GSM VOIP GATEWAY

GB 400 010 with GSM module Two-way converter between VoIP and GSM



User Guide



Dear customers,

Congratulations on purchasing our product - GSM Gateway GB 400 010. You have acquired high quality and highly professional product.

GSM gateway GB 400 010 enables to save costs on phone calls by connecting call from your VoIP device (VoIP PBX or VoIP phone) to the GSM network and vice versa – to enable mobile number calls to be connected from GSM to VoIP.

The most significant savings with VoIP GSM gateway GB 400 010 can be achieved especially:

- By purchasing a SIM with a flat rate or very low calling rate between numbers inside a group.

- For international calls, where you can cross the borders with free call over VoIP, you will only pay the national charge of GSM to GSM calls. This can be very useful especially for companies with branches abroad.

Please read this User Guide carefully before putting the gateway into operation. It provides not only the information about correct installation and operation of this product but it will also help you use all the technical features.

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1. Introduction

GB 400 series products provide you the best connect solution for heterogeneous network including WLAN GSM or PSTN. You may use a SIP-protocol VoIP phone or software to connect to the GB 400, then reach this call to the mobile network, and vice versa. With multiple sets of GB 400, you may even build an international call network.

2. Functions

- VoIP (SIP)-GSM conversion.
- Voice response for setting and status enquiring. (Dial in GSM numbers of GB 400 to get voice information or to operate.)
- 50 sets of LAN->MOBILE routing, and 50 sets of MOBILE->LAN routing.
- Series connections to save bills.
- Standard SIP (RFC2543, RFC3261) protocol to communicate with other gateways or PC.
- Settings and managing via web page

3. The contents in package

- GB 400 gateway
- AC-DC Adaptor (220V AC 12V DC)
- Network cable
- Antenna
- User's Manual

4. Panel description

- PWR: Power indicator light. Light is on when system's power supply is normal.
- GSM: GSM indicator light. Light flashes when GSM status is normal; light turns on constantly when GSM is called.
- LAN: LAN indicator light. Light flashes when Lan is called; light turns off when GSM answers.
- LINK: Link indicator light. Light is on when network is connected correctly.

5. Accessory attachment

- Connect the network cable both to your Hub and to LAN socket of GB 400.
- Connect the antenna and place it in a good receiving location (not too close to the device).
- Insert a SIM card into left bank of GB 400 and close the cover.

- Plug the adapter in DC 12V socket and PWR socket. The PWR light should turn green at the moment.
- Click reset button 3 sec. GB 400 will restore default IP. Other setting as usual.

6. Setting and managing via web page

The default IP address of GB 400 is http://192.168.0.100. Before accessing the web page, please confirm this address is available in your network.

Login VoIP				
Enter your username and password to login VoIP server				
Username				
Password				
	Login Clear			

Enter the default username and password to login. Default username: voip Default password: 1234

7. System Information

After login, you could see the system information such as: model name, firmware version, codec version name, etc. in this page.

LEVEL®		GSM Vol This page illustrate the	P v6.0 system related information.	
Route	•			
Mahila		Company:	LEVEL Ltd.	
IVIO011e	•	Model Name:	GB 400 010	
Notrioul-		Firmware Version:	Mon Nov 12 15:17:30 2007.	
Ivetwork	•	Codec Version:	Mon Jul 24 10:55:05 2006.	
CTD Cattin an		Tel:	+420 491 44 66 11	
our settings	•	Fax:	+420 491 44 66 44	
NTATE T		Web Site:	http://www.levelna.cz	
NAT Trans.	•			

You could also see the setting table in the left side. Please click on the option you would like to set. The setting methods are indicated as the following chapters, please input the value or select the item according to your situation.

Note:

Please remember to save change whenever you submit any setting. Click "Save Change" then "Save" button, the system will restart and make the changed function/setting operative.

8. Route

8.1. Mobile to LAN Settings

In this page: Mobile To Lan Table, you could set the routing rules to transfer the calls incoming from GSM to LAN. Maximum 50 sets.



When the GSM number of the GB 400 is called, this device transfers the call to URL according to the caller ID of the incoming call.

8.1.1 CID: caller ID, the numbers of incoming call

You could set the CID as the following formats:

- The complete number, e.g. 0911111111
- The prefix part plus *, e.g. 0911*. This format means any number starting with 0911 will be accepted to transfer.
- *, this means any incoming call is accepted to transfer.
- N, this means the incoming call without showing its CID is accepted to transfer.

