GSM VOIP GATEWAY

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

User Guide

LEVEL

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.

Content

1. Introduction 5
2. Functions 5
3. The contents in package 5
4. Panel description 5
5. Accessory attachment 5
6. Setting and managing via web page 6
7. System Information 6
8. Route 7
9. Mobile 12
10. Network 16
11. SIP Setting 19
12. NAT Trans 24
13. System Auth. 25
14. Save Change 25
15. Update 25
16. Reboot 26
17. Setting and checking via IVR 27
18. Specification 28

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.

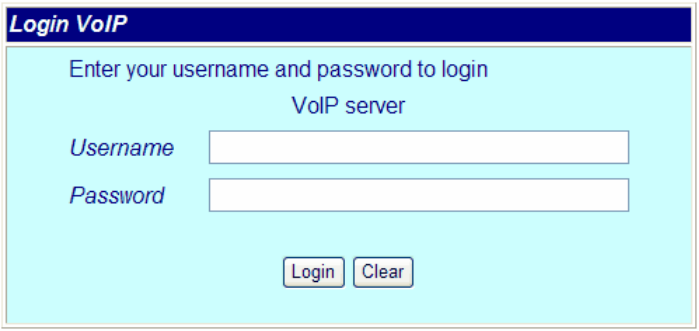
5. Accessory attachment

- 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

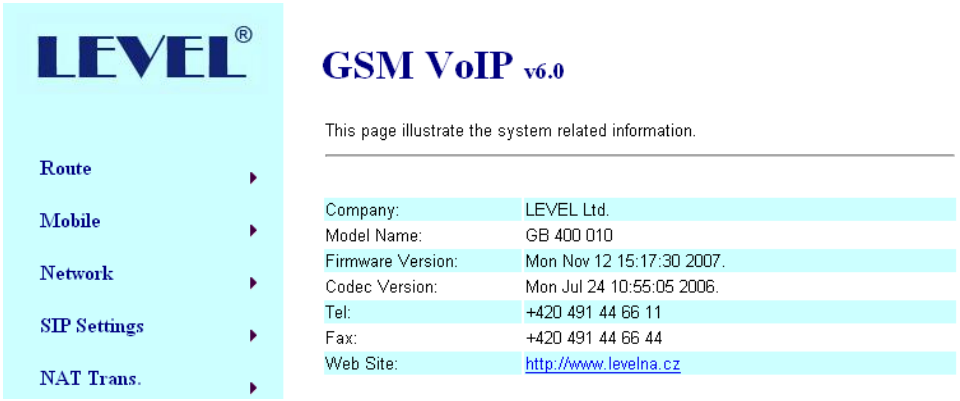
The default IP address of GB 400 is http://192.168.0.100. Before accessing the web page, please confirm this address is available in your network.



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.



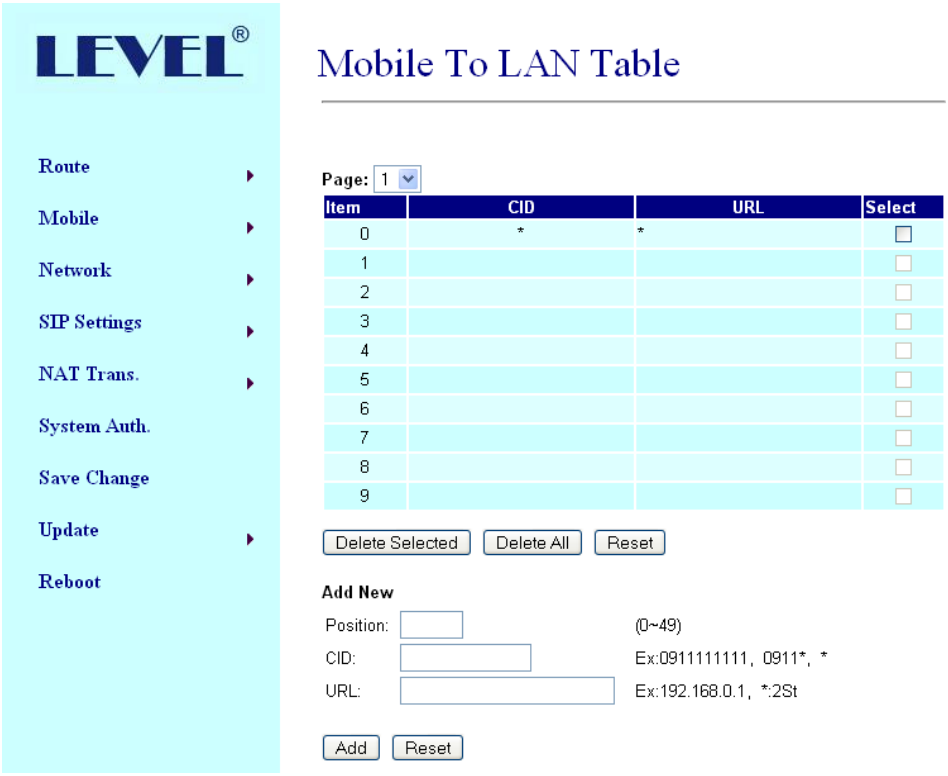
You could also see the setting table in the left side. Please click on the option you would like to set. The setting methods are indicated as the following chapters, please input the value or select the item according to your situation.

