DINSTAR

DAG3000 FXS Analog VoIP Gateway

Overview

The DAG3000 Series 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 DAG3000 is compact and structured, supporting RJ21 interface standards and up to 312 FXS voice interfaces via user boards plugging at a size of 3U. In addition, DAG3000 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 corporate telephone systems.

DAG3000 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 DAG3000 to achieve interoperability between branches, which can provide an efficient, stable, and cost-effective voice communication solution for large enterprises.

Front View

Rear View

Key Features

- High density gateway, 24 to 312 FXS ports in 3U size via user boards plugging
- 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 FreeSWITCH platforms, and compatible with BroadSoft
- Fully compatible with leading IMS/NGN, SIP based IP telephony system

Physical Interfaces

- **Hardware**

  1*MCU Boards

  13*User Boards
- **FXS Boards**
- RJ21: One user board supports 24 FXS ports, up to 13 user boards
- **MCU Board**

  Network Interfaces:

  2* 10/1000 Base-T

  1* 10/100 Base-T

  Console: 1* RS232, 115200bps

Call Features

- Digit map
- IMPU
- CDR
- 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

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

Maintenance

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

Physical Features

- Power Supply: 100-240VAC, 50/60Hz
- Dual power supplies
- Operating Temperature: 0℃ ~ 45℃
- Storage Temperature: -20℃ ~80℃
- Humidity: 10%-90% Non-Condensing
- Dimensions(W/D/H): 440*360*132mm(3U)
- Consumption (Idle/With full channel used): 150W/400W
- Unit Weight: 12 kg
- Compliance: CE and FCC

FXS

- Connector: 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

Security

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

VoIP

Protocols:

- SIP v2.0 (UDP/TCP), RFC3261
- SDP(RFC2327),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

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DINSTAR

About Us

Founded in 2011 in Shenzhen, DINSTAR is a leading global provider of IP Unified Communication products including VoIP Gateways, IP PBXs, IP Phones and SBCs, we have been delivering more agile, efficient and affordable communication solutions and unparalleled communication experiences to our customers with our reliable, innovative and future-proof products for years. Through our value-added distributors and resellers worldwide, now DINSTAR serves telecom operators, service providers, system integrators, enterprises, SMBs and OEM partners in over 100 countries.