web Browser configuration
Upgrade	• TFTP/FTP software upgrade
Power	• Input AC 100V~240V Output DC12V.
Operation Temp	• 5° C to 40° C
Humidity	• 10% to 90% (Non-condensing)
Chassis Dimension	• 223mm(W) x 35mm(H) x 152mm(D)
Weight (unit)	• 1.5 kg

ORDERING INFORMATION

Model	Description
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