DINSTAR

MTG3000 Trunk Gateway

Overview

MTG3000 is a carrier-grade VoIP gateway, which is designed for telecom operators, and ITSPs with high reliability and performance. Focusing on a concept of maintainable, manageable, and operable, MTG3000 adopts the 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 (Optional)
  2* Standard LC SDH, 155M
  1+1 Redundancy Channels Protection
- Main Control Unit(MCU)
  1+1 Redundancy, Hot Plug
- Digital Processing Unit (DTU)
  Up to 4 DTU Boards
  Support 512 Voice Channels Each Board
- Ethernet Interface
  GE1: 10/100/1000 BaseT Adaptive Interface
  GE0: 10/100/1000 BaseT Adaptive Interface
- Serial Port
  1* RS232, 115200bps

VoIP Protocol

- SIP v2.0 (UDP/TCP/TLS), RFC3261
  SDP, RTP(RFC2833), RFC3262,
  RFC3263, RFC3264, RFC3265,
  RFC3515, RFC2976, RFC3311
- RTP/RTCP, RFC2198, RFC1889
- SIP-T, RFC3372, RFC3204, RFC3398
- SIP Trunk Work Mode: Peer/Access
- SIP/IMS Registration: with up to 2,000 SIP Accounts
- NAT: Dynamic NAT, Rport

Call Features

- Flexible Route Methods
  PSTN-IP, IP-IP, IP-PSTN
- Intelligent Routing Rules
- Call Routing based on Time
- Call Routing based on Caller/Called Prefixes
- 512 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, G.726
- Silence Suppression
- Comfort Noise
- Voice Activity Detection
- Echo cancellation (G.168 up to 128ms tail)
- Adaptive Dynamic Buffer
- Voice, Fax Gain Control
- FAX: T.38 and Pass-through
- Support Modem/POS
- DTMF Mode: RFC2833/Signal/In-band

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_CRC4, MF
- Line Code: HDB3
- Clock Source:
  Local/Remote Clock Source

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

Software Features

- Local/Transparent Ring Back Tone
- Overlapping Dialing
- Dialing Rules, with up to 8000
- PSTN group by E1 port or E1 Timeslot
- IP Trunk Group Configuration
- Voice Codecs Group
- Caller/Callee Black List
- Caller/Callee White List
- Access Rule Lists
- IP Trunk Priority
- Full Concurrency Recording
- Call Frequency Control

Maintenance

- Web GUI Configuration
- Data Backup/Restore
- PSTN Call Statistics
- SIP Trunk Call Statistics
- Firmware Upgrade via TFTP/FTP/Web
- SNMP v2
- Network Capture
- Syslog: Debug, Info, Warning, Notice
- Call History Records via Syslog
- NTP Synchronization
- Centralized Management System

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DINSTAR

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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.