An STL Product

MTG2000 VoIP Trunk Gateway

Overview

MTG2000 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, MTG2000 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.

MTG2000 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 with modular slots, 1+1 hot swappable power supply
- Supports IPv4 and IPv6 in addition to Routing Protocols, static routes, PPP, PPPoE, Frame Relay, 802.1 p/q VLAN Tagging, QoS
- High-integrated structure, up to 20 PRI/E1 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.726, G.729A/B and iLBC/Opus
- Replaceable media module and fan trays
- Futuristic support for multiple PSTN Interfaces
- Supports SIP Trunks
- Inbuilt SBC functionalities

Physical Interfaces

E1/T1 Ports

4/8/12/16/20 E1/T1

DTU Module :

4 E1/T1

Interface Type

RJ48(Impedance 120Ω）

Ethernet Interfaces (LAN and WAN)

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.726

G.729A/B, iLBC, AMR, Opus

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

Self Cooling mechanism

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/R2

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 Codes:

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 v3

Syslog:

Debug, Info, Error, Warning , Notice

Call History Records via Syslog

NTP Synchronization

Centralized Management System

Supports SSH/SCP

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

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 Features

Flexible Route Methods

PSTN-PSTN, PSTN-IP, IP-PSTN

Intelligent Routing Rules and policies

Call Routing base on Time

Call Routing base on Caller/Called Prefixes

256 Route Rules for each Direction Caller and Called Number Manipulation

DSP Support for devices not supporting VoIP

802.1q VLAN, PPP, MLPPP, FR, MLFR, HDLC

NEOX

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About Sterlite Technologies Ltd. (STL)

STL is a leading global optical and digital solutions company providing advanced offerings to build 5G, Rural, FTTx, Enterprise, and Data Centre networks.The company, driven by its purpose of ‘Transforming Billions of Lives by Connecting the World’, designs and manufactures in 4 continents with customers in more than 100 countries. Telecom operators, cloud companies, citizen networks, and large enterprises recognise and rely on STL for advanced capabilities in Optical Connectivity, Global Services, and Digital and Technology solutions to build ubiquitous and future-ready digital networks. STL’s business goals are driven by customer-centricity, R&D and sustainability. Championing sustainable manufacturing, the company has committed to achieve Net Zero emissions by 2030.