User Media Gateway with Modular Telephony

Main features

- It is possible to have all telephony interfaces in a single device: E1/T1, FXS, FXO, and GSM
- Register up to 10 VoIP accounts
- SBC – routing between VoIP channels*
- Stand Alone Survivability – SAS*
- Up to 51 simultaneous calls**

Applications

- VoIP telephony carriers
- Corporate environments

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

** *Check the comparative table for call capacities for more information.*

Overview

The UMG Modular 300 is the voice gateway developed by Khomp for its media gateways line. It performs the conversion of telephony networks into VoIP, and it is also capable of making calls between VoIP channels (SBC). Designed to meet the needs of small and medium-sized businesses, the UMG Modular 300 performs up to 46 simultaneous calls between telephony and VoIP interfaces, which can be divided into 3 slots for the combination of the most convenient telephony interfaces for the business model to which it will be applied. It also enables calls between VoIP (SBC) channels.

It has a robust architecture, with processors that are exclusively used for handling the critical telephony tasks, as well as signaling and echo canceling, resulting in high-quality audio. It supports the major signaling systems and codecs in the market, and it also controls and routes calls according to predefined rules.

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.

Available telephony modules

The UMG Modular 300 can be purchased with the following telephony modules (which can also be purchased separately):

- UMG 1E1/T1 module
- UMG 2GSM module
- UMG 4FXO module
- UMG 8FXS module
- UMG 4FXS module
- UMG 2FXO/2FXS Bypass module

Table of simultaneous calls

The UMG Modular 300 can make up to 46 simultaneous calls through the physical telephony channels, which can be used by the E1/T1, FXS, and GSM technologies. Altogether, there are 57 VoIP channels to be used with physical channels or calls between VoIP channels, with each physical channel occupying one VoIP channel. This is what makes the UMG a flexible gateway, as it allows you to have routes between physical channels and VoIP, as well as VoIP/VoIP (SBC).

As indicated in the last line of the table below, if there are 46 physical channel calls in use, you can still make 5 additional simultaneous calls between VoIP channels using the G.711 codec, or 4 calls using transcoding, or 3 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
35	11	7	5
40	8	5	4
46	5	4	3

***** *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

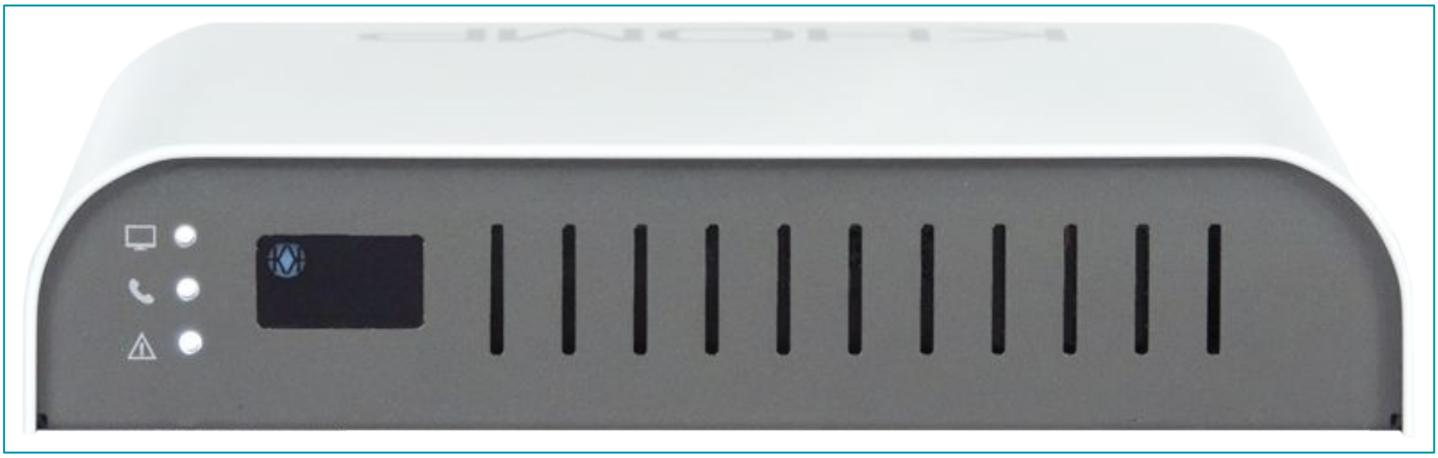
The UMG offers route failover to avoid downtime in call processing in case of a VoIP server failure. The failover function is implemented by using routes along with VoIP server monitoring through the Keep Alive feature. When the Keep Alive function is active, the UMG sends OPTIONS messages to the VoIP 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.

