



MX60E-G

Max. FXS/FXO ports: 48

Max. concurrent calls: 48

## Features

- SIP/MGCP protocols
- 500 routing and number manipulation rules
- Fax over IP using T.38 & G.711 pass-through
- Dual redundant Ethernet ports
- Dual DC/AC power supplies available
- PSTN failover
- TLS/SRTP
- Auto provisioning
- Remote access via New Rock Cloud (US: Youbiquity)
- Management with New Rock or third-party EMS (TR-069, SNMP)
- Interoperability with popular SIP servers, such as Cisco CallManager/ CUCM, Broadsoft, Microsoft Skype for Business (Lync), Huawei IMS, VoS, FreeSWITCH and Asterisk/Elastix
- Class I lightning protection

The MX60E-G is a reliable, cost-effective, high-performance VoIP gateway with up to 48 FXS/FXO ports, designed to meet the needs of high-traffic contact center and enterprise applications.

The feature-rich MX60E-G allows users to consolidate analog phones, fax machines, and POS terminals, configurable with up-to-full concurrency over SIP trunks and the PSTN.

### Comprehensive Feature Set

As an intelligent gateway running on embedded Linux operating system, the MX60E-G supports an advanced feature set such as call forward, call transfer, 3-way calling, caller ID, DND, color ringback tones, call forking, automatic routing, digit map, and much more, providing a complete and highly reliable VoIP solution.

### Carrier-class Reliability

The MX60E-G provides high-availability configurations with SIP registration failover, load balancing and PSTN failover with optional 1+1 redundancy of Ethernet ports and power supplies, virtually eliminating downtime.

### Robust Protection against Intrusions

MX60E-G provides a rich set of features to support perimeter protection against malicious intrusions, such as brute force logins and SIP attacks, ensuring a high level of security in your IP telephony environment.

### Remote Management and NAT Traversal

The MX60E-G includes the New Rock Cloud client, providing easy access with NAT traversal to endpoint devices and value-added applications for remote, real-time monitoring, maintenance, software upgrades, troubleshooting, data analysis, alarm notification, remote and packet capture.

## Specification

### Protocols

<b>Call control</b>	SIP/UDP and SIP/TCP (RFC3261), IMS (3GPP), MGCP (RFC3435)
<b>Network</b>	SSH, HTTP, HTTPS, DHCP/PPPoE client, Radius, DNS (A/SRV record), STUN

### Media Processing

<b>Caller ID</b>	Bellcore/Telcordia, ETSI, BT, DTMF
<b>Codec</b>	G.711 (a/μ), G.729a, G.723.1, GSM, iLBC
<b>DTMF</b>	In-band audio, RFC2833, SIP-INFO
<b>Hook-flash</b>	Local processing, RFC2833, SIP-INFO
<b>Fax over IP</b>	T.38, G.711 pass-through T.38 compliant Group 3 Fax Relay Maximum fax rate of 33,600 bps (pass-through)
<b>Disconnect modes</b>	Polarity reversal, Busy tone detection, Loop current
<b>QoS</b>	DiffServ, TOS, 802.1P/Q VLAN tagging
<b>Voice-quality enhancement</b>	Echo cancellation (G.168-2004), Jitter buffer, Silence suppression (VAD, CNG), PLC

### Voice

<b>Call transfer</b>	Blind transfer, Consultative transfer
<b>Call forward</b>	Call forward all, Call forward no answer, Call forward busy
<b>Call recording storage</b>	Remote recording server for storage
<b>Call settings</b>	Routing based on the caller or callee number, Digit map, 3-way calling, Speed dialing, Do not disturb, Call forking, Color ringback tones, Hunt group, Ring cadence, Message Waiting Indicator (MWI)

### Security

<b>Protocol</b>	HTTPS, SSH, TLS, SRTP
<b>Encryption</b>	Encrypted password/PIN
<b>Intrusion prevention</b>	Brute force login protection, Access list, Firewall, SIP-allowed IP addresses, Alarm and notification*
<b>VLAN</b>	Auto discovery via DHCP, Manual configuration
<b>VPN client*</b>	L2TP, OpenVPN

### High Availability

<b>Redundancy</b>	Primary-Standby, Active-Standby, Load balancing
<b>PSTN failover</b>	Power failure, Network failure

### Provisioning, Administration and Maintenance

<b>Device management</b>	New Rock UMS, TR-069 management (TR-069, TR-104 and TR-106), SNMP
<b>Remote access over TCP</b>	New Rock Cloud (US: YoubiquiTy)
<b>Auto provisioning</b>	Download configuration file via TFTP/FTP/HTTP/HTTPS, Obtaining ACS address via DHCP option 66 or redirection

<b>Log management</b>	8-level logs, Syslog
<b>Data capture</b>	Port capture, Packet capture
<b>Status and statistic</b>	Call status and history, Device status monitoring and statistics collection
<b>Upgrade</b>	Firmware upgrade via Web GUI or Auto provisioning

\* Roadmap feature

## Included Software

### Software

<b>Finder</b>	Obtains IP addresses of devices in a LAN environment
<b>Telegreeting</b>	Formats pre-prepared IVR audio files for uploads them into a device
<b>Recording Agent</b>	Receives recording streams in PCM and converts them into MP3 format. Includes management functions such as search, play, delete and backup

## Hardware

CPU	1GHz
Voice DSP module	200 MHz, 2 to 6 modules
RAM	128 MB, DDR3
Flash	32 MB
H×W×D (1U)	44 × 440 × 260 mm
Weight (net)	3 kg maximum
Ethernet	10/100/1000M
Single/Dual AC power supplies	100 to 240V AC, 50/60 Hz, 1.2 A maximum
Single/Dual DC power supplies	-36 to -72V DC, 2.5 A
Power consumption	60 W
Mounting	Rack
Operating	Temperature: 0 to 40°C Humidity: 10 to 90% RH (non-condensing)
Storage	Temperature: -40 to 70°C Humidity: 5 to 90% RH (non-condensing)

