UNIWAY2200
Hybrid VoIP Gateway

In hybrid VoIP & TDM era with diverse and rapidly-growing needs of unified communications and services, Synway UNIWAY 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 SP and application developers. One of major advantage of UNIWAY is to reduce time to market and introduce innovative applications more efficiently and effectively. Tailored to satisfy diverse customers' needs, UNIWAY, with its open and standardized format, enables users to develop a range of applications.

The hybrid architecture of UNIWAY allows for standard protocols between different network components and ensures high independence and interoperability, which better caters to sophisticated communications. Also, in the Mobile Internet Era, UNIWAY brings about efficiency and unparalleled cost advantages for developers by optimizing R&D and integrating an array of data, voice, video and other applications.

Key Features and Benefits

- Flexible configuration for any network

Compliant with diverse networks (FXO, FXS, T1, E1, J1, GSM, CDMA, 3G, VoLTE, IP): support various multimedia processing capability (conferencing, fax, compression and SuPerFormTM echo cancellation for voice enhancement and an array of Protocols (SS7, SIGTRAN, ISDN PRI, CAS, R1, R2, Wireless)

- Compliant with any IP-Based applications

With optional inbuilt industrial server, UNIWAY series are compliant with any IP-based applications; it also even supports any category of third party software, including UC, IP-PBX, Contact Center and more. In legacy PSTN network, UNIWAY could converge applications via internal modules.

- Low to High Scalability

Modular architecture ensures flexibility and expandability from low density and high density. Modular design allows for easy configurations, system upgrading or general maintenance

- Multimedia Convergence

Adopt 100M-Ethernet switching chipset, UNIWAY's media stream exchanges in IP packets, and access to soft switching system via Media Gateway Controller, ensuring high-level applications are streamlined

- Diverse Media Resources

Support high-capacity voice playback and Codecs, conferencing, faxing; Support T.38/T.30; optimized for IP-PBX, IVR and ACD applications, with EXT IVR server or GUI management.

UNIWAY2200
Hybrid VoIP Gateway

- Carrier-Grade Reliability

Special power system with standby redundancy; advanced cooling system to reassure long-standing robustness ; special air cleaner to protect against dust accumulation inside chassis; Inside temperature control and alert system; No need to change wiring when changing functional modules.

Technical Specification

Functional module available:

UMG-1016: 16*FXO, 16*FXS, or hybrid 8*FXS+8*FXO

UMG-2120: 1, 2, or 4E1(T1)

UMG-4008: 4 or 8 Wireless Ports(GSM/CDMA/WCDMA/3G/4G)

Notice: a total of 8 slots for all these modules

Optional inbuilt server to run applications:

Option 1: UMG-CPU01

Intel 1037U Dual Core, 1.7G, DDR1333, 4G RAM(Upgradeable to 8G);

Option 2: UMG-CPU04

Intel i5 3210U Dual Core, 2.5~3.1G, DDR1600, RAID, 4G RAM (Upgradeable to 8G)

Multimedia & Signaling

Voice Processing

CODECs: support A-law, μ-law, PCM8, PCM16, IAM-ADPCM, VOX, MP3, GSM, G.729A/B,G.722,G.723, iLBC etc;

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/reception;

Support (standard/self-defined) tone transmission and detection;

Support R2 transmission and reception;

Support Barge-in function;

Support simultaneous recording/replay;

Compliant 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;

Signaling Protocols

E1/T1/J1: support R1, R2, CAS, SS7, SIGTRAN and ISDN PRI;

Signaling interface: CAS supports MFC, R2 and so on;

Signaling interface: SS7 supports TUP and ISUP ;

Support signaling redundancy, changeover and reset; SS7/ISDN data links could be on any timeslot, not only on timeslot 16;

Support 64Kbps SS7 signaling links or 2M Hi-Speed Links;

Support calling distribution among signaling links or among signaling link groups;

Support call transfer among various signaling points or direct-connectivity mode;

Support multiple source signaling point-codes as well as projected signaling point-codes;

Support real-time signaling link's adding, removing, activating, reset, normal setup, emergency setup, anti-congestion;

Support multiple signaling points and signaling point transference;

ISDN supports network terminal and subscriber terminal;

Support overlapping in reception/transmission of called number;

VoIP Resources

RTP Protocol

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

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

UNIWAY2200
Hybrid VoIP Gateway

Support signaling 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 tunneling);

Support REFER call forwarding;

Allow DTMF tone transmission/detection in three modes: inner-band/SIP-INFO/out-of-band (RFC2833);

Support REGISTER messaging and authentication;

Inner multiple-threads mechanism;

Support SIP server;

Support UDP "pulse-holding" mechanism;
Support INFO messaging;

Conference/Fax resource

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 33.6Kbps (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, MR format;

Network Interface

E1 interface: Compliant with G.703, including 75Ω unbalanced interface and 120Ω balanced interface;

T1/J1 interface: DSX-1 and CSU line compensation available for different extents of signal losses, including 100Ω and 110Ω balanced interfaces;

Analog interface: Optional functional modules for FXO interface, FXS interface or high-impedance logging;

2 *TCP/IP 1,00M Ethernet (RJ-45);

2 *LAN Ethernet (RJ-45);

1 *Console (Mini-USB);

Development Environment

Windows OS: Windows2000/XP/2003/Vista/NT;

Linux OS: Including RH7.2/RH9.0/AS4/FC4/SUSE10;
Programming language: ANSI C/C++, Microsoft Visual C++, C#, Delphi;

Security and Certifications

Lighting-proof grade: Level 4;

Certification: FC/CE/China CCC
For RoHS compliance, please contact Synway's sales representatives;

Physical Characteristics

Dimensions: 2U form factor: 88mm (H) x 472mm (W) x 440mm (L)
Net Weight: about 8Kg (different for the number of optional modules)

Power Requirement

AC: 100-240V AC

Power consumption: different for configuration, less than 350Watt;
Environment Requirement

Ventilation: normal;

Operating temperature: 0°C～ 40°C ;

Relative humidity: 10% ～ 85%;

Avoid dust accumulation;

Anti-electrostatic: please Grounded;

Installation recommendation: mounted on standard 19-inches rack;

About Synway

As a major manufacturer and supplier of communication products and solutions, Synway specializes in providing superior Multimedia Gateway, Integrated Multimedia Switch, Telephony Hardware in use for Telecom communications. www.synway.net