DAG1000-4S(GE) FXS Analog VoIP Gateway

Overview

DAG1000-4S(GE) is a multi-functional VoIP gateway with 1000Mbps of full duplex connection through its combo copper/ SFP uplinks. The gateway offers 2 fast ethernet ports, 1 SFP port and 4 FXS RJ11 Telephone ports, which helps to seamless connectivity between IP-based telephony networks and legacy telephones (POTS), fax machines and PBX systems. It is ideally suited for active FTTx application, small and medium businesses, service providers and multi-location environments that need VoIP services.

DAG1000-4S(GE)

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
- Combo copper / SFP slot interface runs 1000Mbps (Gigabit) on up-link port
- 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, 100/1000Mbps, RJ-45

   1* LAN, 100/1000Mbps, RJ-45

   1* SFP Slot (uplink port)

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

Speed Dial

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

OpenVPN

Digitmap

Routing Rules based Prefixes

Caller/Called Number Manipulation

Maintenance

SNMP v1/v2/v3

TR069, TR181

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,

   3263,3264,3265,3515,2976,3311

   ETC (3GPP TS24.629), 3891,3892

SIP over TLS

RTP/SRTP/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): 170*118*25mm

Unit Weight: 0.5kg

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