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SMART UNIFIED COMMUNICATIONS

PRODUCT **BROCHURE**

Introduction

OMX Smart Unified Communication Server is an Integrated Communication Platform that manages and controls your communications and collaboration resources to reach their full potential and enhance productivity, while reducing initial and operational cost by combing cutting edge open-standard technology, ease of use, and flexibility.

OMX Smart Unified Communications has an extensive and rich set of features and functions available in one box:
IP Telephony, E-Mail Services, Fax to Mail, Web Conferencing, Call Center, Call Recording, Instant Messaging, Presence, and VPN for Mobility and Remote Access.

OMX Call Center module is designed to simplify management of inbound and outbound calling campaigns, allowing the interaction between system agents and clients seamlessly and efficiently. Key features include:
IVR (Auto Attendant), Call Queues, Team Operations (barge / whisper / listen to calls), SLAs and quality reports, live performance monitoring and more.

OMX Call Center Pro is a powerful, robust, flexible, and easy to use solution designed for automation and efficient management of call centers, allowing real-time collaboration and improving productivity between agents and supervisors through a unified and renowned application as OMX and the expertise of our Call Center Pro Module.

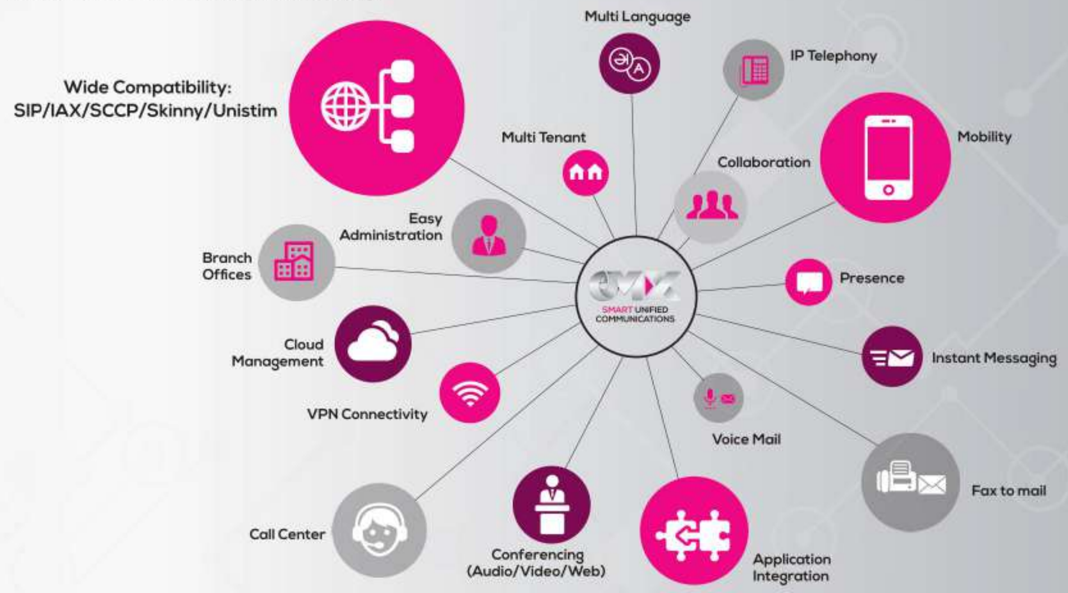
- It is designed to manage inbound and outbound call campaigns, a call management interface, and OMX proprietary communication protocol.
- OMX Call Center Pro has a myriad of features that enable the management and implementation of your Call Center and complementing the features offered in our basic version of the Call Center.

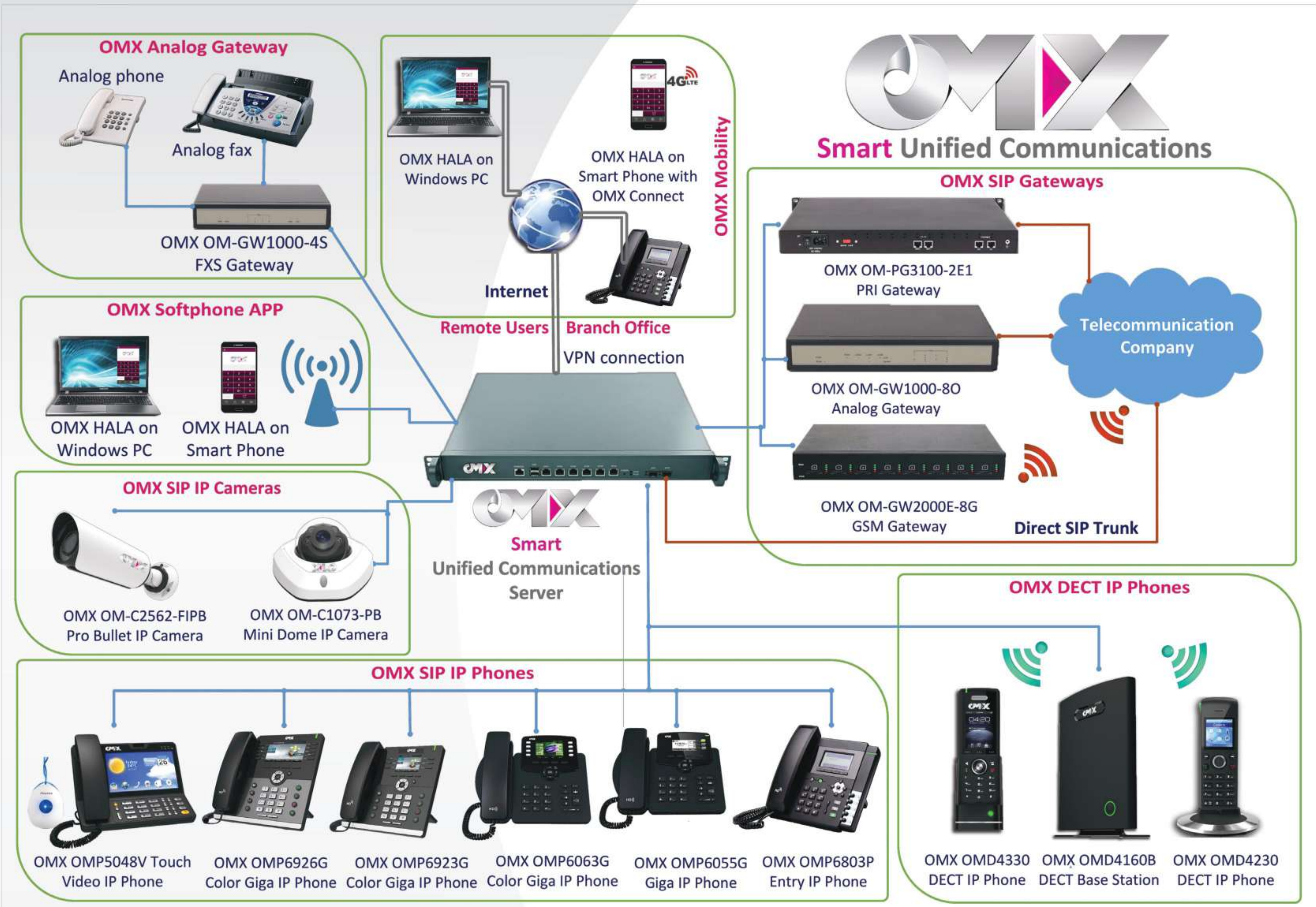
Models:

OMX offers a wide set of appliances to suit your needs. Ranging from small office with 50 extensions up to multi branch enterprises with thousands of extensions.

