Data sheet

\Orchestrating a brighter world

NEC

Hybrid E-SBC and Media Gateway

UNIVERGE BX800

Benefits

- 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 outag

Key features

- Rich and powerful SIP normalization and routing mechanisms for seamless interoperability
- Hybrid SBC that connects to PSTN / PBX trunks for fallback and gradual enterprise migration to SIP
- Support for analog (FXO, FXS) and digital (PRI, BRI) interfaces
- Perimeter defense against denial of service, fraud and eavesdropping
- VoIP quality monitoring and enforcement
- High Availability using two box redundancy
- Media Processing for Transcoding, Gain Control, DTMF/Fax, etc.

The NEC UNIVERGE BX800 Enterprise Session Border Controller (E-SBC) and Media Gateway offers a complete connectivity solution for small-to-medium sized enterprises.

The BX800 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 BX800 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 BX800 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

The BX800 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 with E911) result in minimum communications downtime.

Applications

- SIP trunking
- Hosted PBX & UC as a Service
- Remote and mobile worker support
- SIP mediation between UC and IP-PBX systems

Specifications

Capacities			
Max. Signaling/Media Sessions	250	Max. SRTP/RTP Sessions	180
Max. Transcoding Sessions	57	Max. Registered Users	800
Telephony Interfaces			
Analog	4 FXS ports; 4 FXO ports		
Digital	Two E1/T1 interfaces with an option for PSTN Fallback / 8 Basic Rate interfaces		
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 or 4 GE + 8 FE interfaces configured in 1+1 redundancy or as individual port		
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	SIP 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		
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 ga n control		
Direct Media (No Media Anchoring)	Hair-pinning of local calls to avoid unnecessary media delays and bandwidth consumption		
Voice Quality Monitoring	RTCP-XR		
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 UA		
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		
Redundancy	Detection of proxy failures and subsequent routing to alternative proxies		
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, E		
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
Environmental	Operational: 5 to 40 °C (41 to 104 °F); Storage: -25 to 85 °C (-13 to 185 °F) Relative Humidity: 10 to 90% non-condensing		

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