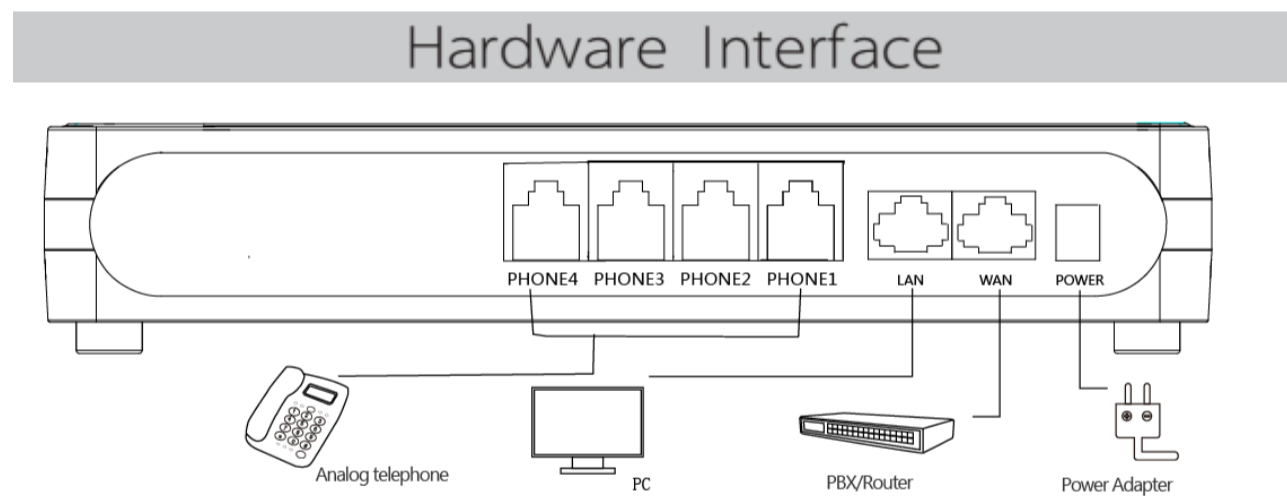


## 4 FXS Ports VoIP Phone Adapter G504



### Feature Keys

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



### Overview

G504, which has 4 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 4 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, G504 support T.38 real time FAX and T.30 FAX with G.711. G504 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. G504 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	<ul style="list-style-type: none"> <li>AC/DC Adapter</li> <li>AC Input: 100~240V, 50~60Hz</li> <li>DC Output: 12V, 2A</li> </ul>
Operating System	<ul style="list-style-type: none"> <li>Linux 2.6.36</li> </ul>
I/O Interfaces	<ul style="list-style-type: none"> <li>2 RJ-45 for 10/100/1000Ethernet Ports</li> <li>4 RJ-11 for FXS Ports</li> </ul>
Environmental	<ul style="list-style-type: none"> <li>Operation Temperature: 0~50 Degree C</li> <li>Storage Temperature: -25~ 85 Degree C</li> <li>Relative Humidity: 10%~90% No Condensing</li> </ul>
Audio Codec	<ul style="list-style-type: none"> <li>G.711(A/u),PAMS&gt;4.3</li> <li>G.729A/AB,PAMS&gt;4.0</li> <li>T.30 FAX with G.711</li> <li>Real time FAX over IP via T.38</li> <li>Adaptive Jitter Buffer</li> <li>Voice Activity Detection</li> <li>Comfort Noise Generation</li> <li>Echo Cancellation</li> </ul>
Management	<ul style="list-style-type: none"> <li>Firmware Upgrade</li> <li>Web Management Interface</li> <li>IVR Management Interface</li> <li>Local and Remote Syslog (RFC3164)</li> <li>Auto Provisioning</li> <li>SNTP Time Synchronization</li> <li>Multi User Level</li> <li>SNMPv2</li> <li>TR069</li> </ul>
Protocols	<ul style="list-style-type: none"> <li>SIP V2 (RFC 3261,RFC3262,RFC3263,RFC3264,RFC3265,RFC3515, RFC3891, RFC3892,3GPP,IMS)</li> <li>Backward Compatible with RFC2543</li> <li>Session Timer (RFC4028)</li> <li>SDP (RFC2327)</li> <li>RTP/RTCP (RFC1889 and RFC1890)</li> <li>NAPTR for SIP URI Lookup (RFC2915)</li> <li>STUN (RFC 3489)</li> <li>ARP/RARP (RFC 826/903)</li> <li>SNTP (RFC 2030)</li> <li>DHCP/PPPoE</li> <li>PPTP/L2TP VPN</li> <li>HTTP Server for Web Management</li> <li>TFTP/HTTP/HTTPS for Auto Provisioning</li> <li>DNS/DNS SRV (RFC1706 and RFC 2782)</li> </ul>

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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

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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

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SIZE And Weight

- 180mm(L)x110mm(W)x30mm(H)
  - 295g(N.W)
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