		OMX SMB		OMX Corporate		OMX Enterprise	
		Small Business		Branch Offices or Small to Mid-Business		Enterprise and Large Contact Centers	
	OMX Appliance Models	OMX Nano OM-SUC10	OMX Mini OM-SUC15	OM-SUC30/60	OM-SUC120/250	OM-SUC500	OM-SU1000
Telephony Features	Extensions (SIP/IAX/Skinny/SCCP/Unistim)	50	75	120 / 250	500 / 1000	2500	5000
	Concurrent Calls	10	15	30 / 60	120 / 250	500	1000
	VoIP Trunks Supported (Analog/GSM)	8	12	24	32	128	256
	Digital Ports (PRI)/SIP Trunks	-	1	1 / 2	2 / 4	8	16
	Recording Capacity*	-	6 months	12 months			
OS	Operating System	Linux 64 bit					
Warranty	Warranty	2 years Hardware Replacement Warranty 2 Years Application Updates & Maintenance Releases					
Hardware	RAID	-	-	-	Optional	✓	✓
	Network Interfaces	1	Redundant Gigabit		Redundant Gigabit with optional Fiber	Redundant Gigabit, with optional Fiber, 10 GB Ethernet	
	Power Redundancy	-	-	-	Optional	✓	✓
	Form Factor	Small Form Factor	Desktop	1U Rack Mount			2U Rack Mount
Advanced Features	HA Option	-	-	✓	✓	✓	✓
	Call Center Option	✓	✓	✓	✓	✓	✓
	Billing & Call Costing	✓	✓	✓	✓	✓	✓
	Integration with CRM/ERP	✓	✓	✓	✓	✓	✓

* *Calculated at 1 hour talk/user/business day*




OMX Analog Gateway
Analog phone
Analog fax
OMX OM-GW1000-4S
FXS Gateway
OMX HALA on
Windows PC
OMX HALA on
Smart Phone with
OMX Connect
OMX Mobility
Internet
Remote Users
Branch Office
VPN connection
Smart Unified Communications
OMX SIP Gateways
OMX OM-PG3100-2E1
PRI Gateway
OMX OM-GW1000-8O
Analog Gateway
OMX OM-GW2000E-8G
GSM Gateway
Telecommunication
Company
Direct SIP Trunk
OMX Softphone APP
OMX HALA on
Windows PC
OMX HALA on
Smart Phone
OMX SIP IP Cameras
OMX OM-C2562-FIPB
Pro Bullet IP Camera
OMX OM-C1073-PB
Mini Dome IP Camera
Smart
Unified Communications
Server
OMX DECT IP Phones
OMX SIP IP Phones
OMX OMP5048V Touch
Video IP Phone
OMX OMP6926G
Color Giga IP Phone
OMX OMP6923G
Color Giga IP Phone
OMX OMP6063G
Color Giga IP Phone
OMX OMP6055G
Giga IP Phone
OMX OMP6803P
Entry IP Phone
OMX OMD4330
DECT IP Phone
OMX OMD4160B
DECT Base Station
OMX OMD4230
DECT IP Phone

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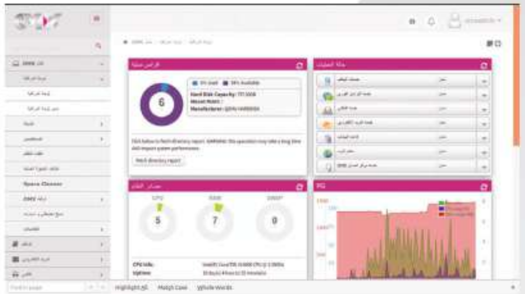
OMX
SMART UNIFIED
COMMUNICATIONS

Technical Specs

Voice Features

- Protocols: SIP, IAX, MGCP, IETF, H323, Google Talk, Jingle/XMPP, Skinny, SCCP, Unistim, NMS, mISDN
- Audio Codec : G.711 (a-law & μ-law), G.719 (passthrough), 722 (Wideband), G.723.1 (pass through), G.726, G.729a, GSM, iLBC, Linear, LPC-10, Speex, ADPCM, CELT (passthrough), and SILK
- Video Codecs: H.261, H.263, H.263+, H.264, mpeg4, vp8
- Fax over IP support
- VAD/CNG/G.168 Echo Cancellation



- NAT Transversal
- DTMF: In-band, Out-of-band (RFC-2833), SIP Info
- ISDN Protocols: AT&T 4ESS, EuroISDN PRI and BRI, Lucent 5ESS, National ISDN 1, National ISDN 2, NFAS, Nortel DMS100, Q.SIG.

Telephone Features

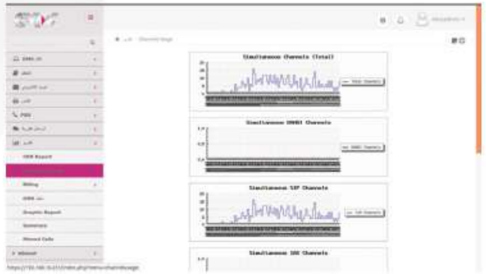
- Call Hold /Forward / Transfer
- DND (Do Not Disturb)
- Call Park / Call Pickup
- Music on Hold
- Voicemail / Voicemail-to-email
- Follow-me
- Ring groups
- Call Paging and Intercom
- Call Forward Rules (Busy / No Answer / Time / Caller ID / Call Type)
- Call Routing (DID)
- Caller ID
- BLF Status Update
- WMI (Message Waiting Indicator)
- Auto Attendant / Digital Receptionist
- Flexible and configurable IVR
- Call Recording
- Callback support
- Call Queues
- Conference center with virtual rooms
- Video Phone support
- DISA (Direct Inward System Access)
- Support for voice synthetization
- Integrated echo canceller
- Incoming and outgoing routes with support for dial pattern matching
- Time-based rules

Management

- Web-based Management Console
- Dashboard for Real Time Status Monitoring
- Integrated Web Server
- Web-based operator panel
- Endpoint configurator
- Hardware detection interface
- VoIP provider configuration
- Built in Firewall
- Integrated Enterprise Database (MySQL)

