

PG-101 Personal Gateway

User Manual



PG-101

PORTech Communications Inc.

【Content】

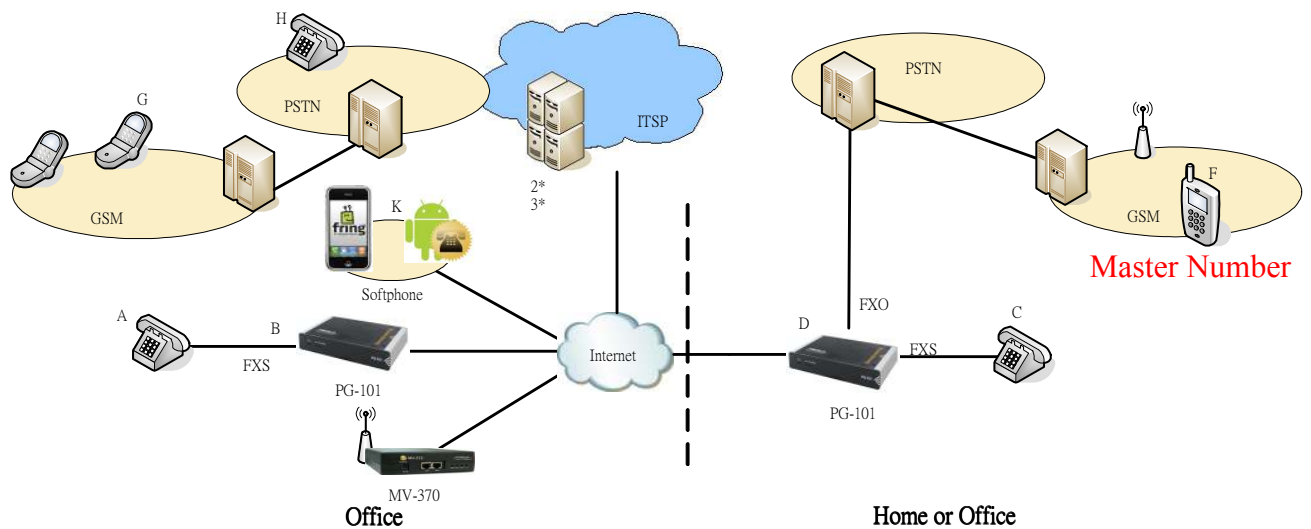
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1. Introduction

PG-101 can be used with soft-phone support SIP, residential VOIP gateway and connect to MV-37X VoIP GSM Gateway. The setting of soft-phone and residential gateway is beyond the scope of this document and user should refer to the user manual of original manufacturer. The related configuration of peer MV is also included. For the detail configuration of MV product, user should refer to MV user manual.

2. System Topology



3. Function Description

1. Incoming call support for VOIP (A-B-D-C or A-B-D-F or K-D-C or K-D-F) and FXO.
2. Outgoing call support for dialing plan with VOIP and local FXO routing.
3. Support concurrent ring(C and F) for VOIP incoming call.
4. Support for caller ID authentication for VOIP outgoing call.
5. Support Call-Transfer.
6. Support A(or K)-C-F conference call.
7. Select specific ITSP service by Realm Prefix (1*, 2* or 3*).

4. Parts list

Please check the parts for any missing parts. If do, please contact our agents :

4.1 「PG-101」 main body

4.2 Power adaptor AC-DC (110V AC – 12V DC) or (220V AC – 12V DC)

4.3 Network cable

4.4 Phone connecting Line

4.5 User Manual



(4.1)



(4.2)



(4.3)



(4.4)

5. Dimension: 14*9*3 cm

6. Chart of the device

Front Panel

In normal situation, the register LED only blink at power-on stage for 3~6 times. If register LED keep blinking, the system is not in normal state and the registering of SIM may not be completed. Once the register LED keep blinking for several minutes, check ether-net connection or try another STUN server which can be set in the WEB page.



***Power:** LED on after power on.

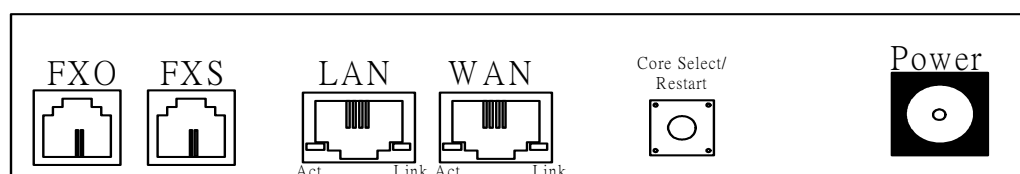
***Hook:** Indicate the status of phone

***PSTN:** Indicate the status of PSTN Line

***Register:**

LED status	Behavior
OFF	Default
ON	Registered to SIP proxy
Blinking	Checking NAT

Back Panel



***Restart button**

Action	Behavior
Short Click	System restart
Long Press (over 5 second)	System factory reset and restart

7. Web Page Setting

When the IP setting is done, the operator may setup all the rest parameters via web page.

Item	Value
HTML Port	9999
System User Name	ata
System Password	1234
Normal User Name	user
Normal Password	1234
WAN port IP	Default DHCP
LAN Port IP	192.168.123.1

Via WAN port

Step1: Use the FXS port phone set to get the WAN IP by IVR (#126#).

