KHOMP

Server with integrated modular gateway

Main features

- This solution integrates a server and a modular voice gateway in a single device
- The server is powered by a processor to embed custom platforms and applications
- The voice gateway can optionally have all telephony interfaces: E1/T1, FXS, FXO, and GSM
- The gateway supports calls between VoIP channels (SBC)*

Aplicações

- PBX IP installaton
- Customized applications with telephony interfaces
- Firewall

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

Overview

The UMG Server Modular 300 is an appliance composed of a modular voice gateway — which can be assembled using different telephony interfaces — and a server with a motherboard and a processor dedicated to the installation of any Windows, Linux, or FreeBSD-based platform.

This appliance allows you to design a complete solution, such as a Unified Communications Center or a Telephone Exchange with call routing, and you can also create firewall solutions, with the option to configure alarm triggers per IP call or cell phone number. All this comes in a single 1U device, and you can also customize the enclosure with a custom logo that is imprinted in the Khomp factory

Flexibility for your business

The UMG Server Modular 300 can include several storage options, as well as the telephony interfaces that best fit your business application. There are three telephony interface modules available which support the E1/T1, FXS, FXO and 2G/3G GSM technologies, a RAM memory stick of up to 8GB, and two SATA ports for connecting 2.5” SSDs or HDDs.

A processor that is exclusive to your solution

All signal convergence and call routing is processed by the voice gateway embedded in the UMG Server Modular 300, thus freeing the motherboard processing for the exclusive use of the operating system and installed application.

User-friendly Web Interface

The UMG Modular 300 gateway embedded in the appliance has a friendly web interface for system monitoring, configuration, diagnosis and management. This allows for time optimization and greater autonomy for the user. The UMG can be remotely accessed, which allows you to manage multiple UMG gateways, if applicable.

Route failover

The UMG device offers route failover to avoid call processing downtime in case of SIP server failures. The failover function is implemented using routes along with SIP server monitoring through the Keep Alive feature. When Keep Alive is enabled, the UMG sends OPTIONS messages to the SIP server in order to monitor its status. When the SIP server does not respond to an OPTIONS message, the UMG then ignores the route where this server is and searches for another compatible route.

Comparative table for call capacities

The UMG Modular 300 gateway embedded in the appliance can make up to 46 simultaneous calls through the physical telephony channels, which can be shared by the E1/T1, FXS, FXO, and GSM technologies. Altogether, there are 57 VoIP channels to be used with the physical channels or calls between VoIP channels (SBC), which makes the UMG a flexible voice gateway.

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

*** Using the G.729 codec reduces the ability for simultaneous calls. Refer to your product manual or contact our consultants for more information.*

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

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.

Server hardware specifications

- IPX4020 motherboard
- Intel Celeron J1800 dual-core processor 2.41 GHz 64-bits
- 2 GB DDR3 RAM Memory (expandable to 8 GB)
- One 2.5" 120 GB SSD (supports up to two 2.5" SATA disks)

Optional itens *

- RAM memory expansion up to 4GB or 8GB DDR3
- Storage discs:
  - 64GB 2.5" SSD
  - 120GB 2.5" SSD
  - 500GB 2.5" HDD
- Supports set of storage disks with a maximum of 2TB

Telephony module specifications E1/T1

- One link
- Allows you to select the number of channels to match the telephone carrier
- ISDN or R2 signaling (R2 only for E1)
- 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
- Limit of one E1/T1 link per media gateway

Smart modular 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

VoIP

- Allows adding of up to 10 VoIP accounts with or without logging
- Supported CODECs
  - G.711 (a-law and μ-law)
  - G.711A
  - G.723
  - 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

Telephony module specifications GSM

- Two channels per module. Supports two SIM cards per module
- Supports SIM cards from different carriers in the same module
- Available band:
  - 2G Quad-band: 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
- Up to six GSM channels per media gateway

8 Telephony module specifications FXS

- Eigth channels per module
- Two ports RJ45: Four FXS channels per connector
- 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
- Up to twenty-four FXS channels per media gateway

4 FXS telephony module specifications

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

Telephony module specifications 2FXS / 2FXO Bypass

- Two FXS channels and two FXO channels
- Four ports RJ11: Two FXS and two FXO
- Bypass: Toggles between FXO and FXS channel in the case of power failure
- They have the same technical specifications as the FXS and FXO modules
- Up to 6FXS/6FXO channels per media gateway

Security

- Password-protected access to the Web Interface
- Access via HTTP or HTTPS protocol
- ACL - Access Control List for the web interface
- Network topology hiding for VoIP/VoIP routing (SBC)

Warranties and Certifications

- Total warranty (legal + Khomp warranty): 1 year
  - Legal warranty: 90 days
  - Khomp warranty: 9 months
- Anatel (Brazilian National Telecommunications Agency) Certification
- ISO 9001 certified industry

