Data sheet

\Orchestrating a brighter world

NEC

Hybrid E-SBC and Media Gateway

UNIVERGE BX800

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

Scaling up to 400 concurrent sessions, the UNIVERGE BX800 connects IP-PBXs to any SIP trunk service providers and offers superior performance in connecting any SIP to SIP environment. In addition, the BX800 supports up to 124 channels in a 1U platform to enable versatile connectivity between TDM and VoIP networks, such as connecting legacy TDM PBX systems to IP networks and IP-PBXs to the PSTN.

Benefits

- Fully integrated device for secure SIP trunks 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

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
- Media Processing for Transcoding, Gain Control, DTMF/Fax, etc.
- AC/DC dual power supply (optional for BX800C)

Vast mediation capabilities and proven interoperability

The UNIVERGE BX800 supports a wide range of voice codecs and is capable of transcoding between narrowband and wideband voice coders, providing SIP normalization, fax handling, gain control and numerous other media processing features. It offers certified inter-operability with leading unified communications solutions and SIP trunk providers.

Security

The UNIVERGE BX800 provides robust IP communications infrastruc-ture protection, preventing Denial of Service, fraud and service theft and guarding against cyber-attacks and other service-impacting events.

Reliability

The UNIVERGE BX800 offers active/standby high availability and main-tains high voice quality to deliver reliable enterprise VoIP communi-cations. Advanced call routing mechanisms, network voice quality monitoring and branch survivability capabilities (including PSTN fallback with E911) result in minimum communications downtime.

Applications

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

Specifications

Capacities	Max. Signalling	Max. RTP/SRTP Sessions	Max. Transcoding Sessions	Max. Registered Users
BX800	250	250/250	57	1500
BX800C	400	400/300	114	2000

Telephony Interfaces	
Analog	4 FXS ports; 4 /FXO ports
Digital	Up to 4 E1/T1 interfaces; 4/8 BRI ports
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	4GE or 4GE + 8FE interfaces configured in 1+1 redundancy or as individual ports
Security	
Access Control	DoS/DDoS line rate protection, bandwidth throttling, dynamic blacklisting (Intrusion Detection System)
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	Automatic 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 and more
Registration and Authentication	SIP Registrar, registration on behalf of users/servers, SIP Digest access authentication
Transport Mediation	SIP over UDP/TCP/TLS/WebSocket, IPv4 / IPv6, RTP / SRTP (SDES/DTLS)
Header Manipulation	Add/modify/delete SIP headers and message body using simple WireShark-like language with powerful capabilities such as variables and utility functions
Number 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, AMR-WB (G.722.2), SILK-NB/WB, Opus-NB/WB, iLBC
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, T.38 V3, V.34, packet-time conversion V150.1
WebRTC Gateway	Interworking between WebRTC endpoints and SIP networks Supports WebSocket, Opus, VP8 video coder, lite ICE, DTLS, RTP multiplexing
NAT	Local and far-end NAT traversal for support of remote workers
Voice Quality and SLA	
Call Admission Control	Limit number and rate of concurrent sessions and registers per peer for inbound and outbound directions
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)
Voice Monitoring and Enhancement	Transrating, RTCP-XR, acoustic echo cancellation, replacing voice profile due to impairment detection, fixed & dynamic voice gain control, packet loss concealment, dynamic programmable jitter buffer, silence suppression/comfort, noise generation, RTP redundancy, broken connection detection
Direct Media	Hair-pinning (No Media Anchoring) of local calls to avoid unnecessary media delays and bandwidth consumption
High Availability	SBC high availability with two-box redundancy, active calls preserved
Test agent	Ability to remotely verify connectivity, voice quality and SIP message flow between SIP UAs
SIP Routing	
Routing Criteria	Incoming SIP trunk, DID ranges, host names, any SIP headers, codecs, QoE, bandwidth
Querying External Databases	Routing based on customized queries of ENUM, LDAP, HTTP server (REST API)
Route to	Configured SIP peers, registered users, IP address, request URI
Advanced Features	Alternative destinations, load balancing, LCR, call forking, E911 emergency call detection and prioritization
SBC Media Types	Audio/Video/Fax/Text/Message Session Relay Protocol (MSRP)/Binary Floor Control Protocol (BFCP)
SIPREC	IETF standard SIP recording interface, supporting both audio and video SBC sessions
Management	
Operation & Management	Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, OVOC

Physical/Environmental			
Dimensions	45 x 345 x 320mm (HxWxD)	Weight	Approx. 5.95lb (2.7kg)
Mounting	Desktop or 1U 19" rack mount	Operating Temperature	5 to 40 °C (41 to 104 °F)
Power	Internal AC power supply rated: 100-240 VAC ~50- 60Hz 1.5A maximum (Optional) Additional 12V 10A DC power, via an external AC/DC power adapter		

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