Please note the priority of the routing rules; the CID with more digits gets the priority.

8.1.2 URL The IP address of destination

You could set the URL as the following formats:

- The complete IP address, e.g. 192.168.0.101
- The proxy extension numbers
- The phone numbers.

Note:

If the device has registered at proxy server/Asterisk, you can enter any destination phone number. Also note that in the proxy server/Asterisk, you need to set the route of destination phone number.

Leave it blank or 'N', this mean to refuse to transfer.

*, this means to transfer via 2-stage-dialing. The call will be answered with a prompt dial tone for the caller to press the IP address, proxy extension, or any phone number as destination. The caller press the IP address on the phone keys: 192*168*0*101# as 192.168.0.101

Example of Mobile to Lan setting:

• Mobile to Lan: 0932*, 0911123456

When the GSM numbers of the device is called, if the caller's prefix numbers are 0932, GB 400 transfers the call to 0911123456, then 0911123456 rings (while available).

Precondition:

- a) GB 400 has registered at proxy server/Asterisk
- b). The proxy server/Asterisk has the route of "09"
- Mobile to Lan: *, *

Any incoming call gets a prompt dial tone; so the caller can enter any IP address, sip extension, or phone number.

Precondition:

a) SIP extension or phone number needs to register at SIP Proxy Server or Asterisk.

b) Phone number, SIP Proxy Server or Asterisk needs to set the route of destination phone number.

8.2. Mobile to LAN Speed Dial Settings

When you set both Mobile to LAN Speed Dial Settings and Mobile to LAN settings at the same time, Mobile to LAN Speed Dial Settings gets higher priority. Mobile to Lan setting will be not available.

LEVE	R	Mobil	le To]	LAN Speed Dia	al	
		You could s	et the speed	dial in this page.		
Route	•					
Mobile		Num	Name	URL	Select	
	•	0	pokus	491446637		
Network		1				
	- 1 - C	2				
SIP Settings		3				
		4				
NAT Trans.	•	5				
System Auth		6				
system Auth.		7				
Save Change		8				
		9				
Update		Delete Ce				
		Delete Selected Delete All Reset				
Reboot						
		Add New P	hone			
		Position:	n	~9)		
		Name:		-/		
		ivanie.				
		URL:				
		Add F	Reset			

The call is answered with a prompt dial tone for the caller to press the "Num", and then the device connects the "URL" as destination.

Example: after you call the GSM number of the device and hear a dial tone, you press 0, then the lan phone of IP address: 192.168.0.107 rings.

8.3. LAN to Mobile Settings

In this page: Lan To Mobile able, you could set the routing rules to transfer the calls incoming from Lan to Mobile. Maximum 50 sets.

	LAN To Mobile Table			
Route	Page: 1 💌			
Mobile	ltem URL	Call Num Select		
Network	0 * 1	*		
SIP Settings	2			
NAT Trans.	4			
System Auth.				
Save Change	7			
Update	9			
Reboot	Delete Selected Delete All Reset			
	Add New			
	Position:	(0~49)		
	URL: Ex:192.168.0.1, 192.168.0.* Call Ex:0911, *:2St, #, #d?, #d?A??:19			
	Add Reset			

When the Lan of the GB 400 is called, this device transfers the call to Call Num according to the URL of the incoming call.

8.3.1. URL: The IP address or proxy extension numbers of the incom. call.

You could set the URL as the following formats:

- The complete IP address, e.g. 192.168.0.101
- The proxy extension numbers, e.g. 103

- Part of an IP address plus *, e.g. 192.168.0.*. This means the IP address starting with 192.168.0 would be accepted to transfer,
- Part of the proxy extension numbers plus, e.g. 10*. This means the extension numbers starting with 10 would be accepted to transfer.

8.3.2. Call Num: the phone numbers of destination.

You could set the Call Num as the following formats:

- The complete number, e.g. 091111111
- *, this means to transfer via 2-stage-dialing. The call will be answered with a prompt dial tone for the caller to press the destination phone numbers, e.g. 0911111111
- *#*, this allow the caller with lan phone dial directly the destination numbers. Precondition:
 - a) GB 400 and incoming lan Phone are both registered at proxy server or Asterisk.
 - b) Proxy server/asterisk has set the routing rules to assign specific prefix of numbers to be transferred from GB 400.
 - c) Lan to Mobile routing sets: *, # Usage: You could dial on your lan phone call any destination number with prefix of "09". When your lan phone and GB 400 had registered and "09" prefix is setted the routing rules at proxy server or Asterisk.
 - d) #['d'n]['a'ppp], this means to do the above routing, and to modify the numbers.

