Internet Telephony Gateway Expandable RVP - 208

RealTime™

RVP-208;Internet Telephony Gateway Expandable 4FXS,4FXO & 1 10/100 Base-T port

Overview

To meet the increasing Internet telephony service demands, **RVP-208** delivers the next-generation platform for voice services and applications. The **RVP-208** is a reliable, efficient, TOC-saving, widely interoperable, and scalable and manageable solution for your VoIP network.
RVP-208 not only inherits traditions of quality voice communications and real-time fax data over IP networks, but also eliminates the human resources of deploying VoIP network. With optimized SIP architecture, **RVP-208** is the ideal choice for P2P voice chat and ITSP cost-saving. It also provides network-converting features to translate the packet network into conventional PBX system. With 2-modular slot design and built-in PPPoE/DHCP/DDNS clients up to 8 concurrent connections in

RVP-208, voice communications can be established from anywhere around the world. **RVP-208** comes with intuitive user-friendly yet powerful management interface (web/telnet/console). Equipped with remote management capability, the product reduces IT personnel resource, completes VoIP deployment in a short time, VoIP administrators can monitor machine/network status and proceed with maintenance/troubleshooting service via Internet browser or telnet session.

Key Features

Standards compliant & surpassing voice quality
RVP-208 is SIP version 2 compliant. Moreover, **RVP-208** provides Call forwarding, Call waiting, Call Transfer features for rich telephony communication without changing user's habit and the **RVP-208** is interoperable with major SIP gateways/agent/proxy servers. Retaining excellent traditions in VoIP products, **RVP-208** prioritizes voice packets using IP precedence, and combine state-of-the-art technology of voice packet handling, including echo, noise reduction, voice reconstruction and redundancy to provide customers toll quality VoIP communication.

VoIP, FoIP and Network conversion
Via configurable voice codec: G.723, G.729ab and FAX: T.30, T.38, **RVP-208** supports multiple algorithms to meet different VoIP application demands. **RVP-208** supports converting telephony protocols (SS7, Analog (FXS/FXO)) into packet switching network between making and receiving calls via voice gateways or PBX system in heterogeneous signal-switching environment.

Domain name call , DDNS supports
By either IP or URL addressing, **RVP-208** is able to find and communicate with destination SIP gateway/agents. Meanwhile, DDNS service is useful to those VoIP gateways deployed in a dynamic IP environment. Collaborate connection agents (PPPoE, or DHCP clients) with built-in DDNS client in **RVP-208**, those who don't have static IP address can map dynamic IP address to an easy-to-remember URL, and then the 3rd party SIP compatible voice gateways/agents will allocate position of VIP much easier than ever. (To establish voice communication via domain name, please make sure the third party gateway/agents supports domain name call).

Various field applications compliant
The **RVP-208** is capable of handling peer-to-peer as well as SIP proxy registration and authentication to interact with major SIP gateway/IP Phone on the market. **RVP-208** offers the most flexibility and interoperability with REALTIME and 3rd party VoIP vendors, allowing the deployment of both simple and complex VoIP networks such as ITSP, PC-to-Phone/Phone-to-PC and enterprise VoIP environments.

Specifications

Specifications:	
Model	**RVP-208**
Product	Expendable SIP Internet Telephony Gateway
Ports (LAN & VOICE)	One (l0/100 Base-TX Auto-Negotiation) Two open slots (up to 8 RJ-11)
SIP Specification	
Call Signaling Control	SIP version 2.0 (RFC3361)/SIP over UDP (IETF RFC2543)
Voice Codec	G.711 PCM 64kbps (A-law and u-law) G.723.1AACELP/M-MLQ (5.3,6.3kbps) G.729AB CS-ACELP (8kbps
Fax supports	T.30, T.38
Simultaneous Connection	Up to 8 channels voice/FAX
Voice processing	Gain/Attenuation Settings, Programmable Dynamic Jitter Buffer Call ,progress detection Dial tone, busy tone, call-holding tone and ring-back tone
Protocols/Standards	
Protocol	TCP/IP, HTTP, DNS, RTP, RTCP, SDP
Managements	RS-232 Console/Telnet, HTTP
LED Indicators	System: 2, PWR, CPU LAN: 4, ACT, 100, LNK, COL Voice: 4 In-Use/Ringing
Dimension	Dimension (LxWxH mm) 275 x445 x44 (Metal Case)
Environmental	Temperature: 0~50 degree C (operating) Humidity: up to 90% (non-condensing)
Emission	EMI: FCC part 15, CE / PTT: FCC part 68 Humidity: 10 to 95% (non-condensing)
Module Specifications	
Model	**RVP-304(**4 FXS port module) **, RVP-404 (**4 FXO port module)
Telephony specifications	
Voice channels per card	4
Interface Connectors	4 RJ-11 2-pin modular jacks
Line Impedance	600 ohms, 900 ohms
Insertion Loss	2 dB nominal
Frequency Response	300Hz ~ 3400Hz +/- 2dB w.r.t. 1004Hz
Return Loss	>=18Db

Input Level adjustment	3-6 dB to +6 dB
Output Attenuation	0dB tol3dB
Longitudinal Balanced	>=45Db
Loop current	25mA nominal
Ring Voltage	50Vrms nominal
Ringing Tone	16.67Hz, 20Hz(default), 25Hz or 50Hz
Signaling	Loop Start/DTMF

OrderingInformation:
RVP-208: Expendable SIP Internet Telephony Gateway (up to 8 x RJ-11)
RVP-304: 4 FXS port module
RVP-404: 4 FXO port module