DB2400A 24 Port FXS Cohesive VoIP Gateway

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The Cohesive 24 ports analog FXS gateway provides a simple and affordable solution for legacy telephone, fax machines, and PBXs to interconnect with an IP network. This enables call centers and multi-branch enterprises to leverage powerful, versatile, and efficient VoIP solutions with significant cost advantages. By connecting the 24 ports FXS VoIP Gateway between a PBX, LAN, or WAN, the analog PSTN messages can be converted into a format suitable for transmission over standard IP networks.

The 24 Ports Analog VoIP Gateway DB2400A is specifically designed for voicemail and unified messaging applications and features a 10/100/1000M (optional) Base-T Ethernet connection to connect legacy PBXs 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).

DB2400A

Features

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

Benefits of

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

- **Support 24 FXS Ports, Field Approved Globally**
- **Superior Voice Quality by Designated DSP Chipsets**
- **User-Friendliness and Web-based Administration**

Technical Specifications:

Physical Interface

Phone Interface: 24 Ports FXS, RJ-11 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 SIP channels

24 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 Inbound/Outbound Routing Number

Manipulation

Digit maps
TDM to IP or IP to TDM
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 for transmission over a packet network

T.38/Pass-through, up to 14.4kbps

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

Network Capability

Static 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

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
Hotline Call 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.
Operating temperature range: -10 °C ~55°C

Physical Dimension

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

Warranty/Certifications

1 year warranty
On the Second & Third free to repair.
CE, FCC or Any other Certificates Customizable
Broadsoft, Elastix, Asterisk, Teams and other UC platform