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

The BX1000 connects IP-PBXs to any SIP trunking service provider, scaling up to 150 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 192 voice channels in a modular 1U platform.

Vast mediation capabilities and proven interoperability
The BX1000 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 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 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 trunking
- Hosted PBX & UC as a Service
- Remote and mobile worker support
- SIP mediation between UC and IP-PBX systems
## Specifications

### Capacities

<table>
<thead>
<tr>
<th>Feature</th>
<th>Value</th>
</tr>
</thead>
<tbody>
<tr>
<td>Max. Signaling/Media Sessions</td>
<td>150</td>
</tr>
<tr>
<td>Max. Transcoding Sessions</td>
<td>96</td>
</tr>
<tr>
<td>Max. Registered Users</td>
<td>600</td>
</tr>
</tbody>
</table>

### Telephone Interfaces

#### Modularity and Capacity
- 6 slots for hosting voice processing and PSTN termination modules (up to 192 channels)
- Each module supports 1 or 2 E1/T1/J1 spans, with an option of PSTN Failback

#### 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 Failback

#### Digital PSTN Protocols
- Supporting various ISDN PRI protocols such as EuroISDN, North American NI-2, Lucent™ 4/SESS, Nortel™ DMS-100 and others. It also supports different variants of CAS protocols, including MFC R2, E&M immediate start, E&M delay start / start and others.

#### BRI Module
- Up to 20 BRI ports provided on BRI modules.
- Each module supports 4 BRI ports, with PSTN Failback. Providing S/It 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 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, 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/32UA
- 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, IPv4 / IPv6, RTP / SRTP (SDES)

#### 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.728, G.729, GSM-FR, AMR-NB/WW, SILK-NB/WW, ILBC, OCELP, GSM EFR

#### 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
- Based on bandwidth, session establishment rate, number of connections/rегистraions

#### 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 failback 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

#### 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
- IETF standard SIP recording interface

#### Advanced Routing Criteria
- Request URL, IP address, FQDN, ENUM, advanced LDAP, third-party routing control through REST API

#### Redundancy
- QOE, bandwidth, SIP message (SIP request, coder type, etc.), Layer-3 parameters

#### Routing Features
- Detection of proxy failures and subsequent routing to alternative proxies

#### SIPRegister
- Least-cost routing, call forking, load balancing, E911 gateway support, emergency call detection and prioritization

### Management

#### OAM&P
- Browser-based GUI, CLI, SNMP, INI Configuration file, REST API, E

#### OSN Server Platform (Optional)
- Embedded, open Network Solution Platform for third-party services

### Memory
- Up to 8 GB RAM

### Storage
- HDD or SSD

### Physical / Environmental

#### Dimensions
- 1U x 444 mm x 355 mm (HxWxD)

#### Mounting
- Desktop or 19" rack mount

#### Environment
- Operational: 0 to 40°C (32 to 104°F); Storage: -20 to 70°C (-4 to 158°F)

- Relative Humidity: 10 to 85% non-condensing

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About NEC Corporation - NEC Corporation is a leader in the integration of IT and network technologies that benefit businesses and people around the world. By providing a combination of products and solutions that cross utilize the company’s experience and global resources, NEC’s advanced technologies meet the complex and ever-changing needs of its customers. NEC brings more than 100 years of expertise in technological innovation to empower people, businesses and society. For more information, visit NEC at http://www.nec.com

Please note that not all features described are necessarily available in all regions.

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