

Q-1FXS-SIP-PP & Q-2FXS-SIP-PP

**Proven effective, low-cost VoIP gateway for
IP-based Voice and Real-Time Fax Communication**

**Designed for use with Ecocarrier's service platform to do
unblockable VoIP operation by running proprietary protocol**

**Ideal for use in
Call Shops, Small Office and Home Office**



Applications

For use as a standalone VoIP gateway for a telephone service provision business in a Call Shop operation

- connect individual regular analog telephones into Q-1FXS-SIP-PP or Q-2FXS-SIP-PP and use it to provide prepaid or post-paid Long Distance Telephone Call service to customers for 1 or 2 calls simultaneously in a Call Shop operation
- all Long Distance Calls are automatically placed through Ecocarrier service platform to call destination telephone numbers at extremely low rates – providing the Call Shop operator with significant competitive advantage
- small offices of an enterprise and home offices of employees of an enterprise may use Q-1FXS-SIP-PP or Q-2FXS-SIP-PP for a low-cost way of equipping the offices for IP telephony
- creation of a virtual private telephone network that is implemented with facility for inter-company dialing and call traffic data record and accounting that enhance the business operation of the enterprise

Product Key Features

- Supports TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP/RARP, DNS, DHCP (both client and server), PPPoE, TFTP etc
- Built-in router, NAT and Gateway
- Supports SIP
- Multiple(1/2/4/8/16/24) FXS and necessary FXO ports with independent telephone numbers
- Simultaneous signaling and media encrypting and mangling
- Free access to customer network
- Intelligent voice routing and discovery
- Interoperable with various market-leading third party Softswitch or Sip Server
- Supports popular voice-coders including G.723, G.729AB and G.711 (A law and U-law), FAX and Modem pass-through
- Supports PSTN-Relay in case of power failure and external control
- Supports DIGEST authentication (MD5 only)
- Supports standard voice features such as Caller Id, Consultation Hold, Call Waiting, Blind Call Transfer, Attended Call Transfer, Call Forward, Mid-Call DTMF, Flash, Three-way Conference
- Supports Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), AGC and Line Echo Cancellation G.168 128ms
- Supports layer-3 (DiffServ, ToS) QoS
- Supports automated NAT traversal without manual manipulation of firewall/NAT
- Supports remote automated provisioning and software upgrade even through firewall/NAT to enable "zero configuration" and "plug-and-dial" for end users
- Supports device configuration via built-in IVR by dialpad, Web browser or central configuration file
- Intelligent powerful proprietary AVMP management scheme
- Standards-based implementation (ITU-T, IETF compliant)

Software Specification

VoIP Protocols

- Supports popular voice-coders including G.723, G.729AB and G.711 (A law and U-law), FAX and Modem pass-through
- Supports PSTN-Relay in case of power failure and external control
- SIP (RFC 3261~3266 and various SIP draft extensions)
- SDP(RFC2327)
- RTP/RTCP(RFC 3550, 3551)

Call Features

- Call Transfer (Attended)
- Call Forward (Busy / No Answer /Unconditional)
- Call Hold / Retrieve
- Call Waiting
- Caller ID Display
- Caller ID Blocking
- Anonymous Call Blocking
- Hook Flash

FAX Support

- G.711 pass- through
- T.38

DTMF

- In-band DTMF
- Out-of-band DTMF(RFC 2833)

QoS

- ToS field

Dial Signal

- Pulse dialing 10/20 pps
- DTMF dialing

Codec

- G.711μ-law
- G711a-law
- G.729, G.729a/b
- G.723

Security

- HTTP 1.1 basic/digest
- authentication for Web setup
- MD5 for SIP authentication (RFC2069/ RFC 2617)

Dial Method

- Direct IP call without SIP proxy
- Dial number via SIP server
- Dial URI via SIP server
- Flexiable Dial Plan

Router

- Virtual Server
- NAT
- DHCP Server
- DMZ

IP Assignment

- Static IP
- DHCP

NAT Traversal

- STUN
- Proprietary tunnel-based TCSP
- Fresh Open

Voice Quality

- VAD (Voice Activity Detection)
- CNG (Comfort Noise Generation)
- AEC (Acoustic Echo Cancellation)
- AGC
- Packet Loss Concealment
- Each Line G.168 up to 128ms
- Dynamic Jitter buffer
- Bell 202 and V.23 Caller ID
- Tone
 - DTMF
 - Ring Tone
 - Ring Back Tone (local and remote)
 - Dial Tone
 - Busy Tone
 - MWI
- TCP/IP
 - IP/TCP/UDP/DHCP/RTP/RTCP
 - ICMP/HTTP/SNTP/TFTP/DNS
- Configuration & Management
 - Web browser
 - Touch-tone telephone keypad configuration with voice prompt
 - Auto-provisioning scheme with Aegis QuickStep
 - Reverse Control even residing after NAT/Firewall
 - Periodic "HeartBeat" Monitor
- Firmware Upgrade
 - TFTP
 - AQS extensible remote agent

© Copyright 2007, QiiQ Communications Inc. All Rights Reserved. Features and specifications subject to change without notice.