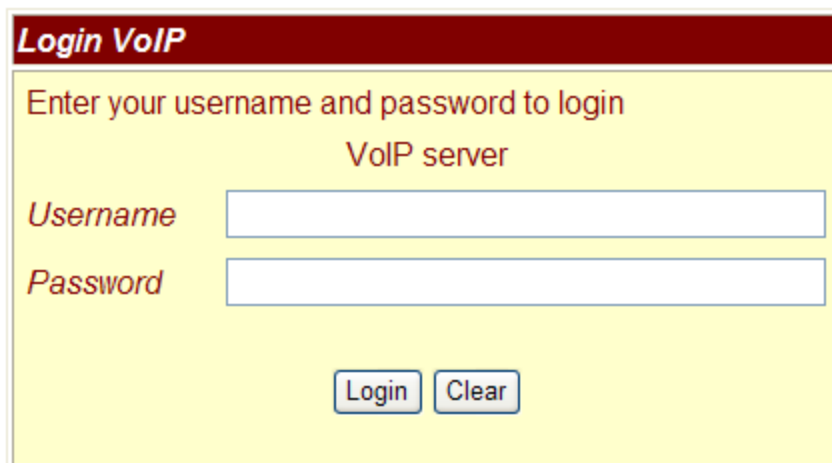
Step2: Start a WEB browser window and enter the following line to access WEB page.

http://wan_ip:9999

Via LAN port

Step1: Start a WEB browser window and enter the following line to access WEB page.

http://192.168.123.1:9999



Login VoIP

Enter your username and password to login
VoIP server

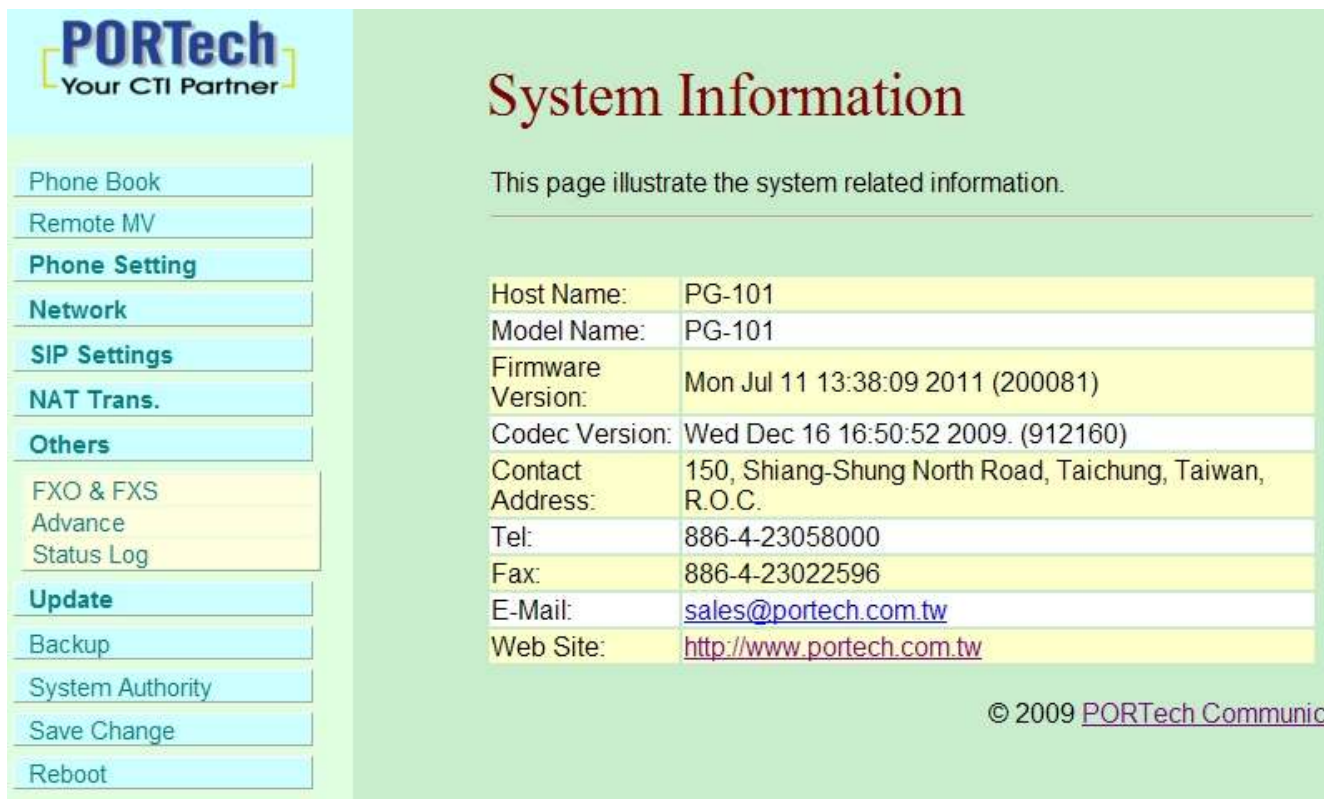
Username

Password

Login Clear

8. System Information.

- 8.1 When you login the web page, you can see the demo system current system information like firmware version, company... etc in this page.
- 8.2 Also you can see the function lists in the left side. You can use mouse to click the function you want to set up.



The screenshot displays the PORTech web interface. On the left is a navigation menu with the PORTech logo and tagline 'Your CTI Partner'. The menu items include Phone Book, Remote MV, Phone Setting, Network, SIP Settings, NAT Trans., Others, Update, Backup, System Authority, Save Change, and Reboot. The main content area is titled 'System Information' and contains a table of system details. Below the table is a copyright notice for 2009 PORTech Communication.

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System Information

This page illustrate the system related information.

Host Name:	PG-101
Model Name:	PG-101
Firmware Version:	Mon Jul 11 13:38:09 2011 (200081)
Codec Version:	Wed Dec 16 16:50:52 2009. (912160)
Contact Address:	150, Shiang-Shung North Road, Taichung, Taiwan, R.O.C.
Tel:	886-4-23058000
Fax:	886-4-23022596
E-Mail:	sales@portech.com.tw
Web Site:	http://www.portech.com.tw

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9. Phone Book

You could add/delete items in current phone books.

The screenshot shows the PORTech web interface for managing a phone book. On the left is a navigation menu with options like Phone Book, Remote MV, Phone Setting, Network, SIP Settings, NAT Trans., Others, Update, Backup, System Authority, Save Change, and Reboot. The main content area is titled 'Phone Book' and contains a table with 10 rows (0-9) for phone entries. Each row has columns for 'Phone', 'Name', 'Number or URL', and 'Select'. Below the table are buttons for 'Delete Selected', 'Delete All', and 'Reset'. At the bottom, there is an 'Add New Phone' section with input fields for 'Position' (0-99), 'Name', and 'Number or URL', along with 'Add Phone' and 'Reset' buttons.

