PRODUCT DATASHEET

PRI Gateway

High Availability Digital VoIP Gateway

UB1000

Product Description

UB1000 Series E1/T1 Digital VoIP Gateways with 1/2 ports E1/T1 is a compact and cost-effective trunk gateway designed to interconnect between PSTN and IP networks. With powerful hardware design, MTG1000 series has comprehensive PSTN access capabilities as well as SIP to SIP interworking features that enables the interconnection between all these elements.

UB1000 series trunk gateway with high-efficient design and strong DSP processor ensures high-performance of the interconversion of PCM voice signal and IP packets, even when the gateways are fully loaded.

Product Features

- 1/2 ports E1/T1 in 1U chassis
- Dual Power Supplies
- Up to 60 simultaneous calls
- Flexible routing
- Multiple SIP trunks
- Fully compatible with mainstream VoIP platforms

UB1000 Digital VoIP Gateway

Physical Interfaces

E1/T1 Ports
1 to 2 E1/T1
Interface Type
RJ48(Impedance 120Ω)
Ethernet Interface
2X RJ45 Adaptive Ethernet Interface
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

R2 MFC
standard 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

Call 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

Voice Capabilities

Codecs:
G.711a/μ law,G.723.1,
G.729A/B, iLBC 13k/15k,AMR
Silence Suppression
Comfort Noise
Voice Activity Detection
Echo Cancellation (G.168), with up
to128ms
Adaptive Dynamic Buffer
Voice ,Fax Gain Control
FAX:T.38 and Pass-through
Support Modem/POS

DTMF Mode:
RFC2833/SIP Info/In-band
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 TLS/SRTP
SIP-T,RFC3372, RFC3204, RFC3398
SIP Trunk Work Mode :Peer/Access

SIP/IMS Registration :
with up to 256 SIP Accounts
NAT: Dynamic NAT, Rport

Environmental
1+1 Redundancy Power Supply
Input 100-240VAC, 50-60 Hz
Power Consumption:15W
Operating Temperature:0 ℃ ~ 45 ℃
Storage Temperature: -20 ℃ ~80 ℃
Humidity:10%-90% Non-Condensing
Dimensions(W/D/H):
436*300*44.5mm(1U)
Unit Weight: 2.0kg
Compliance: CE, FCC

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

Maintenance
Web GUI Configuration
Data Backup/Restore
PSTN Call Statistics
SIP Trunk Call Statistics
Firmware Upgrade via TFTP/Web
SNMP v1/v2/v3
Network Capture

Syslog:
Debug, Info, Error, Warning , Notice
Call History Records via Syslog
NTP Synchronization
Centralized Management System

UBIQCOM INDIA PVT. LTD.

Corporate Office: D-92, Sector-63, Noida, UP-201301 | Website: www.ubiqcom.in | Email: info@ubiqcom.in
Call Us: +91 9999 284 312, 9811 417 703
Branch Offices: Delhi, Noida, Mumbai, Pune, Chandigarh, Bengaluru, Hyderabad, Ahmedabad, Chennai, Kolkata, Kochi