RP-FSO522

2-Line FXO, 2-Line FXS SIP IP Gateway

RP-FSO522 is an 2-Line FXO plus 2-Line FXS gateway with SIP protocol IP device which allows to connect 2 Lines of analog PSTN telephone line and connect to 2 Lines analog telephone set (for instance, analog trunk line of PABX) to make or receive VoIP call over Internet or VPN network. This device is suitable for office PABX to enable VoIP call without changing cabling, dial plan, extension number.

Feature

- Dual IP Stack : IPv6 and IPv4 Simultaneously
- Support up to 4 SIP proxy Servers
- Enhance existing PABX to VoIP Call
- Auto HTTP Provision feature
- Flexible Routes Plan and Dial Plan, Digit Manipulation
- Redundant Firmware Image

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Specification

Interface	• Ethernet port (RJ-45, 10/100 base-T): ▪ WAN port, connect to IP Network ▪ LAN port connect to PC with NAT • Support Bridge, NAT and Gateway mode • Telephony port connect to local PSTN line (RJ-11 x 4 pcs) • DC +12V power input Jack • Reset key to return Factory setting • LED Indicator for System, SIP, FXS and FXO status
IP Network connection	• IPv4 (RFC 791) and IPv6 Simultaneously • IPv6 Auto Configuration (RFC 4862) • IPv6 Only, IPv4 Only or dual stack • MAC Address (IEEE 802.3) • MAC Clone Setting • Vendor Class ID • IP/ICMP/ARP/RARP/SNTP • Static IP • DHCP Client (RFC 2131), WAN port • DHCP Server, LAN port • NAT Server (RFC 1631) • PPPoE Client • DDNS (DynDNS) • DNS Client • Firewall • URL Filter • IP Filter • MAC Address Filter • Application program Filter • Port Filter • Port Forwarding (TCP, UDP or both) • Bandwidth Control (Download and Upload), Maximum Bandwidth priority setting • UPnP Server at LAN port • Behind NAT, use DMZ for NAT traversal • SNTP with time zone and Daylight Saving • TCP/UDP (RFC 793/768) • RTP/RTCP (RFC 1889/1890) • IPV4 ICMP (RFC 792), • TFTP Client • VoIP VLAN Support 802.1Q, 802.1P • VLAN ID Range : 2 to 4094 • VLAN Priority : 0 to 7 (Highest Priority) • QoS : DiffServ (RFC 2475), TOS (RFC791, 1394)
SIP Protocol	• RFC3261 compliance • Support up-to 4 SIP Trunk to Register • SIP UDP Protocol • Support SIP compact Form • Support SIP HOLD Type: Send Only, 0.0.0.0 or inactive • SIP Session Timer (RFC 4028) • SIP Session Refresher: UAC or UAS • SIP Encryption • MD5 Digest Authentication (RFC2069/RFC2617) • Reliability of provision response PRACK (RFC3262) • Early/Delay Media support • Offer/Answer (RFC3264) • Message Waiting Indication (RFC3842)