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Your CTI Partner

Phone Book

You could add/delete items in current phone book.

Phone Book Page: page 1

Phone	Name	Number or URL	Select
0			<input type="checkbox"/>
1			<input type="checkbox"/>
2			<input type="checkbox"/>
3			<input type="checkbox"/>
4			<input type="checkbox"/>
5			<input type="checkbox"/>
6			<input type="checkbox"/>
7			<input type="checkbox"/>
8			<input type="checkbox"/>
9			<input type="checkbox"/>

Delete Selected Delete All Reset

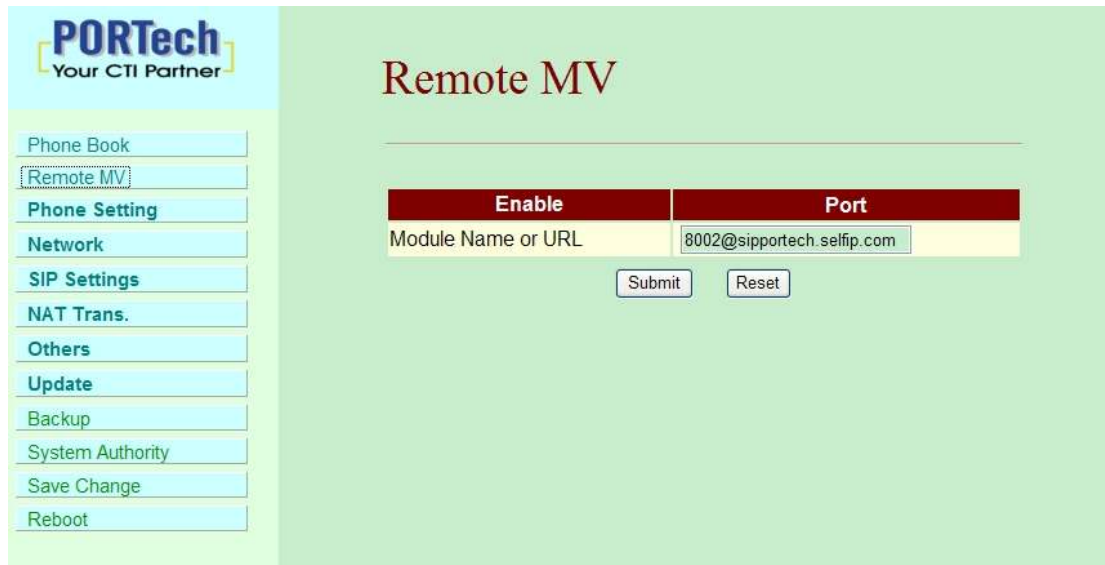
Add New Phone

Position: (0~99)
Name:
Number or URL:

Add Phone Reset

10. Remote MV

PG-101 can register to MV as optional. The setting is for one-stage or two-stage outgoing call in case of MV environment. As for the case of soft-phone environment, please leave this part blank



The screenshot shows the 'Remote MV' configuration page. On the left is a navigation menu with the following items: Phone Book, Remote MV (highlighted), Phone Setting, Network, SIP Settings, NAT Trans., Others, Update, Backup, System Authority, Save Change, and Reboot. The main content area is titled 'Remote MV' and contains a table with two columns: 'Enable' and 'Port'. The 'Enable' column has a radio button that is currently selected. The 'Port' column has a text input field containing '8002@sippotech.selfip.com'. Below the table are 'Submit' and 'Reset' buttons.

Enable	Port
<input checked="" type="radio"/>	<input type="text" value="8002@sippotech.selfip.com"/>

***Module Name or URL:** The peer SIP account used when dialing VOIP outgoing call(i.e. The SIP account of B used to register in the system topology).

11. Phone Setting

11.1 Master Setting

PORTech
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Phone Book
Remote MV
Phone Setting
Master Setting
Volume Settings
DND Settings
Caller ID
Dial Plan
Flash Time
Alarm
Network
SIP Settings
NAT Trans.
Others
Update
Backup
System Authority
Save Change
Reboot

Master Setting

You could set the master number of your phone in this page.

	Number
Master Number:	<input type="text" value="0975227590"/>
No Answer Fwd Time Out:	<input type="text" value="0"/> (2~8 Ring, 0 for forward directly)

***Master Number:** The number that FXO dial out (i.e. the phone number of F in the system topology). System uses this number to authenticate the caller ID of PSTN incoming call. If the caller ID matches to the setting in the master number, it has the permission to dial out via VoIP in MV (i.e. F-D-B-G or F-D-B-H). If the matching result is failed, the PSTN incoming call will ring the local FXS directly (F-D-C).

***No Answer Fwd Time Out:** Setup the FXS ringing period. For setting 2~8 rings, the incoming VOIP call from MV always rings local FXS first. After ringing period, system change to dial master number via FXO port. (Parameter "0": forwarding to PSTN directly.) The caller side will hear a greeting at first then a ring back tone from PSTN while dialing master number. To Set 0 for concurrent ring, which is for ringing FXS, dialing master number via FXO port are concurrence. The empty of master number will disable dialing master number function which means the incoming VoIP call will ring FXS only till caller side end the call.

11.2 Volume Settings

You could set the volume of your phone in this page.

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Your CTI Partner

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Master Setting
Volume Settings
DND Settings
Caller ID
Dial Plan
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Volume Setting

You could set the volume of your phone in this page.

Handset Volume:	<input type="text" value="10"/>	(0~12)
PSTN-Out Volume:	<input type="text" value="10"/>	(0~12)
Handset Gain:	<input type="text" value="10"/>	(0~15)
PSTN-In Gain:	<input type="text" value="10"/>	(0~15)

11.3 DND Settings

You could set the do not disturb period of your phone in this page

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Your CTI Partner

Phone Book
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Master Setting
Volume Settings
DND Settings
Caller ID
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Others
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Save Change
Reboot

DND Setting

You could set the do not disturb period of your phone in this page.

