



MTG3000 Trunk Gateway

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 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.

Product Picture



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.

Honesty Concentration

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

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

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

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, Rport

Innovation Open-mind

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
- RTP and Signaling Encryption (VOS RC4)

Call Features

- Flexible Route Methods
PSTN-PSTN, PSTN-IP, IP-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

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

Environmental

- Redundant Power Supply
- Power Supply: 100-240VAC, 50-60 Hz
- Power Consumption: 110W
- Operating Temperature: 0 °C ~ 45 °C
Storage Temperature: -20 °C ~ 80 °C
- Humidity: 10%-90% Non-Condensing
- Dimensions(W/D/H): 437*320*88mm(2U)
- Unit Weight: 6.5kg
- Compliance: CE, FCC, CCC