

HS-AT9600

FXS Analog VoIP Gateway



Front View



Rear View

Key Features:

- High density gateway, 96FXS ports in 1U size
- Support IPv4 and IPv6
- 5 km Line Length(MAX)
- Support SNMP and TR069
- Multiple codecs: G.711A/U,G.723.1,G.729A/B and iLBC
- Support Asterisk and FreeSWITCHplatforms, and compatible with BroadSoft
- Fully compatible with leading IMS/NGN, SIP based IP telephony system

Overview

The HS-AT9600 FXS analog gateway is multi-functional voice gateway that provides seamless connectivity with analog phones, fax machine or analog PBXs to VoIP gateway. The overall frame of HS-AT9600 is compact and structured, supporting RJ45/RJ21 interface standards and up to 48 FXS voice interfaces at a size of 1U. In addition, HS-AT9600 provides high-performance hardware solution with good voice/fax processing capability. It is able to provide efficient and high-quality IP voice services for service provider, enterprises, community residents, and corporatetelephone systems.

HS-AT9600 supports standard SIP and RTP protocols and is compatible with multiple well-known platforms, for example, compatible with BroadSoft. It also works as extensions with various IPPBX products based on Asterisk and FreeSWITCH, and is also perfectly compatible with IMS/NGN softswitch platforms and call center systems. In addition, customers can build an enterprise branch communication network by using the HS-AT9600 to achieve interoperability between branches, which can provide an efficient, stable, and cost-effective voice communication solution for largeenterprises.

Physical Interfaces

- Phone Connector
96 FXS, RJ 45 or RJ21, 50 PIN
- Console: 1* RS232, 115200bps
- Network Interfaces:
2* 10/1000 Base-T
1* 10/100 Base-T

Call Features

- Digit map
- IMPU
- Action URL
- Voicemail
- Custom IVR
- Call Waiting
- Call Holding
- Do-not-disturb
- Blind Transfer
- Attend Transfer
- Call Forward on Busy
- Call Forward on No Reply
- Ringing Group, Hotline
- 3-way Conference
- Early Media/Early Answer
- Abbreviated Dialing
- Access and Peer Mode
- SIP PnP Auto Provisioning
- PBX IVR
- Unconditional Call Forward
- SIP registration to different servers
- Flexible Call Routing Policy
- Caller/Called Number Manipulation

FXS

- Connector: RJ45 or RJ21
- Max Line Length: 5 km
- Reversed Polarity
- Pulse: 10 and 20 PPS
- Dial Mode: DTMF and Pulse
- Programmable Call Progress Tone
- Caller ID: DTMF/FSK CLI Presentation

VoIP

- Protocol:
- SIP v2.0 (UDP/TCP), RFC3261
- SDP,RTP(RFC2833), RFC3262, RFC3263, RFC3264, RFC3265, RFC3515, RFC2976, RFC3311
- RTP/RTCP, RFC2198, RFC1889
- IPv4 and IPv6
- Outbound Proxy
- Master/Slave Server
- RFC2806 TEL URI
- RFC3581 NAT,rport
- VLAN 802.1P/802.1Q
- RFC4028 Session Timer
- DNS SRV/A Query/NATPR Query
- NAT:STUN, Static/Dynamic NAT

Maintenance

- CDR
- Ping/Tracer Test
- Network Capture
- Outward Test(GR909)
- SNMP, TR069, Provision
- NTP/Daylight Saving Time
- IVR local Maintenance
- RADIUS Authentication Management
- Cloud-based Management (NMS)
- Syslog: Debug, Info, Error, Warning, Notice
- Configuration Backup/Restore
- Firmware Upgrade via
HTTP/HTTPS/TFTP/FTP/HTTP

Voice & FAX

- Recording
- T.38/Pass-through
- Modem/POS
- Silence Suppression
- Bandwidth Optimization
- Comfort Noise Generation(CNG)
- Voice Activity Detection(VAD)
- Packet Loss Concealment
- Adaptive (Dynamic) Jitter Buffer
- Programmable Gain Control
- G.711A/U law, G.723.1, G.729A/B, G.726, iLBC, AMR, AMR-GSM
- DTMF mode:
Signal/RFC2833/INBAND
- Echo Cancellation(G.168), with up to 128ms

Security

- QoS
- TLS, SRTP
- Web ACL/SSH ACL
- RADIUS Authentication
- Certification Management
- Service Network Port:
Isolated/BOND/Bridge Mode
- Signaling/Voice Encryption

Physical Features

- Power Supply: 100-240VAC, 50/60Hz
- Dual power supplies (only 48S/72S support)
- Power Consumption: 95W
- Operating Temperature: 0°C ~ 45°C
- Storage Temperature: -20°C~80°C
- Humidity:
10%-90% Non-Condensing
- Dimensions(W/D/H): 440*328*44mm (1U)
- Unit Weight: 4.4kg
- Compliance: CE and FCC