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:

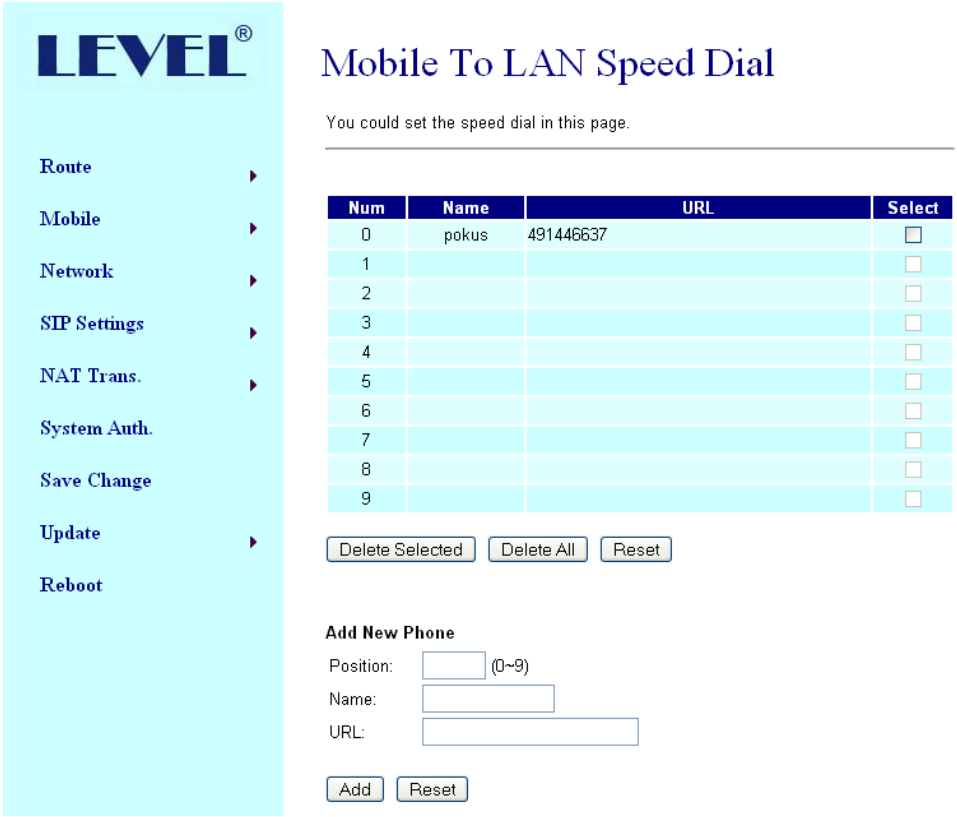
- Mobile to Lan: 0932*, 0911123456
  When the GSM numbers of the device is called, if the caller's prefix numbers are 0932, GB 400 transfers the call to 0911123456, then 0911123456 rings (while available).
  Precondition:
  a) GB 400 has registered at proxy server/Asterisk
  b). The proxy server/Asterisk 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.