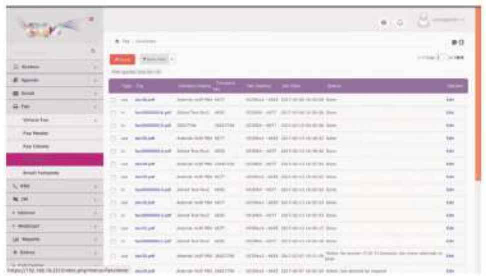
GENERAL

- Online embedded help
- System resources monitor
- Network configurator
- Access control to the interface based on ACLs
- Centralized updates management
- Backup/Restore support via Web (Local /OMX cloud Services)
- Heartbeat Module



FAX

- Fax admin with downloaded PDFs
- Fax to email application with customization
- Access control for fax clients
- Fax send through Web Interface or applications



EMAIL

- Built in mail server with multi-domain support
- Web based management / Web based email client
- Support for mail relay
- Remote SMTP Module

MOBILITY

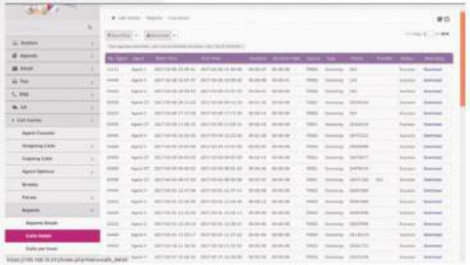
- OMX Hala Full featured Android Client with built in OMX VPN client

INSTANT MESSAGING

- Built-in instant messaging server with web based management
- Support for other IM gateways like MSN, Yahoo Messenger, GTalk and ICQ, XMPP/Jabber
- LDAP support

OMX Call Center Features

- Web Administration
- Do-Not-Call List Support
- Support for breaks generation and configuration
- Support for integration of external application (CRM, forms) during campaigns
- Support for forms design
- Support for generation of scripts by campaign and by queues
- Web-based agent console
- Support for call transfer from console
- Ability to place a call on hold
- Support for ingoing and outgoing campaigns
- Support for call schedule in outgoing campaigns
- Support for call schedule assigned to the same agent
- Support for call back login
- Execution of multiple simultaneous campaigns
- Monitoring of agent assigned to a call
- Call recording by queues
- Maximum wait configuration of a dialed call
- Support for activation/deactivation of prediction
- Automatic calling from a list of numbers
- Asynchronous events assignment

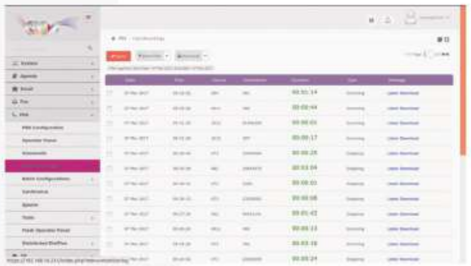


Predictive Dialer

- Interacts directly with calls.
- The dialer asks regularly about the status of the queue to find out how many registered agents are free.
- The number of free agents regulates how many calls are initiated simultaneously.
- The dialer estimates the average length of call, to try to predict if the calls are in progress to finish to place new calls proactively.
- The prediction model is a cumulative Erlang distribution.
- The Web interface lets you start and stop the dialer

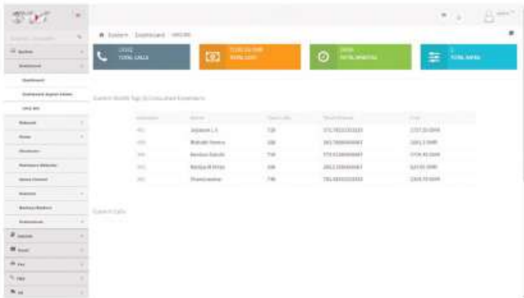
OMX Call Center Reports

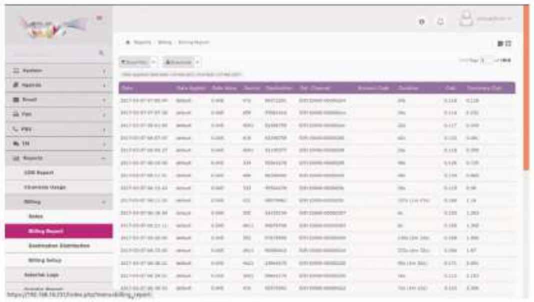
- Export to spreadsheets, PDF, and CSV
- Breaks/Restore reports
- Calls detail
- Calls per hour • Calls per agent
- Wait time, Login / Logout
- Incoming successful calls
- Calls per hour graph
- Agents' Information & Monitoring
- Ingoing calls monitoring
- Campaign monitoring



Call Billing and Cost Control

- Multi Tenant / Multi companies
- Each company supports multiple users with multiple extensions per user
- Generate and email invoices automatically
- OMX Rate Engine integrated for outbound billing
- Inbound / Outbound Billing support
- Summary reports extension wise or user wise
- Call Detail reports extension wise or user wise





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OMX SMART UC **ADVANCED MODULES**

OMX Telephony High-Availability

Overview

- OMXTHA (OMX Telephony High-Availability) is an Add-in Module which creates a high-availability cluster out of any pair of OMX Smart UC servers.
- OMXTHA is a 100% software solution, with a typical Smart UC full switchover in seconds when a failure is detected.
- OMTHA has built-in intelligent network control that allows a single IP address to be shared between peers, so clients/phones automatically connect to the active OMX Smart UC server.
- OMXTHA is ideal for demanding telephony environments like call centers, medical facilities, and mid-to-large size businesses, looking for high telephony uptime.
- OMXTHA does not share any hardware / logical device between peers, so there is no single point of failure.
- OMXTHA is robust in functionality yet simple to set up and use, with web management interface.

Health Sensors

OMXTHA health sensors can detect a range of failures/degradations in OMX Smart UC (not just a stopped process), the environment, external network devices, ITSP/carrier routes, and more. The sensors contribute to an overall health score which allows OMXTHA to automatically take corrective action, notify an administrator, or start an orderly transition to the standby peer.

Peer Synchronization

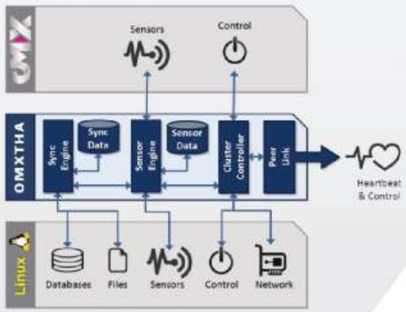
OMXTHA has the ability to synchronize files, directories, databases (MySQL) between peers, when the active peer is healthy, and when scheduled by admin. OMXTHA can instruct the standby peer to overwrite settings/data after synchronization (for differences between peers).

Intelligent Fallback

OMXTHA intelligently transfers control from the backup peer to the primary peer only once the systems are stable and the cluster is reconnected, and can be scheduled when users are least affected.

Encrypted Peer Communications

OMXTHA communications between peers are encrypted. OMXTHA uses 256-bit encryption to ensure that control of the cluster is never at risk, and performance/call data is never exposed.



OMX Integration & Collaboration

OMX Integration

OMX has two features to integrate with external systems. These are: OMX Management Interface (OMI) and OMX Gateway Interface (OGI).

