DINSTAR

DAG1000-2S2O Hybrid Analog VoIP Gateway

Overview

DAG1000 series hybrid analog gateways are multi-purpose IP-based voice and fax gateways. DAG1000 supports kinds of work places, for small business, work at home, remote office and branch businesses and provides a low cost, simple operation VoIP solution. FXS/FXO hybrid gateway can support network failure and power failure lifeline feature, flexibly achieve interoperability with simulation PBX and offer reliable voice quality assurance for the traditional voice transition to IP voice. It also supports standard SIP protocol and can be compatible mainstream IPPBX and softswitch platform.

Pictures

Front View

Back View

Key Features

- 2 FXS Port(s) and 2 FXO Port(s)
- SNMP and TR069
- Power and network failure life line
- Supports call detection and reverse polarity detection
- Multiple codecs: G.711, G.729, G.723, G.726, AMR
- Fully compatible with leading IMS/NGN, SIP-based IP telephony system
- Supports connecting with various Open Source and Commercial IPPBX platforms

Physical Interfaces

- Phone Interfaces: 2 FXS and 2 FXO, RJ-11
- Ethernet Interfaces:
  1* LAN, 10/100Mbps, RJ45
  1* WAN, 10/100Mbps, RJ45

FXS

- Connector: RJ11
- Dial Mode: DTMF and Pulse
- Pulse: 10 and 20 PPS
- Caller ID: DTMF/FSK CLI Presentation
- Reversed Polarity

FXO

- Connector: RJ11
- Pulse: 10 and 20 PPS
- Caller ID: FSK, DTMF
- Polarity Reversal
- Busy Tone Detection
- No Current Detection
- Automatic Impedance Matching
- Dial Mode: DTMF/Pulse
- Dial Mode: Primary Dial/ Secondary dial
- Call Detection: Bellcore Type 1&2, ETSI, DTMF

Software Features

- Port Group
- Web ACL
- Telnet ACL
- Action URL
- Digitmap
- Speed Dial
- Routing Rules based Prefixes
- Caller/Called Number Manipulation

VoIP

- Protocol: SIP v2.0 (UDP/TCP),RFC3261 SDP,RTP(RFC2833), RFC3262, RFC3263, RFC 3264, RFC3265, RFC3515, RFC2976, RFC3311
- SIP Trunk
- SIP TLS/SRTP
- RTP/RTCP, RFC2198, 1889
- RFC4028 Session Timer
- RFC2806 TEL URI
- RFC3581 NAT, rport
- Outbound Proxy
- DNS SRV/ A Query/NATPR Query
- Early Media/Early Answer
- NAT:STUN, Static/Dynamic NAT

Voice & FAX

- Modem/POS
- T.38/Pass-through
- Silence Suppression
- VLAN 802.1P/802.1Q
- Layer3 QoS and DiffServ
- Programmable Gain Control
- Comfort Noise Generation(CNG)
- Voice Activity Detection(VAD)
- Adaptive (Dynamic) Jitter Buffer
- Echo Cancellation(G.168), with up to 128ms
- Audio Codec: G.711A/U law, G.723.1, G.729A/B,
- DTMF mode: Signal/RFC2833/INBAND

Network

- Static/Dynamic IP
- PPPoE
- Action URL
- DHCP Client
- IPv4, TCP, UDP,TFTP, FTP, ARP,RARP, Ping, NTP, SNTP, HTTP/HTTPS, DNS
- Ping / Tracert
- DHCP Option 66,120,121
- OpenVPN

Maintenance

- CDR
- Syslog
- Web/Telnet
- SNMP v1/v2/v3
- TR069, TR181
- Auto Provisioning
- Configuration Backup/Restore
- Firmware Upgrade via Web
- Network Capture
- NTP/Daylight Saving Time
- IVR local Maintenance
- Cloud-based Management

Environmental

- Power supply:
  Input 100~240VAC Output 12VDC, 2A
- Power consumption(W): 10W
- Operator Temperature: 0 ℃ ~ 45 ℃
- Storage Temperature: -20 ℃ ~80 ℃
- Humidity: 10%-90% no condensation
- Dimensions(W/D/H): 195*135*35mm
- Weight: 0.6kg
- Compliance: CE, FCC

Shenzhen Dinstar Co.,Ltd.

Floor 18, Building 7A, Vanke Cloud City Phase 1, Xingke 1st Street, Xili Sub-district, Nanshan District, Shenzhen, P.R. China 518000

www.dinstar.com
sales@dinstar.com

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.