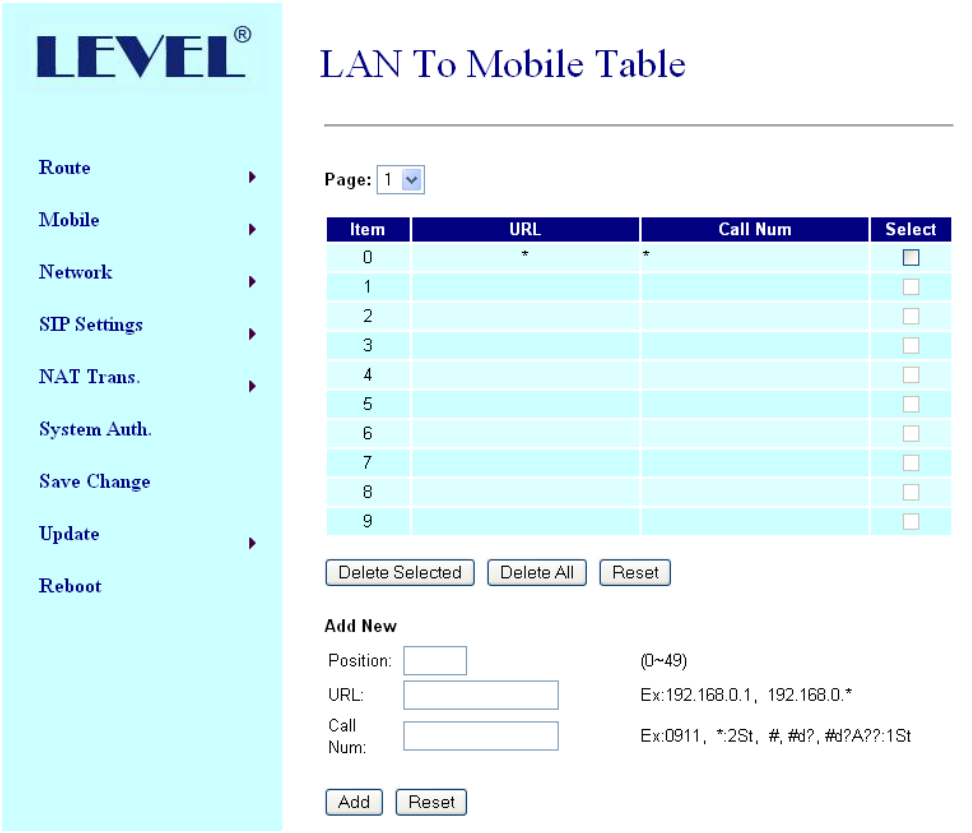


The call is answered with a prompt dial tone for the caller to press the “Num”, and then the device connects the “URL” as destination.

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.



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. 0911111111
- *, 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.

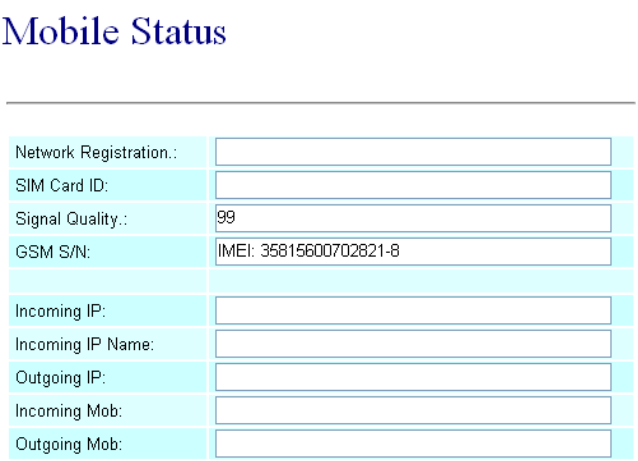
  Note:
  'd'n means to delete the number of prefix,
  'a'ppp means to add 'ppp' prefix.

  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

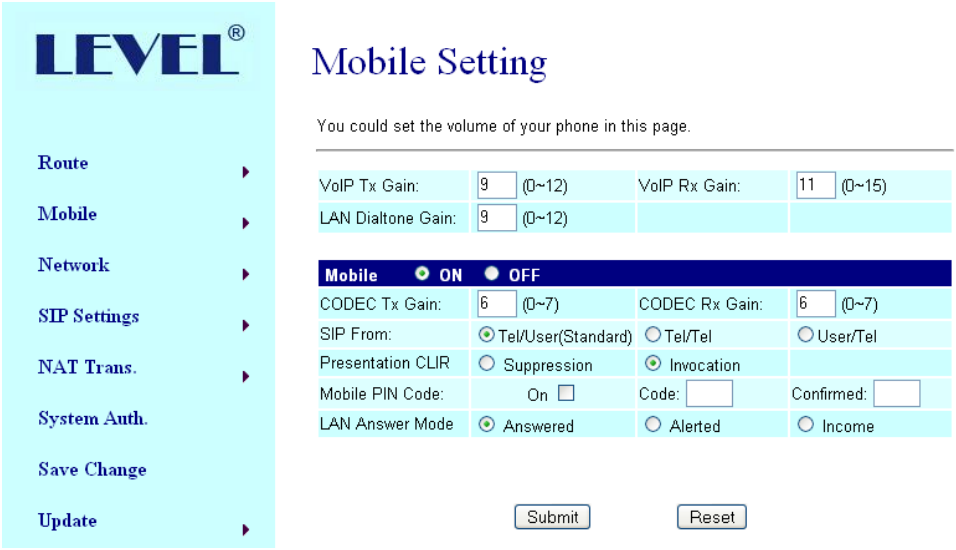
In this page: Mobile Status, you could get the information of your GSM network and the latest operation.

