Voice gateway that supports up to 4x E1/T1 links and SBC

Main features

- 3x gigabit Ethernet network ports
- Up to 4x E1/T1 links
- Supports SS7/SIGTRAN*
- SBC - Routing between VoIP* channels
- Survivability - SAS*
- Register Authorization*

** 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 the main office and branches using an IP network (VoIP).

Overview

The UMG Modular 1200 is a voice gateway created by Khomp for its line of media gateways. It is shipped ready to be connected to a public switched telephone network (STFC), VoIP interconnections, soft-switches and PBX devices, meeting the needs of small and medium businesses.

Robust and effective, it has up to 4x E1/T1 links. With 3x Ethernet Gigabit ports, it allows for the creation of up to 40 SIP accounts, using more than one VoIP telephony carrier in the same network or separate 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 of these features come in a device that was developed with a footprint that allows for easy installation and has a user-friendly web interface for configuration and monitoring.

Routing and Customer Loyalty Scheme

Obtain greater 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 Modular 1200 performs 120 simultaneous calls through an E1/T1 link, which can be divided between TDM and VoIP channels and between VoIP (SBC) channels, making the UMG a flexible voice gateway.

As indicated in the next to last line of the table below, if there are 90 physical channel calls in use, you can still make 15 additional simultaneous calls between VoIP channels regardless of the codec used.

Maximum number of calls between a physical channel and VoIP	**Maximum number of simultaneous SBC calls****		
	Codec G.711 ↔ G.711	**Codec G.729 ↔ G.711**	**Codec G.729 ↔ G.729**
0	60	60	60
30	45	45	45
60	30	30	30
90	15	15	15
120	0	0	0

** *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 is implemented by using the routes together with monitoring of the VoIP server with the Keep Alive resource, which, when enabled, makes the UMG send OPTION type messages to the VoIP server 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.

Survivability

The Stand-Alone Survivability (SAS) feature 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, such as: 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.

Product Images

Subtitle: *Front View.*

Subtitle: *Rear View.*

Technical specifications

Attention

- Product hardware may be replaced without notice.
- Replacement happens when raw material is not available on the market or when better hardware comes along.
- When the hardware is replaced, the product will operate at the same potential as the previous configuration.

Operation Interfaces

- Configuration, monitoring, administration and diagnostics via web interface
- Module for diagnostics via Intercafe Web
- User Interface Web access control
- Packet capture via web interface

System status

- System status via Interface Web
- Status of trunks and channels via Interface Web
- SNMP support

E1/T1 Link

- Up to 4 links
- You can select the number of channels to match the telephony carrier
- ISDN or R2 signaling (R2 only for E1)
- ISDN PRI
- Connector options
  - Coaxial BNC - electrical resistance: 75 Ohms
  - RJ45 - electrical resistance: 120 Ohms
- Clock setting
- Supports an 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 by double answering in R2 signaling
- Collect call blocking through signaling in ISDN
- SS7 and SIGTRAN (optional license)

VoIP

- Up to 40 VoIP accounts with or without registration
- Supported codecs:
  - G.711 (a-law and μ-law)
  - G.729A, G.723.1 and G.726
- Network port selection for SIP protocol and RTP for each VoIP account
- SIP using TCP
- Keep Alive support (SIP OPTIONS)
- Option to ignore source port
- Use of the called number through the 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
- Handling of destination number (to) and source number (from)
- Destination monitoring with Keep Alive (sends UDP packets to the router to indicate that the port is in use, without affecting bandwidth)
- Selection of DTMF sending mode: In band, Out band - RTP (RFC 2833) or Out band - SIP Info
- Addition, removal and retransmission of headers
- Transcoding (conversion between G.711, G.729, G723.1 and G726 codecs)

Smart routing

- Route selection by prefix
- Route selection by regular expressions
- Modification of called and caller numbers
- Imposition 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
- Up to 120 simultaneous records (Resource shared between survival and Records Authorization)

Security

- Password-protected access to the web interface
- Access via HTTP or HTTPS
- Network topology hiding for VoIP/VoIP routing (SBC)*
- Detection of intrusion (fail2ban)
- TLS and SRTP support
- Fraud prevention: call blocking by destination and origin
- DoS / DDoS Protection
- Hiding network topology
- SIP TLS and SRTP protocols (SDES, DTLS and AES)
- Access control - ACL (whitelist and blacklist)
- Protection against malformed packages
- Rogue RTP protection
- Register authorization * (separately licensed item)

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

Survivability feature - SAS*

- Up to 120 extensions can be registered in this mode
- Digit manipulation in survival

Register Authorization*

- Up to 10 remote extensions can be registered

Other features

- User friendly 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
- Provisioning (exporting and importing configurations)
- Zero-touch provisioning
- TR-069 support
- Support ITU-T G.165 and G.168 standards
- Acoustic signaling treatment performed by hardware through DSPs
- Automatic fax tone detection (2100 Hz) automatically enabling echo cancellation

