Connect 128S/H VoIP Gateway adopts the latest modular architecture with built-in server, and opens a new milestone to maximize VOIP and TDM network's value for Service Providers and application developers. One of major advantage to reduce time to market and introduce innovative applications more efficiently and effectively. Tailored to satisfy diverse customers' needs.

The hybrid architecture of Connect UMGW allows for standard protocols between different network components and ensures high independence and interoperability, which better caters to sophisticated communications.

Key Features and Benefits

Flexible Configuration for any network

Complaint with diverse networks (FXO, FXS) support various multimedia processing capability (conferencing fax, compression and SuPerFormTM echo cancellation for voice enhancement and an array of protocols.

Low to High Scalability

Modular architecture ensures flexibility and expendability from low density and high density. Modular design Allow for Easy configuration system upgrading or general maintenance.

Multimedia Convergence

Adopt 100M Ethernet switching chipset, Connect UMGW media Stream exchanges in IP packets and across to soft switching system via Media Gateway Controller ensuring high-level application are streamlined

Diverse Media Resources

Support high-capacity voice playback and codecs, conferencing, faxing; support T.38/T.30; IVR and ACD applications, with EXT IVR server or GUI management

Carrier Grade Reliability

Special power system with standby redundancy, advanced cooling system to reassure long standing robustness, special air clear to protect against dust accumulation inside classic, inside temperature controller and alert system; No need to change wiring changing functional modules.

Technical Specifications

Functional Module Available

- UMGW 160/16S 16 FXO or 16 FXS
- Notice – a total of 8 slots for all these modules optimal inbuilt server to run application
- Option 1: UMGW-CPU01
- Intel 1037U Dual Core 1.7 G DDR1333 4GB RAM (Upgradable to 8 GB)
- Option 2: UMGW-CPU04
- Intel i5 3210U Dual Core, 2.5x 3.1G DDR16000, RAID, 4GB RAM, Upgradable to 8GB

Multimedia & Signalling Voice Processing

- Codecs support A-law, u-law, PCM8, PCM 16. 1AM
- Voice file format: support standard WAV format file and any non-format file;
- Support conversion among various (de)coding formats;
- Support real time file replay from RAM and server.
- Support real time recording to RAM and server (Dynamic Storage);
- Support DTMF and FSK transmission and detection;
- Support R2 transmission and reception;
- Support Barge-in function
- Support simultaneous recording /replay; complaint with G.168 echo cancellation, with up to 128ms tail length
- Support AGC/ALS;
- Support Answer Machine Detection
- Support voice call recording (on-demand or permanent);
- Support full-duplex recording and replay;
- All voice channels could be converted to conferencing channels;
- Two voice channels could be converted to a fax session on demand;

Conference/Fax resource

SIP Protocol

- Supported SIP standards:
- IETF RFC 3261 (SIP: Session Initiation Protocol)
- IETF RFC 2327 (SDP-Session Description Protocol);
- IETF RFC 3550 and 3551 (RTP/RTCP);
- IETF RFC 2833 (DTMF);

SIP Protocol Stacks

- Support signalling transmitting over UDP;
- Support call holding;
- Support Digest Authentication;
- Intelligent URL Scheme analysis algorithm;
- Support INVITE/REINVITE in calling processing;
- Support via rPort setting (for NAT/Firewall tunnelling);
- Allow DTMF tone transmission/detection in three modes: inner-band/SIPINFO/out-of-band (RFC2833);
- Support REGISTER messaging and authentication;
- Inner multiple threads mechanism; Support SIP Sever;
- Support UDP "pulse holding" mechanism;
- Support INFO messaging;

Network Interface

- Analog interface: Optimal functional modules for FXO interface, FXS interface or high impedance logging;
- 2*TCP/IP 100 M Ethernet (RJ-45);
- 2* LAN Ethernet (RJ-45);
- 1* Console
- 1*USB

- Support distributed conferencing mode, with conferencing resource in each voice channel;
- Fully support SIP-based Fax T.38 standard
- Support V29/V27/V17 standards, with faxing rate up to 33.6 kbps (automatically slowing down);
- Support ECM (Fax/Error Correction Mode) for reception /transmission (optional for EXM/non-ECM mode);
- Support TIFF files input in MH/MR/MMR format and transmission/reception in MH format;

VoIP Resources
RTP Protocol
- Compliant with RTP/RTCP protocol (RFC3551, RFC3552);
- Coding/Decoding: G.711 (A-law/u-law)/GSM /G.729A;
- Self -adaptive echo cancellation (voice enhancement);
- RTP DTMF loading (RFC2833);
- Support NAT /Firewall monitoring and tunnelling;