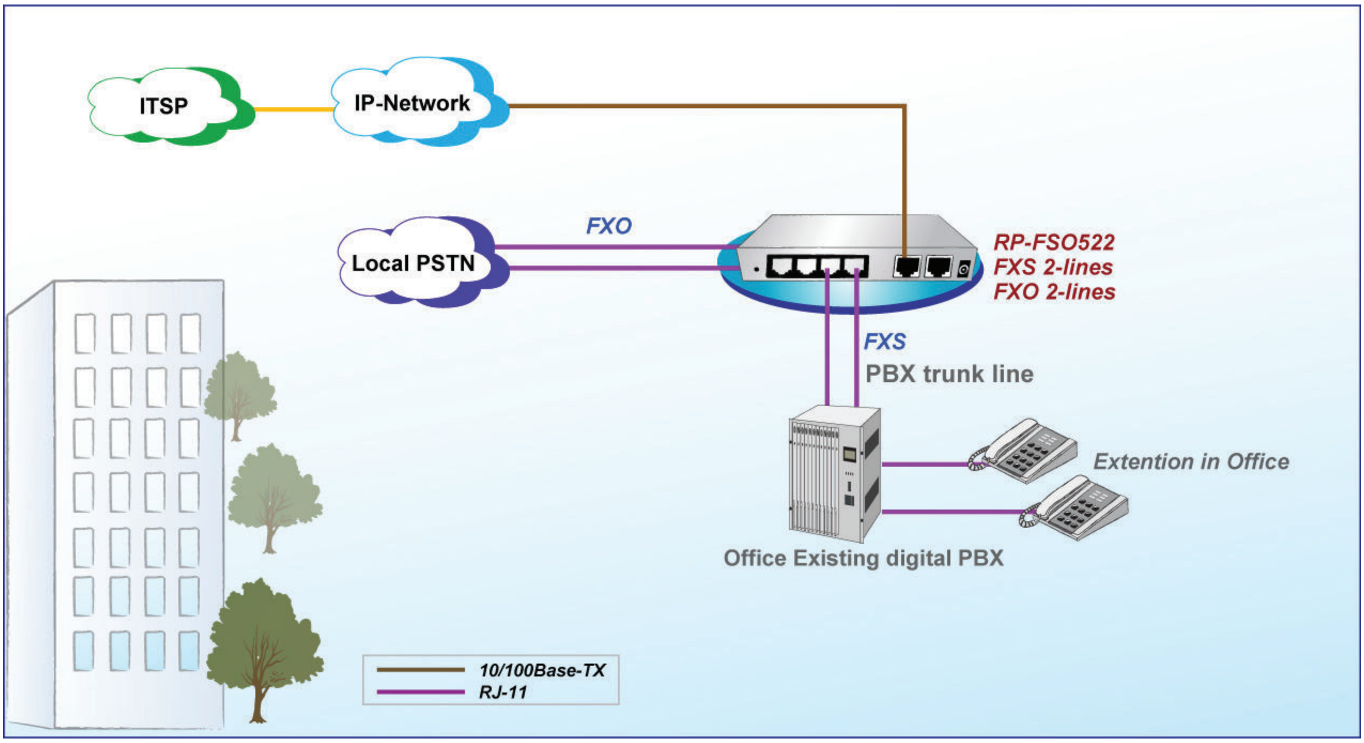
	• Event Notification (RFC3265) • REFER (RFC3515) • Support Outbound Proxy • Support Primary and Backup SIP Server • Support STUN NAT Traversal • Support “rport” parameter (RFC 3581) • Configure SIP local Port • SIP QoS Type: DiffServe or QoS • Accept Proxy Only : YES or NO
Audio Codec	• G.711 A-law/µ-law, G.729A, G.723.1 (6.3K, 5.3K) • Select voice codec priority : Local or Remote • Voice Payload size (ms) configuration • Silence Suppression • VAD/CNG • LEC : Line Echo Canceller • Max Echo Tail Length (G.168): 32, 64 and 128ms • Packet Loss Compensation • Automatic Gain Control • In-band/out of band DTMF (RFC4733, RFC2833 / SIP INFO) • Adaptive/Configurable Jitter Buffer • G.168 Acoustic Echo Cancellation • Configure RTP basic Port • RTP QoS Type : DiffServ or TOS • Phone Book (50 records) for peer to peer calls • Dialing Plan with drop, replace, Insert dialing digits • Select First digit and Inter digit timeout duration (Sec) • Selectable Call Progress Tone • Support Specified Line Calling
Call Features	• Support Peer to Peer Dialing • 2-Line FXO connects to PSTN Line • 2-Line FXS connects to analog phone set or PABX. • Caller ID recognition DTMF (before/after 1st ring) and FSK (before 1st ring), ETSI and Bellcore • DTMF Caller ID start and stop BIT configurable • Current Drop Detection to release FXO port • Disconnect tone recognition to release FXO port • Tone Generation: Ring Back, Dial, Busy, call waiting, ROH, Warning, Holding, Stutter dial tone and disconnect tone • Configure Tone Frequency, Cadence, Level and Cycle • Select Tone specification by Country name List • Global Country Based Tone Specification • NAT Traversal support STUN, UPNP and Behind NAT • Out-Band DTMF : RFC2833 and SIP Info • RFC2833 Payload type : 101 or 96 • DTMF send out ON and OFF Time configure • DTMF incoming recognition Minimum ON and OFF time • DTMF Relay Volume configuration • T.38 FAX Volume configuration • Flash Time transmit via SIP Info (Enable or Disable) • Message Waiting Indication (Stutter Tone Notice) • Block Anonymous Call • Call Hold • Call Transfer
FXO/FXS Line Configuration	• Activate or deactivate • Line ID • Line Phone number • Polarity Reversal detection or generation for call establish and Billing

