MTG2000B Trunk Gateway

Overview

MTG2000B is a new-generation intelligent VoIP gateway, which is designed for enterprises, telecom operators and various industries. Focusing on a concept of maintainable, manageable and operable, MTG2000B 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.

MTG2000B

MTG2000B 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 access networks of SMEs, call centers, telecom operators and large-scale enterprises.

Key Features

- Carrier grade hardware design, 1+1 power supply and MCU, hot plug
- High-integrated structure, up to 16E1 ports in 1U 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

E1/T1 Ports

4/8/12/16 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

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 256

SIP Accounts

NAT: Dynamic NAT, Rport

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

RTP and Signaling Encryption(VOS RC4)

Environmental

1+1 Redundancy Power Supply

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

Power Consumption:45W

Operating Temperature:0 ℃ ~ 45 ℃

Storage Temperature: -20 ℃ ~80 ℃

Humidity:10%-90% Non-Condensing

Dimensions(W/D/H): 436*300*44.5mm(1U)

Unit Weight: 3.8kg

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 :

4-Frame Multi-frame (F4,FT),

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

Extended Super-frame (F24, ESF) ,

Remote Switch Mode (F72, SLC96)

Line Co. des:

E1:NRZ,CMI,AMI,HDB3

T1:NRZ,CMI,AMI,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

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DINSTAR

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.