User Manual

SIP-GW3CM

SIP Analog Telephone Adaptor

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1. Key Feature

New feature

- Make/Receive both VoIP calls and regular PSTN calls.
- Auto-detection and auto-switch to VoIP or PSTN calls.
- Phone rings for all incoming VoIP and PSTN calls.

Support Caller ID/Name display or block.

- Call hold,call waiting,call forwarding and call transfer.
- Support three service domain and auto-provision.

1.1 Network Feature

- Supports SIP 2.0 (RFC3261)PPPoE, STUN, DHCP.
- Support in-band and out-of-band DTMF transfer.
- Support automated NAT traversal without manual manipulation of firewall/NAT.
- Support firmware upgrade via TFTP, FTP and HTTP.

1.2 Voice Feature

Support VAD,CNG,AEC, PLC,AJB and Volumn adjustment.
Voice Codec: G.711,G.723.1,G.726,G.729A,G.729B.

1.3 Feature Description

- Make/Receive both VoIP calls and regular PSTN calls.

- Auto-detection and auto-switch to VoIP or PSTN calls.
- Phone rings for all incoming VoIP and PSTN calls.
- VoIP or PSTN calls selected by phone key.
- Support Caller ID/Name display or block.
- Support call hold, call waiting, call forwarding, and call transfer.
- Dial plans, dial tone, busy tone, ring back tone, alert tone can be set flexible.

1.4 Management Feature

- Configured ATA via web browser or remote.
- Firmware can be upgraded through HTTP,FTP or TFTP

1.5 Physical Feature

- Two RJ45 ports: Dual 10M/100M auto-sensing, with router built-in, one for internet, one for PC.
- Two RJ11 Port: One FXS telephone Port and One lifeline for PSTN line.
- LED Indication: There are three LED indicators in the TA to show the Status, and Off-Hook indication.
- Power adaptor:Input:AC 100~240V,Output:DC12V/0.5A.
- Operating Temperature: 0℃~40℃.

1.6 Packeage content

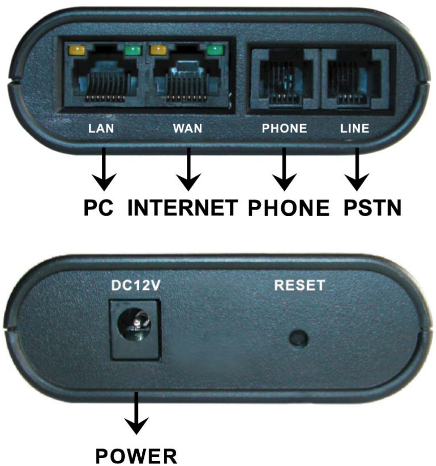
- One SIP ATA main body

- One Ethernet cable
- One user Manual
- One Universal Power Adaptor
- One PSTN cable

2. Quick install

2.1 Connecting to the TA

Please install the TA as the connection chart below:



2.2 Configuring by webpage

2.2.1 Login the webpage

Login via LAN port

a) Connect the PC to LAN port of TA.Default IP address of LAN port is 192.168.123.1, the port number ":8000".

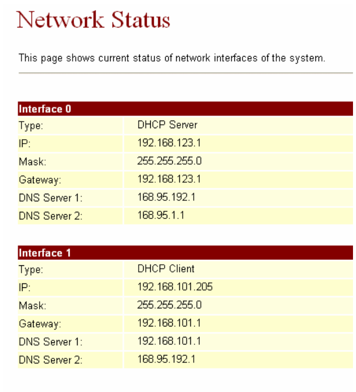
b) Open web browser and input http://192.168.123.1:8000

c) Enter the account and password(default account is **root** and password is **test**).

2.3 Network Configuration(WAN)

2.3.1 In Network you can check the Network status, configure the WAN Settings, LAN Settings.

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



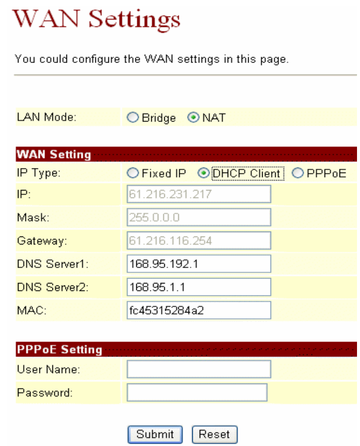
2.3.3 WAN Settings: The WAN port is for you to connect to the ADSL Router, Broadband Router. Also you can use PPPoE to get the WAN IP address from your ISP.

a) The IP Phone's default setting is NAT mode. If you don't need to use the NAT Mode, you can chang to Bridge Mode.

b) The WAN port default is DHCP Client mode, You can change the setting to Fixed IP Mode, or PPPoE Mode.

c) If you change the WAN port's setting to Fix IP Mode, then you have to make sure the IP address. Net Mask, Gateway, and DNS setting is suitable in your current network environment.

d) If you change the WAN port's setting to PPPoE Mode, you have to input a correct username/password to get the IP address from your Internet Service Provider.