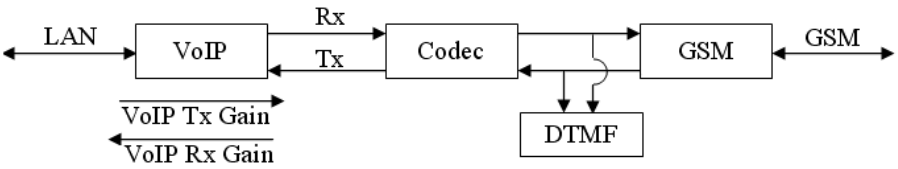


- Network Registration: The telecom carrier which the SIM card been registered.
- SIM Card ID: SIM card ID.
- Signal Quality: Signal quality.
- GSM S/N : IMEI Number
- Incoming IP: The IP address of the last incoming call from LAN.
- Incoming IP: Name: proxy server name
- Outgoing IP: The IP address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





- 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
- 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.
    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

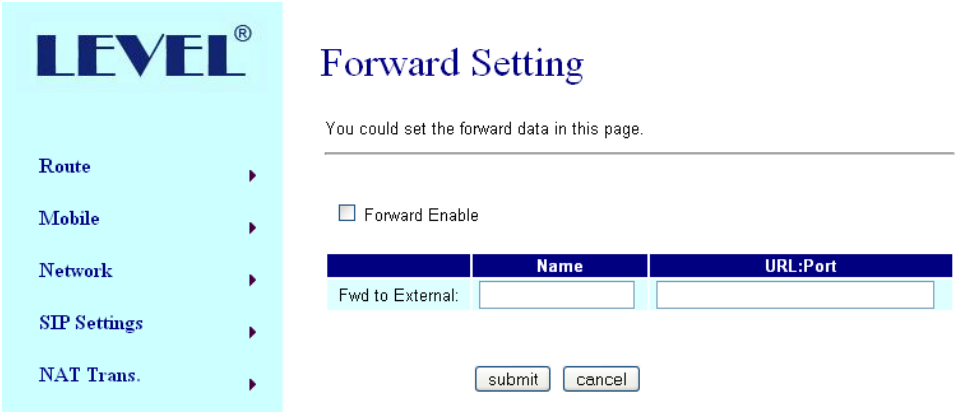
Please note:If you choose this option, please don't register to Asterisk and proxy server. Please only fill proxy server ip and choose Active: on (else field empty) in sip setting/service demain

- 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:
  - Answered: when mobile answer, then connect the call
  - Alerted: when the mobile is ringing back tone, then connect the call
  - Income: when lan dial out, then connect soon

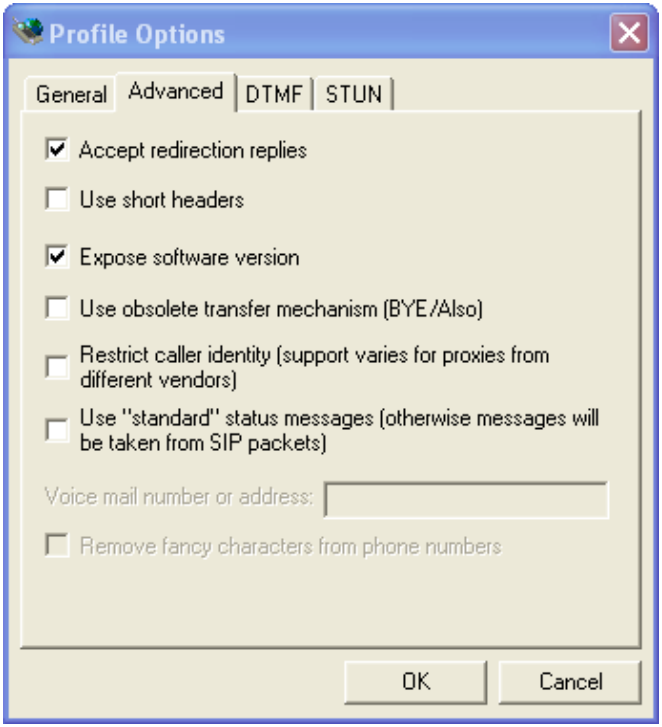
9.3. Forward Setting

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).



* "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.)

