VoIP Solutions Redefining Connection!

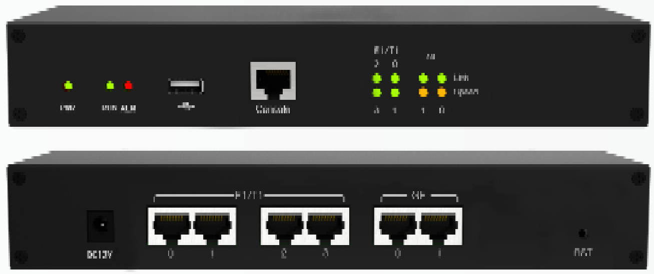
UBIQCOM™

PRODUCT DATASHEET

UBG200

PRI Gateway

1/2/4 E1/T1



Product Description

UBIQCOM UBG200 Trunk Gateway enables direct routing of calls between the fixed line ISDN and the cost-effective IP networks for capitalizing on low cost VoIP telephony. By integrating UBG200 Trunk Gateway with existing PBX / PABX telephone systems, businesses of all sizes can benefit from low-cost Voice over IP calls and achieve substantial cost savings without the high upfront costs associated with communications infrastructure changes related with the migration to VoIP.

UBG200 offering PBX/IP-PBX users a powerful telephony cost saving device and advanced flexible routing capabilities between ISDN PRI to SIP, and also can be situated between the PBX and the PSTN, saving the cost of an additional PBX E1/T1 port. It is ideally fit for various access networks of SMEs, call centers, telecom operators and large-scale enterprises.

Key Features

- Up to 60 simultaneous calls with 1/2/4 E1/T1 ports
- Support echo cancellation, DJB, CNG, VAD and QoS
- Use of existing E1/T1 ISDN PRI interface of the PBX
- Use of existing VoIP interface of the IP-PBX
- Maintain existing dialing habits and business communication patterns
- Built-in Web-based management, SNMP, command line interface (CLI)
- Standard ISDN compliant and Interoperable with a wide range of IP-PBXs

PRODUCT FEATURES AND SPECIFICATIONS

UBG200 PRI Gateway

Physical Interfaces

E1/T1 Ports

1/2/4 E1/T1 Ports

Interface Type

RJ48 (Impedance 120Ω)

Ethernet Interface

GE1: 10/100/1000 Base-T Adaptive Interface

GE0: 10/100/1000 Base-T Adaptive Interface **Serial Port**

1* RS232, 115200bps

Voice Capabilities

Default codec:G.711a/μ law, AMR G.723.1, G.729AB, iLBC
In-band G.711, RFC2833 (DTMF out-of-band), SIP INFO Voice Activity Detection

Support CAS (R2, MFC/R2) vai SDN PRI (Q.921, Q.931, Q.Sig) –30B+D(E1), NT/TE
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/SIP Info/In-band

Software Features

Local/Transparent Ring Back Tone
Overlapping Dialing
Dialing Rules, with up to 8000
Voice Codecs Group
Access Rule Lists
Caller/Callee Black List
Caller/Callee White List
IP Group Management
Route direction: PSTN-IP, IP-PSTN, PSTN-PSTN, IP-IP

CPS = 15
30 PCM channels (64 kbps per channel) / 1 E1 stream
Support: PPPoE, DHCP, static IP
Support 60 IP routing tables.

PSTN

ISDN PRI

23B+D(T1),30B+D(E1),NT or TE ITU-T Q.921, ITU-T Q.931, Q.Sig

SS7 (optional)

ITU-T,ANSI, ITU
MTP1/MTP2/MTP3, TUP/ISUP

R2 MFC (optional)
China and other 22 variants standard

E1 Frame Type :

DF,CRC-4, MF-CRC4, MF

T1 Frame Type :

Super-frame (F12, SF) Extended Super-frame (F24, ESF)

Line Codes:

E1: HDB3, T1: B8ZS

Maintenance

Web GUI Configuration, HTTP/HTTPS 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

VoIP Protocol

SIP v2.0 (UDP/TCP),RFC3261
SDP,RTP(RFC2833), RFC3262, 3263,3264,3265,3515,2976,3311
RTP/RTCP, RFC2198, 1889
TLS/SRTP
SIP-T,RFC3372, RFC3204, RFC3398
SIP Trunk Work Mode :Peer/Access

SIP/IMS Registration :

with up 1000 SIP Accounts,
Services: SIP trunk, Ethernet LAN port configuration, Call routing, Device reboot, reset, Web interface access via LAN, Custom web access TCP port NAT: Static/Dynamic NAT, Rport
TACACS, RADIUS

Environmental

Power Adapter: 100-240VAC@DC12V2A
Power Consumption:15W
Operating Temperature:0 ℃ ~ 45 ℃
Storage Temperature: -20 ℃ ~80 ℃
Humidity:10%-90%Non-Condensing
Dimensions(W/D/H):225*150*38mm
Unit Weight: 0.8kg
Compliance: MTCTE

Voltage stability: Load regulation = (Vno load Vfull load) / Vnoload ×100% ≤ 5%

Ripple voltage: Vpp≤ 150 mV (under 100% load across the entire input voltage range)

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