2.4 Register account(Sip Setting)

2.4.1You may get account information from your service provider.

2.4.2 In Service Domain Function you need to input the account and the related informations in this page, please refer to your ISP provider. You can register three SIP account in the TA. You can dial the VoIP phone to your friends via first enable SIP account and receive the phone from these three SIP accounts. For the second phone you can use the same way to register.

a) Click Active to enable the Service Domain

b) Display Name:input the name you want to display.

c) User Name: input the User Name get from your ISP.

d) Register Name: input the Register Name get from your ISP.

e) Register Password: input the Register Password from your ISP.

f) Domain Server: input the Domain Server get from your ISP.

g) Proxy Server: input the Proxy Server get from your ISP.

h) Outbound Proxy: input the Outbound Proxy get from your ISP. If your ISP does not provide the information, then you can skip this item.

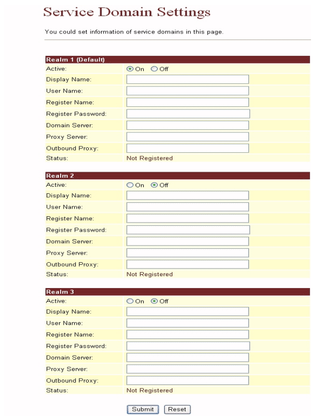
i) Register Period: input the Register Period get from your ISP. This is count in minute.

k) If you have more than one SIP account, you can following the steps to register to the other ISP.

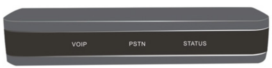
l) When you finished the setting, click the Submit button.

m) If there is nothing need to change, please click the Save

Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.



2.5 The LED of a SIP-GW3CM.



STATUS's LED	PSTN's LED	VOIP's LED
Unregistered:0.5s Flash Registered:Lighted Register Fail:0.5s Flash Reboot: Lighted	Offhook: Lighted Reboot: Lighted	Offhook: Lighted Reboot: Lighted Reboot: Lighted

3. Basic Functions

Note1:

1) When SIP-GW3CM is out of power, the RJ11 line jack will act as a pass through jack. The user will be able to use the same analog phone for PSTN calls.

2) When SIP-GW3CM doesn't register service,Pick up the anologe phone and press the number of "0*", the RJ 11 line jack will act as a pass through jack. The user will be able to use the same analog phone for PSTN calls.

3) When SIP-GW3CM registered service, the SIP line jack will act as a pass through jack. The user will be able to use the same analog phone for VoIP calls.

Note2:When you finished the setting, please click the Submit button. If there is nothing need to change, please click the Save Change Item in the left side, then click the Save button. The change you made will save into the system and the system will Reboot automatically.

3.1 Make Phone Calls

3.1.1 Make VoIP Calls

Note: You can make a VoIP call only when LED "STATUS" is lighted.

Pick up the analog phone, LED "SIP" light.

There are currently two methods to call out:

a) Dial the numbers directly and wait for 4 (general) seconds.

b) Dial the numbers directly, and press #

3.1.2 Make PSTN Calls

Note: You can make a PSTN call only when PSTN line is connected.

Pick up the analog phone, if LED "SIP" light, Press "0*" to switch to PSTN line.

LED "PSTN" light, and get a PSTN line dial tone, dial the numbers directly.

3.2 Answer Phone Calls

3.2.1 Answer VoIP Calls

When somebody calls in from VoIP, LED "SIP" flashed, the analog phone shows the caller ID.

Pick up the analog phone, LED "SIP" light, you are in conversation.

3.2.2 Answer PSTN Calls

When somebody call in from PSTN, LED “PSTN” flashed, the analog phone show the caller ID.

Pick up the analog phone, LED “PSTN” light, you are in conversation.

3. 3 Phone Setting

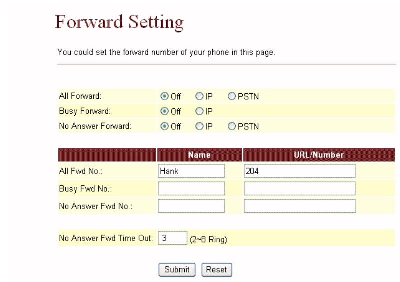
3.3.1 In Phone Setting: contains Call Forward, SNTP Settings, Volume Settings, Block Setting, Caller ID, Dial Plan Setting, Flash Time Setting, Call Waiting Setting, and T.38 (FAX) Setting functions.