Product images

Legend: *Front view.*



Legend: *Front view - With display.*

Legend: *Rear view - 1E1 + 4FXO + 2GSM.*

Especificações técnicas

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 Interface 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

Link E1/T1 Module

- 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:
  - 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

2 GSM Module

- 2 channels per module. Supports 2 SIM cards per module
- Supports SIM cards from different carriers in the same module
- Available band:
  - Quad-band 2G: 850/900/1800/1900 MHz
  - 3G Penta-band (optional)*****: 850/900/1700/1900/2100 MHz with fallback to 2G quad-band
- SIM Card size: mini SIM (2FF)
- SMS receipt, confirmation, and error notifications
- API for SMS sending
- Ability to control the minutes spent per SIM card group
- Cyclic allocation of GSM channels

4 FXO Module

- 4 channels per module (4x RJ11)
- Minimum ring sensor: 13.5 Vrms @ 13–68 Hz
- Caller ID Detection
- Line impedance
- Collect call blocking

8 FXS Module

- 8 channels per module (2x RJ11)
- Ring voltage: 50–70 Vpp/25 Hz
- Extension numbering plan
- Dialing time-out setup
- End-of-dial marker
- Known numbers registering (Dial Plan)
- Ring cadence configuration Distinctive ring
- Internal and external ring tone setting
- Caller ID generation through DTMF or FSK
- Flash validation time
- Operations available at extensions:
  - Call on-hold
  - Assisted transfer
  - Blind transfer
  - Alternate Call Answering

4 FXS Module

- 4 FXS channels (4x RJ11)
- It has the same features as the FXS module

2 FXS / 2 FXO Bypass Module

- 2 FXS channels and 2 FXO channels (4x RJ11)
- Bypass: toggles between the FXO and FXS channels in case of power failure
- They have the same features as the FXS and FXO modules

VoIP

- Up to 10 VoIP accounts with or without registration
- Supported codecs:
  - G.711 (a-law and μ-law)
  - G.729A, G.723.1 and G.726
- Selection of network port for SIP and RTP protocol for each VoIP account
- SIP using the TCP protocol
- Keep Alive support (SIP OPTIONS)
- Option to ignore source port
- Use of the destination number via the URI
- Q.850 Cause Report
- Selection of DTMF sending mode:
  - In band
  - Out band - RTP (RFC 2833)
  - Out band - SIP Info
- T.38 fax and pass-through support
- Echo cancellation
- 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 modular routing

- Route selection by prefix or regular expressions
- Modification of destination and origin number
- Force destination profile on route with VoIP output
- Route failover
- Use of the "Display name" as a caller ID
- Registration of up to 50 routes
- LCR call routing - lowest cost routing
- 120 extensions passing through the survival proxy (respecting the resource table)
- 120 extensions for registration authorization (respecting the resource table)

Security

- Access to the web interface via password
- Access via HTTP or HTTPS protocol
- Access control - ACL (whitelist and blacklist)
- Hiding network topology in VoIP / VoIP (SBC) routing*
- Intrusion detection (fail2ban)
- TLS and SRTP support
- Fraud prevention: call blocking by destination and origin
- DoS / DDoS Protection
- SIP TLS and SRTP protocols (SDES, DTLS and AES)
- 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

- Full warranty (legal + Khomp warranty): 3 years
  - Legal warranty: 90 days
  - Khomp Warranty: 2 years and 9 months
- Anatel Certification
- ISO 9001 certified industry

Stand Alone Survivability – SAS*

- Supports the registration of up to 120 extensions in this mode
- Digit manipulation in survival