Note:

'd'n means to delete the number of prefix,

'a'ppp means to add 'ppp' prefix.

E.g. #d2a09 means to call the registered numbers via one-stage-dialing. The numbers are modified to: delete 2 digits of prefix of the original numbers, then add 09 to be new prefix of the destination numbers.

9. Mobile

9.1. Mobile Status

In this page: Mobile Status, you could get the information of your GSM network and the latest operation.

LEVE	R	Mobile Stat	us
Route	•	Network Registration.:	
Mobile		SIM Card ID:	
	· ·	Signal Quality.:	99
Network	•	GSM S/N:	IMEI: 35815600702821-8
SIP Settings	•	Incoming IP:	
NAT Trans.		Incoming IP Name:	
		Outgoing IP:	
System Auth.		Incoming Mob:	
Save Change		Outgoing Mob:	

- Network Registration: The telecom carrier which the SIM card been registered.
- SIM Card ID: SIM card ID.
- Signal Quality: Signal quality.
- GSM S/N : IMEI Number
- Incoming IP: The IP address of the last incoming call from LAN.
- Incoming IP: Name: proxy server name
- Outgoing IP: The IP address of the last outgoing call to LAN.
- Incoming Mob: The caller ID of the last incoming call from MOBILE.
- Outgoing Mob: The called number of the last outgoing call to MOBILE.

9.2. Mobile Setting

In this page: Mobile setting, you could adjust the parameter and click on the option to fit your need. You could leave those default value before you had tried the complete operation of this device.

LEV	E	Mobile Se	etting		
		You could set the volume of your phone in this page.			
Route	•	VolP Tx Gain:	9 (II~12)	VolP Bx Gain:	11 (0~15)
Mobile	•	LAN Dialtone Gain:	9 (0~12)		
Network	•	Mobile ON	• OFF		
SIP Settings		CODEC Tx Gain:	6 (0~7)	CODEC Rx Gain:	6 (0~7)
NTATE TELESCO		SIP From: Presentation CLIP	Tel/User(Standard) Suppression	Tel/Tel	O User/Tel
NAI Irans.	•	Mobile PIN Code:	On D	Code:	Confirmed:
System Auth.		LAN Answer Mode	 Answered 	 Alerted 	Income
Save Change					
Update	•		Submit	Reset	
	VoIP	Rx Tx Co	dec	GSM	GSM →

- LAN Dialtone Gain: DTMF Reciver is not good, you can adjust gain down.
- ON/Off: If you use this channel, please click on. Otherwise, please click off.
- CODEC Tx Gain: as above

VoIP Rx Gain

- CODEC Rx Gain: as above
- SIP From: Caller ID transfer
 - Tel/User(Standard): If you need to register to Asterisk and proxy server, please choose this option. GB 400 will send the message as follows in the Packet.

DTMF

From: " caller number " <sip:3001@192.168.0.228>;tag=51088abb

• Tel/Tel:GB 400 will send the message as follows in the Packet. From: "caller number" <sip: caller number @192.168.0.228>;tag=6ac93f7c Please note: If you choose this option, please don't register to Asterisk and proxy server. Please only fill proxy server ip and choose Active: on (else field empty) in sip setting/service demain

- User/Tel GB 400 will send the message as follows in the Packet.
 From: "Username " <sip: caller number @192.168.0.228>;tag=7f130947
 If you choose this option, please don't register to Asterisk and proxy server. Please only fill proxy server ip, Username and choose Active: on (else field empty) in sip setting/service demain
- Presentation CLIR : If you need to block the Caller Id for call termination, please choose Suppression
- Mobile PIN Code: If you need to unlock pin code via GB 400, you can click "On" and enter pin code.
- LAN Answer Mode:
 - $\circ\;$ Answered: when mobile answer, then connect the call
 - o Alerted: when the mobile is ringing back tone, then connect the call
 - o Income: when Ian dial out, then connect soon

9.3. Forward Setting

When the first route are busying, SIP can transfer phone call to another free route. When the device are busying, the phone call can be transfer to another device (external equipments).

