DAG1000-4S FXS Analog VoIP Gateway

Overview

DAG1000-4S is a multi-functional analog gateway offering seamless connectivity between IP-based telephony networks and legacy telephones (POTS), fax machines and PBX systems. The device offers 4 FXS ports, fax over IP and flexible dial plans. It is ideally suited for small and medium businesses, call centers and multi-location environments that need VoIP services.

DAG1000-4S

DAG1000-4S supports the standard SIP protocol and it's compatible with leading IMS/NGN platforms and SIP-based IP telephony systems.

Key Features

- Cost effective gateway with 4 FXS ports
- Fax over IP (T.38 and Pass-Through)
- Support IPv4 and IPv6
- TR069 and SNMP
- Multiple codecs: G.711A/U,G.723.1,G.729A/B, iLBC
- Fully compatible with leading IMS/NGN, SIP based IP telephony system

Physical Interfaces

Capacity

4 FXS, RJ11

Ethernet Interfaces:

1* WAN, 10/100Mbps, RJ-45

1* LAN, 10/100Mbps, RJ-45

Console: N/A

Voice & FAX

G.711A/U law, G.723.1, G.729A/B,G.726,iLBC

Silence Suppression

Comfort Noise Generation(CNG)

Voice Activity Detection(VAD)

Echo Cancellation(G.168), with up to 128ms

Adaptive (Dynamic) Jitter Buffer

Hook Flash

Programmable Gain Control

T.38/Pass-through

High speed fax up to 14.4kbps

Modem/POS

DTMF mode: Signal/RFC2833/INBAND

VLAN 802.1P/802.1Q (Voice/data/management VLANs)

Layer3 QoS and DiffServ

Supplement Service

Call Waiting

Blind Transfer

Attend Transfer

Call Forward on Busy

Call Forward on No Reply

Unconditional Call Forward

Warm/Immediately Hotline

Call Hold

Do-not-disturb

3-Way Conference

Message Waiting Indicator

FXS

Connector: RJ11

Dial Mode: DTMF and Pulse

Pulse: 10 and 20 PPS

Caller ID: DTMF/FSK CLI Presentation

Max Cable Length: 3KM

Reversed Polarity

Programmable Call Progress Tone

Software Features

Hunting Group

Web ACL

Telnet ACL

Action URL

PPPoE/IPv4/IPv6

Digitmap

Bandwidth Optimization

Routing Rules based Prefixes

Caller/Called Number Manipulation

Maintenance

SNMP v1/v2/v3

TR069

Auto Provisioning

Web/Telnet

Configuration Backup/Restore

Firmware Upgrade via Web

CDR

Syslog(Emerg,alert, critical,error warning,notice,info, debug)

Ping/Tracert Test

Network Capture

Outward Test(GR909)

NTP/Daylight Saving Time

IVR local Maintenance

Cloud-based Management

VoIP

Protocol:

SIP v2.0 (UDP/TCP),RFC3261

SDP,RTP(RFC2833), RFC3262, 3263,3264,3265,2976,3311

ETC (3GPP TS 24.629, RFC 3515, RFC 3891, RFC 3892)

SIP over TLS

RTP/RTCP, RFC2198, 1889

RFC4028 Session Timer

RFC3266 IPv6 in SDP

RFC2806 TEL URI

RFC3581 NAT,rport

Primary/backup SIP server

Outbound Proxy

DNS SRV/ A Query/NATPR Query

SIP Trunk

Early Media/Early Answer

NAT:STUN, Static/Dynamic NAT

Environmental

Power Supply:

100-240VAC, 50-60 Hz@DC12V 1A

Power Consumption:5W(Typical)

Storage Temperature: -20 ℃ ~80 ℃

Humidity:10%-90% Non-Condensing

Dimensions(W/D/H): 140*86*25mm

Unit Weight: 0.2kg

Compliance: CE, FCC

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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.