Mitel GX Gateway

Survivable Branch Gateway for Multi-Site Enterprises

Key Features

- Robust multi-service platform
- Flexible architecture
- Survivability – service continuity
- Communication service access for branch offices and remote sites
- 120 simultaneous VoIP channels
- Ideal for remote locations with up to 200 remote SIP users
- Legacy and IP systems integration
- Easy configuration and management

Multi-Service Business Appliance

Mitel GX Gateway is a combination of Session Border Controller and Media Gateway, a robust multi-service 1U appliance for remote office/branch office survivability needs. The system provides capacity for up to 120 simultaneous VoIP channels and offers local survivability and PSTN or SIP trunk access for up to 200 SIP and 24 FXS users in a remote site/branch office location situated outside the main site data center where the MiVoice call manager is located. The GX Gateway is a reliable and fully autonomous solution for remote site SIP deployment with incomparable QoS monitoring, security, survivability and interoperability. The GX also supports ISDN PRI, E&M, and R2 E1/T1 CAS providing access to the local PSTN services. The built in SBC service also allows for secure local SIP trunk access as well.

SURVIVABILITY

Should there be a loss of service toward the main site, the Mitel GX Gateway ensures service continuity by providing users with local phone services and allowing them to establish external calls via the local PSTN/SIP trunks. It will also manage rerouting of internal calls to the central site via the public network when access to the corporate WAN network is temporarily unavailable.

REMOTE USERS

Mitel GX Gateway provides communication service access to branch offices, as if they were on the same site as the MiVoice Call Manager and UCC application servers.

NETWORK SEPARATION

The GX creates a clear separation between the enterprise's and the operator's networks by hiding the topologies and credentials, and by blocking unauthorized users.

LEGACY AND IP SYSTEMS INTEGRATION

With its flexible configuration of FXS, FXO, and PRI telephony ports, call-switching, and user-defined call properties (including caller/calling ID), the GX Gateway smoothly integrates legacy CPE into IP systems.

Branch Office / Survivability GW

The GX Gateway is designed to provide survivability for Branch Office users in a centralized call manager deployment scenario.

In a normal situation, the local SIP and analog phone users register to the main site call manager through the GX gateway that acts as a proxy and provides local PSTN hop-off for MiVoice users.

Should the main site system be unreachable, the GX gateway appliance will take over basic call control functions locally to allow internal traffic and PSTN access for the local SIP or analog phone users in the branch office.

When the main site system or WAN connection comes back up, then the GX gateway will hand over the call control back to the central site call manager and resume its role as a local PSTN gateway.

The built in SBC service allows local secure connections for SIP trunks as an alternative to traditional PSTN connections. This enables customers to create a shield of confidentiality between the enterprise and the Internet for users in all remote locations.

Key Features

CARRIER-GRADE FEATURES

- T.38 and clear channel fax over IP
- High performance processing of up to 120 voice channels

ROBUST SECURITY

- Enterprise communication encryption
- SIP-enabled firewall inspects and authorizes communications and prevents DoS attacks

EASY CONFIGURATION AND MANAGEMENT

- Zero-touch configuration
- Intuitive Web GUI
- Customizable factory settings

NETWORKING

- Dual-stack IPv6 and IPv4
- Multiple IP addresses and VLANs NAT, firewall, and router capabilities

Technical Specifications

SESSION BORDER CONTROLLER

- Back-to-Back user agent
- SIP header manipulation
- SIP registrar
- SIP authentication
- SIP failover
- Registration throttling/caching
- Advanced, rule-based, call routing
- Dynamic call routing based on peer monitoring state and registration cache
- Call Admission Control, per trunk, based on call volume, bandwidth usage and concurrent calls
- Near and far-end NAT traversal
- Audio and video media relay
- Codec filtering
- SIP and media encryption
- UDP/TCP/TLS interworking
- DTMF interworking

MEDIA PROCESSING

- G.711 (A-law and µ-law) and G.729a/b;
- G.168 echo cancellation
- DTMF detection and generation
- Carrier tone detection and generation
- Silence detection/suppression and comfort noise
- Configurable de-jitter buffer and packet length

ENHANCED SECURITY

- SIP over TLS
- SRTP with AES cipher – 128 bits
- SDES key management protocol (RFC 4568)
- TLS-encrypted configuration and management
- X.509 certificate management
- OCSP (Online Certificate Status Protocol) revocation status verification
- Supported TLS key exchange mechanism:
    - RSA
- Supported TLS ciphers (minimum):
    - AES (128 and 256 bits)

MANAGEMENT

- Web GUI
- SSHSMNP v2c, and v3
- Scripts/firmware files uploaded via HTTP, HTTPS, FTP, and TFTP
- Multiple levels of management access rights
- Event notifications via Syslog, SIP, log file, and SNMP traps
- Remote activation of service licenses

QUALITY OF SERVICE (QOS)

- Bandwidth limitation and traffic shaping
- TOS/DiffServ
- IEEE 802.1p/Q

IP TELEPHONY PROTOCOL

- SIP (RFC 3261) over UDP, TCP, and TLS
- RTP (RFC 3550)
- SDP (RFC 4566)
- Multi-part body support
- Redundancy support via DNS SRV
- Multiple trunk support
- IPv4 and IPv6 dual stack signaling and media

DIGITAL TELEPHONY

- Euro ISDN EDSS-1/ETSI PRI/NET5
- ISDN NI-2 (US T1 PRI)
- ISDN DMS100 (US T1 PRI)
- ISDN 5ESS (US T1 PRI)
- ISDN speech, audio, and data (Fax Gr 4, UDI 64, and RDI 64)
- ECMA-143 (QSIG-BC)
- E1 R2 digital line signaling (ITU-T Q.421)
- E1 R2 MFC inter-register signaling (ITU-T Q.441)
- Presets for: Brazil, Argentina, Mexico, Saudi Arabia, Venezuela, Philippines, and ITU-T
- T1/E1 E&M (Immediate, Wink-Start, Feature Group-B, and Feature Group-D), MF-R1, DTMF
- Advice of Charge AOC-D, AOC-E (ETS 300 300 182)

ANALOG TELEPHONY

- Support for call forward, call transfer, conference call, call waiting, CCNR, and CCBS
- Multiple country presets
- Customizable tones and ring patterns
- Echo cancellation
- Message Waiting Indication (MWI), via FSK
- Caller ID detection (name & number) as per Bell-core FSK
- On-hook/off-hook caller ID generation (name & number) as per Bell-core DTMF or FSK and Telebras BINA
- Answer and disconnect signaling

NETWORKING

- IPv4 – IPv6
- Multiple IP addresses per link or VLAN
- Multiple VLANs per link
- PPPoE (RFC 2516)
- IEEE 802.1q + DSCP QoS tagging (media, signaling, and management)
- IEEE 802.1x wired authentication
- LLDP-med (ANSI/TIA-1057)
- QoS traffic shaping
- Static routing

POWER SUPPLY

- Internal 100-240 VAC power supply

PHYSICAL INTERFACES

- 5 x 10/100/1000 BaseT Ethernet RJ-45 connectors
- 2 x TDM sync RJ-45 connectors
- 1 x USB 2.0 Type-A connector

OPERATING ENVIRONMENT

- Operating temperature: 0°C to 40°C
- Storage temperature: -20°C to 70°C
- Humidity: up to 85%, non-condensing

DIMENSIONS

- Height: 4.4 cm
- Width (mounting brackets): 48.3 cm
- Depth: 19.5 cm

SBC LICENSING

- A license is needed for each concurrent call
- No restrictions on number of users/registrations