LEVEL®	Forward Setting	
Route		_
Mobile	Forward Enable	
Network	Name URL:Port	
SIP Settings	Fwd to External:	
NAT Trans.	submit cancel	

* "Forward Enable" is not motivate on Defualt value.

So please, mark "Forward Enable" this blank to motivate this function.

Take SJ Phone for example: Profiles -> Edit -> Advanced -> Accept redirection replies (Turn on the "Forward Enable", therefore the SJ Phone can designate a port which are free to use.)

Profile Options	K			
General Advanced DTMF STUN				
Accept redirection replies				
Use short headers				
Expose software version				
Use obsolete transfer mechanism (BYE/Also)				
Restrict caller identity (support varies for proxies from different vendors)				
Use "standard" status messages (otherwise messages will be taken from SIP packets)				
Voice mail number or address:				
Remove fancy characters from phone numbers				

10. Network

In Network, you could check the Network status; configure the WLAN Settings, LAN Settings and SNTP settings.

10.1. Network Status

Information of current Network in this page.

LEVE	®
Route	•
Mobile	•
Network	•
SIP Settings	•

Network Status

This page shows current status of network interfaces of the system.

Ethernet 0				
Interface				
Туре:	Fixed IP Client			
IP:	192.168.1.165			
Mask:	255.255.255.0			
Gateway:	192.168.1.1			

10.2. WAN Settings

You can check the current Network setting in this page.

The default IP is 192.168.0.100; you could change it to any available IP address, or select different IP type to suit your environment.

	WAN Settings		
	You could configure the WAN settings in this page.		
Route			
Mobile	WAN Setting		
· · · · · · · · · · · · · · · · · · ·	IP Type:	Fixed IP O DHCP Client O PPPoE	
Network	IP:	192.168.1.165	
	Mask:	255.255.255.0	
SIP Settings	Gateway:	192.168.1.1	
NAT Trans.	DNS Server1:	195.146.100.5	
•	DNS Server2:	195.146.100.100	
System Auth.	MAC:	00037e001236	
Save Change	PPPoE Setting		
The Jacks	User Name:		
opuate	Password:		
Reboot			
		Submit Reset	

- Fixed IP: the TCP/IP Configuration item is to setup the WAN port's network environment. You may refer to your current network environment to configure the system properly.
- DHCP client: you could refer to your current network environment to configure the system properly
- PPPoE: If you have the PPPoE account from your Service Provider, please input the Username and the Password correctly.

After you input or modify the value, click the Submit button.

10.3. SNTP Settings

You could select "On" to give SNTP function to this device.

Input the primary and secondary IP Address of SNTP Server to get the date/time information. Also you could set the Time Zone according to your location; and set the time to synchronize. After setting, remember to click the Submit button.

LEVEL®		SNTP Settings You could set the SNTP servers in this page.		
Route	•	SNTP		
Mobile		5		
Network		Primary Server:	time.windows.com	
		Secondary Server:	208.184.49.9	
SIP Settings				
		Time Zone:	GMT 🕂 🕶 08 💌 : 00 💌 (hh:mm)	
NAT Trans.	•	Sync. Time:	1 : 0 : 0 (dd:hh:mm)	
System Auth.			Submit Reset	

11. SIP Setting

If you need, you could setup the Service Domain, Port Settings, Codec Settings, RTP setting, RPort Setting and Other Settings.

If ISP provides the VoIP service, you need to input the related information correctly to register at SIP Proxy Server.

11.1. Service Domain

In this page, you should input the data refer to your ISP. Maximum is 3 accounts (Realm 1 to 3). You could dial out via first SIP account, and receive via the three SIP accounts.

LEVE	®	Service Domain Settings You could set information of service domains in this page.		
Route	•	Realm 1 (Default)		
Mobile		Active:	💿 ON 🔘 OFF	
	- F	Display Name:		
Network	•	User Name:	491110104	
CTD Catting		Register Name:	491110104	
off bettings	•	Register Password:	kolokiok	
NAT Trans.		Domain Server:	212.71.146.177	
System Auth.		Proxy Server:	212.71.146.177	
		Outbound Proxy:		
Save Change		Status:	Registered	

- Active: click "On" to enable the function in Service Domain, then input the following items.
- Display name: input the name you would like to display.
- User name: input your user name in ISP.
- Register Name: input your register name in ISP.
- Register Password: input your password in ISP.
- Domain Server: input the Domain Server IP address.
- Proxy Server: input the Proxy Server IP address.
- Outbound Proxy: input the Outbound Proxy IP address. If your ISP does not provide the information, you could skip this item.
- After setting, click the Submit button. Remember to click "Save Charge"
- You can see the Register Status in the Status item.

