Denwa

SWITCHING YOUR LIFE

Digital VoIP Gateways

DW-GTW-AC-E1060

The **DW-GTW-AC-E1060** Enterprise Session Border Controller (E-SBC) and Media Gateway offers a complete connectivity solution for small-to-medium sized enterprises.

The **DW-GTW-AC-E1060** connects IP-PBXs to any SIP trunking service provider, scaling up to 250 concurrent SBC sessions. It offers superior performance in connecting any SIP to SIP environment, legacy TDM-based PBX systems to IP networks, and IP-PBXs to the PSTN, supporting up to 60 voice channels in a 1U platform.

Vast mediation capabilities and proven interoperability

The **DW-GTW-AC-E1060** supports a wide range of voice coders and is capable of transcoding between narrowband and wideband voice coders, providing SIP normalization, fax handling, gain control and numerous additional media processing features. It offers certified interoperability with leading unified communications solutions and SIP trunking providers.

Security

The **DW-GTW-AC-E1060** provides robust protection for the IP communications infrastructure, preventing Denial of Service, fraud and service theft and guarding against cyber-attacks and other service-impacting events.

Reliability

DW-GTW-AC-E1060 offers active/standby high availability and maintains high voice quality to deliver reliable enterprise VoIP communications. Advanced call routing mechanisms, network voice quality monitoring and branch survivability capabilities (including PSTN fallback.

MAIN FEATURES

- **Fully integrated device for secured SIP trunking and PSTN access.**
- **Hybrid SBC and Media Gateway platform lowers CAPEX and reduces space and power footprints**
- **Extensive interoperability and partnerships that extend across multiple vendor devices and protocol implementations.**
- **Offers comprehensive security, interoperability and reliability.**
- **Delivers high service performance and voice quality.**
- **Branch office survivability in the event of a WAN outage.**

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System Specification

Capacities *	Regular / Max.		Regular / Max.
SBC Sessions	0 / 250	SRTP/RTP Sessions	180
Transcoding Sessions	0 / 30	Registered Users	0 / 800

Telephony Interfaces	
Analog	4 FXS ports
Digital	2 E1/T1 interfaces with an option for PSTN Fallback
Clock Source	5 ppm High Precision
Digital PSTN Protocols	Supporting various ISDN PRI protocols such as EuroISDN, North American NI-2, Lucent™ 4/5ESS, Nortel™ DMS-100 and others. It also supports different variants of CAS protocols, including MFC R2, E&M immediate start, E&M delay dial / start and others
Network Interfaces	
Ethernet	4 GE, support 1+1 redundancy or as individual ports
Security	
Access	Control DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting
VoIP Firewall	RTP pinhole management, rogue RTP detection and prevention, SIP message policy, advanced RTP latching
Encryption/Authentication	TLS, DTLS, SRTP, HTTPS, SSH, client/server SIP Digest authentication, RADIUS Digest
Privacy	Topology hiding, user privacy
Traffic Separation	VLAN/physical interface separation for multiple media, control and OAMP interfaces
Intrusion Detection System	Detection and prevention of VoIP attacks, theft of service and unauthorized access
Interoperability	
SIP B2BUA	Full SIP transparency, mature and broadly deployed SIP stack, stateful proxy mode
SIP interworking	3xx redirect, REFER, PRACK, session timer, early media, call hold, delayed offer
Registration and Authentication	User registration restriction control, registration and authentication on behalf of users, SIP authentication server for SBC users
Transport Mediation	TSIP over UDP/TCP/TLS/WebSocket, IPv4 / IPv6, RTP / SRTP (SDES/DTLS)
Message Manipulation	Ability to add/modify/delete SIP headers and message body using advanced regular expressions (regex)
URI and Number Manipulations	URI user and host name manipulations, ingress and egress digit manipulation
Transcoding and Vocoders	Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.726, G.729, GSM-FR, AMR-NB/WB, SILK-NB/WB, Opus-NB/WB
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion, V.150.1
WebRTC Controller	Interworking between WebRTC devices and SIP networks Supports WebSocket, Opus, VP8 video coder, lite ICE, DTLS, RTP multiplexing, secure RTCP with feedback
NAT	Local and far-end NAT traversal for support of remote workers
Voice Quality and SLA	
Call Admission Control	Based on bandwidth, session establishment rate, number of connections/registrations
Packet marking	802.1p/Q VLAN tagging, DiffServ, TOS
Standalone Survivability	Maintains local calls in the event of WAN failure. Outbound calls can use PSTN fallback for external connectivity (including E911)
Impairment Mitigation	Packet Loss Concealment, Dynamic Programmable Jitter Buffer, Silence Suppression/Comfort, Noise Generation, RTP redundancy, broken connection detection
Voice Enhancement	Transrating, RTCP-XR, Acoustic echo cancellation, replacing voice profile due to impairment detection, Fixed & dynamic voice gain control
Direct Media (No Media Anchoring)	Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption
Voice Quality Monitoring	RTCP-XR, Denwa Session Experience Manager (D-SEM)
High Availability (Redundancy)	SBC high availability with two-box redundancy, active calls preserved
Quality of Experience	Access control and media quality enhancements based on QoE and bandwidth utilization
Test agent	Ability to remotely verify connectivity, voice quality and SIP message flow between SIP UAs
SIP Routing	
Routing Methods	Request URL, IP address, FQDN, ENUM, advanced LDAP, third-party routing control through REST API
Advanced Routing Criteria	QoE, bandwidth, SIP message (SIP request, coder type, etc.), Layer-3 parameters
Routing Features	Least-cost routing, call forking, load balancing, E911 gateway support, emergency call detection and prioritization
SIPRec	IETF standard SIP recording interface
Management	
OAM&P	Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, EMS
Physical / Environmental	
Dimensions	1U x 320mm x 345mm (HxWxD)
Weight	Approx. 5.95lb (2.7kg) loaded with OSN
Mounting	Desktop or 19" rack mount
Power	100-240V 4A 50-60 Hz
Operating Temperature	5°-40° C
Regulatory Compliance	
Telecommunications	TIA/EIA-IS-968 (FXO, T1) interface, ETSI ES203 021 (FXO interface), TBR-4 (ISDN over E1 interface), TBR13/13 (E1 lines), TBR-3 (BRI interface)
Safety and EMC	IEC60950-1, UL60950-1, FCC Part 15 Class A, EN55022 Class A, EN55024, EN300 386
Environmental Storage	ETS300019-2-1 class T1.2
Transportation	ETS300019-2-2 class T2.3
Operating	ETS300019-2-3
Optional Licenses	
Enterprise Session Border Controller (E-SBC)	5 sessions: MDE-SBC-005P
Cloud Resiliense Package (CRP)	25 registered users: DW-GTW-AC-CRP-025 / 50 registered users: DW-GTW-AC-CRP-050 100 registered users: DW-GTW-AC-CRP-100

*See combinations table