

## *Data Sheet*

For Speed VoIP SPE Gateway

**The SpeedVoIP SPE voice gateway is a IP-to-PSTN adaptor that turns traditional telephone devices into IP devices. Customers can take advantage of the many new and exciting IP telephony applications by connecting their analog phone to SpeedVoIP SPE gateway.**

The SpeedVoIP SPE gateway is capable of delivering a cost-effective solution for commercial and residential applications worldwide. Consisting of voice coprocessor, network processor and voice codec, the gateway converts your voice from an analog signal to a digital signal which is then sent out over a high speed Internet connection. Supporting multiple FXS & FXO voice ports and two 10/100BaseT Ethernet ports, the gateway can make use of existing Ethernet LANs in addition to xDSL or cable broadband modems. Connect a traditional telephone and a FAX machine to your computer's VoIP network; or deploy a second-line service to take advantage of emerging telephony applications.

The SpeedVoIP SPE gateway extracts the maximum benefit from Voice over IP (VoIP) and Fax over IP (FoIP) technologies. It interfaces analog telephones with IP-based telephony networks. A typical usage/deployment scenario would be to use the SPE gateway as a separate box behind a home-broadband SOHO router or xDSL modem.

## **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 2.0
- Multiple(1/2/4) 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.726, 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 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 DIGEST authentication (MD5 only)
- 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

- 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

### Codec

- G.711 $\mu$ -law
- G711a-law
- G.729, G.729a/b
- G.726

### DTMF

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

### 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
- Flexible Dial Plan

### **Router**

- Virtual Server
- NAT
- DHCP Server
- DMZ

### **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

### **QoS**

- ToS field

### **Dial Signal**

- Pulse dialing 10/20 pps
- DTMF dialing

### **Tone**

- DTMF
- Ring Tone
- Ring Back Tone (local and remote)
- Dial Tone
- Busy Tone
- MWI

### **IP Assignment**

- Static IP
- DHCP
- PPPoE

### **NAT Traversal**

- STUN
- Proprietary tunnel-based TCSP
- Fresh Open

### **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
- SpeedVoIP IntelliManage® Reverse Control even residing after NAT/Firewall
- Periodic "HeartBeat" Monitor

### **Firmware Upgrade**

- TFTP
- AQS extensible remote agent

## Hardware Specification

- IP Network Port 1 x 10/100Base-T WAN port, 1 x 10/100Base-T LAN port
- Analog Telephone Port FXS port(Loop Start and Polarity Reversal) and FXO port
- Exterior Black ABS plastic, tri-color LED button
- External USB for Flash programming
- Universal Power Supply 100-240V input, +12VDC/1A output
- US/Euro/UK/Japan/Australian style available
- Operating Temperature 0-50C
- Humidity 10-95% non-condensing
- Compliance FCC/CE/C-Tick/RoHS.