VoIP2ALL Gateway
8 / 16 / 24 Channel

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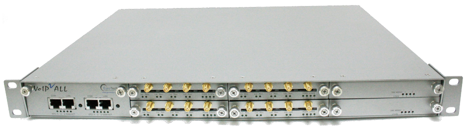
GSM UMTS CDMA
DISCOVERY
telecom



Increased Efficiency. Saves You Money.

VoIP²ALL™ is an innovative line that integrates Cellular (GSM UMTS CDMA) Networks with both Internet (VoIP) communication and PSTN Networks. The Device may be used with an analog or IP PBX - in each case it increases the capabilities connection in a cost efficient manner. The user benefits lower day-to-day telecom bills by routing the call via the least expensive route - VoIP, Cellular or PSTN Network.

This innovative VoIP²ALL™ product line is modular, flexible and compatible to the current and future communication needs of small to large-scale business.

It simply enables telephone communication at lower cost - inside and out of the organization. The design features AudioCodes renowned, high quality VoIP chip.

VoIP²ALL™ grants flexibility in channel allocation to incoming and outgoing calls, between cellular / Internet / landlines from within the same unit. VoIP²ALL™ is able to simultaneously work and monitor each individual network protocol with a powerful management program and user-friendly interface software that you can remotely manage or configure the in real time, from any global location.

How You Save On Your Telecom Bill

- VoIP²ALL™ reduces costs by choosing to route the telephone call via the least expensive network to any VoIP / Cellular / PSTN Network by means of call forwarding / call back / follow me features.
- IP connection: Your organizations IP Gateway will enable cellular calls through its network.
- Call Back option: Allow traveling employees to call through your organizations network at your known local low cost.
- ‘Follow-Me’ feature – No more lost calls: If a number is unreachable the Gateway will direct the incoming call to any open number.
- Agility & Reliability of the Gateway: If a landline call fails, it will pass direct to the cellular network. Mobile connection problems will be handled via the PSTN network.

Call Management Features	
Call Routing	Block all incoming calls on a port (GSM) All incoming calls receive a dial tone, then the user may dial the destination number using DTMF, the calls are routed according to user defined prefix groups. Incoming calls are routed to destination number automatically, options: 1. Fixed destination number for each incoming port. 2. Destination number according to user-defined list, selection cyclic. 3. Destination number according to user-defined list, selection according to priority.
Routing Groups	The user can define multi-prefix options for each port. The user can dedicate a ports utilizing the same prefix - the system will route the call according to free port selection. The user can define default port/s that will be used if no other port prefix is defined.
Internal Users Database	Each user has capabilities definitions that define the unit handling of his calls.
CDR	Supports internal CDR file up to 400KB. The CDR file is downloaded using FTP – optional.
DISA	Holds list of extensions (VOIP) for automatic direct routing of incoming call – optional.
Call Back	Available - Option
Tone Definitions	Home feeling, the user can choose its own country call progress tones definitions.
SIP Client	Support of register and route calls to other SIP servers (can connect up to 10 V2G together, asterisk, other SIP providers).
Call Forward Features *(In Development)*	Support of CFU, CFB, CFNR for the internal VOIP users.
Conference Call *(In Development)*	Up to three (3) participants/
Fax *(In Development)*	Able to transfer facsimile (fax) data.
SMS	Send & Receive SMS using e-mail.

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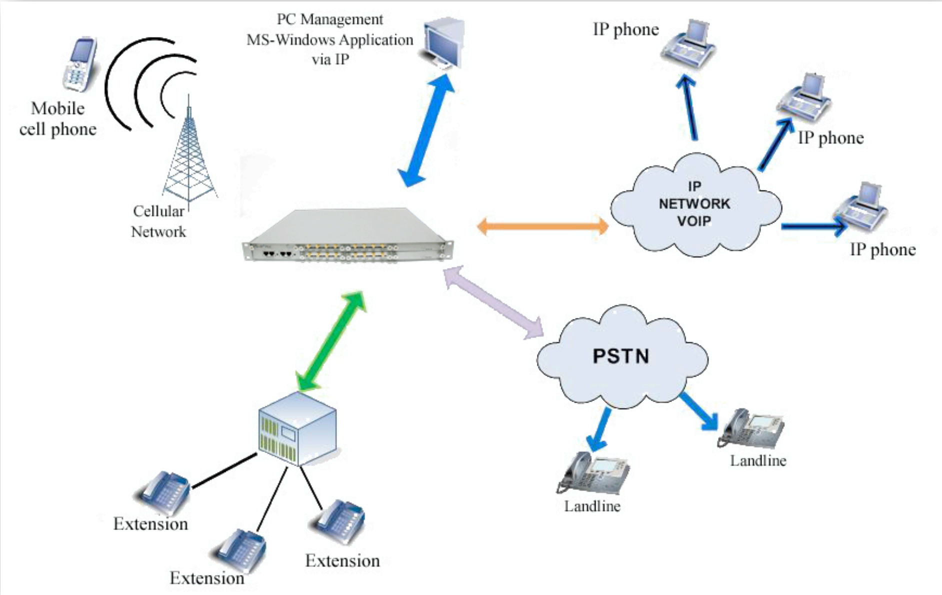
GSM UMTS CDMA



TARGET USERS

- Corporate - SMB, & SOHO using IP
- Integrators - VARs for PBX upgrading to VoIP or GSM
- Telecom Equipment Distributors
- Telecommunication Service Firms
- Companies with International branches

VoIP Parameters	
Voice channels	8 / 16 / 24 Simultaneous
Codecs	G.711 PCMA/U , G.729A , G.723 , G.726 , G.727
Signaling	SIP – RFC 3261
Echo cancellation	G.168-2002
SIP account	Management with Authentication
SIP Server	Up to 32 SIP clients
Interfaces	
Internal SIM Server	Up to 64 Additional SIMs (Optional)
24 Channel	1 SIM per Channel
LAN	RJ-45
Administration	
User Management program via LAN interface	
Includes version update capabilities for firmware or management Software	
Other	
Dimensions	Metric: 420 x 360 x 45 mm (1U)
Weight	4.5 kg (9.92 lbs.)
Main Power	
Power Supply	100 - 240V AC, 50 - 60MHz
Power Input	Max - 230 VA

GSM Parameters	
GSM channels	8 / 16 / 24 Channels
Network types	850 / 900 / 1800 / 1900 MHz (quad-band)
GSM engine	Wavecom (P5186), SIMCom, Siemens (TC35i)
Transmitter power	+33dBm(2W) 850/900MHz,+30dBm(1W) 1800/1900MHz
SIM card	1 SIM per channel, Small plug-in, 3V
Antenna connector	SMA (female), Impedance 50Ω
UMTS Parameters	
UMTS channels	8 / 16 / 24
Network types	UMTS 2100MHz, GSM 850 / 900 / 1800 / 1900
UMTS engine	SIMCom
USIM	1 USIM per channel, Small plug-in, 3V
Antenna connector	SMA (female), Impedance 50Ω
CDMA Parameters	
CDMA channels	8 / 16 / 24
Network types	800/1900 MHz
CDMA engine	Wavecom, AnyDATA
R-UIM Card	1 R-UIM per channel, Small Plug in, 3V
Antenna Connector	SMA (female), Impedance 50Ω

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