

Mitel MiVoice Border Gateway

Creating Secure Workspaces for Mobile And Remote Employees

Key Features

- **Teleworker Service:** connects remote IP phones, softphones and Mitel applications, such as collaboration and contact center tools, securely to the corporate voice network providing full access to Mitel services
- **Web Proxy Service:** provides a secure method for remote web browser users, such as web conferencing users, to connect with Mitel tools on the corporate LAN
- **Secure Recording Connector:** facilitates the recording of Mitel-encrypted voice streams for call recording equipment
- **SIP Trunk Proxy Service:** connects your Mitel communications platform securely to the traditional PSTN via SIP trunks
- **WebRTC Gateway:** enables both anonymous and subscribed connections to your Mitel communications platforms through WebRTC-enabled browsers



The way we conduct business has changed. Enable an in-office communications experience for your employees when they are not physically in the office.

The traditional workplace has transformed. Employees require the workplace to be flexible, to enable them to be mobile and work from remote locations and on different devices whenever and wherever they need to.

Whether it's for MiVoice IP phones, softphones on laptops, or softphones on mobile devices, Mitel@ MiVoice Border Gateway can help through the creation of secure, encrypted voice connections across the Internet to your company's internal network, all without the need for a VPN.

Easily build upon your Mitel communications investment and extend unified communications capabilities; simply, transparently, and securely to remote workers with the MiVoice Border Gateway - the session border controller built specifically for your Mitel communications infrastructure.

MiVoice Border Gateway - Technical Specifications

SECURITY

Access Control	DoS/DDoS protection, traffic shaping, whitelisting/blacklisting
VoIP Firewall	RTP/SRTP pinhole management, rogue RTP prevention, Advanced RTP latching, Source Validation, Packet Validity checking
Encryption and Authentication	TLS, DTLS, SRTP, HTTPS, SSH, client/server SIP Digest Authentication
Privacy	Topology hiding, user privacy
Traffic Separation	Multiple interface (Physical or Virtual) separation for ingress / egress and different network topology requirements

INTEROPERABILITY

SIP B2BUA	SIP Normalization, 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 remote users
Transport Mediation	SIP over UDP/TCP/TLS, RTP/SRTP, WebRTC to SIP conversion
Header Manipulation	Ability to add/modify/delete SIP headers and message body using SIP Adaptation scripting language
URI and Number Manipulations	Arbitrary modifications through SIP Adaptation scripting language
Transcoding and Vcoders	Coder normalization including transcoding, coder enforcement and re-prioritization, extensive vocoder support: G.711, G.729, G.722, G.722.1, Opus-NB/WB, G.722, others as pass-through; video support; T.38 fax
Signal Conversion	DTMF/RFC 2833/SIP, packet-time conversion, T.38, reframing
WebRTC Controller	Interworking between WebRTC devices and SIP networks, supports WebSocket, Opus, G.711A/u, VP8 & H.264 video coder, lite ICE, DTLS, RTP multiplexing, secure RTCP with feedback
NAT	Local and far-end NAT traversal for support of remote workers
Mitel IP Sets	Full MiNet, Secure MiNet and TLS MiNet support, stateful proxy mode

VOICE QUALITY AND SLA

Call Admission Control	Call admission based on bandwidth, Emergency Calls pre-emption, call denial without security
Packet Marking	DSCP marking
Stand Alone Survivability	Clustering with load balancing and resiliency, maintain active calls as long as possible in the event of PBX failure
Impairment Mitigation	Packet Loss Concealment, Dynamic Jitter Buffer
Direct Media/ Media Anchoring	Support for local streaming or anchored media on a per user basis
Voice Quality Monitoring	Integration with Mitel Performance Analytics
High Availability (Redundancy)	High Availability with MiVoice Border Gateway Clustering, plus VMware support
Test Agent	Ability to remotely verify connectivity and, voice quality with Teleworker Network Analyzer tool, built in Diagnostics panel, packet traces, and concurrent signaling trace
Load Balancing	MiNet set automatic load balancing with nodal weighting

SIP ROUTING

Routing Methods	Request URL, IP address, FQDN, ENUM
Redundancy	Detection of proxy failures and subsequent routing to alternative proxies
Call Recording	Interoperable with multiple call recording vendors (Redbox ASC, Mitel, etc.)
Routing Features	Emergency call detection/prioritization

MANAGEMENT

OAM&P	Browser-based GUI, CLI, SNMPv3, REST API, integrated provisioning with Mitel solutions
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SYSTEM CAPACITY

Up to six MiVoice Border Gateway servers can be clustered together in a single service.

A customer network may have several MiVoice Border Gateway clusters working together to provide combined services, for example, one cluster dedicated for SIP Trunks and another cluster dedicated for remote Teleworkers

TELEWORKER SERVICE CAPACITY (SINGLE SERVER)

	Mitel IP Set Registrations	SIP Set Registrations	Concurrent G.711 Calls
Physical Server	10,000	10,000	3,000
Virtual Server	8,000	8,000	830

Up to six servers can be clustered together supporting 50,000 users in a N+1 configuration

WEB PROXY SERVICE CAPACITY (SINGLE SERVER)

	MiContact Center Ignite / MiCollab
Physical Server	1,200
Virtual Server	1,200

Up to six servers can be clustered together supporting 6,000 users in a N+1 configuration

SECURE RECORDING CAPACITY (SINGLE SERVER)

	Mitel IP Set Registrations	SIP Set Registrations	Concurrent G.711 Calls
Physical Server	10,000	10,000	3,000 with 0 active recordings; 2,750 with 500 active recordings; 2,500 with 1000 active recordings; 2,000 all active recordings
Virtual Server	8,000	8,000	830 with 0 active recordings; 750 with 100 active recordings; 700 with 200 active recordings; 630 with 300 active recordings

SIP TRUNK PROXY SERVICE CAPACITY (SINGLE SERVER)

	Concurrent G.711 Calls
Physical Server	1,100
Virtual Server	500

SIP Trunk capacity is calculated as a dedicated application

WEBRTC SERVICE CAPACITY (SINGLE SERVER)

	Audio	Audio & Video (no transcoding)	Audio & Video (with transcoding)
Physical Server	2,000	600	36
Virtual Server	500	150	10