DND Always:	<input type="radio"/> On	<input checked="" type="radio"/> Off
DND Period:	<input type="radio"/> On	<input checked="" type="radio"/> Off
From:	<input type="text" value="00"/> : <input type="text" value="00"/>	(hh:mm)
To:	<input type="text" value="00"/> : <input type="text" value="00"/>	(hh:mm)

11.4 Caller ID

In order to correctly generate caller ID, user should set the caller ID format of FXS. You could enable/disable the caller ID setting in this page.

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Caller ID
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Others
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Caller ID Setting

You could enable/disable the caller ID setting in this page.

Caller ID:	Caller ID after 1st Ring (FSK) ▼
Single Caller ID:	<input type="radio"/> Yes <input checked="" type="radio"/> No
CID Without Time:	<input type="radio"/> Yes <input checked="" type="radio"/> No
CID Type 2:	<input type="radio"/> Yes <input checked="" type="radio"/> No

11.5 Dial Plan



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Dial Plan

You could the set the dial plan in this page.

Transit Method : One Stage Two Stage Proxy

Routing to : IP FXO Disable

Routing rule : 2+*+0975

Drop prefix : Yes No

Replace rule 1: 0916 + 0981+*0988

Drop prefix : Yes No

Replace rule 2: +

Drop prefix : Yes No

Replace rule 3: +

Drop prefix : Yes No

Replace rule 4: +

Realm 1 prefix: 1*

Realm 2 prefix: 2*

Realm 3 prefix: 3*

Auto Dial Time: 5 (3~9 sec)

Use # as send key: Yes No

***Transit Method:** Three types of methods provided to user.

- One Stage: The destination phone number is sent in the SIP INVITE message with transiting a VOIP Call. It should be supported by SIP Server while dialing via a SIP proxy. This is the default setting of the system.
- Two Stage: The destination phone number is sent to peer MV by 2833 DTMT packet or SIP INFO depending on PG-101 setting SIP Setting→DTMF Setting page. It should be supported by SIP server while the 2833 packet or SIP INFO is ready by SIP proxy.
- Proxy: The destination phone number is sent in SIP INVITE message and routed by outbound proxy. For outbound service, please consult ISP for more details.

***Routing to:** Define the call direction (IP call or PSTN call) if the dialed digit match the rule set in Routing rule field.

***Routing rule:** Define the digit string of routing rule. Only digit 0~9 are valid in rule string, as well as “x” denotes wildcard digit. Different patterns is separated by sign “+”.

***Drop prefix:**

Item	Description
Yes	The following replace rule define the prefix dropping rule
No	The following replace rule define the prefix adding rule

***Replace Rule:** The digit string defined in the first field will be added if the dialed string match the digit string defined in the second field of the replacement rule. If the selection of Drop prefix is “yes”, the matched digit string in the dialed string will be removed before adding prefix. Otherwise, if “No” is selected, the prefix is added at the beginning of dialed string without removing any digits.

Only digit 0~9 are valid in rule string, as well as “x” denotes wildcard digit and “,” denotes pause for one second. Different patterns is separated by sign “+”.

Replace rule 1: +

Example 1:

Drop prefix : Yes No

Replace rule 1: +

For example, the dialed string is 86621742885. After the processing of replacement rule, the result dialing string will become 002+86621742885.

Example 2:

Drop prefix : Yes No

Replace rule 1: +

For instance, the dialed string is 00286621742885. After the processing of replacement rule, the result dialing string will become 006 86621742885 since “Yes” of Drop prefix is select and “002” is dropped and replaced by “006”.

Example 3:

Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 1:	<input type="text" value="007"/> + <input type="text" value="5xxx+35xx+21xx"/>

In this example, any 4-digits string begin with 5, 35 or 21 will be add a “007” prefix. e.g. 5171 will become 0075171.

Example 4:

Drop prefix :	<input type="radio"/> Yes <input checked="" type="radio"/> No
Replace rule 1:	<input type="text" value="9,.."/> + <input type="text" value="5"/>

In this example, any string begin with 5 will be add a “9” prefix. After the prefix “9” digit, it will be hold in 3 seconds and then the succeeded digits.

***Realm Selection:**

The call behavior of PG-101 is act as a number router if Module Name or URL in Remote MV page is set. The IP call will route to peer MV, thus the dialed number is called by GSM network. User can specify an ITSP or a SIP proxy by adding Realm Prefix before dialing destination number or SIP account. The configuration of Realm Prefix can be found in the Dial Plan page.

For the case in MV coexist with soft-phone, user should use Realm Prefix method to a specific SIP call.

Example: If you put Module Name or URL: **8002@sippotech.selfip.com**, which means it's the SIP account of mobile channel 1 on MV. And dial 0937123456 will route call to MV and send 0937123456 to the operator MV registered. Dial 1*4002 will call SIP user 4002 at SIP proxy of realm 1, which is sippotech.selfip.com.

11.6 Flash Time

You could set the flash time in this page

The screenshot shows the 'Flash Time Setting' page. On the left is a navigation menu with 'Phone Setting' selected. The main content area has a title 'Flash Time Setting' and a subtitle 'You could set the flash time in this page.' Below this are two sections: 'FXO Flash Time' with a 'Generate Flash Signal' of 10 x 10 ms (9~120), and 'FXS Flash Time' with 'Flash Signal Detect (MAX): 60 x 10 ms (4~255)' and 'Flash Signal Detect (MIN): 7 x 10 ms (3~12)'. At the bottom are 'Submit' and 'Reset' buttons.

11.7 Alarm

You could set the alarm time in this page.

The screenshot shows the 'Alarm Settings' page. On the left is a navigation menu with 'Alarm' selected. The main content area has a title 'Alarm Settings' and a subtitle 'You could set the alarm time in this page.' Below this is an 'Alarm' section with radio buttons for 'ON' and 'OFF', where 'OFF' is selected. The 'Alarm Time' is set to 0:0 (hh:mm). The 'Current time' is 2011-07-07 09:16. At the bottom are 'Submit' and 'Reset' buttons.

