Data sheet

\Orchestrating a brighter world

NEC

Hybrid E-SBC and Media Gateway

UNIVERGE BX1000

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
- Scalable “pay-as-you-grow” modular architecture
- 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 enables seamless migration and PSTN fallback
- Modular support for analog and digital TDM interfaces
- Perimeter defense against denial of service, fraud and eavesdropping
- VoIP quality monitoring and enforcement
- Media Processing for Transcoding, Gain Control, DTMF/Fax, etc.
- Optional Open Solution Network (OSN) server module for hosting value-added applications

The UNIVERGE BX1000 Enterprise Session Border Controller (E-SBC) and Media Gateway offers a complete connectivity solution for small-to-medium sized businesses.

Scaling up to 150 concurrent sessions, the UNIVERGE BX1000 connects IP-PBXs to any SIP trunk service providers and offers superior performance in connecting any SIP to SIP environment. In addition, the BX1000 supports up to 192 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.

Vast mediation capabilities and proven interoperability

The UNIVERGE BX1000 supports a wide range of voice codecs and is capable of transcoding between narrowband and wideband voice codecs, 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 UNIVERGE BX1000 provides robust protection for IP communications infrastructure, preventing Denial of Service, fraud and service theft and guarding against cyber-attacks and other service-impacting events.

Reliability

The UNIVERGE BX1000 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 trunks
- Hosted PBX & UC as a Service
- Remote and mobile worker support
- SIP mediation between UC and IP-PBX systems

Specifications

Capacities			
Max. Signaling Sessions	150	Max. SRTP/RTP Sessions	120
Max. Transcoding Sessions	96	Max. Registered Users	600

Telephony Interfaces	
Modularity and Capacity	6 slots for hosting voice processing and PSTN termination modules (up to 192 channels)
Digital Module	Up to 6 E1 or 8 T1/J1 spans provided on trunk modules. Each module supports 1 or 2 E1/T1/J1 spans, with an option of PSTN Fallback
Digital PSTN Protocols	Various ISDN PRI protocols such as EuroISDN, North American NI-2, Lucent™ 4/5ESS, Nortel™ DMS-100 and others, different CAS protocols, including MFC R2, E&M immediate start, E&M delay dial / start and others.
BRI Module	Up to 20 BRI ports provided on BRI modules. Each module supports 4 BRI ports, with PSTN Fallback. Providing S/T interfaces; NT or TE termination; 2W per port (power supplied)
Analog Module	Up to 24 FXS/FXO interfaces, provided on 4 ports FXO / FXS modules, ground / loop start
Media Processing Module	Up to 4 Media Processing modules (MPM), providing additional DSP resources
Network Interfaces	
Ethernet	Up to 6GE interfaces configured in 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, 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
Interoperability	
SIP B2BUA	Full SIP transparency, stateful proxy mode
SIP interworking	3xx redirect, REFER, PRACK, session timer, early media, call hold, delayed offer
Registration and Authentication	SIP Registrar, registration on behalf of users/servers, SIP Digest access authentication
Transport Mediation	SIP over UDP/TCP/TLS, IPv4 / IPv6, RTP / SRTP (SDES)
Message 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	Codec normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.723.1, G.726, G.729, GSM-FR, AMR-NB, iLBC, QCELP, GSM EFR, G.727
Signal Conversion	DTMF/RFC 2833/SIP, T.38 fax, V.34, packet-time conversion
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 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
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 Routing Features	Alternative routes, load balancing, least-cost routing, call forking, E911 emergency call detection and prioritization
Management	
Operation & Management	Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, OVOC
OSN Server Platform (Optional)	
Single Chassis Integration	Optional embedded, x86, Intel-based Open Solution Network platform for third-party applications

Physical / Environmental			
Dimensions	45 x 444 x 355 mm (HxWxD)	Weight	Approx. 9.7lb (4.4kg)
Mounting	Desktop or 1U 19” rack mount	Power Supply	Single power supply 100-240V, 50-60 Hz, 1.5A max. optional redundant power supply
Operating Temperature	0 to 40° C (32 to 104°F)		

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