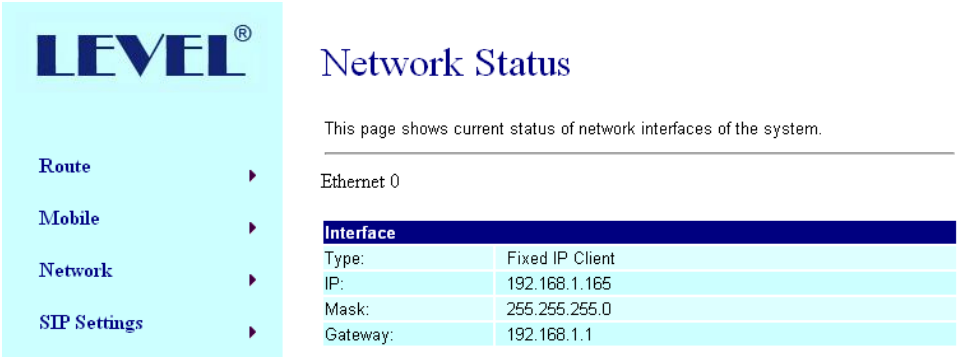


10. Network

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

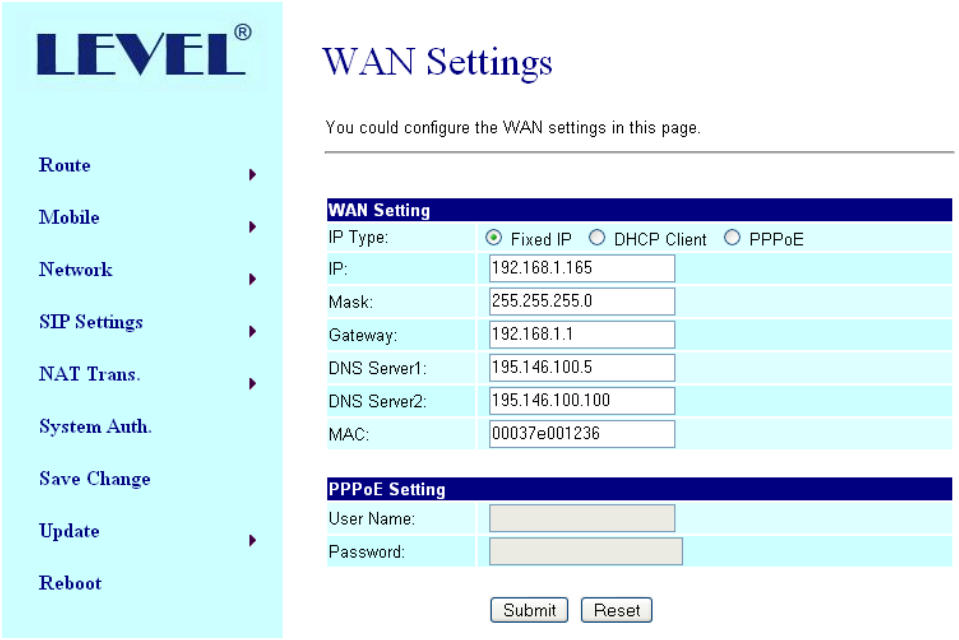
10.1. Network Status

Information of current Network in this page.



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.

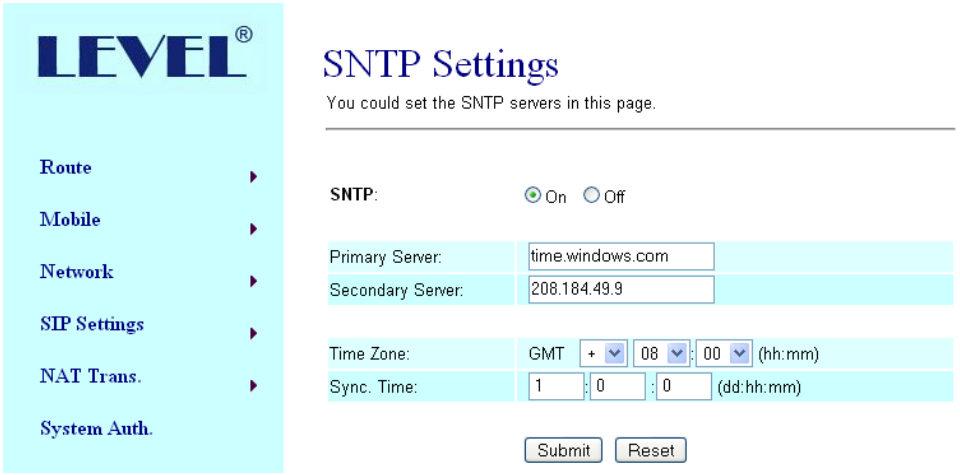


- Fixed IP: the TCP/IP Configuration item is to setup the WAN port's network environment. You may refer to your current network environment to configure the system properly.
- DHCP client: you could refer to your current network environment to configure the system properly
- PPPoE: If you have the PPPoE 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.

