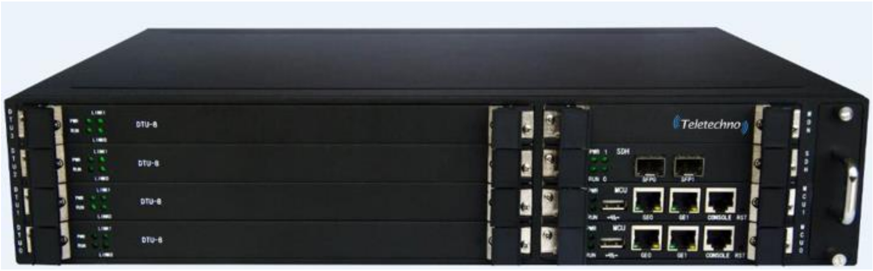
TELETECHNO GATEWAY SIP TO SS7/PRI
TG-63E1

TG-63E1 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, TG-63E1 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.

TG-63E1 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.

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

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

- Clock Source Local/Remote

  Clock Source

- Line Code

  HDB3

- Clock Source Local/Remote
- Clock Source

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

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

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 ℃ ~ 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