Techroutes
NETWORK

Career Class High Density VoIP Gateways

Protocols Support
SIP/
SS7 / PRI

Product Overview

TG5200 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, TG5200 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. TG5200 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

- 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.729AB, iLBC and AMR
- 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 4/8/12/16/20 E1/T1
- Five of DTU card slots
- DTU Module :4 E1/T1
- Interface Type RJ48(120Ω)
- Ethernet Interface
- GE1: 10/100/1000 BaseT Adaptive
- GE0: 10/100/1000 BaseT Adaptive
- Serial Port : 1* RS232, 115200bps

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 Capabilities

- Local/Transparent Ring Back Tone
- Dial Overlapping
- Upto 2000 Dialing Rules
- 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)
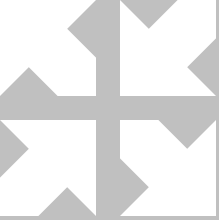
Technical Specifications

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

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

Maintenance

- Radius
- 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

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

PTSN

- 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)