TG-20E1 Trunk Gateway

TG-20E1 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, TG-20E1 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-20E1 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 voicerecognition 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.

TG-20E1 CHASSIS

Key Features

Multi-port and high-integrated structure; up to 20 E1/T1 with 1U size Provide various services, such as VoIP, FoIP, Modem and POS 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 softswitch of Huawei,Cisco and ZTE etc

TDM TO SIP CONCENTRATION

Physical Interfaces

- **E1/T1 Ports**
  4/8/12/16/20 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

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,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

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
- 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