

D32S VOICE GATEWAY

With a simple and economical way to help legacy telephones, fax machines, and PBXs interconnect with an IP network, Bitttel's 32 ports analog FXS gateway enables call center and multi-branch enterprises to process powerful, versatile, and efficient VoIP solutions with unparalleled cost advantages. Connected between a PBX, LAN, or WAN, the 32 ports FXS VoIP Gateway converts analog PSTN messages into a format suitable for transmission over standard IP networks.

Designed for voicemail and unified messaging applications, the 32 Ports Analog VoIP Gateway D32S has a 10/100/1000M (optional) Base-T Ethernet connection for connecting legacy PBX to a LAN. The analog loop start functionality supports integration via in-band signaling (DTMF or FSK), serial protocols, as well as T.38 for fax transmissions over IP (FoIP).



BENEFITS

- High-performance VoIP connectivity for SMBs.
- Voice optimization to ensure better user experiences.
- Enhanced call routing ability with high voice quality.
- Easy to install, configure, and maintain.
- Support IPv4 and IPv6 international networks.
- Data/voice/management VLAN and more.
- Build-in firewall and access rules.
- Support SNMP/TR069/Auto-Provision.
- Cloud-based management and bandwidth optimization.
- Support SIP, MGCP, or other customizable protocols.
- Primary/Backup SIP Servers.
- Flexible routing and manipulation.

KEY FEATURES

- Completely non-blocking architecture and Scalable System.
- Easy integration with existing telephony interfaces.
- Open-standard SIP support and register to multiple SIP proxy servers.
- Make and receive IP calls from analog extensions.
- Call budgeting is based on the allocated amount, minutes, and call count.
- Manageable-based call routing TDP-IP/IP-TDM.
- Restrict unwanted calls with a list of denied numbers.
- Real-time call record sends to CDR server.
- Caller ID presentation and restriction.
- Hotline extension setting.
- Web-based remote administration.
- Consol access via Telnet, SSH.

PART#: D32S

- Support 32 FXS Ports, Field Approved Globally
- Superior Voice Quality by Designated DSP Chipsets
- User-Friendliness and Web-based Administration

TECHNICAL SPECIFICATIONS:

PHYSICAL INTERFACE

Phone Interface: RJ21 Amphenol connector available as well Ethernet Interface: 2* RJ-45 10/100Mbps Base-T Ethernet, Female RJ-45 1000M LAN/WAN available for some product models while required.

SESSION CAPACITY

24, 32 or 48 SIP channels
24, 32, or 48 FXS channels

CONNECTIVITY

Dial Mode: DTMF and Pulse
Pulse: 10 and 20PPS
Caller ID:DTMF/FSK
Max Cable Length:5KM
Reversed Polarity
OpenVPN

VOIP PROTOCOLS

TLS/ SRTP
OpenVPN
SIP V2.0 (RFC 3261, 3262, 3264)
IMS/3GPP
SDP
REFER (RFC 3515)
RTP/RTCP
STUN (RFC3489)
ARP/RARP (RFC 826/903)
SNTP (RFC 2030)
DHCP/PPPoE
TFTP/HTTP/HTTPS
DNS/DNSSRV (RFC 1706/RFC2782).
VLAN802. 1P/802.1Q.

CALL & ROUTING

Port Groups
IP Trunks
Primary and Secondary SIP Account
24, 32 or 48 Inbound/OutboundRouting
Number Manipulation
Digit maps
TDMtoIPtoTDM
IP load balancing
IP fault tolerance

VOICE CAPABILITY

G.711A/U law, G.723.1, G.729A/B,G.726,iLBC,AMR
Comfort Noise Generation(CNG)
Echo Cancellation(G.168)
DTMF mode: Signal/RFC2833/INBAND
Silence suppression with comfort noise
G.168 automatic echo cancellation
Call Progress Analysis (CPA), including Positive Voice Detection, Positive

ANSWERING

Machine Detection (PAMD), DTMF detection, and fax tone detection
Manageable based call routing TDP-IP/IP-TDM.
Restrict unwanted calls with list of denied numbers.
Voice Activity Detection (VAD)
Adaptive (Dynamic) Jitter Buffer
Programmable Gain Control
Hook Flash

FOIP PROTOCOL & FAXING

T.38 FoIP: transcode fax from T.30 fax protocol (supporting V.17) modulation schemes

NETWORK CAPABILITY

TStatic IP, PPPoE, DHCP Client IPv4, IPv6
Static/dynamic ARP DIFFServ, ToS
NAT (Rout and Bridge)+ MAC Address Clone Static routing+
Built-in Firewalls
QoS, Traffic Shaping
Voice/Data/Management Vlan

TECHNICAL SPECIFICATIONS CONTD:

MAINTENANCE & UPGRADING

SNMP/TR069.
Auto Provision
Action URL
Digit map
Web/Telnet. ACL
Configuration Backup/Restore
Bandwidth Optimization
Routing Rules based Prefixes
Firmware Upgrade via WEB
Syslog and CDR.
Access Rule list.
Network Capture
Outward Test(GR909).
Automatic Time Synchronization
IVR local Maintenance.
Cloud-based Management
Caller/Called Number Manipulation
Open-standard SIP support and register to multiple SIP proxy servers.

APPLICATION CAPABILITIES

Call waiting
Blind Transfer
Attend Transfer
Call forward on Busy
Call forward on No Reply
Unconditional Call Forward
HotlineCall hold
DND
Call Pickup
3-way conference
Voicemail

CONFERENCING RESOURCE

Call budgeting based on allocated amount, minutes and call count
Complete non-blocking architecture and Scalable System
Hotline extension setting
Support 3-Way and Multi-Way Conferencing

ENVIRONMENT & POWER

Power Supply: 100-240V, 50-60Hz+
Power Consumption: Approximately 50W
Temperature(Operation):0 °C ~ 45°C
(Storage): -20 ~85°C
Humidity: 10%-90% No condensation.
Operatingtemperaturerange:-10 °C~55°C

PHYSICAL DIMENSION

L*W*H 440(mm)*202(mm)*44(mm)
Weight Approximately 5.95ibs(about 2.7kg)
L*W*H 440(mm)*202(mm)*44(mm)
Weight Approximately 5.95ibs(about 2.7kg)

WARRANTY/CERTIFICATIONS

1 year warranty. CE, FCC
Broadsoft, Elastix, Asterisk, Teams and other UC platform

CONTACT US