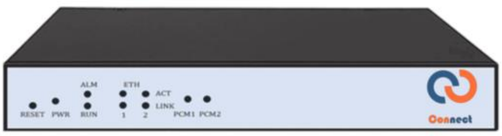
UMTG 100 Lite

Connect Unified Media Trunk Gateway 100 Lite

The Connect UMTG 100L, with mini size for better space and shipment efficiency, are new members of Connect's VoIP gateway family that enables service providers and enterprises to maximize value of their networks and services. It converts digital EI/TI PSTN message into IP formats and secures sessions across IP and mixed network boundaries to support the seamless delivery of services.

Connect UMTG 100L are unparalleledly cost effective and compliant with PRI ISDN, R2 and SS7 packets, and adopt the equivalent hardware architecture, with features high reliability and unparalleled cost efficiency to delivers a perfect alternative option for enterprises, operators and system integrators. Key Features and Benefits

- 30/60 simultaneous SIP sessions with multimedia transcoding, and 30/60 channels of ISDN signalling
- Help configure SIP, SIP trunking, SIP Mediation, PCM and ISDN Routing and more
- Combined IP and TDM gateway features on a single platform
- SIP profiler and web-based user configuration
- Integrated transcoding support for voice, tone and faxing
- Support for ISDN, SIP signalling, and SIP UMTG 100 Lite interworking along with voice and transcoding



Flexible and efficient VoIP Gateway Solution

The Connect UMTG IOOL With its scalable density and versatility can help enable wireless and wireline service providers to add new Value-Added Services (VAS) quickly, and provide a clear migration path to an all-IP network. It can scale up to 60 simultaneous IP sessions and at the same time provide media transcoding and impressive sessions per second.

Connect UMTG 100L support voice densities ranging from 30 to 60 channels, call routing, call translation and IP transcoding in a single mini chassis for gateway operations. The integrated gateway functionality not only provides interworking between IP and TDM domains, but also automated failover from IP to TDM for outbound routing. These features help service providers looking to improve network and routing resiliency and lower TCO. These capabilities make the UMTG100L an excellent option for mobile VAS, SIP trunking, contact center and emergency service deployments, as well as for retail, wholesale, business, and enhanced Voice over IP (VOIP) services.

Any-to-Any Signalling and Multimedia Connectivity

The connect UMTG 100L provides any-to-any network connectivity through its ability to interwork multiple protocols used by telecommunications providers to deliver services to their retail, business and wholesale customers. It can provide interworking between ISDN, SS7, SIP formats.

Connect UMTG 100L also supports any-to-any media transcoding for popular voice codecs. T.38 and G.711 fax interworking and support for RTP, INBAND and SIPINFO method based tones and event handling complement the media transcoding capabilities to provide a high degree of flexibility to help deliver value added services economically.

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User-friendly management and configuration toolkits

The Web graphical user interface (Web UI) is a real-time web toolkit to configure, monitor UM TG 100L . It allows operators to configure and perform real-time monitoring and maintenance. Flexible SIP and Protocols configuration enable services providers and enterprises to seamlessly connect in hybrid networks, Helping configure SIP, SIP trunking, SIP Mediation, PCM, SS7and ISDN, Routing and more; And a broad range of gateway toolkits also help gateway's maintenance and software
upgrading for Web UI, gateway services and firmware.

Specifications

Routing Features

- Call Routing and translation (from PCM to IP reversely)
- IP Bearer Features
- Coder support: G.711A, G.711U, G.729 GSM, iLBC, RFC 2833, RF
- Compliant with TCP/UDP, HTTP, ARP/RARP, DNS, NTP, TFTP, TELNET, STUN and more IP protocols
- Echo cancellation: G. 168 128 ms tail length
- Voice activity detection and packet loss concealment
- Comfort noise generation
- T.38 real-time fax, T.38 - G. 711 interworking
- Digit transmission via RFC 2833 (SIP)
- Hosted NAT

OAM&P

- Network Time Protocol (NT P)
- Web User Interface (Web UI) supports configuration via browser
- SNMP MIBs

IP Protocols

- Connect SIP Specifications and Notable Extensions
- RFC 3261 SIP Basic
- RFC 3262 SIP PRACK
- RFC 3265 SIP Subscribe/Notify

QOS

- Adaptive jitter buffer
- Packet loss compensation
- Configurable Type of Service (TOS) fields for packet
- prioritization and routing

Notable SIP Extensions

- RFC 3398 ISUP/SIP Mapping

I/O Interfaces — Rear I/O — T1/E1

- Telephony
- Fiber Optical 1 ~ 2 T 1/ EI for timing (BITS clock),
- T1 and E1
- signalling and bearer traffic (T1 - 100 ohms and E1 - 120 ohms)
- Clock Sync Stratum

IP Interfaces

- Dual redundant 2 *100 Base-T Ethernet for VoIP payload and signalling

TDM Signalling Protocols

- ISDN PRI
- SS7 ISUP
- SS7 MTP1 ~ 3
- SS7 SIGTRAN
- SS7 TCAP

Other Physical Features

- AC Power Supply Range 100 ~ 240VAC
- The power supply frequencies: - 47 Hz and 63 Hz
- Power consumption: - 15W (Normal Conditions)
- Operating humidity: 8 to 90% (non-condensing)
- Storage temperature range: -20 to +85 ^{0}C, 8-90% relative humidity non-condensing

Management & Physical Dimensions

- Power supplies field installation
- High: 1.18 in (30 mm)
- Wide: 7.48in (190 mm)
- Deep: 4.72in (120 mm)
- Weight 1.43 1b (Approx. 0.65kg)

• RFC 3711 SRTP (for SIP) • Tel URI - RFC 3966 • IP and ISUP interworking and more	

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