MTG5000 VoIP Trunk Gateway

Overview

MTG5000 is a new-generation intelligent VoIP gateway, and it is purposely designed for large enterprise network, call center and Telcom service provider to connect with E1/T1 network interfaces.

It is developed with the aspect of powerful call control features and maintenance tools. MTG5000 supports high density calls with a very stable system support. It also provides carrier-grade VoIP and FoIP services, as well as value-added functions such as fax modem and voice recognition service.

MTG5000

MT5000 supports a range of signaling protocols, offers the connection between SIP and traditional PSTN signaling like SS7 and PRI. It supports multiple codec formats, offers signaling encryption and ASR interface. MTG500 can be implemented with large call center solution, telephone system or IP PBX and service provider for IP/PSTN call services.

Key Features

- Carrier grade hardware design, 1+1 power supply and hardware-based HA
- Highly scalable and compact design structure, support up to 64 E1 ports(MAX) in 3.5U size
- Support flexible dialing rules and operations, allowing users to customize dialing rules according to different call routing environments
- Support multiple codec formats: G.711A/U, G.723.1, G.729A/B and iLBC
- Very good compatibility, and be workable with Avaya PBX, NEC and Alcatel, and large soft-switch from Huawei, Cisco and ZTE etc.

Physical Interfaces

E1/T1 Ports: 4~64 E1/T1

DTU Module: 4 E1/T1

Interface Type:

RJ48(Impedance 120Ω）

Ethernet Interface:

GE1: 10/100/1000 BaseT Adaptive Ethernet

GE0: 10/100/1000 BaseT Adaptive Ethernet

Serial Port:

1* RS232, 115200bps

1* USB2.0

Voice Capabilities

Codecs:G.711a/μ law,G.723.1, G.729A/B, iLBC, AMR

Silence Suppression

Comfort Noise

Voice Activity Detection

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

Adaptive Dynamic Buffer

Voice ,Fax Gain Control

FAX:T.38 and Pass-through

Support Modem/POS

DTMF Mode: RFC2833/Signal/In-band

Clear Channel/Clear Mode

Environmental

1+1 Redundancy Power Supply

Power Supply: 100-240VAC, 50-60 Hz

Power Consumption:125W

Operating Temperature:0 ℃ ~ 45 ℃

Storage Temperature: -20 ℃ ~80 ℃

Humidity:10%-90% Non-Condensing

Dimensions(W/D/H): 437*345*154mm(3.5U)

Unit Weight: 12.8kg

Compliance: CE, FCC

PSTN

ISDN PRI

23B+D(T1),30B+D(E1),NT or TE

ITU-T Q.921, ITU-T Q.931, Q.Sig

Signal 7/SS7

ITU-T, ANSI，ITU-CHINA

MTP1/MTP2/MTP3, TUP/ISUP

E1 Frame Type : DF,CRC-4,CRC_ITU

T1 Frame Type :

2-Frame Multi-frame (F12, D3/4)

Extended Super-frame (F24, ESF)

Line Codes:

E1:HDB3 T1:B8ZS

Clock : Local/Remote Clock Source

Maintenance

Web GUI Configuration

Data Backup/Restore

PSTN Call Statistics

SIP Trunk Call Statistics

Firmware Upgrade via TFTP/FTP/Web

Network Capture

SNMP v2

Syslog:

Debug, Info, Error, Warning , Notice

Call History Records via Syslog

NTP Synchronization

Centralized Management System

Call Features

Local/Transparent Ring Back Tone

Overlapping Dialing

Dialing Rules, with up to 2000

PSTN group by E1 port or E1 Timeslot

IP Trunk Group Configuration

Voice Codecs Group

Caller and Called Number Whitelists

Caller and Called Number Blacklists

Access Rule Lists

IP Trunk Priority

VoIP Protocol

SIP v2.0 (UDP/TCP),RFC3261

SDP,RTP(RFC2833), RFC3262,

3263,3264,3265,3515,2976,3311

SIP TLS/SRTP

RTP/RTCP, RFC2198, 1889

SIP-T,RFC3372, RFC3204, RFC3398

SIP Trunk Work Mode : Peer/Access

SIP/IMS Registration :With up to 2000

SIP Accounts

NAT: Dynamic NAT, Rport

Call Routing Features

Flexible Route Methods

PSTN-PSTN, PSTN-IP, IP-PSTN

Intelligent Routing Rules

Call Routing base on Time

Call Routing base on Caller/Called Prefixes

256 Route Rules for each Direction

Caller and Called Number Manipulation