

E1 and SIP User Media Gateway



Main features

- 1x E1 link
- 1x gigabit Ethernet network port
- Route failover
- Register up to 10 SIP accounts

Applications

- VoIP telephony carriers
- Call routing between branches and headquarters over the IP network

Overview

The UMG 100R is a voice gateway that is part of Khomp's media gateways line. Designed to meet the needs of small-sized businesses, this gateway can be connected to the Public Switched Telephone Network (PSTN), SIP trunks, soft-switches, and PBX equipment. It has one E1 link and can connect to up to 10 different SIP servers, with or without registration. This way, the UMG can be used along with traditional PBX systems for making calls using VoIP carriers or to enable the IP PBX system to make calls using landline carriers.

Robust and effective, it features dedicated processors to handle critical telephony tasks and echo canceling. It supports industry-leading signaling and codec solutions, and it handles the control and routing of calls according to predefined rules. All this comes in a device that was developed to meet the users' needs, with a footprint that allows for easy installation and a user-friendly Web Interface for configuration and monitoring.

Routing and customer loyalty plan

Obtain enhanced control over your telephony costs by configuring routing rules according to phone number prefixes and/or carrier loyalty attributes. This way, you can route calls to the carriers that offer the best cost-benefit relation for each call, resulting in lower-cost telephony rates.

Route failover

The UMG offers route failover to avoid downtime in call processing in case of a VoIP server failure. The failover function is implemented by using routes along with VoIP server monitoring through the Keep Alive feature. When the Keep Alive function is active, the UMG sends OPTIONS messages to the VoIP server in order to monitor its status. When this server does not respond to the OPTIONS command, the UMG then ignores the route through which this server is being used and searches for another compatible route.

Technical specifications

Operation Interfaces

- Configuration, monitoring, administration and diagnostics via web interface
- Module for diagnostics via Interface Web
- User Interface Web access control
- Packet capture via web interface
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- **System status**
- System status via Interface Web
- Status of trunks and channels via Interface Web
- SNMP support

E1 link

- One E1 link
- Allows you to select the number of channels to match the telephony carrier
- ISDN or R2 signaling
- ISDN PRI
- Connector options:
- Coaxial BNC - electrical resistance: 75 Ohms
- RJ45 - electrical resistance: 120 Ohms
- Clock setting
- Supports error checking method (CRC-4)
- Channel allocation algorithm selection (first free or balanced channel)
- Channel allocation sorting
- ISDN and R2 signaling advanced settings
- Collect call blocking through double answering in R2 signaling
- Collect call blocking through signaling in ISDN

VoIP

- Up to 10 VoIP accounts with or without registration
- Supported Codecs:
 - G.711 (a-law and μ -law)
 - G.729A (up to 29 simultaneous calls in this configuration)
 - G.723.1 e G.726
- VoIP account
- SIP and RTP using the TCP protocol
- Keep Alive support (SIP OPTIONS)
- Option to ignore source port
- Use of a destination number through URI
- Q.850 Cause Report
- DTMF sending mode selection:
 - In band
 - Out band – RTP (RFC 2833)
 - Out band – SIP Info
- Supports fax T.38 and pass-through
- Echo canceling:
- Standard filter: G.168/2002
- Dual filter: G.168/2004
- Tail-length adjustment up to 128 ms

Call Admission Control

- Based on local resources
- Call rate limiting QoS (Quality Control)

Warranties and certifications

- Total warranty (legal + Khomp warranty): 3 years
 - Legal warranty: 90 days
 - Khomp warranty: 2 years and 9 months
- Anatel (Brazilian National Telecommunications Agency) certification
- ISO 9001 certified industry

Smart routing

- Route selection by prefix
- Route selection by regular expressions
- Modification of destination and source number
- Imposing of the codec and the destination profile along the route with VoIP output
- Route failover
- Use of the "Display name" as the caller ID
- Registration of up to 50 routes
- LCR call routing - lowest cost routing

Other features

- Simplified web configuration
- Single-step initial configuration wizard
- Diagnostics interface
- Line volume setting
- DTMF suppression
- Customizable CDR
- SNMP Support
- Log recording on a remote server or local site
- FTP access
- Provisioning (exporting and importing configurations)
- Zero-touch provisioning
- Remote terminal with advanced CLI (Command Line Interface)
- TR-069 support
- Support ITU-T G.165 and G.168 standards
- Acoustic signaling treatment performed by hardware through DSPs
- Automatic fax tone detection (2100Hz) automatically enabling echo cancellation