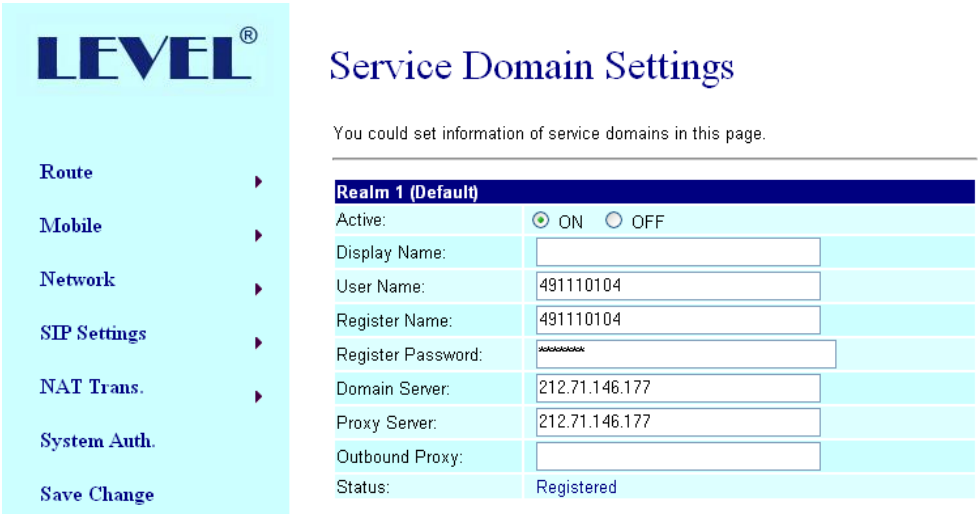


11. SIP Setting

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

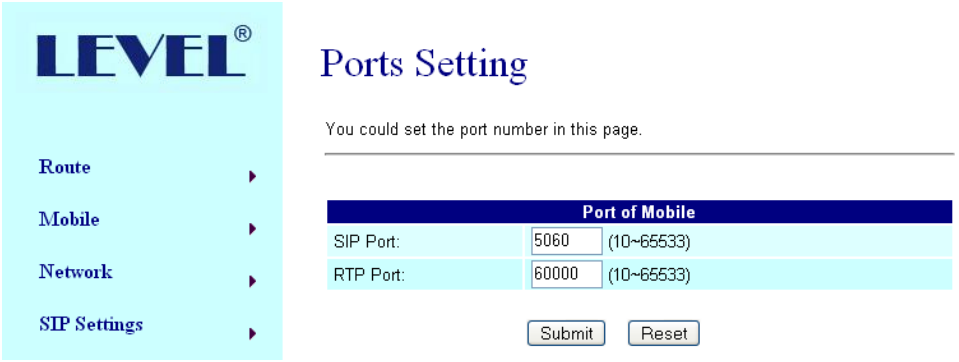
In this page, you should input the data refer to your ISP. Maximum is 3 accounts (Realm 1 to 3). You could dial out via first SIP account, and receive via the three SIP accounts.



- 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.

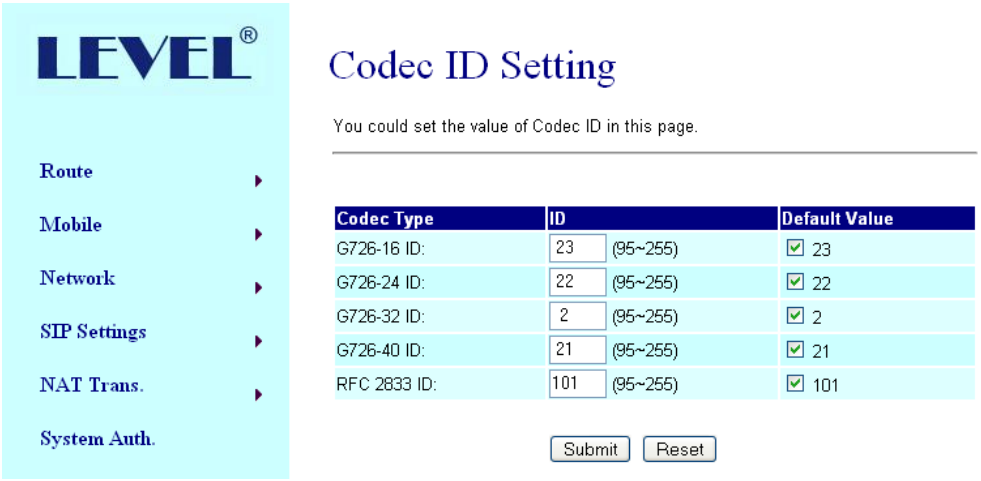


11.3. Codec Settings

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.