OMI is mostly used to control and manage the OMX system from external system. While OGI is an API gateway to external systems.

OGI communicates with the dial plan through STDIN (Standard Input) and STDOUT (Standard Output).

OMX Gateway Interface (OGI)

The OMX Gateway Interface, abbreviated as OGI, is an interface for adding functionality to OMX UC with many different programming languages. Perl, PHP, C, Pascal, Bourne Shell.

OMX management API

The OMX Management Interface (OMI) allows a client program to connect to an OMX instance and issue commands or read events over a TCP/IP stream. Integrators will find this particularly useful when trying to track the state of a telephony client inside OMX, and directing that client based on custom (and possibly dynamic) rules.

OMX CRM/ERP Integration

OMX can be integrated with different software solutions to enhance, improve and facilitate general users. OMX-CRM Integration can provide your OMX Smart UC with the ability to manage your customers directly. The CRM integration to OMX provides notes uploading, call recording, customer management, number lookups and more.

OMX UC integration can send main call data to the CRM/ERP App like: Call ID, Caller ID, Call Agent no., Call Type (In/Out) and Campaign ID (for marketing campaigns)

OMX Collaboration

- Integrated calendar with support for voice notifications
- Central PhoneBook with click-to-dial capabilities
- Integration with Microsoft Outlook
- Web Conference / WebRTC
- Integrated CRM Support (vTigerCRM and SugarCRM)
- Click to Call

OMX HALA

Smart SIP softphone application for Android and iPhone

NEW

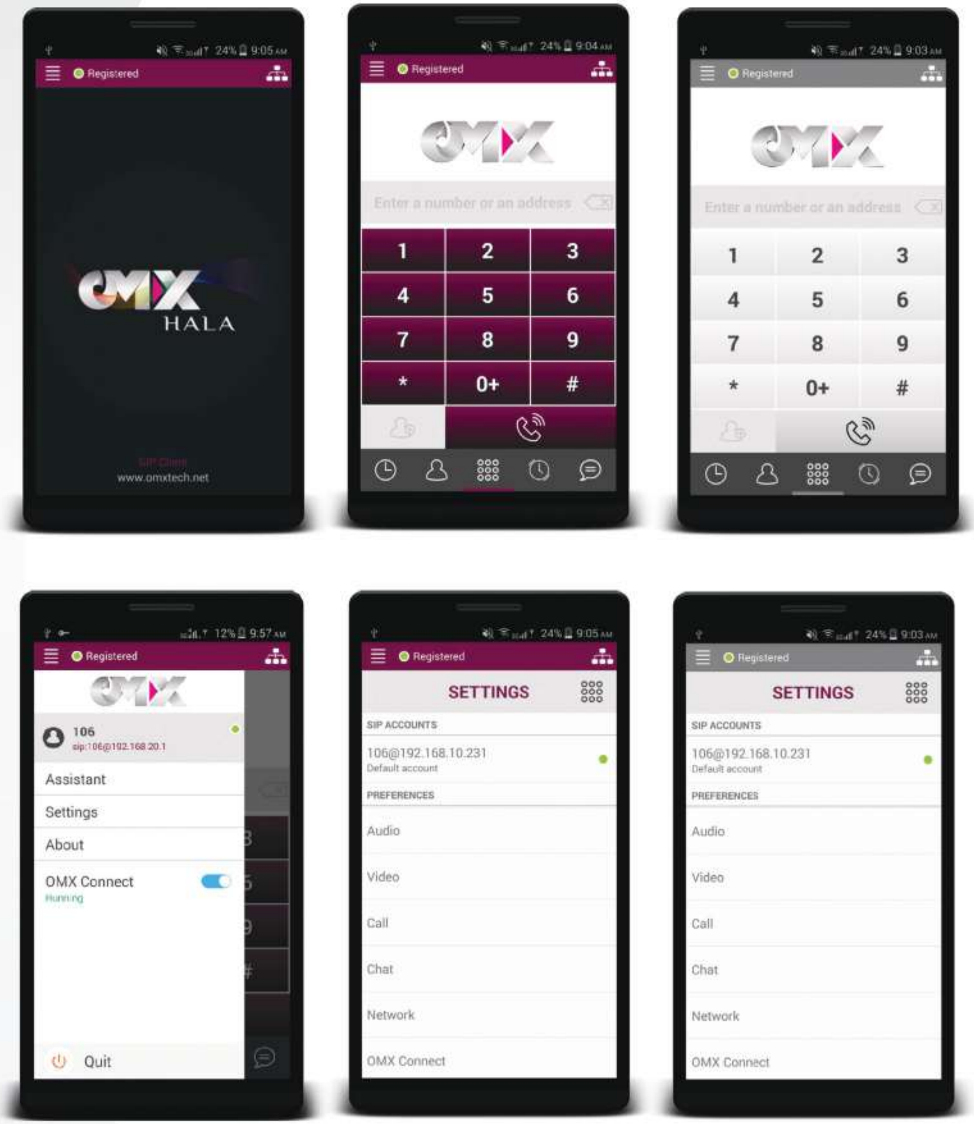
Key Features

- High definition audio and video calls
- Supports calling over 3G and WIFI
- Built-in encryption tunnel for remote user (OMX-Connect)
- Call cost based project selection
- Multiple calls management (pause & resume)
- Call transfer
- Presence and peer status
- Audio conference calls with participants
- Instant Messaging
- Pictures and files sharing
- Contact list
- Call History
- Display of advanced call statistics
- Echo Cancellation
- Call quality indicator
- Secure communication (encryption options)
- Multi Lingual
- Easy account creation using profile import

SIP

ANDROID APP ON Google play

Available on the iPhone App Store



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OMX **SIP SOFT PHONE**

Seamleass Integration with OMX Smart Unified Communications

OMP6926G

Enterprise HD Gigabit Color IP Phone - Managers

Key Features

- 4.3" TFT-LCD, 480 x 272 pixel
- 6 VoIP accounts, 10 Line Keys with LED
- HD Voice: HD Handset, HD Speaker
- 5-way conference
- Multi-language
- PoE, Dual-port Gigabit

NEW

OMP6923G

Enterprise HD Gigabit Color IP Phone - Executive

Key Features

- 2.8" TFT-LCD, 320 x 240 pixel
- 3 VoIP accounts, 6 Line Keys with LED
- HD Voice: HD Handset, HD Speaker
- 5-way conference
- Multi-language
- PoE, Dual-port Gigabit
- Wall mountable

SIP

NEW

OMP6063G

Enterprise HD Gigabit Color IP Phone for SMB

Key Features

- 2.8" TFT-LCD, 320 x 240 pixel with backlight
- 3 VoIP accounts, 8 Line Keys with LED
- HD Voice: HD Handset, HD Speaker
- 3-way conference • Multi-language
- PoE, Dual-port Gigabit