11.2. Port Setting

You can setup the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTP port setting, please refer to the ISP to setup the port number correctly. After setting, remember to click the Submit button.



11.3. Codec Settings

You can setup the Codec priority, RTP packet length in this page. You need to follow the ISP suggestion to setup these items. After setting, remember to click the Submit button.

	Codec Set	ttings
	You could set the co	dec settings in this page.
Route		
Mobile	Codec Priority	
Network	Codec Priority 1: Codec Priority 2:	G.711 a-law
SIP Settings	Codec Priority 3:	G.723 💌
NAT Trons	Codec Priority 4:	G.729 V G.726-16 V
	Codec Priority 6:	G.726 - 24 💌
System Auth.	Codec Priority 7:	G.726 - 32 💌
Save Change	Codec Priority 8: RTP Packet Length	G.726 - 40 💌
Update 🕨	G.711 & G.729:	20 ms 💌
Reboot	G.723:	30 ms 💌
	G.723 5.3K	
	G.723 5.3K:	On ⊙Off
	Voice VAD	
	VOICE VAD.	
		Submit Reset

11.4. Codec ID Setting

You can setup the Codec ID in this page.

LEVE	®	Codec ID Setting You could set the value of Codec ID in this page.			
Route	•				
Mobile		Codec Type	ID	Defi	ault Value
	- *	G726-16 ID:	23 (95~2	255) 🔽	23
Network	•	G726-24 ID:	22 (95~2	255) 🔽	22
STD Cottings		G726-32 ID:	2 (95~2	255) 🔽	2
or seconds	•	G726-40 ID:	21 (95~2	255) 🔽	21
NAT Trans.		RFC 2833 ID:	101 (95~2	255) 🔽	101
System Auth.			Submit	Reset	

11.5. DTMF Setting

You can setup the DTMF Setting in this page.

LEVE	®	DTMF Setting You could set the DTMF setting in this page.		
Route	•			
Mobile	•	○ 2833		
		Inband DTMF		
Network	•	○ Send DTMF SIP Info		
SIP Settings	•	Mobile DTMF debounce: <mark>80 (</mark> (range:40~200, default:80) step:10ms.		
NAT Trans.	•	Submit Reset		

Note:

If this device has registered at SIP Proxy Server/Asterisk, please select "2833". If not, please select "Inband DTMF".

11.6. RPort Setting

You can setup the RPort Enable/Disable according to your ISP information. After setting, remember to click the Submit button.

	RPort Setting You could enable/disable the RPort setting in this page.	
Route		
Mobile	RPort of Mobile : 💿 On 🔘 Off	
Network	Submit Reset	

11.7. SIP Setting: SIP Responses

	SIP Responses Setting You could set the Lan to Mobile sip message in this page.		
Route			
Mohile		Response on port busy.	
		Busy here	
Network	0 503	Service unavailable	
SIP Settings		SIP Responses	
-	💿 ON 🔘 OFF	180 Ringing (Auto force to ON, if 183 was OFF.)	
NAT Trans.	○ ON ⊙ OFF	183 Session Progress	
System Auth.		Submit	

- 486(busy here), 503(Service unavailable): When Device are busying, you can select 486 or 505 to response to SIP.
- 180 Ring on/off: LAN TO MOBILE two stage dialing can be turn off, therefore there will be no the Ring Back Tone, all the phone call will be transferred to Voice-Mail directly. (For this function, 183 must be turn on)
- 183(Session Progress)-->[It means"on progressing"] : When you turn 183 on, it means you can hear voicemail while GMS side are busying. We recommend you to turn this on if you use SIP Proxy.

11.8. Other Settings

You could setup the RFC and QoS according to your ISP information. After setting, remember to click the Submit button.

The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets get the higher priority to the Internet. But the QoS function still need to cooperate with the others Internet devices.

LEVEL®	Other S	Other Settings You could set other settings in this page.		
Route				
Mobile	Hold by RFC o	f Mobile : 🛛 🔘 Or	n 💿 Off	
Network	Voice QoS:	40	(0~63)	
SIP Settings	SIP QoS:	40	(0~63)	
· · · ·	SIP Expire Tim	ne: 300	(60~86400 sec)	
NAT Trans.		Sub	mit Reset	

12. NAT Trans

In this page: NAT Trans./ STUN, you could setup the STUN Enable/Disable and STUN Server IP address. This function helps your VoIP device work properly behind NAT. Change these settings according to your ISP information. After setting, remember to click the Submit button.

