User Media Gateway with 4 Gigabit Ethernet ports

Features

- 4x gigabit Ethernet network ports
- 1x E1/T1 link
- Register up to 10 SIP accounts
- SBC - routing between VoIP channels*
- Stand Alone Survivability – SAS*

* *Optional feature - optional items are available at an additional cost.*

Applications

- VoIP telephony carriers
- Corporate environments
- Companies with traditional PBX systems that need to route calls between branches and the main office over an IP network

Overview

The UMG 104 is a voice gateway that is part of Khomp's media gateways line. It offers out-of-the-box connection to the Public Switched Telephone Network (PSTN), VoIP links, soft-switches and PBX devices, in order to meet the needs of small scale scenarios.

This robust and effective device includes one E1/T1 link and allows you to add up to 10 different SIP accounts.

It has four Ethernet ports, allowing you to directly connect the UMG to more than one VoIP carrier, and you can also place those carriers in the same network or in different networks. 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 rates.

Table of simultaneous calls

The UMG 104 makes up to 30 simultaneous calls via the E1/T1 link. There are 57 VoIP channels available to be used in TDM-VoIP calls and also in VoIP-VoIP calls (SBC), which makes the UMG a flexible voice gateway.

As indicated in the last line of the table below, if there are 30 physical channel calls in use, you can still make 13 additional simultaneous calls between VoIP channels using the G.711 codec, or 9 calls using transcoding, or 6 calls using the G.729 codec.

Maximum number of calls between a physical channel and VoIP - with G.711 codec*	Maximum number of simultaneous SBC calls**		
	G.711 ↔ G.711 codec	G.729 ↔ G.711 codec	G.729 ↔ G.729 codec
0	28	19	14
5	26	17	13
10	23	15	11
15	21	14	9
20	18	12	8
25	16	10	7
30	13	9	6

* *The use of the G.729 codec reduces the number of possible simultaneous calls. Refer to your product manual or contact our consultants for more information.*

** *The SBC feature requires the purchase of an additional license.*

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.

Stand Alone Survivability – SAS

The SAS ensures the continuity of telephone communications in case the IP PBX system becomes unavailable. When the UMG has an installed SAS license applied, it assumes the basic functions of the IP PBX system: making and receiving external calls, making calls to extensions, and transferring calls. This way, you don't have to wait for the IP PBX to be available again to restore your telephone communications.

Technical specifications

E1/T1 Link

- 1 link
- Allows you to select the number of channels to match the telephony carrier
- ISDN or R2 signaling (R2 only for E1)
- ISDN PRI
- Connector options:
  - BNC coaxial - 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 and G.726
- Network port selection for SIP protocol and RTP for each 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 and dual filter
  - Tail-length adjustment up to 128 ms

Stand Alone Survivability - SAS*

- Up to 120 extensions can be registered in this mode

Security

- Password-protected access to the web interface
- Access via HTTP or HTTPS
- ACL - Access Control List for the web interface
- Network topology hiding for VoIP/VoIP routing (SBC)*
- Register authorization * (separately licensed item)

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
- 120 extensions for registration authorization (respecting the resource table)

Other features

- Simplified web configuration
- Single-step initial configuration wizard
- Diagnostics interface
- Dashboard with channel status and call statistics
- Line volume setting
- DTMF suppression
- Customizable CDR
- SNMP Support
- Log recording on a remote server or local site
- FTP access

Physical / Environmental

- Polarized power source connector 12 VDC
- Power adapter:
  - Input: 100–240 VAC, 50/60 Hz
  - Output: 12 VDC/2.5 A
- Maximum power consumption: 12 W
- 4x RJ45 Gigabit Ethernet 10/100/1000 Mbps
- Gateway status LED
- Telephony channel status LED
- Error warning LED
- Reset button
- Dimensions: 206.2 mm x 41.8 mm x 102 mm
- Approximate weight: 621 g (without packaging)
- Operating temperature: 0 °C to 50 °C
- Operating humidity: 10–90% non-condensing
- Storage temperature: 0 °C to 85 °C
- Storage humidity: 10–90% non-condensing
- Grounding screw