SIP

NEW

OMP6046EXT

Expansion module for OMP6926G Color IP Phones - Operator

Key Features

- Large 800*480 color graphic LCD
- 20 programmable Keys Each with a dual-color LED, 2 pages switching
- Daisy-chain 6 modules for up to 240 DSS Keys
- Rich features, each with different Icon

OMP6926G **OMP6923G** **OMP6063G**



Color Business SIP IP Phones

Models	OMP6926G	OMP6923G	OMP6063G
Line Keys / SIP Accounts	10 Line Keys/ 6 SIP Accounts	6 Line Keys/ 3 SIP Accounts	8 Line Keys/ 3 SIP Accounts
Display	4.3" Color TFT-LCD, 480 x 272 pixel, 24bit depth colors	2.8" Color TFT-LCD, 320 x 240 pixel, 24bit depth colors	2.8" Color TFT-LCD, 320 x 240 pixel, 24bit depth colors
OS	Linux		
Keys	42 keys, 20 programmable	38 Keys, 16 programmable	39 Keys, 18 programmable
Interfaces	Dual Gigabit LAN/WAN		
BLF	10 line keys, up to 36 features (4-page view)	6 line keys, up to 20 features (4-page view)	8 line keys can be programmed
Other Features	Optional OMP6046EXT expansion module		Supports Wireless Headset Adapter Supports Action URI/URL
Voice Codecs	Wideband G.722, G.711 μ-law/A-law, G.723.1, G.726, G.729A/B		
Call Features	Call hold, call waiting, call forward, call return, Redial, Call transfer, Caller ID display, DND, Auto-answer		
Telephony Features	Menu-driven user interface, XML Idle Screen, Theme, Screen Sleep, Mute, Speed dial, SMS, Voicemail, Message Waiting, Indication (MWI) LED, call history, BLF/BLA, Hot-desking, Tone scheme, Volume control, Ring tone selection/Import/Delete, Broad and Deep Interoperability, Soft keys programmable, Phonebook, Black list XML/LDAP phonebook, Multi-Languages Support		
Voice Features	DTMF (In-Band, RFC2833, SIP Info), Acoustic Echo Cancelation(AEC), Acoustic Gain Control(AGC), Voice Activity Detection(VAD), Comfort Noise insertion. Full-duplex Hands-free Speakerphone with AEC		
Network Protocols	TCP/UDP/DNS-SRV, ICMP, RARP, ARP, NTP, SNTP, STUN, UPnP, SNMP, TR069, Static IP/DHCP, TFTP/PPPoE client, HTTP/HTTPS Web server, OpenVPN		

OMP6055G

Enterprise HD Business IP Phone for SMB

NEW

Key Features

- 2.3",132x64 pixel Graphical LCD with backlight
- 3 VoIP accounts, 3 Line Keys with LED
- HD Voice: HD Handset, HD Speaker
- 3-way conference
- Multi-language
- PoE, Dual-port Gigabit

SIP

OMP6803P

HD Business IP Phone for SMB

Key Features

- Backlit 132*64 graphic LCD
- 2 VoIP accounts, 2 Line Keys
- HD Voice: HD Handset, HD Speaker
- 5 BLF keys
- Multi-language
- PoE, Dual 10/100 Ethernet
- Wall Mountable

OMP6055G

OMP6803P

Business SIP IP Phones		
Models	**OMP6055G**	**OMP6803P**
Line Keys / SIP Accounts	2 Line Keys/ 2 SIP Accounts	2 Line Keys/ 2 SIP Accounts
Display	2.3" 132x64 backlit Graphical LCD	132x64 backlit Graphical LCD
OS	Linux	
Keys	34 keys, 13 programmable keys	34 Keys, 7 programmable Keys
Interfaces	Dual Gigabit LAN/WAN	10/100 LAN/WAN
BLF		5 BLF
Voice Codecs	G.722, G.711 μ-law/A-law, G.723.1, G.726, G.729A/B,	
Call Features	Call hold, call waiting, call forward, call return, Redial, Call transfer, Caller ID display, DND, Auto-answer, 5-Way Conference	
Telephony	Menu-driven user interface, XML Idle Screen, Theme, Screen Sleep, Mute, Speed dial, SMS, Voicemail, Message Waiting, Indication (MWI) LED, call history, BLF/BLA, Hot-desking, Tone scheme, Volume control, Ring tone selection/Import/Delete, Broad and Deep Interoperability, Soft keys programmable, Phonebook, Black list XML/LDAP phonebook, Multi-Languages Support	
Voice Features	DTMF (In-Band, RFC2833, SIP Info), Acoustic Echo Cancelation(AEC), Acoustic Gain Control(AGC), Voice Activity Detection(VAD), Comfort Noise insertion. Full-duplex Hands-free Speakerphone with AEC	
Network Protocols	TCP/UDP/DNS-SRV, ICMP, RARP, ARP, NTP, SNTP, STUN, UPnP, SNMP, TR069, Static IP/DHCP, TFTP/PP PoE client, HTTP/HTTPS Web server, OpenVPN	
Management	Transport Layer Security (TLS), SRTP (RFC3711), SIPS, VLAN QoS (802.1pq). Auto Provision: TFTP/HTTP/HTTPS/PnP	
POE	Yes	

Seamless Integration with OMX Smart Unified Communications

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VIDEO
SIP IP PHONES

OMP5048V

Android-based Multimedia 7" Touch Screen IP Phone

Key Features

- 7-inch capacitive touch screen TFT LCD, 1024*600 pixels;
- 1.3M pixels CMOS camera
- Dual switched 10/100/1000Mbps Ethernet ports
- Built-in Bluetooth, USB and Wi-Fi feature
- Android 5.1 Operating System
- Voice and Video Codec: PCMA, PCMU, G.722, G.723, G.729 OPUS, ILBC, AMR-NB, AMR-WB, H.263, H.264, H.265;
- Protocol: SIP RFC3261, TCP/UDP/IP, PPPoE, RTP/RTCP, HTTP/HTTPS

SIP

OMP5048V(869)

Android-based multimedia 7" Touch Screen IP Phone for Medical Sector / Intelligent Health Care Solution

SIP

Key Features

- Same Specifications as above model
- Easy to use
- Suitable for Patients / Elderly people / Kids
- Built-in Radio Frequency Module
- Support up to 10 pendants
- Emergency Call
- Remote Monitor
- Health Data transmission
- Remote Diagnosis

OMP5048V

OMP5048V(869)