Register Authorization*

- Supports the registration of up to 120 remote extensions

Other features

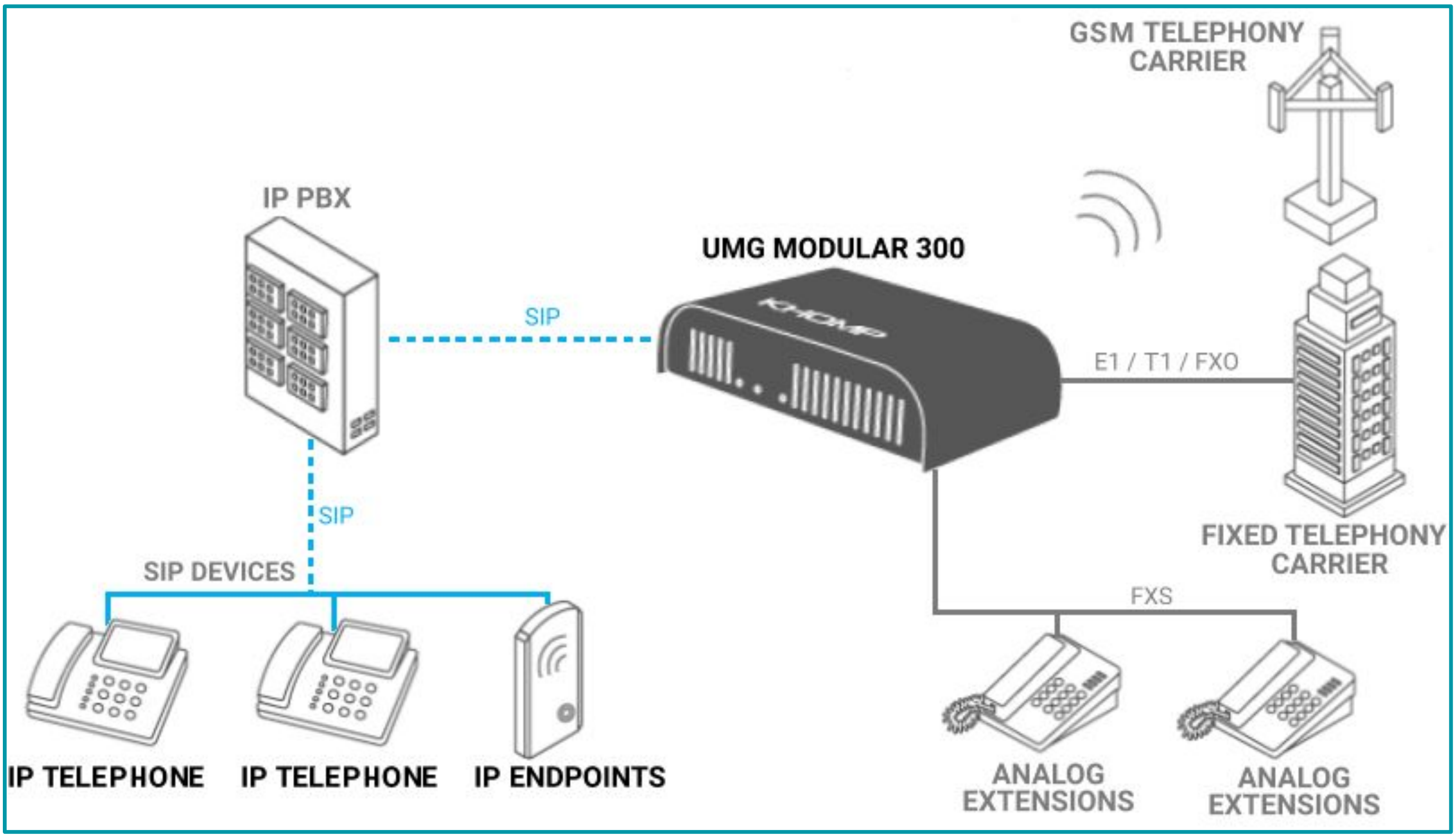
- Initial one-step setup wizard
- Diagnostic interface
- Dashboard with channel status and call statistics
- Line volume adjustment
- DTMF suppression
- Customizable CDR
- SNMP support
- Local or remote server logging
- Access to Logs and CDR via FTP
- Provisioning (exporting and importing configurations)
- 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 (2100 Hz) automatically enabling echo cancellation

* *Optional feature - Optional items incur additional costs.*

Physical/Environmental

- 12 VDC polarized power supply connector
- Energy Adapter:
  - Input: 100–240 VAC 50/60 Hz
  - Output: 12 VDC / 3.5 A
- 2x RJ45 Gigabit Ethernet 10/100/1000 Mbps
- 3x slots for telephony modules: E1 / T1, FXS, FXO and GSM
- Gateway status LED
- Telephony channel status LED
- Error alert LED
- Reset button
- Dimensions: 211x185x46 mm
- Transport box dimensions: 286x252x87 mm
- Gross weight:
- Net weight:
- OLED graphic display (available in DY model)
- Operating temperature: 0–50 ºC
- Operating humidity: 10–90% non-condensing
- Storage temperature: 0–85 ºC
- Storage humidity: 10–90% non-condensing

Application model



Legend: *CIP PBX connection with carrier through E1, FXO, GSM and SIP connections.*