12. Network

In Network you can check the Network Status, WAN, LAN, STNP Settings, and Virtual Server.

12.1 Network Status: You can check the current Network setting in this page.

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Network Status

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

System Up Time:	0 day(s) 17 hour(s) 12 minute(s)
Network Link Up Time:	0 day(s) 17 hour(s) 12 minute(s)

WAN

Type:	DHCP Client
IP:	192.168.0.189
Mask:	255.255.255.0
Gateway:	192.168.0.254
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

LAN

Type:	Fixed IP Client
IP:	192.168.123.1
Mask:	255.255.255.0
Gateway:	192.168.123.1
DNS Server 1:	168.95.192.1
DNS Server 2:	168.95.1.1

12.2 WAN

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WAN Settings

You could configure the WAN settings in this page.

LAN Mode: Bridge NAT

WAN Setting

IP Type: Fixed IP DHCP Client PPPoE

IP: 192.168.0.189

Mask: 255.255.255.0

Gateway: 192.168.0.254

DNS Type: Fixed Auto

DNS Server1: 168.95.192.1

DNS Server2: 168.95.1.1

MAC: 00037e00e14e

Host Name: PG-101

PPPoE Setting

User Name:

Password:

Service Name:

- (1) The DHCP Client Configuration item is to setup the WAN port's network environment.
- (2) The PPPoE Configuration item is to setup the PPPoE Username and Password. If you have the PPPoE account from your Service Provider, please input the Username and the Password correctly.
- (3) When you finished the setting, please click the Submit button.

12.3 LAN

You could configure the LAN settings in this page.

The screenshot shows the PORTech web interface for LAN settings. On the left is a navigation menu with the following items: Phone Book, Remote MV, Phone Setting, Network, Status, WAN, LAN (highlighted), SNTP Settings, Virtual Server, SIP Settings, NAT Trans., Others, Update, Backup, System Authority, Save Change, and Reboot. The main content area is titled "LAN Settings" and contains the text "You could configure the LAN settings in this page." Below this are two configuration sections: "LAN Setting" and "DHCP Server".

LAN Setting

IP:	<input type="text" value="192.168.123.1"/>
Mask:	<input type="text" value="255.255.255.0"/>
MAC:	<input type="text" value="00037e00e14f"/>

DHCP Server

DHCP Server:	<input type="radio"/> On <input checked="" type="radio"/> Off
Start IP:	<input type="text" value="150"/>
End IP:	<input type="text" value="200"/>
Lease Time:	<input type="text" value="1"/> : <input type="text" value="0"/> (dd:hh)

Submit Reset

12.4 SNTP Settings

SNTP Setting function: you can setup the primary and second SNTP Server IP Address, to get the date/time information. Also you can base on your location to set the Time Zone, and how long need to synchronize again. When you finished the setting, please click the Submit button.

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SNTP Settings

You could set the SNTP servers and Daylight Saving Time (DST) in this page.

SNTP: On Off

Primary Server: north-america.pool.ntp.org
Secondary Server: asia.pool.ntp.org

Time Zone: GMT + 08:00 (hh:mm)
Sync. Time: 0:6:0 (dd:hh:mm)

Daylight Saving: On Off

DST Offset: -2

DST Start Date: Jan
 Day of Month 01
 Week of Month Week 1 Sun
Start Time: 00

DST End Date: Jan
 Day of Month 01
 Week of Month Week 1 Sun
End Time: 00

Submit Reset

12.5 Virtual Server

You could set your virtual servers in this page. The usual port numbers are WEB [TCP 80], FTP(Control) [TCP 21], FTP(Data) [TCP 20], E-mail(POP3) [TCP 110], E-mail(SMTP) [TCP 25], DNS [UDP 53] and Telnet [TCP 23].

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Virtual Server Settings

You could set your virtual servers in this page. The usual port numbers are WEB [TCP 80], FTP(Control) [TCP 21], FTP(Data) [TCP 20], E-mail(POP3) [TCP 110], E-mail(SMTP) [TCP 25], DNS [UDP 53] and Telnet [TCP 23].

Virtual Server Page: page 1

Num	Enable	Protocol	In Port	Ex Port	Server IP	Select
0	<input type="checkbox"/>					<input type="checkbox"/>
1	<input type="checkbox"/>					<input type="checkbox"/>
2	<input type="checkbox"/>					<input type="checkbox"/>
3	<input type="checkbox"/>					<input type="checkbox"/>
4	<input type="checkbox"/>					<input type="checkbox"/>
5	<input type="checkbox"/>					<input type="checkbox"/>
6	<input type="checkbox"/>					<input type="checkbox"/>
7	<input type="checkbox"/>					<input type="checkbox"/>

Enable Selected Delete Selected Delete All Reset

Add Virtual Server

Server IP:
Protocol:
Internal Port Start: Internal Port End:
External Port Start: External Port End:

Add Server Reset

13. SIP Settings

In SIP Setting you can setup the Service Domain, Port Settings, Codec Settings, Codec ID, DTMF Setting and Other Settings.

13.1 Service Domain

Service Domain Settings

You could set information of service domains in this page.

Realm No.:

Realm	
Active:	<input checked="" type="radio"/> On <input type="radio"/> Off
Display Name:	<input type="text" value="4006"/>
User Name:	<input type="text" value="4006"/>
Register Name:	<input type="text" value="4006"/>
Register Password:	<input type="password" value="....."/>
Domain Server:	<input type="text"/>
Proxy Server:	<input type="text" value="sipportech.selfip.com"/>
Outbound Proxy:	<input type="text"/>
Subscribe for MWI:	<input checked="" type="radio"/> On <input type="radio"/> Off
Status:	Registered

User need to set the registering information of SIP proxy at this page. PG-101 cannot work correctly without registering to SIP proxy.

First you need to click Active to enable the Service Domain, and then you can input the following items.

- (1) Display name: you can input the name you want to display.
- (2) User name: you need to input the User Name get from your ISP.
- (3) Register Name: you need to input the Register Name get from your ISP.
- (4) Register Password: you need to input the Register Password get from ISP.

