

# 8 FXS Ports VoIP Phone Adapter G508

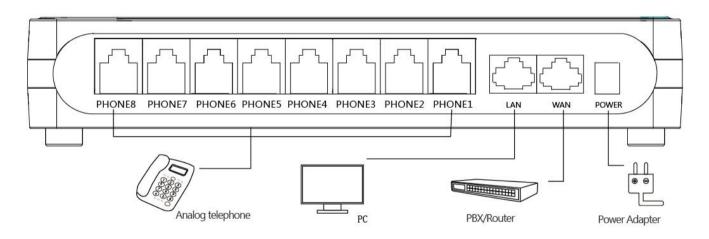




### **Feature Keys**

- 8 FXS Ports
- T.38 And T.30 FAX
- TR069,SNMP
- 10/100/1000Mbps Ethernet

#### Hardware Interface



### **Overview**

G508, which has 8 FXS ports , one LAN port and one WAN port, is one of the most popular VoIP ATA researched and produced by FlyingVoice. This product can not only provide 8 SIP lines for users to make calls, but also it is a wire-speed NAT router, make you enjoy easy network atmosphere. What' s more, G508 support T.38 real time FAX and T.30 FAX with G.711.G508 is a stand-alone device, which requires no PC to make Internet calls. This ATA guarantees clear and reliable voice quality on Internet, which is fully compatible with SIP industry standard and able to interoperate with many other SIP devices and software on the market.Their compact size, excellent

voice quality, packed feature functionality and best-in-class price-performance point enable consumers to maximize the power of IP voice and data connectivity.

G508 is based on SIP V2.0 standard and compatibility with most service providers. It features 4 FXS telephone ports, TR069 CPE management & monitoring protocols and a base stand for vertical



# **Technical Parameters**

Power	AC/DC Adapter
	AC Input: 100~240V, 50~60Hz
	DC Output: 12V, 3A
Operating System	• Linux 2.6.36
I/O Interfaces	• 2 RJ-45 for 10/100/1000Ethernet Ports
	8 RJ-11 for FXS Ports
Environmental	<ul> <li>Operation Temperature: 0~50 Degree C</li> </ul>
	Storage Temperature: -25~ 85 Degree C
	Relative Humidity: 10%~90% No Condensing
Audio Codec	• G.711(A/u),PAMS>4.3
	• G.729A/AB,PAMS>4.0
	• T.30 FAX with G.711
	Real time FAX over IP via T.38
	Adaptive Jitter Buffer
	Voice Activity Detection
	Comfort Noise Generation
	Echo Cancellation
Management	Firmware Upgrade
	Web Management Interface
	IVR Management Interface
	Local and Remote Syslog (RFC3164)
	Auto Provisioning
	SNTP Time Synchronization
	Multi User Level
	SNMPv2
	• TR069
Protocols	<ul> <li>SIP V2 (RFC 3261,RFC3262,RFC3263,RFC3264,RFC3265,RFC3515, RFC3891, RFC3892,3GPP,IMS)</li> </ul>
	Backward Compatible with RFC2543
	Session Timer (RFC4028)
	• SDP (RFC2327)
	RTP/RTCP (RFC1889 and RFC1890)
	NAPTR for SIP URI Lookup (RFC2915)
	• STUN (RFC 3489)
	• ARP/RARP (RFC 826/903)
	• SNTP (RFC 2030)
	• DHCP/PPPoE
	PPTP/L2TP VPN
	HTTP Server for Web Management
	TFTP/HTTP/HTTPS for Auto Provisioning
	DNS/DNS SRV (RFC1706 and RFC 2782)



• T.30 FAX with G.711

• 180mm(L)x110mm(W)x30mm(H)

• Dial Plan

• Black List

Call Log

• 295g(N.W)

Applications

**Call Features** 

SIZE And Weight

NAT/NAPT Router function
MAC Address Cloning
DHCP Server
PPTP/L2TP VPN
• PPPoE
SIP proxy redundancy
Dynamic via DNS SRV, A records
NAT Traversal by STUN
• DMZ
• QoS with Layer 3
DHCP Client and DHCP Server
IP conflict detectionv
• 3-way Conference
<ul><li> 3-way Conference</li><li> Music on hold</li></ul>
-
Music on hold
<ul> <li>Music on hold</li> <li>DTMF mode: In-band, RFC2833 and SIP INFO</li> </ul>
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<ul> <li>Music on hold</li> <li>DTMF mode: In-band, RFC2833 and SIP INFO</li> <li>Call Hold</li> <li>Call Forwarding</li> <li>Call Mute</li> <li>Call Transfer</li> </ul>
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