MTG3000 Trunk Gateway

Overview

MTG3000 is a carrier grade VoIP gateway, which is designed for telecom operators, ITSPs with high reliability and performance. Focusing on a concept of maintainable, manageable and operable, MTG3000 adopts STM-1 interface which features high integration and large capacity. It provides carrier-grade VoIP and FoIP services, as well as value-added functions such as modem and voice recognition. Thus it constructs a flexible, high-efficient, future-oriented communication network for users.

MTG3000

MTG3000 supports a range of signaling protocols, realizing the interconnection between SIP and traditional signals like SS7 and PRI. It supports multiple codec methods, offers signal encryption technology and smart voice recognition technology, and improves the utilizing efficiency of trucking resources while ensuring voice quality. The trunk gateway is ideally fit for various networks of ITSPs, telecom operators and large-scale enterprises.

Key Features

- Carrier grade hardware design, 1+1 power supply and MCU, hot plug
- High-integrated structure, STM-1 155M (63*E1) in 2U size
- Support flexible dialing rules and operations, allowing users to customize dialing rules according to different working environments
- Support multiple coding standards: G.711A/U, G.723.1, G.729A/B and iLBC
- High compatibility, interoperable with PBX of Avaya, NEC and Alcatel, and also leading soft-switch of Huawei, Cisco and ZTE etc.

Physical Interfaces

SDH Interfaces

2* Standard LC SDH, 155M

1+1 Redundancy Channels Protection

Master/Slave Clock Source

Main Control Unit(MCU)

1+1 Redundancy, Hot Plug

Digital Processing Unit (DTU)

4* DTU Maximum

Support 512 Voice Channels Each Board

Ethernet Interface

GE1: 10/100/1000 BaseT Adaptive Ethernet

GE0: 10/100/1000 BaseT Adaptive Ethernet

Serial Port

1* RS232, 115200bps

VoIP Protocol

SIP v2.0 (UDP/TCP),RFC3261

SDP,RTP(RFC2833), RFC3262,

3263,3264,3265,3515,2976,3311

RTP/RTCP, RFC2198, 1889

SIP-T,RFC3372, RFC3204, RFC3398

SIP Trunk Work Mode :Peer/Access

SIP/IMS Registration:

with up to 256 SIP Accounts

NAT: Dynamic NAT, Report

Call 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

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/Inband

Clear Channel/Clear Mode

Software 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 White Lists

Caller and Called Number Black Lists

Access Rule Lists

IP Trunk Priority

Environmental

Redundant Power Supply

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

Power Consumption:110W

Operating Temperature:0 ℃ ~ 45 ℃

Storage Temperature: -20 ℃ ~80 ℃

Humidity:10%-90% Non-Condensing

Dimensions(W/D/H): 437*320*88mm(2U)

Unit Weight: 6.5kg

Compliance: CE, FCC,CCC

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,MF_CRC,MF

Line Code: HDB3

Clock Source:

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