3.3.2 Call Forward function: you can setup the phone number you want to forward in this page. There are three type of Forward mode. You can choose All Forward, Busy Forward, and No Answer Forward by click the icon.

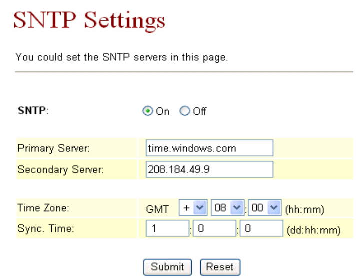
a) All Forward: All incoming call will forward to the number you choosed. You can input the name and the phone number in URL field. If you select this function, then all the incoming call will direct forward to the speed dial number you choose.

b) Busy Forward: If you are on the phone, the new incoming call will forward to the number you choosed. You can input the name and the phone number in URL field.

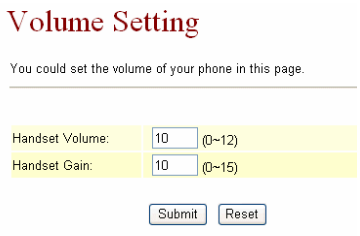
c) No Answer Forward: : If you can not answer the phone, the incoming call will forward to the number you choosed. You can input the name and the phone number in URL field.。



3.3.3 SNTP Setting function: you can setup the primary and second SNTP Server IP Address, to get the date/time information.



3.3.4 Volume Setting function: you can setup the Handset Volume and the Handset Gain.



a) Handset Volume is to set the volume for you can hear from the handset.

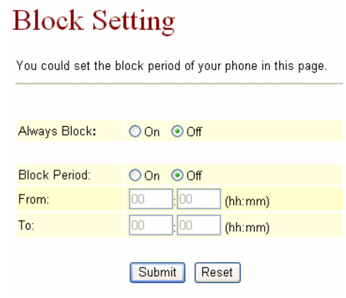
b) Handset Gain is to set the volume send out to the other side's handset.

3.3.5 Block Setting function: you can setup the Block Setting to keep the phone slience. You can choose Always Block or Block a period.

a) Always Block: All incoming call will be blocked until disable this feature.

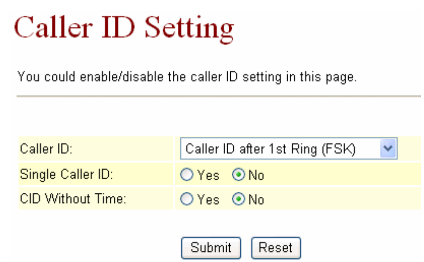
b) Block Period: Set a time period and the phone will be blocked during the time period.

c) When you finished the setting, please click the Submit button.

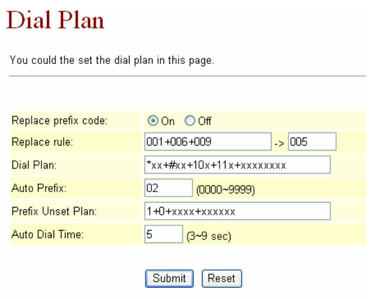


3.3.6 Caller ID function: you can set the device to show Caller ID in your PSTN Phone or IP Phone.

a) There are four selection of Caller ID. You need to base on your environment to set the Caller ID function for FSK or DTMF. When you change the setting, please also double check the PTT seting in Others. You need to choose the correct country code then the Caller ID will be effect.



3.3.7 Dial Plan Setting function: This function is when you input the phone number by the keypad but you don't need to press "#". After time out the system will dial directly.



a) Symbol explan:

x or X	0.1.2.3.4.5.6.7.8.9
+	or

b) Replace rule:If replace prefix code is ON and prefix number is matched with rule then 005 will replace prefix.

c) Auto Dial Time : Stop dialing after seconds then send dial number out.

d) Dial Plan: When match with pattern then send dial number out but if fisrt digit is '0' then dial plan will be ignored.

Example:

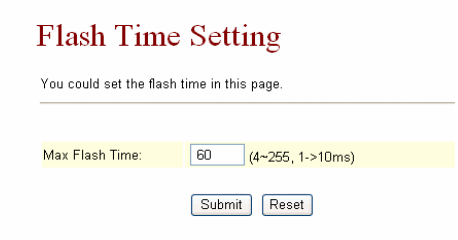
*xx	If matched with one of *00,*01….*99 then will send number out
#xx	If matched with one of #00,#01….#99 then will send number out
10x	If matched with one of 100,101….109 then will send number out
11x	If matched with one of 110,111….119 then will send number out
Xxxxxxxx	If dial with 8 digits then send number out

e) **Auto Prefix : Number for add before dial number.**

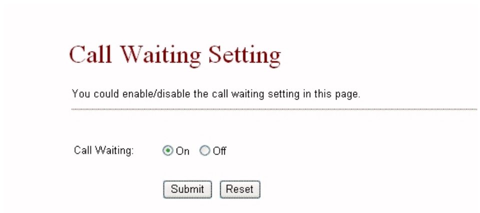
f) Prefix Unset Plan : When first digit or dial numeb match with pattern then ignore auto prefix.