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

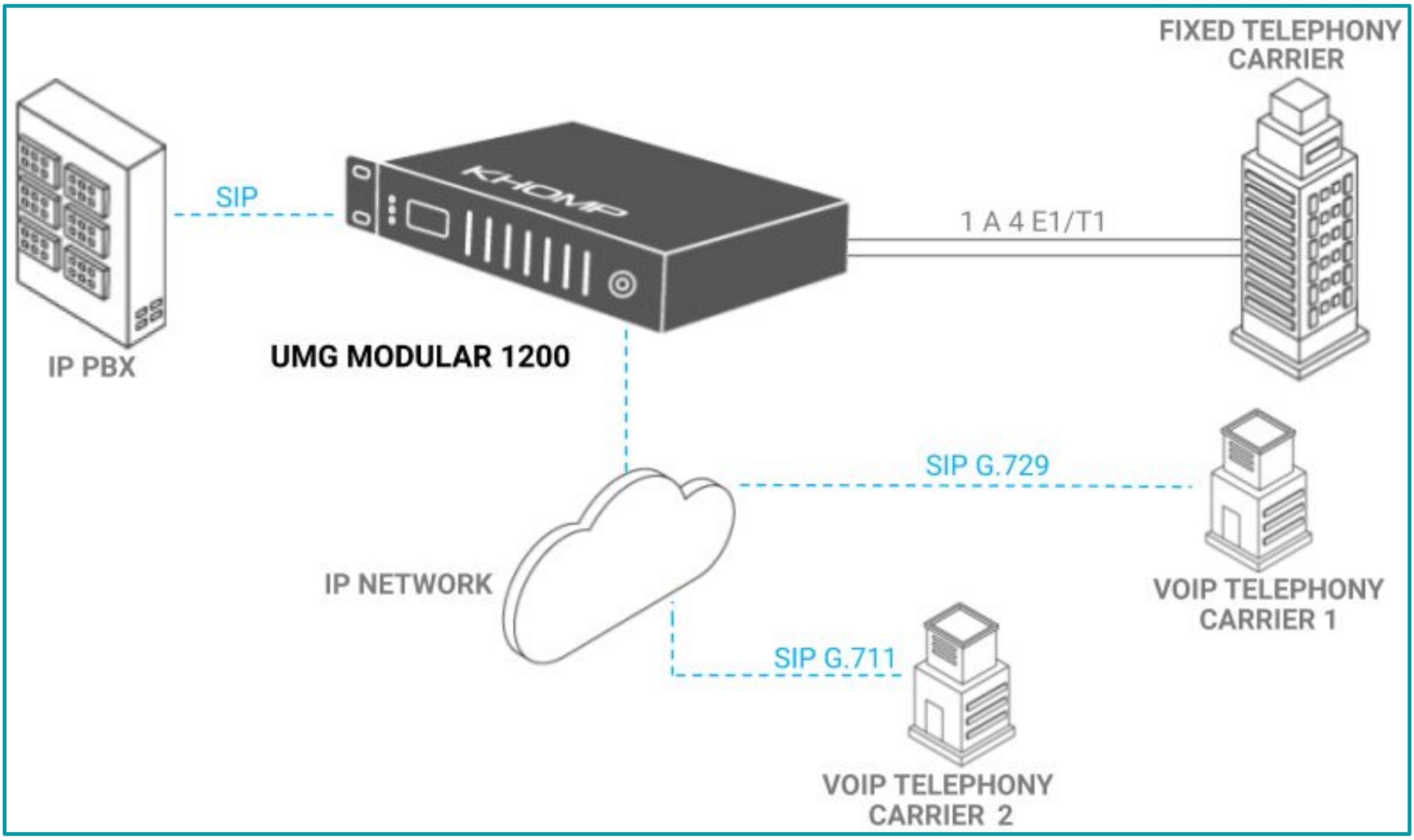
Physical/Environmental

- Polarized power source connector 12 VDC
- Power adapter:
  - Input: 100–240 VAC, 50/60 Hz
  - Output: 12 VDC, 2.5 A
- Internal source -48 VDC*****
- Internal source 100–240 VAC*****

***** *Optional feature*

- Maximum power consumption: 24 W
- 3x RJ45 Gigabit Ethernet 10/100/1000 Mbps
- Gateway status LED
- Telephony channel status LED
- Error warning LED
- Reset button
- Dimensions (W x H x L): 8.68" x 1.75" x 10.95"
- Approximate weight: 5.18 lb (without packaging)
- Operating temperature: 32–122 ºF
- Operating humidity: 10–90% non-condensing
- Storage temperature: 32–185 ºF
- Storage humidity: 10–90% non-condensing
- Grounding screw
- Standard 1U module, 19" half-rack (includes tab for rack mounting)

Application model



Subtitle: *IP PBX connection with carrier through E1 and VoIP.*

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