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

Telephony module specifications FXO

- Four channels per module
- Four ports RJ11
- Minimum Ring sensor: 13.5 Vrms @ 13–68 Hz
- Caller ID Detection
- Line impedance
- Collect call blocking
- Up to twelve FXO channels per media gateway

Physical/Environmental

- Power Source:
  - Input: 100–240 VAC, 50/60 Hz
- Maximum power consumption: 150 W
- Two RJ45 ports gigabit Ethernet 10/100/1000 Mbps
- Three slots that may contain E1/T1, FXS, FXO and GSM channels according to the modularity
- Three USB 2.0 ports (one in the front / two in the back)
- One VGA port
- OLED graphic display (available only on UMG Server Modular DY model)
- Display size: 62.5 x 15 mm
- Reset button
- Equipment status LED
- E1/T1 Link status LED
- Error warning LED
- Appliance size: 482.8x44.45x280 mm
- Approximate weight: 4.3 Kg
- Standard 1U Module for 19” rack

Compatibility

- Windows
- Linux (kernel version 3.10 or higher)
- FreeBSD**
- pfSense**

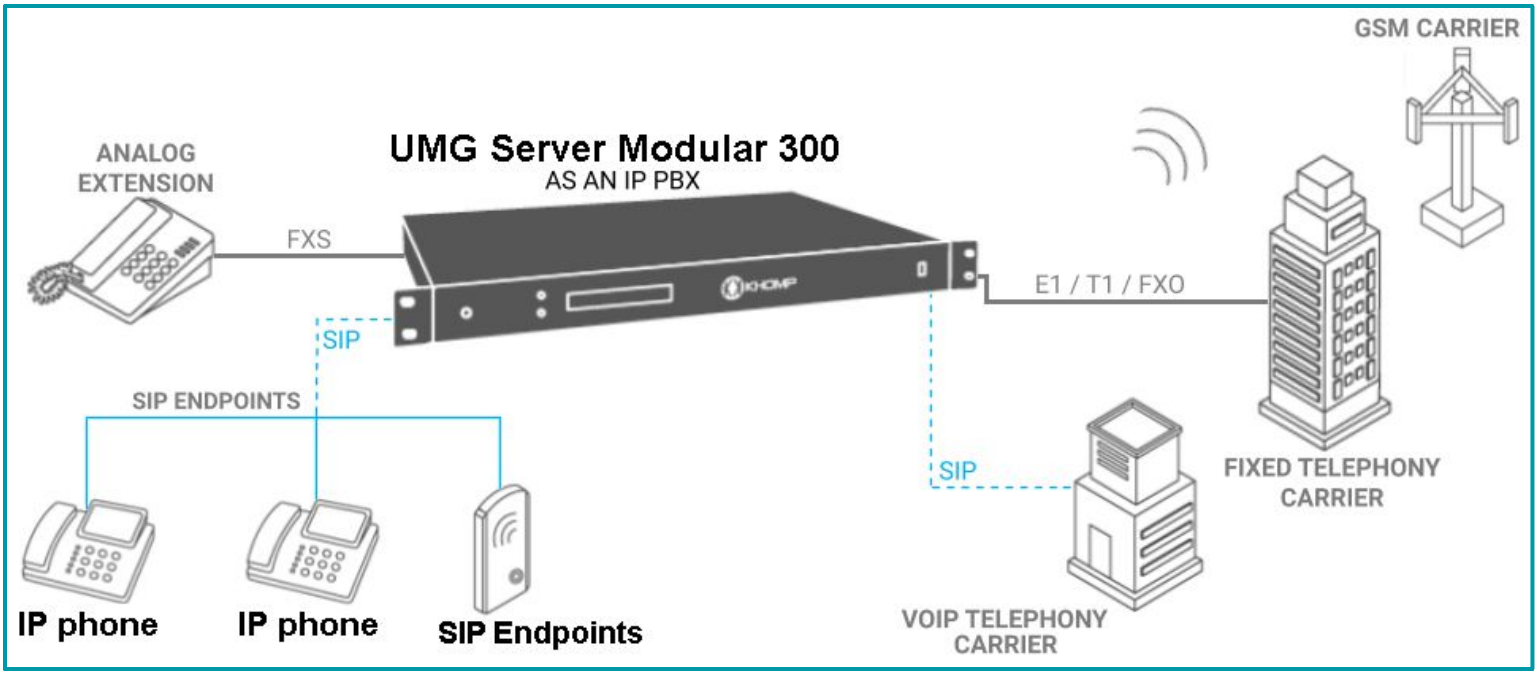
* *Optional items are available at an additional cost.*

** *Does not support use of the display*

Product image

Legend: *Rear view.*

Application model



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