- (5) Domain Server: you need to input the Domain Server get from your ISP.
- (6) Proxy Server: you need to input the Proxy Server get from your ISP. Please note, PG-101-102L must connect "message Server". E.g. the free server we recommend, IPTel, with this feature.
- (7) Outbound Proxy: you need to input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can skip this item.
- (8) You can see the Register Status in the Status item.
- (9) When you finished the setting, please click the Submit button. Remember to click "Save Charge"

13.2 Port Settings

You could set the port number in this page

PORTech
Your CTI Partner

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SIP Settings
Service Domain
Port Settings
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Codec ID
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Other Settings
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Save Change
Reboot

Port Settings

You could set the port number in this page.

SIP Port:	<input type="text" value="5060"/>	(0~65533) (Set 0 for auto, range as bellow)
RTP Port:	<input type="text" value="0"/>	(0~65533) (Set 0 for auto, range as bellow)
SIP Port Range:	<input type="text" value="10000"/> ~ <input type="text" value="10999"/>	(1024~40000)
RTP Port Range:	<input type="text" value="20000"/> ~ <input type="text" value="21999"/>	(1024~40000)

13.3 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. When you finished the setting, please click the Submit button.

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Your CTI Partner

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Codec ID
DTMF Setting
Other Settings
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Reboot

Codec Settings

You could set the codec settings in this page.

Codec Priority	
Codec Priority 1:	G.711 u-law
Codec Priority 2:	G.711 a-law
Codec Priority 3:	G.723
Codec Priority 4:	G.729
Codec Priority 5:	Not Used
Codec Priority 6:	Not Used
Codec Priority 7:	Not Used
Codec Priority 8:	Not Used
Codec Priority 9:	Not Used

RTP Packet Length	
G.711 & G.729:	20 ms
G.723:	30 ms

G.723 5.3K	
G.723 5.3K:	<input type="radio"/> On <input checked="" type="radio"/> Off

Voice VAD	
Voice VAD:	<input type="radio"/> On <input checked="" type="radio"/> Off

Submit Reset

13.4 Codec ID

You can setup the Codec ID in this page.

The screenshot shows the 'Codec ID Setting' page. On the left is a navigation menu with the PORTech logo and 'Your CTI Partner' tagline. The menu includes: Phone Book, Remote MV, Phone Setting, Network, SIP Settings (highlighted), Service Domain, Port Settings, Codec Settings, Codec ID (highlighted), DTMF Setting, Other Settings, NAT Trans., Others, Update, Backup, System Authority, Save Change, and Reboot. The main content area has the title 'Codec ID Setting' and the text 'You could set the value of Codec ID in this page.' Below this is a table with three columns: Codec Type, ID, and Default Value. The table contains five rows of settings. At the bottom of the table are 'Submit' and 'Reset' buttons.

Codec Type	ID	Default Value
G726-16 ID:	23 (95~255)	<input checked="" type="checkbox"/> 23
G726-24 ID:	22 (95~255)	<input checked="" type="checkbox"/> 22
G726-32 ID:	2 (95~255)	<input checked="" type="checkbox"/> 2
G726-40 ID:	21 (95~255)	<input checked="" type="checkbox"/> 21
RFC 2833 ID:	101 (95~255)	<input checked="" type="checkbox"/> 101

13.5 DMTF Settings

You can setup the DMTF Setting in this page

The screenshot shows the 'DTMF Setting' page. On the left is the same navigation menu as in the previous screenshot, with 'DTMF Setting' highlighted. The main content area has the title 'DTMF Setting' and the text 'You could set the DTMF setting in this page.' Below this is a form with three radio button options: RFC 2833 (selected), Inband DTMF, and Send DTMF SIP Info. At the bottom of the form are 'Submit' and 'Reset' buttons.

13.6 Other Settings

Other Settings: you can setup the Hold by RFC and QoS in this page. To change these settings, please follow your ISP information. When you finished the setting, please 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 will get the higher priority to the Internet. But the QoS function still need to cooperate with the others Internet devices.

PORTech
Your CTI Partner

Phone Book
Remote MV
Phone Setting
Network
SIP Settings
Service Domain
Port Settings
Codec Settings
Codec ID
DTMF Setting
Other Settings
NAT Trans.
Others
Update
Backup
System Authority
Save Change
Reboot

Other Settings

You could set other settings in this page.

Hold by RFC:	<input type="radio"/> On <input checked="" type="radio"/> Off
Voice QoS (Diff-Serv):	<input type="text" value="40"/> (0~63)
SIP QoS (Diff-Serv):	<input type="text" value="40"/> (0~63)
SIP Expire Time:	<input type="text" value="60"/> (15~86400 sec, 0=define by Server)
Use DNS SRV:	<input type="radio"/> On <input checked="" type="radio"/> Off
Send Keep Alives Packet:	<input type="radio"/> On <input checked="" type="radio"/> Off
Keep Alives Period:	<input type="text" value="60"/> (15~250 sec)
Jitter Buffer:	<input type="text" value="1"/> (0~32 packets)
SIP Server type:	General <input type="button" value="v"/>
SIP VID (VLAN):	<input type="text" value="0"/> (2~4094, 0:disabled)
RTP VID (VLAN):	<input type="text" value="0"/> (2~4094, 0:disabled)

14. NAT Trans.

In NAT Trans. you can setup STUN and uPnP function. These functions can help your VoIP device working properly behind NAT.

14.1 STUN Setting (optional)

There is a default setting of STUN server in the PG-101 system database; user can ignore this step, which is optional setting. If the STUN server is not connectable or service shutdown, please change STUN setting at this page.

PORTech
Your CTI Partner

Phone Book
Remote MV
Phone Setting
Network
SIP Settings
NAT Trans.
STUN Setting
uPnP Setting
Others
Update
Backup
System Authority
Save Change
Reboot

STUN Setting

You could set the IP of STUN server in this page.