Register Authorization*

- Supports the registration of up to 120 remote extensions

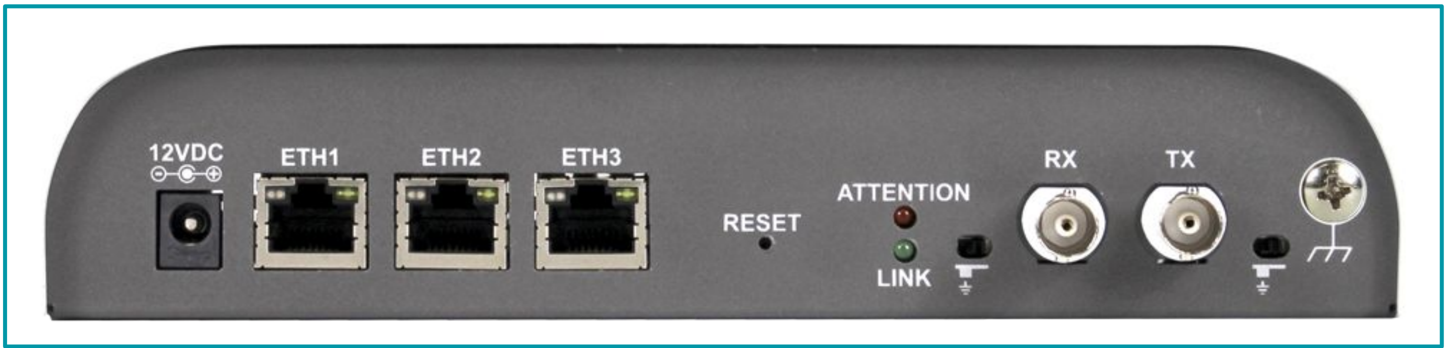
Warranties and Certifications

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

* *Optional feature (optional items are available at an additional cost).*

Product images

Subtitle: *Front view.*

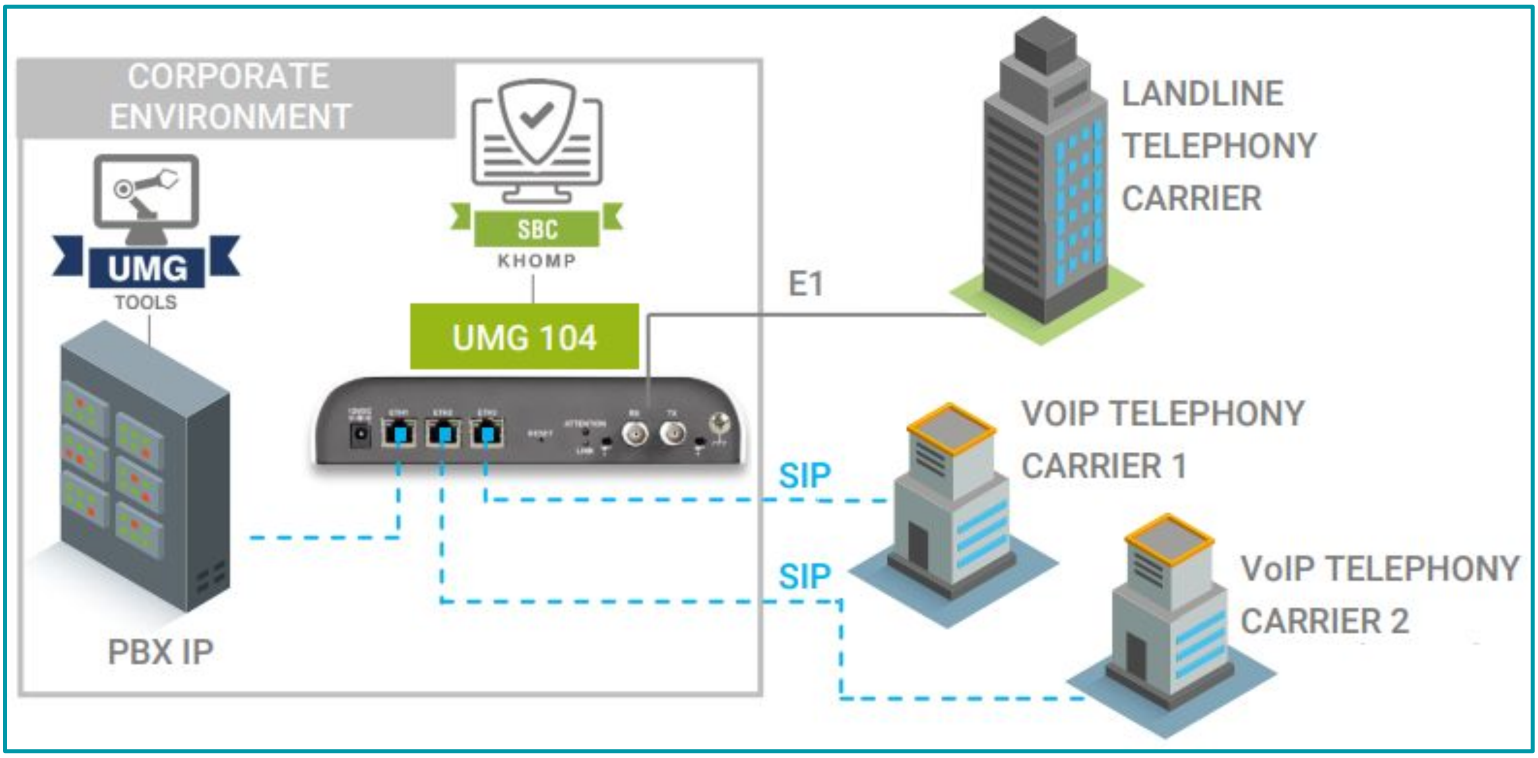


