

UBG5000 PRI Gateway

High Availability Digital VoIP Gateway

4-64 E1/T1



Product Description

UBG5000 is a new-generation intelligent VoIP gateway, and it is purposely designed for large enterprise network, call center and Telecom service provider to connect with E1/T1 network interfaces. It is developed with the aspect of powerful call control features and maintenance tools. UBG5000 supports high density calls with a very stable system support. It also provides carrier-grade VoIP and FoIP services, as well as value-added functions such as fax modem and voice recognition service. UBG5000 VoIP Trunk Gateway Overview the Gateway to VoIP World UBG5000 supports a range of signaling protocols, offers the connection between SIP and traditional PSTN signaling like SS7 and PRI. It supports multiple codec formats, offers signaling encryption and ASR interface. UBG5000 can be implemented with large call center solution, telephone system or IP PBX and service provider for IP/PSTN call services

Product Features

- Carrier grade hardware design, 1+1 power supply and hardware-based HA
- Highly scalable and compact design structure, support up to 64 E1 ports(MAX) in 3.5U size
- Support flexible dialing rules and operations, allowing users to customize dialing rules according to different call routing 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.

UBG5000 Digital VoIP Gateway

Physical Interfaces

E1/T1 Ports 4-64

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

1* USB2.0

Voice Capabilities

Codecs: G.71a/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-band

Clear Channel /Clear Mode

Environmental

1+1 Redundancy Power Supply

Power Supply: 100-240VAC, 50-60 Hz

Power Consumption:125W

Operating Temperature:0 °C ~ 45 °C

Storage Temperature: - 20 °C ~80 °C

Humidity:10%-90% Non-Condensing

Dimensions(W/D/H):

437*340*154mm(3.5U)Unit Weight:

12.8kg

Compliance: MTCTE

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 MTP1

MTP2/MTP3, TUP/ISUP

E1 Frame Type : DF, CRC-4, CRC ITU

T1 Frame Type :

2-Frame Multi-frame (F12, D3 /4),

Extended Super-frame (F24, ESF)

Remote Switch Mode (F72, SLC96)

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

Network Capture

SNMP v2

Syslog:

Debug, Info, Error, Warning, NoticeCall

History Records via Syslog

NTP Synchronization

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 Routing Features

Flexible Route Methods PSTN-

PSTN, PSTN-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