Physical/Environmental

- Polarized power source connector 12 VDC
- Power adapter:
 - Input: 100–240 VAC, 50/60 Hz
 - Output: 12 VDC, 1 A
- Maximum power consumption: 12 W
- 1x RJ45 gigabit Ethernet 10/100/1000 Mbps
- Gateway status LED
- Telephony channel status LED
- Error warning LED
- Reset button
- Product Dimensions: 127x92x33 mm
- Transport box dimensions: 187x113x72 mm
- Gross weight: 300 g
- Net weight: 150 g
- Operating temperature: 0–50 °C
- Operating humidity: 10–90% non-condensing
- Storage temperature: 0–85 °C
- Storage humidity: 10–90% non-condensing

Security

- Access via HTTP or HTTPS protocol
- Fraud prevention: call blocking by destination and origin
- DoS / DDoS Protection
- Hiding network topology
- SIP TLS and SRTP protocols (SDTLS, DES, 3DES, AES-128, or AES256)
- Access control - ACL (whitelist and blacklist)
- Protection against malformed packages
- Rogue RTP protection

Product images



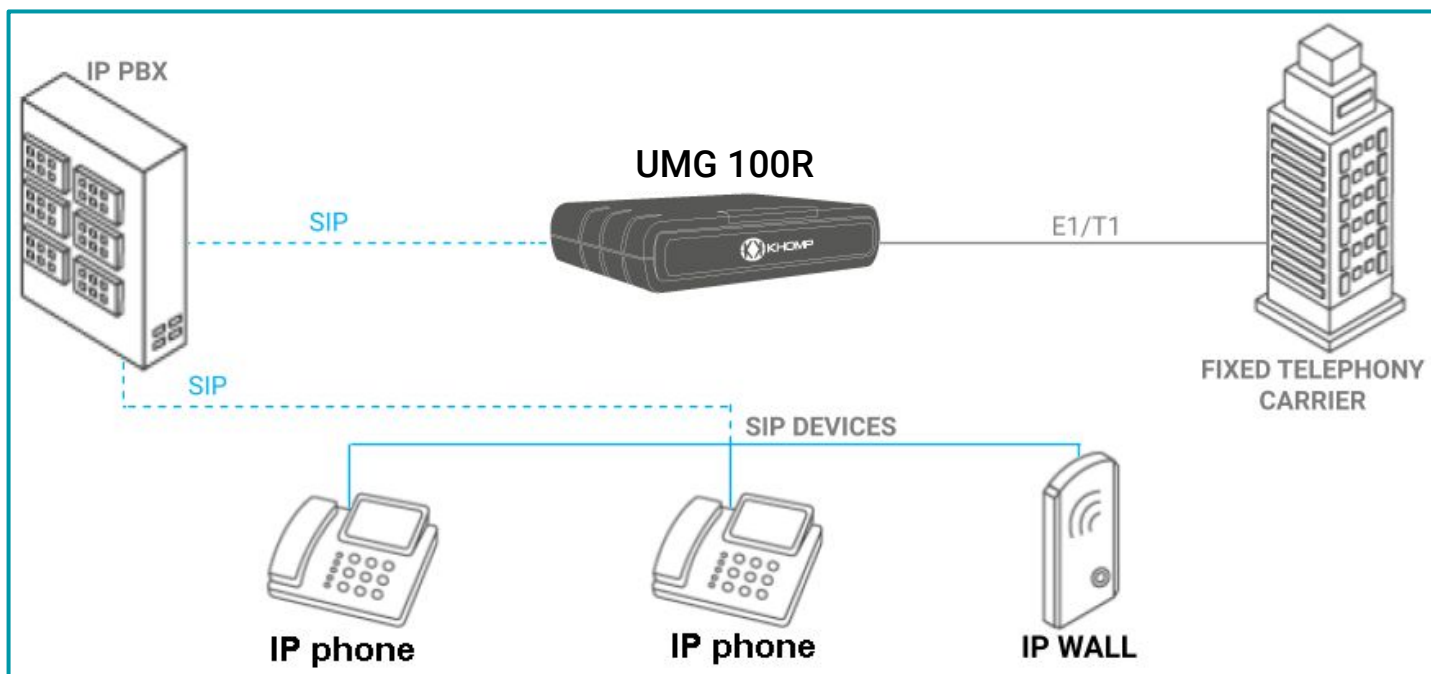
Subtitle: *Front view.*



Subtitle: *Rear view, E1 link with BNC coaxial connector*

Application models

IP PBX connection with landline telephony carrier



Traditional PBX connection with VoIP telephony carrier