Subtitle: *Rear View - E1/T1 link with BNC coaxial connector.*

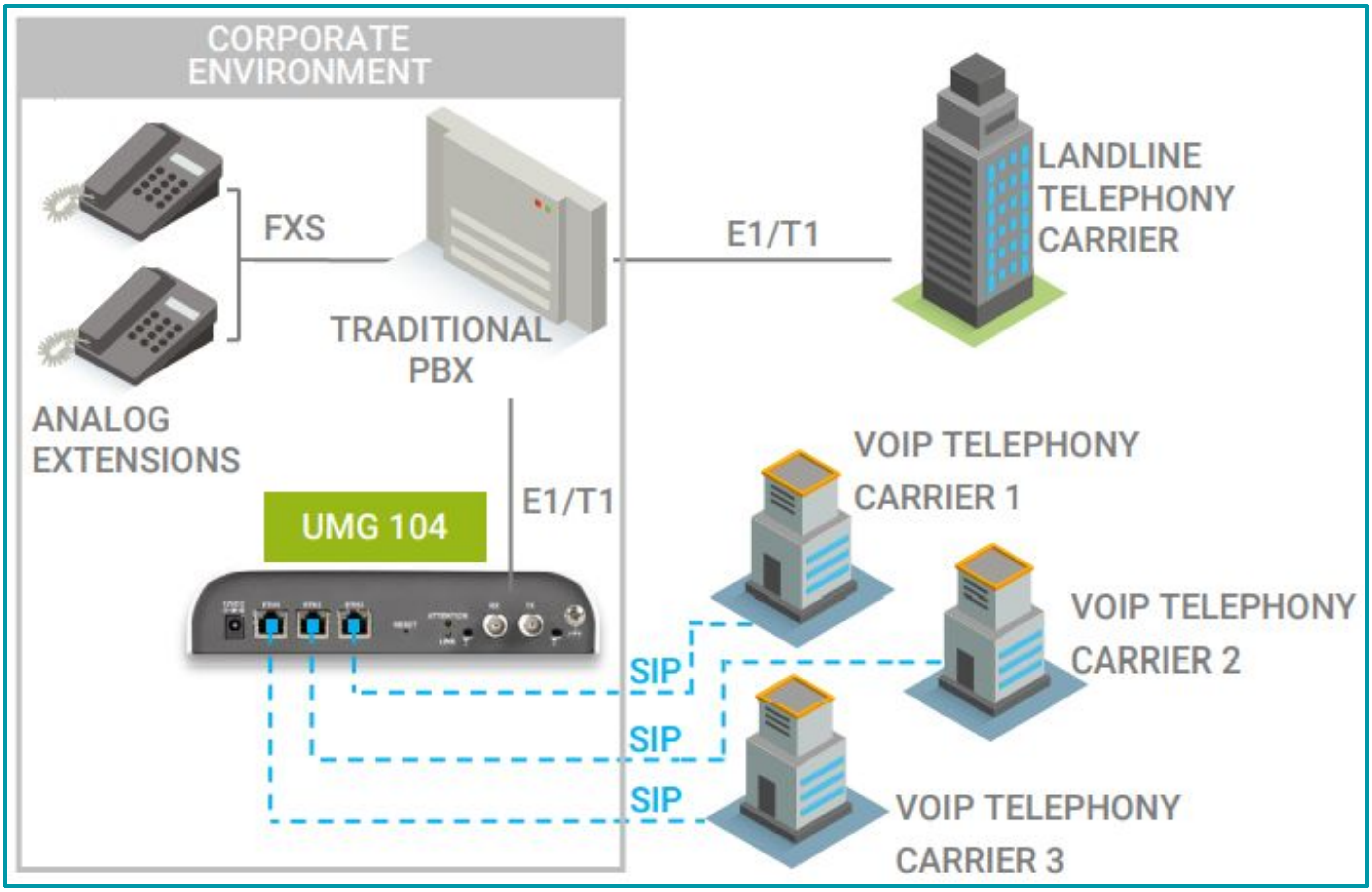
Subtitle: *Rear View - E1/T1 link with RJ45 connector.*

Application model

IP PBX connection with landline carrier and VoIP carrier



Traditional PBX connection with multiple VoIP telephony carriers



KHOMP
Rua Joe Collaço, 253 - Florianópolis, SC, Brazil
+55 (48) 3722.2900
comercial@khomp.com