	• Current drop recognition or generation to release line • Incoming call Handle: Hotline or 2 stage dialing • HOT Line to desired phone number • Play voice file to incoming call • Repeat playing voice file counts • Self-recorded voice files to upload • Generate FLASH TIME to PSTN network • T.38 or FAX Relay Type • Incoming and outgoing dB value configurable • Dialing Answer Delay time to establish call path • Answer PSTN incoming call after how many ring cycles • Caller ID detection mode by Country selection • VoIP dial to FXO/PSTN Line: 1 stage dialing and 2 stage dialing • Outgoing SIP Caller ID Selection • Support 4 SIP Trunk • Accept desired SIP Proxy incoming calls Only
Flexible Routing Plan	• Prefix Match and Length • Priority Ring • Cyclic Ring • Simultaneous Ring • Programmable Hunting Cycle • Backup Routes with Digit Manipulation • Default Routes
Flexible Dial Plans	• Retrieve transfer call from 3rd party by dial Code (default: *#) • Inter digit time out setting • First digit dial out delay time setting • End of dial keypad number • Dial Rule : Match dial Prefix and Maximum digits length (1-15) • Phone Book can be Exported or Imported
Digit Manipulation (Drop and Replace Rule)	• FXO DM Group • FXS DM Group • VoIP DM Group • DM 1 Group • DM 2 Group • DM 3 Group • DM 4 Group • Matched Prefix • Matched digit length • Replace digit start position • Replace digit stop position • Replace number
FXS Analog 2-wire interface	• Flash Time Detection: range from 80 to 800 ms • ON-HOOK Voltage -48Vdc • Configure Ring Cadence, Frequency and Voltage • Support Polarity reversal for Billing • Service Up to 1 Kilo-meter distance to analog telephone set • Generate Current Drop Time (Open Loop Disconnect time)
FXO Analog 2-wire interface	• Incoming Ring frequency recognition range: 10 to 70 Hz • Incoming Ring ON time recognition range: 0 to 8000ms • Incoming Ring OFF time recognition range: 0 to 8000ms • Incoming Ring Level recognition range: 10 to 95Vrms • Flash Time Detection: range from 80 to 800 ms
Management	• Administrative Telnet CLI and HTTP, HTTPS • HTTP provision through MAC address • Multilingual Web User Interface • 3 Levels of User Access Right with Password protection with different Web

	Language (Administrator, Supervisor and User)
	• HTTP/HTTPS Service Access limitation from WAN port
	• Configure Service ports at HTTP, HTTPS and telnet Services
	• Phone Debug Module: Device Control, Call Control, DB, Verbose
	• SIP Debug Module: Register, Call, SIP Message, Others
	• SNTP Debug Module
	• Device Debug Module
	• DSP Debug
	• Provide 8 Debug Levels :
	  ▪ Emergency
	  ▪ Alert
	  ▪ Critical
	  ▪ Error
	  ▪ Warning
	  ▪ Notice
	  ▪ Information
	  ▪ Debug
	• Provides System Status Logs
	• Connect to external SYSLOG Server
	• Status display: Network, Line, SIP Trunk status
	• Diagnostics (debug through Syslog Event Notice)
	• Debug in real time by Telnet
	• Auto Provision via HTTP Server
	• SNMP V2/Trap
	• Configuration Backup/Restore
	• Dual Firmware Image Backup
	• Reset to factory Default

Power Supply	• Input Power adaptor: AC100V~240V, 50/60Hz • Output Power adaptor: DC12V/1.5A
Environment	• Operating temperature : 0°C to 45°C • Operating Humidity:10% to 90% (Non-Condensing)
Dimension	• 175 * 32 * 126 mm
Certification	• CE, FCC, LVD, RoHS

Application



Ordering information

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