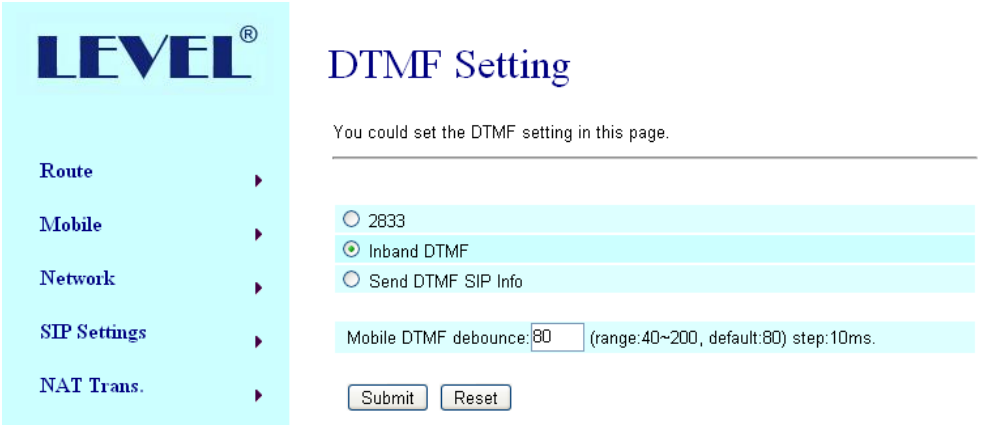
11.4. Codec ID Setting

You can setup the Codec ID in this page.



11.5. DTMF Setting

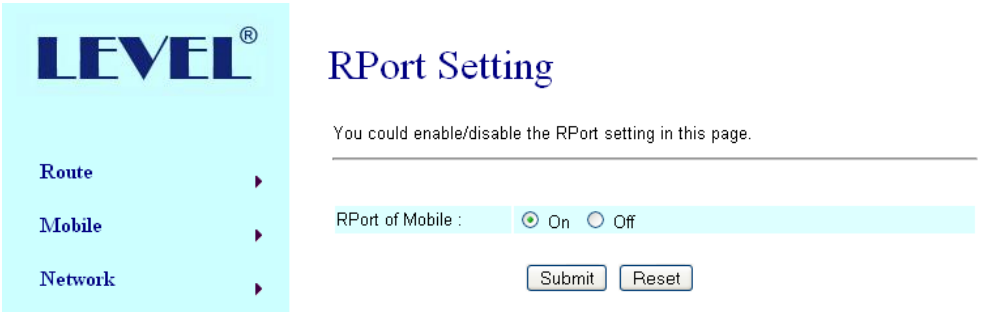
You can setup the DTMF Setting in this page.



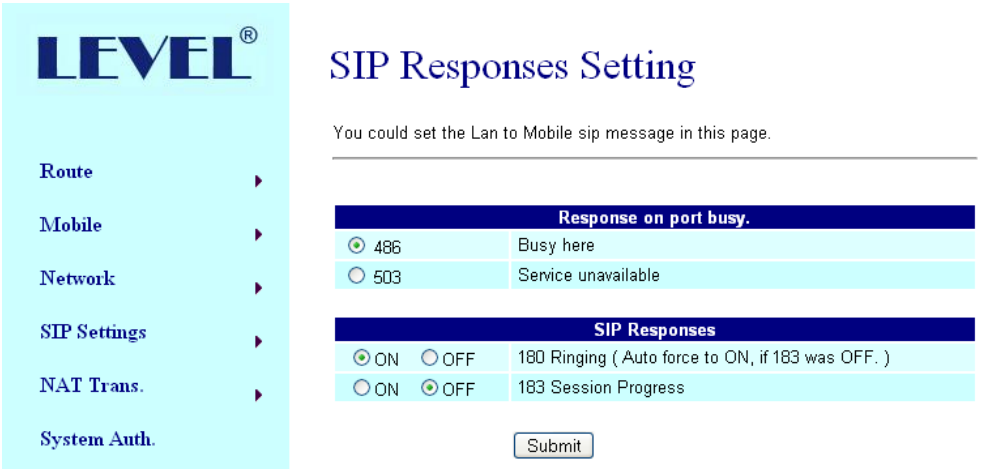
Note:
If this device has registered at SIP Proxy Server/Asterisk, please select “2833”. If not, please select “Inband DTMF”.

11.6. RPort Setting

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



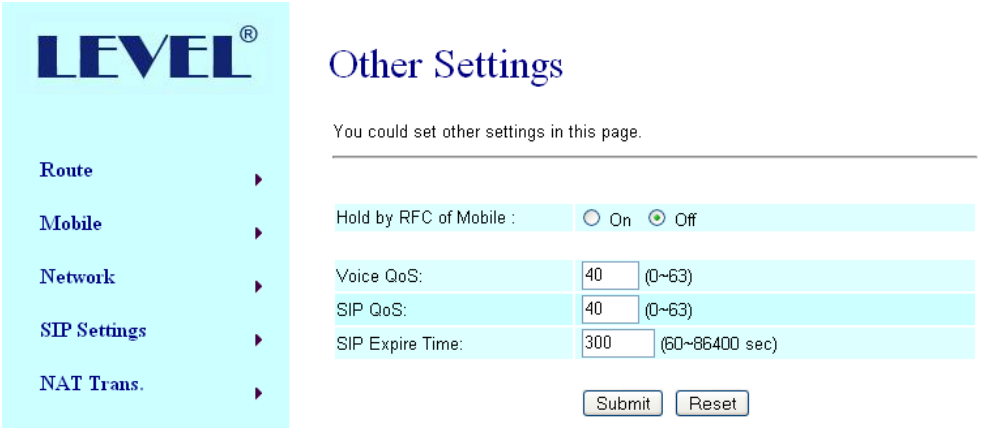
11.7. SIP Setting: SIP Responses



- 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.