Video SIP IP Phones	
Models	**OMP5048V / OMP5048V(869)**
Line Keys / SIP Accounts	6 SIP Accounts
Display	7-inch capacitive touch screen TFT LCD, 1024*600 pixels
OS	Android 5.1 Operating System and support Android API for the 3rd party Android application installation.
Keys	28 keys
Video Features	Video Códec: H.263, H.264, H.265 Video Solution: QCIF, CIF, VGA, 4CIF, 720P Video Bitrate: 64kbps-2Mbps Adaptive Bandwidth Adjustment Video Character: PIP, Full Screen, Local Video ON/OFF Control Image Codec: JPEG, GIF, PNG, BMP
Camera	1.3M pixels CMOS camera with privacy shutter
Voice Codecs	G.722, G.711 μ-law/A-law, G.723.1, G.726, G.729A/B, iLBC, AMR-NB, OPUS, AMR-WB
Call Features	Call Waiting, Call Transfer, Call Hold, Mute, Redial, Speed Dial, Direct IP call,6-Way Conference, DND (Do Not Disturb)
Telephony Features	Volume Adjustable, Ring Tones Selectable Multi-Languages Support Default Apps: Phonebook, Weather, Google Play, Digital Photo Frame, Stock, News Feeds, Calendar, World Clock, Alarm Clock, Email, Camera, File Browser Games. Supports 3rd Party Android Applications
Network Protocols	SIP V2(RFC3261), UDP/TCP/IP, RTP/RTCP, ARP/RARP, NAT Traversal: Static NAT configure, SIP Keep Alive, IP Assignment: Static IP, DHCP, PPPoE, NTP For Auto Time Setting, Network Packet Capture, TFTP/FTP/ HTTP/HTTPS Client API, OpenVPN
Provisioning	Firmware Local Update. Flash Disk Firmware Remote Update: FTP method Auto Provisioning Using FTP/TFTP/HTTP/HTTPS
Management Features	System Log Export Restore to Factory Default Configuration Managements Via LCD
PoE	Yes

DECT Base: OMD4160B

IP DECT Base Station Multi-cell Solution for Small and Medium Enterprise

SIP

Key Features

- DECT GAP / CAT-IQ
- Wideband audio (HDSP) basic and extended
- 12 slot radio with up to 10 voice channels active
- Worldwide radio power levels / frequency bands
- Scalable system from 1 to 254 bases in same network
- 1000 subscriptions (max 1000 handsets)
- Power over Ethernet
- Over the air synchronization
- Support software download to wireless terminals
- LDAP and/or XML phonebook support • Seamless handover
- Repeater support • Auto/Remote provisioning

DECT Phone: OMD4230

OMD4230 Entry Level DECT IP Phone Handset Compliant with OMD4160B Base

SIP

Key Features

- HD audio support (G.722)
- 1.44" TFT display, 128 x 128 • Polyphonic ringtones
- Local phonebook • Central phonebook support • Wideband speakerphone
- Chargeable 2 x AAA batteries

DECT Phone: OMD4330

OMD4330 Mid End DECT IP Phone Handset Compliant with OMD4160B Base

SIP

Key Features

- HD audio support (G.722) • 2" TFT display (176x220)
- Local phonebook (100 entries)
- Remote (LDAP) and Central (3000 entries) phonebooks
- Lithium Ion Battery (1100mAh) • Vibrator / BeltClip

OMD4160B

OMD4230

OMD4330

DECT SIP Base Station

Model	OMD4160B
DECT	Frequency band: 1880 MHz – 1930 MHz (DECT) Four power levels (14, 17, 20 and 24 dBm) Seamless handover using connection handover Authentication / encryption of base and handset
Audio	10 audio channels using G.726 / G711, G.729 5 CAT-IQ wideband audio channels using G.722
Antennas	Internal omni-directional antennas Range: Indoor: 50 m, Outdoor: 300 m
Additional Features	Repeaters supported (up to 100 repeaters) Synchronization via air interface
System	Max 1000 SIP registrations with 254 Bases into one PBX
Network	TFTP, HTTP, HTTPS for remote configuration and firmware download, VLAN, Embedded web server for easy configuration
Etherenet	10/100 BASE-T IEEE802.3
Power Supply	Power over Ethernet (PoE): 36-60 V - IEEE802.3af (Class 2) Max power consumption: 5W

DECT SIP IP Phones

Model	OMD4230	OMD4330
Audio Codecs	G722, G726, HD audio speakerphone, HAC compliant (TIA-1083)	
Display	1.44" TFT Color, 128 x 128 pixels	2" TFT Color, 176x220 pixels
DECT	Power levels: 4 One antenna Encryption supported	Power levels: 4 Two antennas with fast diversity Encryption supported
Call Features	Call list with 20 entries Central phonebook support (one LDAP server) Local phonebook with 50 entries (1 number/name) Date and time in Idle display	Call list with 50 entries Central phonebook support (LDAP or XML) Local phonebook with 100 entries (3numbers/name) Date and time in Idle display
Reliability Spec	Drop height: 1m Keypad lifetime: 40.000 presses Keypad abrasion: Standard	Drop height: 1,7m on wooden floor Keypad lifetime: 100,000 presses
Charging	Charge time: 10 hours EU/US/AUS/UK plugs supported Table charging cradle	Charge time: 5 hours EU/US/AUS/UK plugs supported Table charging cradle

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DECT
SIP IP PHONES

2 E1, PRI/SIP, G.729 Gateway

OM-PG3100-2E1 is a digital trunk gateway based on embedded operating system.

Key Features

- Energy efficiency concurrent processing
- Service oriented architecture, rich services, support voice, IP fax and Modem/POS service
- Flexible dial-up rules and operation, in order to adapt to the different work scene.
- Voice codec: G.711A/U, G.723.1, G.729A/B, iLBC
- Good compatibility, support Asterisk, Elastix, Freeswitch and Small and medium UC/IPPBX platform

SIP

1 E1, PRI/SIP, G.729

OM-PG3600-1E1 is a trunk gateway aimed at small and medium enterprise.

SIP

Key Features

- Strong performance of hardware platform, energy efficiency concurrent processing
- Service oriented architecture, rich services, in addition to support voice and IP fax outside, still can be expanded to fax server
- Flexible dial-up rules and operation, in order to adapt to the different work scene.
- Voice codec: G.711A/U,G.723.1,G.729A/B, iLBC
- Good compatibility, support Asterisk, Elastix, Freeswitch and Small and medium IPPBX

OM-PG3100-2E1 **OM-PG3600-1E1**

PRI (ISDN) E1/T1 Gateways		
Model	**OM-PG3100-2E1**	**OM-PG3600-1E1**
Physical Interfaces	2 x E1/T1, G.703, 120 RJ-48 2 x RJ-45 10/100M 1 x RS232, 9600bps	1 E1/T1, G.703, 120, RJ-48 2 x 10/100M Base-T RJ45 1x RS232,115200bps
Voice & Fax	G.711A/U law, G.723.1, G.729A/B, iLBC, Silence Suppression & Detection, Comfort Noise Generation (CNG), Voice Activity Detection (VAD), Echo Cancellation (G.168), Adaptive (Dynamic) Jitter Buffer, T.38/Pass-through, Modem, QoS: VLAN 802.1p/q, ToS, DTMF:RFC2833, SIP INFO, INBAND	
Maintenance & Upgrade	Web based configuration, HTTP/Telnet Configuration Syslog: Debug, Info, Error, Warning, Notice Cloud Management System	
Routing & Number manipulation	PSTN-PSTN, PSTN-IP, IP-IP, IP-PSTN Intelligent routing 256 routing 128 number manipulation	N/A
Power	100-240V,50-60Hz, 1.1A Max Power consumption: 25W	100-240V,50-60Hz,1.1A Max Power Consumption: 10w

4 Port /8 Port FXS Gateway

OM-GW1000-4S / OM-GW1000-8S analog gateway series is a versatile P-based voice and fax gateway.

