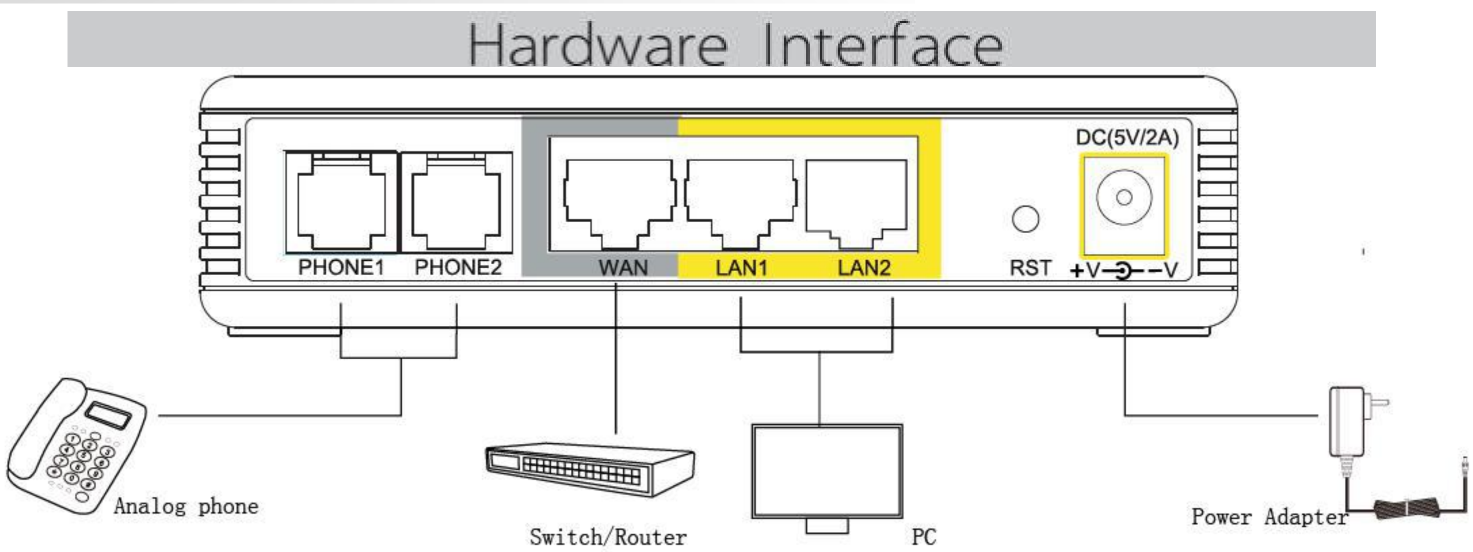
2 FXS Ports VoIP Phone Adapter FTA5102

Feature Keys

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



Overview

FTA5102, which has two FXS port , 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 two 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, FTA5102 support T.38 real time FAX and T.30 FAX with G.711.FTA5102 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. The FTA5102, 2 FXS Analogue Telephone Adapter products in Flyingvoice, which enables customers to register to different SIP Proxy server, IP PBX and establish up to 2 concurrent VoIP calls for more flexibility in the voice communication.heir 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. FTA5102 is based on SIP V2.0 standard and compatibility with most service providers. TR069 CPE management & monitoring protocols and a base stand for vertical positioning.

Technical Parameters

VoIP SoC	• CPU:RT5350
Flash	• 16MB Flash
RAM	• 32MB SDRAM
Power	• AC/DC Adapter AC Input: 100~240V, 50~60Hz DC Output: 12V, 1A
Operating System	• Linux 2.6.21
I/O Interfaces	• 3 RJ-45 for 10/100Ethernet Ports • 2 RJ-11 for FXS Port
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 • Support IPv6 • 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	• 120mm(L)x80mm(W)x32.5mm(H) • 119g(N.W)