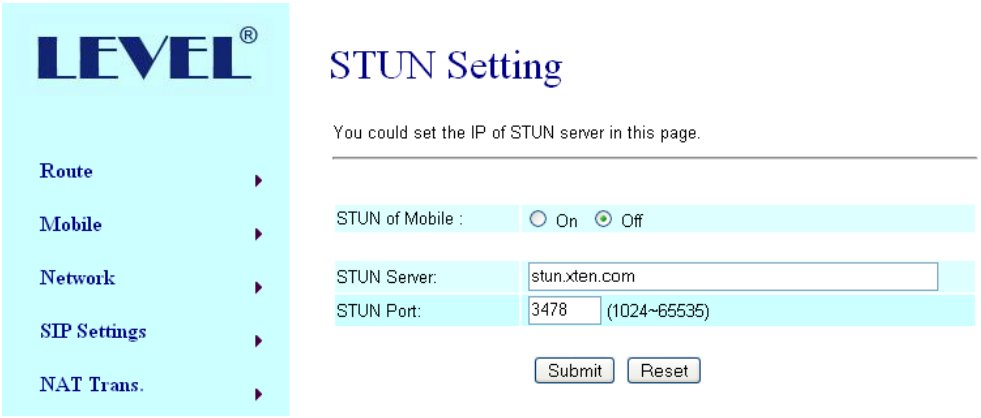
11.8. Other Settings

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.



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.



13. System Auth.

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

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.

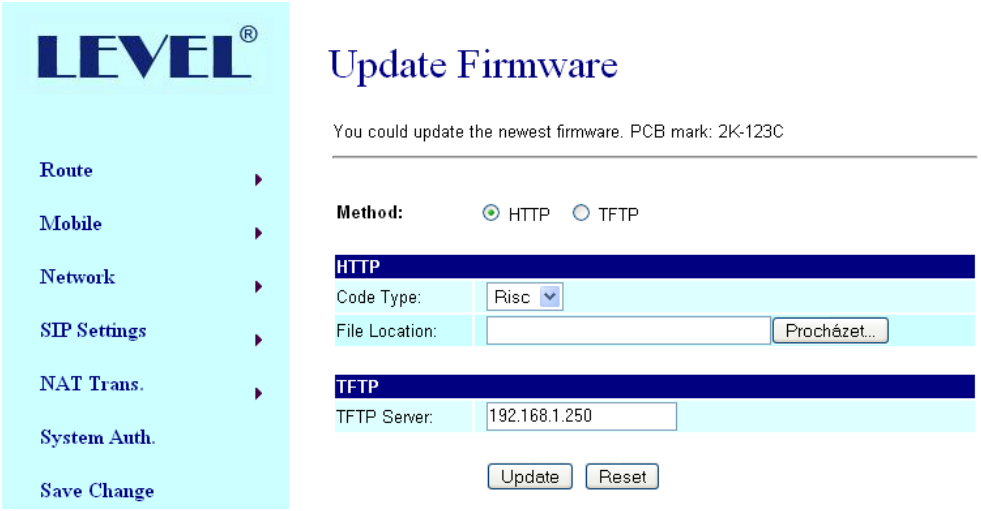
15. Update

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

15.1. Update Firmware

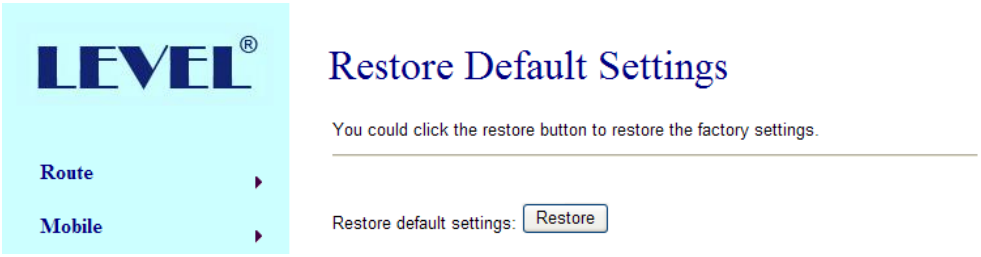
Download the latest firmware, then Method: select “HTTP”
Code Type: select “Risc”.

- 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.



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.

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

GB 400 010, Version 2.00 © LEVEL, 2008
LEVEL Ltd., Plhovská 1997, 547 01 Náchod, Czech Republic
www.levelna.cz