Key Features

- Support SIP/MGCP
- Primary/Backup SIP Servers
- Flexible Routing and Manipulation
- Support Modem/POS
- Elastix/Broadsoft Certification
- Support IPv4 and IPv6
- Data/ Voice/ Management VLAN
- Built-in Firewall and Access Rules
- SNMP/TR069/Provision
- Cloud-based Management System and Bandwidth Optimization

SIP

4 Port / 8 Port FXO Gateway

OM-GW1000-4O FXO / OM-GW1000-8O FXO VoIP analog gateway series is IP-based access gateway to connect Analog lines from the Telephone Operators.

Key Features

- Primary and secondary SIP account
- Flexible FXO port groups
- Multiple SIP trunks
- Caller ID
- Flexible routing and manipulation rules
- Data/Voice/Management VLAN
- T.38andPass-through
- Friendly Web GUI access, easy to maintain

SIP

OM-GW1000-4S **OM-GW1000-4O FXO**

Analog VOIP Gateways		
Model	**OM-GW1000-4S** **OM-GW1000-8S**	**OM-GW1000-4O** **OM-GW1000-8O**
Physical Interface	4 x FXS (OM-GW1000-4S) 8 x FXS (OM-GW1000-8S) 4 x RJ-45 10/100Mbps	4 x FXO (OM-GW1000-4O) 8 x FXO (OM-GW1000-8O) 4 x RJ-45 10/100Mbps
Voice & FAX	G.711A/U law, G.723.1, G.729A/B,G.726,iLBC, AMR Echo Cancellation (G.168) Modem/POS Signal/RFC2833/INBAND	G.711A/U law, G.723.1, G.729A/B Echo Cancellation (G.168) T.38/ Pass-through/Modem FAX mode, Up to 14.4kbps DTMF mode: Signal/RFC2833/INBAND
Protocol	SIP V2.0 (RFC 3261,3262,3264) DHCP/PPPoE VLAN 802.1P/802.1Q STUN (RFC 3489)	
Call & Route	Port Group Primary and Secondary SIP Account 32 Inbound/Outbound Routing	FXO port group Primary and secondary SIP Account 32 Inbound routing 32 outbound routing
Maintenance & Upgrade	SNMP/TR069 Auto Provisioning Syslog and CDR Access Rule List Cloud-based Management	
Power	100-240V,50-60Hz Output: 12VDC,2A Power Consumption: 18W	100-240V,50-60Hz,1.2A Max @External adapter 12VDC Power consumption 4FXO @ 12W 8FXO @ 20W

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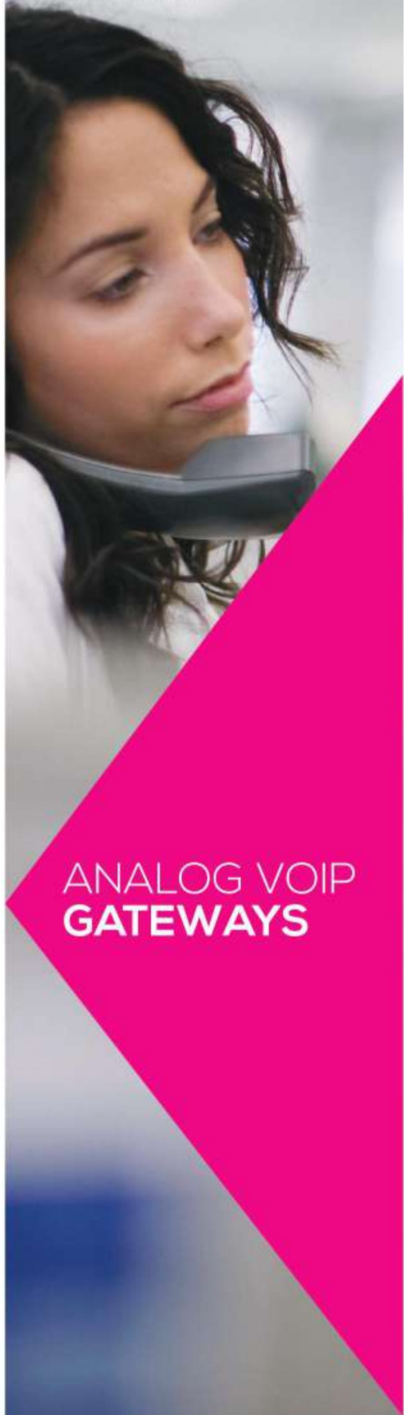

ANALOG VOIP
GATEWAYS

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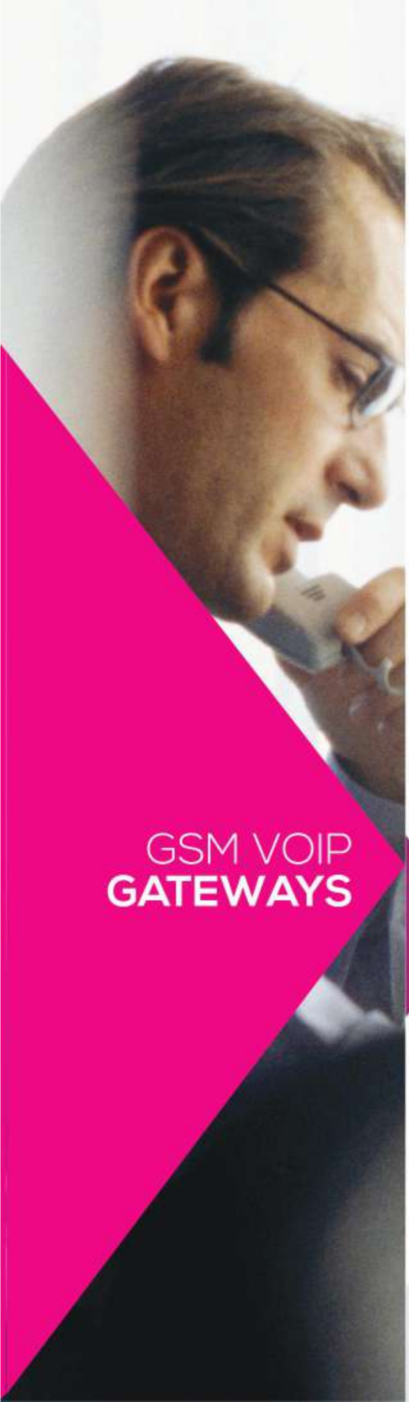