	STUN Setting		
	You could set the IP	of STUN server in this page.	
Route			
Mobile	STUN of Mobile :	◯ On ④ Off	
Network	STUN Server:	stun.xten.com	
	STUN Port:	3478 (1024~65535)	
SIP Settings			
NAT Trans.		Submit Reset	

In this page: System Authority, you could change your login name and password.

	System Au	System Authority		
	You could change the	You could change the login username/password in this page.		
Route				
	New username:			
Mobile	New password:			
Network	Confirmed password:			
	•	Submit Reset		
SIP Settings	•			

14. Save Change

Please remember this step whenever you submit any setting. Click "Save Change" then "Save" button, the system will restart and make the changed function/setting operative.

	Save Changes	
	You have to save changes to effect them.	
Route		
Mobile 🕨	Save Changes: Save	

15. Update

Here you could update the latest firmware and restore the default settings.

15.1. Update Firmware

Download the latest firmware, then Method: select "HTTP" Code Type: select "Risc".

• File Location: Click the "Browse" button in the right side of the File Location for the file.

Please note: no need to unzip the firmware file.

• Click "Update", it takes few minutes to generate new firmware.

LEVEL®	Update Firmware You could update the newest firmware. PCB mark: 2K-123C		
Route			
Mobile	Method: 💿 HTTP 🔘 TFTP		
Network	HTTP		
· · · · · ·	Code Type: Risc 💌		
SIP Settings	File Location: Procházet		
NAT Trans.	ТЕТР		
System Anth	TFTP Server: 192.168.1.250		
Save Change	Update Reset		

15.2. Restore Default Settings

In this page: Update/ Default Settings, you could restore the factory default settings to the system. Click the Restore button, then the system returns to default IP http://192.168.0.100 (the other settings e.g SIP setting, mac address remains), and automatically restart.



16. Reboot

In this page, you could click the Reboot button to restart the system.

LEVEL®	Reboot System
	You could press the reboot button to restart the system.
Route	
Mobile	Reboot system: Reboot

17. Setting and checking via IVR

User could get or set some parameters of the system by dialing in the mobile numbers of the device. The status or result is reported via voice response system. In the first 20 seconds after power-on (when only Mobile light flash), you could dial its mobile numbers. When you hear the dial tone, press the following codes to set or check the device.

Function	Code	Action
Reboot	#195#	Reboot the device
Factory Reset	#198#	Return to default settings
Check IP Address	#120#	IVR announces the current IP address. Default: 192.168.0.100
Check IP Type	#121#	IVR announces DHCP is on or off. Default: off
Check Network Mask	#123#	IVR announces the current network mask. Default: 255.255.255.0
Check Gateway IP Address	#124#	IVR announces the current gateway IP address. Default: 192.168.0.254
Check Primary DNS Server	#125#	IVR announces the current setting in the Primary DNS field. Default: 192.168.0.1
Check Firmware Version	#128#	IVR announces the version of the firmware.
Set as DHCP client	#111#	The system is changed to DHCP Client type
Set Static IP Address	#112xxx*xxx*xxx*xxx#	DHCP is disable and system is changed to static IP type. Enter IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.
Set Network Mask	#113xxx*xxx*xxx*xxx#	Must set Static IP first. Enter value using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.
Set Gateway IP Address	#114xxx*xxx*xxx*xxx#	Must set Static IP first. Enter IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.
Set Primary DNS Server	#115xxx*xxx*xxx*xxx#	Must set Static IP first. Enter IP address using numbers on the telephone keypad. Use the * (star) key when entering a decimal point.

18. Specification

Protocols

SIP (RFC2543, RFC3261)

TCP/IP

IP/TCP/UDP/RTP/RTCP/ CMP/ARP/RARP/SNTP DHCP/DNS Client IEEE802.1P/Q ToS/DiffServ NAT Traversal STUN uPnP IP Assignment Static IP DHCP PPPoE

Codec

G.711 u-Law G.711 a-Law G.723.1 (5.3k) G.723.1 (6.3k) G.729A G.729A/B

Voice Quality

VAD CNG AEC, LEC Packet loss

GSM GB 400

Dual BAND: 900/1800 MHz

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