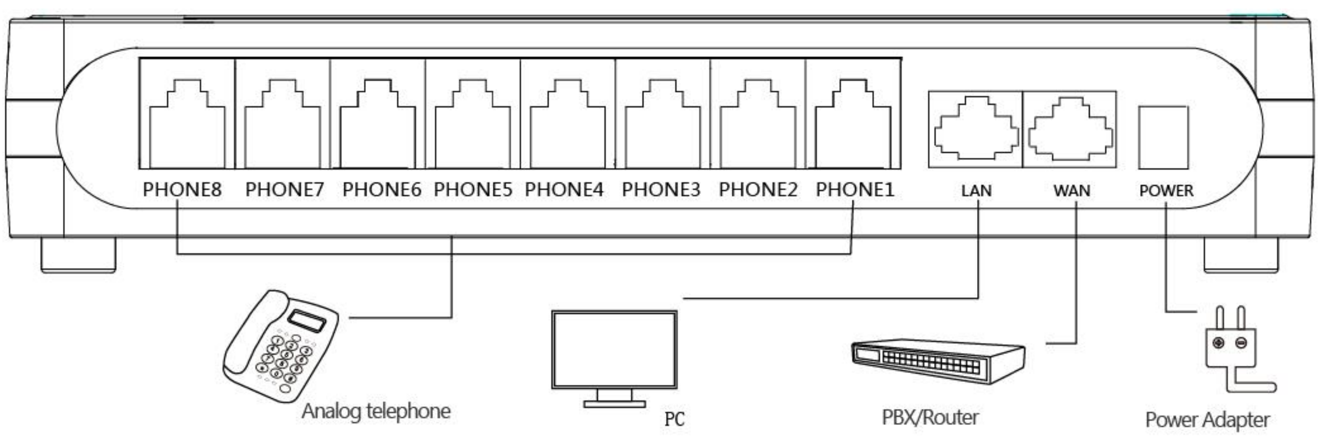
8 FXS Ports VoIP Phone Adapter G508


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 ATAs 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 8 FXS telephone ports, TR069 CPE management & monitoring protocols and a base stand for vertical

Technical Parameters

VoIP SoC	• CPU: MTK MT7621A
Flash	• 16MB Flash
RAM	• 256MB SDRAM
Power	• AC/DC Adapter AC Input: 100~240V, 50~60Hz DC Output: 12V, 2A
Operating System	• Linux 2.6.36
I/O Interfaces	• 2 RJ-45 for 10/100/1000Ethernet Ports • 8 RJ-11 for FXS Ports
Environmental	• Operation Temperature: 0~50 Degree C • 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(Phone 2 port only) • Adaptive Jitter Buffer • Voice Activity Detection • Comfort Noise Generation • Echo Cancellation
Management	• Firmware Upgradeable • Web Management Interface • IVR-driven Management Interface • Local and Remote Syslog (RFC3164) • Auto Provisioning • SNTP Time Synchronization • Multi User Level • SNMPv2 • TR069
Protocols	• SIP V2 (RFC 3261,3262,3263,3264) • 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)

Applications	• 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
Call Features	• 3-way Conference • Music on hold • DTMF mode: In-band, RFC2833 and SIP INFO • Call Hold • Call Forwarding • Call Mute • Call Transfer • Call Waiting • Speed Dial • Caller ID and CWCID • Hotline • Real time fax over IP via T.38 • T.30 FAX with G.711 • Dial Plan • Black List • Call Log
SIZE And Weight	• 180mm(L)x110mm(W)x30mm(H) • 295g(N.W)