GSM VOIP **GATEWAYS**

NEW

1 GSM Channel IP Gateway

OM-GW2000-1G Full featured VoIP gateway based on IP and GSM wireless network

Key Features

SIP

- 1 GSM port
- GSM: quad-band 850/900/1800/1900MHz
- Web-based SMS/SMSC and USSD, open API
- Support digit map, flexible dial rules
- Manual /automatic selection operators
- Syslog way output tracking information
- IP/Password based white/blacklist
- Voice prompts, HTTP Web support for configuration and updates

4 / 8 / 16 / 32 GSM Channel IP Gateway

OM-GW2000E-4G / OM-GW2000E-8G / OM-GW2000E-16 G
OM-GW2000E-32G Multi-functional GSM/CDMA VoIP Gateway series

Key Features

SIP

- 4 GSM/CDMA/WCDMA Channels (OM-GW2000E-4G) / 8 GSM/CDMA/WCDMA Channels (OM-GW2000E 8G)
- GSM: Quad-band 850/900/1800/1900MHz, CDMA: 800MHz
- Up to 4 Concurrent Calls (OM-GW2000E-4G)/ 8 Concurrent Calls (OM-GW2000E-8G)
- Flexible Dial Rules and Manipulation Rules
- SMS Sending and Receiving
- USSD, Open API for SMS/USSD, BCCH
- Carrier Selection, PIN Code Management
- Supports Remote SIM Card Management (SIMCloud)

16 / 32 GSM Channel IP Gateway with SIM Rotation

OM-GW2000F-16G/64SIM / OM-GW2000F-32G/128SIM
Multi-functional GSM/CDMA VoIP Gateway series with local SIM rotation

Key Features

SIP

- 16 GSM Channels with 64 SIM Slots (OM-G2000F-16G/64SIM)/ 32 GSM Channels with 128 SIM Slots (OM-G2000F-32G/128SIM)
- Easy to Conduct SIM Rotation Locally
- GSM: 850/900/1800/1900MHz; CDMA: 800MHz; WCDMA: 850/900/1900/2100MHz
- Built-in Bandwidth Saving, Signaling & RTP Encryption
- HTTP API for Bulk SMS
- SMS Send / Receive. SMS to E-mail
- Auto Call/SMS Generation

OM-GW2000-1G

OM-GW2000E-4G/8G/16G/32G

OM-GW2000F-16G/64SIM 32G/128SIM

GSM VoIP Gateways

Model	OM-GW2000-1G	OM-GW2000E-4G/8G/16G/32G	OM-GW2000E-16G/64SIM 32G/128SIM
GSM Channels	1 port GSM channel	4/8/16/32 port GSM channels	16/32 port GSM channels
SIM Slots	1	4/8/16/32	64 SIM (16G/64SIM) 128 SIM(32G/128SIM)
SIM Card Rotation	No	No	SIM Number and Balance Check
Network ports	1 LAN 10/100M	2 x 10/100M	2 x 10/100M
Console Port	N/A	Console: 1* RS232	Console: 1* RS232
Voice Codec	G.711a/u law, G.723.1, G.729AB		
Voice Processing	Silence Suppression & Detection, Comfort Noise Generation(CNG), Voice Activity Detection(VAD), Echo Cancellation (G.168), Adaptive (Dynamic) Jitter Buffer, Call Progress Tone Generation, Programmable Gain Control		
Network	Network Mode: NAT router or switch mode, Network Protocols: IP, TCP, UDP, TFTP, FTP, RTP, RTCP, ARP, RARP, ICMP Ping, NTP, SNTP, Http, DNS, PPPoE, DHCP, NAT traversal: Static NAT, STUN		

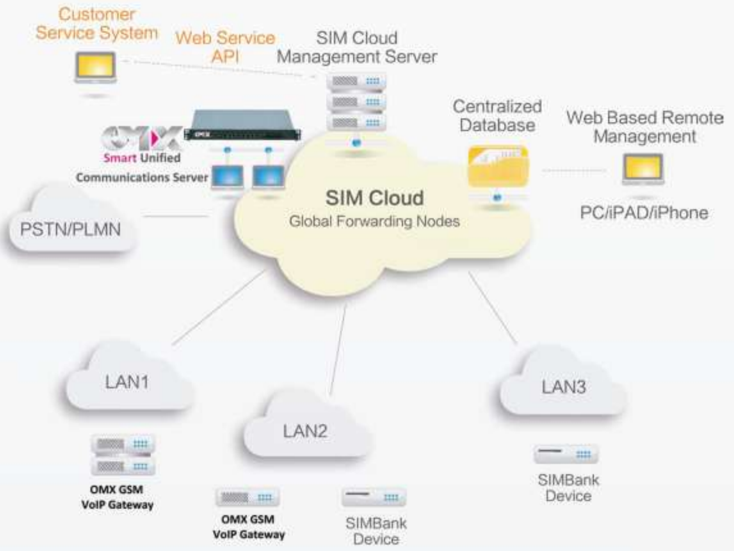
OMX SIMCLOUD

NEW

OMX SIM Cloud is centralized SIM card management system based on the latest cloud communication technologies. Ready-To-Deploy with Public Cloud, or available as SIMCloud Local Server software.
OMX SIMCloud provides device management, SIM card management, human behavior simulation, real-time statistics and Open Web-Service API.

Key Features

- Ready-To-Deploy SIMCloud / On Premise SIMCloud Server
- Commercial Database, Mass Storage and High Security
- Global Forwarding Nodes
- Bandwidth Compression and Signal/Media Encryption
- Flexible SIM Allocation and Credit Usage
- Auto SIM Card Recharge and Promotions Management
- SIM Card Human Behavior Simulation
- Auto SMS/Call Generation
- Real-time SIM Card Availability Detection
- Anti Call Scanning
- 15M/24H Performance Statistics
- Mass CDR/SMS/USSD Records
- Open Web-Service API and Customized Service



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OMX SIMBANK

NEW

OMX SIMBank is a centralized storage device which allowing to store up to 64/128 SIM cards. Together with OMX SIM Server and OMX OM-GW2000E GSM VoIP Gateway, it provides complete wireless VoIP solution for clients based on cloud environment. Moreover, all SIM card information is processed and transmitted through private protocol to ensure high reliability IP network communication between SIMbank and GSM VoIP Gateway.

Key Features

- NAT Traversal
- 128 SIM Slots Maximum
- Dynamic Allocation of SIM Cards
- High Reliability Data Transmit Mechanism
- Remote SIM Card, Easy to Replace SIM Card
- Hot Swap of SIM Cards with No Service Interruption
- Remote Management, Easy to Maintain Devices

SIM	
Capacity	64 or 128 SIM Slots
SIM Type	GSM, USIM-UMTS
Interfaces	
Local	COM Port with RJ-45 Connector
Netework	10/100 Base-T RJ45
Management	
Local	Friendly Web GUI Interface
Remote	Remote Web Access through SIM Cloud System
Firmware Upgrade	Local Firmware Upgrade through Web Interface Remote Upgrade through Auto Provisioning

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