KNGW-E1T1-2000 VoIP Trunk Gateway

Overview

KNGW-E1T1-2000 is a newgeneration intelligent VoIP gateway, which is designed for enterprises, telecom operators, and various industries. Focusing on a concept of maintainable, manageable, and operable, KNGW-E1T1-2000 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

KNGW-E1T1-2000

KNGW-E1T1-2000 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 SiEs, call centers, telecom operators, and large‐scale enterprises.

Key Features

- Carrier-grade hardware design, 1+1 power supply

- High-integrated structure, up to 20 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.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

  4/8/12/16/20 E1/T1 Ports
- DTU Module:

  4 E1/T1 Ports
- Interface Type

  RJ48(Impedance 120Ω)
- Ethernet Interface

  GE1: 10/100/1000 BaseT Adaptive Interface

  GE0: 10/100/1000 BaseT Adaptive Interface
- Serial Port

  1* RS232, 115200bps

Voice Capabilities

- Codecs: G.711a/µ law,G.723.1, G.729A/B, iLBC, AMR
- Silence Detection

  Comfort Noise
- Echo cancellation (G.168 up to 128ms tail)
- Adaptive Dynamic Buffer
- Voice, Fax Gain Control
- FAX: T.38 and Pass-through
- Support Modem/POS
- DTMF Mode: RFC2833/Signal/In-band

Call Features

- Flexible Route Methods

  PSTN-PSTN, PSTN-IP, IP-IP, IP-PSTN
- Intelligent Routing Rules
- Call Routing based on Time
- Call Routing based on Caller/Called Prefixes
- 512 Route Rules for each Direction
- Caller and Called Number Manipulation

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-CRC4, MF
- T1 Frame Type :

  Super-frame (F12, SF)

  Extended Super-frame (F24, ESF)
- Line Codes:

  E1: HDB3, T1: 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
- SNMP v2
- Network Capture
- Syslog: Debug, Info, Warning, Notice
- Call History Records via Syslog
- NTP Synchronization
- Centralized Management System

Software Features

- Local/Transparent Ring Back Tone
- Overlapping Dialing
- Dialing Rules, with up to 8000
- PSTN group by E1 port or E1 Timeslot
- IP Trunk Group Configuration
- Voice Codecs Group
- Caller/Callee Black List
- Caller/Callee White List
- Access Rule Lists
- IP Trunk Priority

VoIP Protocol

- SIP v2.0 (UDP/TCP/TLS), RFC3261 SDP, RTP(RFC2833), RFC3262, RFC3263, RFC3264, RFC3265, RFC3515, RFC2976, RFC3311
- RTP/RTCP, RFC2198, RFC1889
- SIP-T, RFC3372, RFC3204, RFC3398
- SIP Trunk Work Mode: Peer/Access
- SIP/IMS Registration: with up to 2,000 SIP Accounts
- NAT: Dynamic NAT, Rport

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