STUN: On Off

STUN Server:

STUN Port: (80~65535)

Force Public IP: On Off

Public IP address:

Port: (80~65535)

14.2 uPnP Setting

You could enable/disable the uPnP in this page

The screenshot shows the PORTech web interface. On the left is a navigation menu with the following items: Phone Book, Remote MV, Phone Setting, Network, SIP Settings, NAT Trans., STUN Setting, uPnP Setting (highlighted), Others, Update, Backup, System Authority, Save Change, and Reboot. The main content area is titled "uPnP Setting" and contains the text "You could enable/disable the uPnP in this page." Below this text, there is a radio button group for "uPnP:" with "On" and "Off" options. The "Off" option is selected. At the bottom of the form are "Submit" and "Reset" buttons.

15. Others

15.1 FXO&FXS

User should set FXS and FXO type according to country which PG-101 located.

The screenshot shows the PORTech web interface. On the left is a navigation menu with the following items: Phone Book, Remote MV, Phone Setting, Network, SIP Settings, NAT Trans., Others, FXO & FXS (highlighted), Advance, Status Log, Update, Backup, System Authority, Save Change, and Reboot. The main content area is titled "FXO & FXS Setting" and contains the text "You could select the FXO & FXS impedance of the analog telephone by different country in this page." Below this text, there are four form fields: "FXO Port:" with a dropdown menu set to "USA", "FXS Port:" with a dropdown menu set to "USA", "FXO Silence Timeout:" with a text input field containing "30" and "(1-250 minutes)" next to it, and "FXO CID forward:" with radio buttons for "On" and "Off", where "Off" is selected. At the bottom of the form are "Submit" and "Reset" buttons.

15.2 Advance

You could change advanced setting in this page

PORTech
Your CTI Partner

Advanced Setting

You could change advanced setting in this page.

ICMP Not Echo:	<input type="radio"/> Yes <input checked="" type="radio"/> No
Send Anonymous CID:	Disabled
Management from WAN:	<input checked="" type="radio"/> Yes <input type="radio"/> No
Stop feature tone:	<input type="radio"/> Yes <input checked="" type="radio"/> No (MMI,forward,block....)
Billing Signal:	Disabled
CPC Delay:	2 (2~5 Seconds)
CPC Duration:	0 x 10 ms (0~120)
IP Dialing format:	Type 1 (x@x.x.x.x)
Send Flash event:	Disabled
Encryption Type:	Disabled
Encryption Key:
PPPoE retry period:	5 Seconds
System Log Server:	
System Log Type:	None

Submit Reset

15.3 Status Log

PORTech
Your CTI Partner

Status Log

```
<2011-07-07 08:54>REG MSG: 401 is received
<2011-07-07 08:54>REG MSG: REGISTER is sent
<2011-07-07 08:54>REG MSG: 200 is received
<2011-07-07 08:54>Reg Status: REGISTERED
<2011-07-07 08:55>REG MSG: 401 is received
<2011-07-07 08:55>REG MSG: REGISTER is sent
<2011-07-07 08:55>REG MSG: 200 is received
<2011-07-07 08:55>Reg Status: REGISTERED
<2011-07-07 08:56>REG MSG: 401 is received
<2011-07-07 08:56>REG MSG: REGISTER is sent
<2011-07-07 08:56>REG MSG: 200 is received
<2011-07-07 08:56>Reg Status: REGISTERED
<2011-07-07 08:57>REG MSG: 401 is received
<2011-07-07 08:57>REG MSG: REGISTER is sent
<2011-07-07 08:57>REG MSG: 200 is received
<2011-07-07 08:57>Reg Status: REGISTERED
<2011-07-07 08:58>REG MSG: 401 is received
<2011-07-07 08:58>REG MSG: REGISTER is sent
<2011-07-07 08:58>REG MSG: 200 is received
<2011-07-07 08:58>Reg Status: REGISTERED
<2011-07-07 08:59>REG MSG: 401 is received
<2011-07-07 08:59>REG MSG: REGISTER is sent
<2011-07-07 08:59>REG MSG: 200 is received
<2011-07-07 08:59>Reg Status: REGISTERED
<2011-07-07 09:00>REG MSG: 401 is received
```

Submit Reset

16. Update

In Update you can update the system's firmware to the new one or the factory reset to let the system back to default setting.

16.1 New Firmware

PORTech
Your CTI Partner

Update Firmware

You could update the newest firmware.

Method: Local PC TFTP

Local PC

Code Type: OS0 (*.gz, *.gzh) ▼

File Location: 浏览...

TFTP

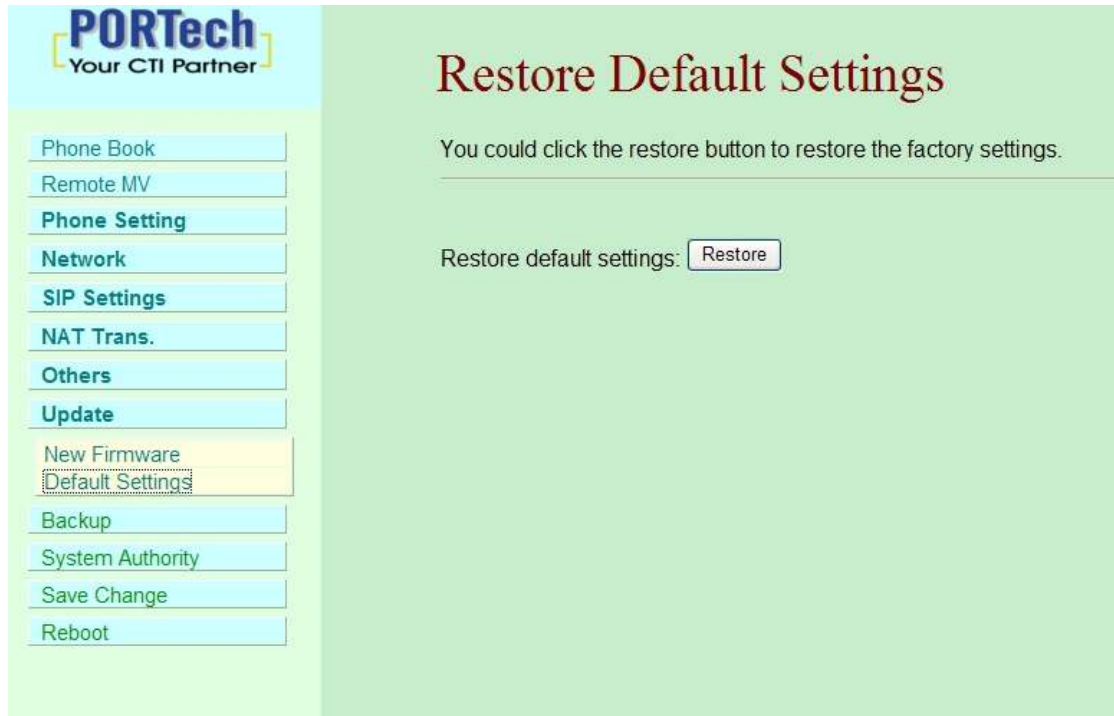
TFTP Server:

- (1) In New Firmware function you can update new firmware via Local PC in this page. You can upgrade the firmware by the following steps:
- (2) Select the firmware code type.
- (3) Click the “Browse” button in the right side of the File Location or you can type the correct path and the filename in File Location blank.
- (4) Select the correct file you want to download to the system then click the Update button.
- (5) Please click update/default setting after update firmware

16.2 Default Settings

In this page: Update/ Default Settings, you could restore the factory default settings to the system. All setting will restore default setting.

IP will retain original IP as usual not default IP.



PORTech
Your CTI Partner

- Phone Book
- Remote MV
- Phone Setting**
- Network
- SIP Settings
- NAT Trans.
- Others
- Update
- New Firmware
- Default Settings**
- Backup
- System Authority
- Save Change
- Reboot

Restore Default Settings

You could click the restore button to restore the factory settings.

Restore default settings:

17. Backup

You could save current system setting to file: System Settings and PhoneBook.



The screenshot shows a web interface for 'PORTech Your CTI Partner'. On the left is a vertical menu with items: Phone Book, Remote MV, Phone Setting, Network, SIP Settings, NAT Trans., Others, Update, Backup, System Authority, Save Change, and Reboot. The 'Backup' item is highlighted. The main content area is titled 'Backup System Settings' and contains the text 'You could save current system settings to file.' Below this is a form with two rows: 'System Settings:' with a 'Save File' button, and 'PhoneBook:' with a 'Save File' button.

18. System Authority

In System Authority you can change your login name and password



The screenshot shows a web interface for 'PORTech Your CTI Partner'. On the left is a vertical menu with items: Phone Book, Remote MV, Phone Setting, Network, SIP Settings, NAT Trans., Others, Update, Backup, System Authority, Save Change, and Reboot. The 'System Authority' item is highlighted. The main content area is titled 'System Authority' and contains the text 'You could change the login username/password in this page.' Below this is a form with three rows: 'New username:' with a text input field, 'New password:' with a text input field, and 'Confirmed password:' with a text input field. At the bottom of the form are 'Submit' and 'Reset' buttons.

19. Save Change

In Save Change you can save the changes you have done. If you want to use new setting in the VoIP system, you have to click the Save button. After you click the Save button, the system will automatically restart and the new setting will effect.



The screenshot shows a web interface for 'PORTech Your CTI Partner'. On the left is a vertical menu with buttons for 'Phone Book', 'Remote MV', 'Phone Setting', 'Network', 'SIP Settings', 'NAT Trans.', 'Others', 'Update', 'Backup', 'System Authority', 'Save Change', and 'Reboot'. The 'Save Change' button is highlighted with a dashed border. The main content area has a light green background and features the title 'Save Changes' in a large, dark red font. Below the title, the text reads 'You have to save changes to effect them.' followed by a horizontal line. At the bottom, it says 'Save Changes:' followed by a 'Save' button.

20. Reboot

Reboot function you can restart the system. If you want to restart the system, you can just click the Reboot button, and then the system will automatically.



The screenshot shows a web interface for 'PORTech Your CTI Partner'. On the left is a vertical menu with buttons for 'Phone Book', 'Remote MV', 'Phone Setting', 'Network', 'SIP Settings', 'NAT Trans.', 'Others', 'Update', 'Backup', 'System Authority', 'Save Change', and 'Reboot'. The 'Reboot' button is highlighted with a dashed border. The main content area has a light green background and features the title 'Reboot System' in a large, dark red font. Below the title, the text reads 'You could press the reboot button to restart the system.' followed by a horizontal line. At the bottom, it says 'Reboot system:' followed by a 'Reboot' button.

21. Specification

21.1 Call transfer

During ongoing call, user can transfer current call by “flash hook”. The call will be transfer to PSTN if the current call is from VOIP, and vice versa.

21.2 Conference call

During ongoing call, user can initiate a conference call by “flash hook”+ “#512#”.

22.3 IVR Command

Group	Action	key	parameter(s)
Function	IVR unlock	#190#	
Function	IVR lock	#191#	
Function	Reboot	#195#	
Function	Factory	#198#	
Function	Set DHCP client	#111#	
Function	Set static IP	#112xxx*xxx*xxx* xxx#	Static IP address
Function	Set network mask	#113xxx*xxx*xxx* xxx#	network mask
Function	Set default gateway IP	#114xxx*xxx*xxx* xxx#	default gateway IP
Function	Set primary DNS	#115xxx*xxx*xxx* xxx#	primary DNS
Info	Check LAN IP Address	#120#	
Info	Check DHCP client status	#121#	
Info	User name of SIP account	#122#	
Info	Check network mask	#123#	
Info	Check default gateway IP	#124#	
Info	Check DNS	#125#	

Info	Check WAN IP Address	#126#	
Info	Software version	#128#	
Info	Check PSTN number of forward all call	#147#	
Info	Check PSTN number of no answer forward call	#149#	
Function	Forward disable	#145#	
Function	Forward all call to PSTN	#147xxxxxxxx#	PSTN number
Function	No answer forward to PSTN	#149xxxxxxxx#	PSTN number
Function	conference	#512#	