0	Iignore auto prefix if first digit is '0'
1	Ignore auto prefix if first digit is '1'
xxxxx	dial numbers are 4 digits ignore auto prefix
xxxxxx	dial numbers are 5 digits ignore auto prefix

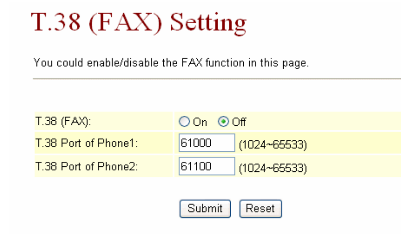
3.3.8 Flash Time Setting function: this function is for you to set the time you press the Hook to represent the Flash function.



3.3.9 Call Waiting Setting function: You can Enable/Disable the Call Waiting function, When you are talking with someone, there is a new incoming call, you will hear the call waiting tone.



3.3.10 T.38 Setting function: You can Enable/Disable the T.38 function.



3.4 IVR Interface for The TA

You can use the PSTN phone to configure the TA. Please follow the instruction to configure your terminal adapter.

3.4.1 Please make sure the TA has connect with Ethernetcable, Phone, and PSTN Line.

3.4.2 If the TA do not register to the SIP Proxy Server, it will switch to PSTN after 30 second. Also if Ethernet or Power fail, the TA will switch to PSTN mode.

3.4.3 Default the IVR setting is disable, if you want to use IVR please input #190# first.

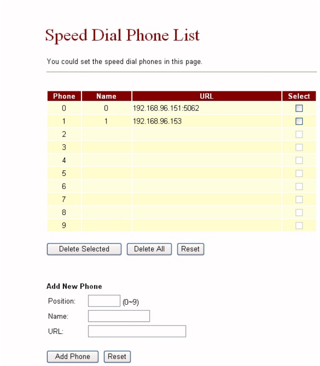
3.4.4 If the TA has connect to PSTN, then you can use**"0*"** to switch from VoIP to the PSTN. If you want to change to VoIP again, you have to "on-hook" and "off-hook "again.

Group	IVR Action	IVR Menu Choice	Parameter(s)	Notes
Function	enable call waiting	**#138#**	None	Enable Call waiting
Function	disable call waiting	**#139#**	None	Disable call waiting
Function	unlock keypad	#190#	None	You have to unlock keypad first, and then you can change the setting by keypad.
Function	lock keypad	**#191#**	None	Lock keypad.
Function	Reboot	**#195#**	None	The system will reboot automatically.
Function	Factory Reset	**#198#**	None	System will automatically Reboot and restore to default setting. WARNING: ALL "User-Changeable" NONDEFAULT SETTINGS WILL BE LOST! This will include network and service provider data.
Info	Check IP Address	**#120#**	None	IVR will report the LAN port IP address
Info	Check IP Type	**#121#**	None	IVR will report the WAN Port DHCP is enabled or disabled.
Info	Check the Phone Number	**#122#**	None	IVR will report current in use VoIP number
Info	Check Network Mask	**#123#**	None	IVR will report the WAN Port network mask
Info	Check Gateway IP Address	**#124#**	None	IVR will report the WAN Port gateway IP address
Info	Check Primary DNS Server Setting	**#125#**	None	IVR will report the WAN Port Primary DNS server IP Address.

Info	Check IP Address	**#126#**	None	IVR will report the WAN port IP address
Info	Check Firmware Version	**#128#**	None	IVR will report the firmware version
Setting	Set DHCP client	**#111#**	None	The system will change the WAN port to DHCP Client type
Setting	Set Static IP Address	**#112xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	WAN port DHCP Client will be disabled and WAN port will change to the Static IP type. Set WAN port IP Address
Setting	Set Network Mask	**#113xxx*xxx*xxx*xxx#**	Enter value using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Must set Static IP first. Set WAN port Network Mask
Setting	Set Gateway IP Address	**#114xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Must set Static IP first. Set WAN port Gateway IP Address
Setting	Set Primary DNS Server	**#115xxx*xxx*xxx*xxx#**	Enter IP address using numbers on the telephone key pad. Use the * (star) key when entering a decimal point.	Must set Static IP first. Set WAN port Primary DNS Server IP Address

Setting	Set Codec	**#130+[1-8]#**	1:G.711 u-Law, 2: G.711 a-Law, 3: G.723.1, 4: G.729a, 5: G.726 16K, 6: G.726 24K, 7: G.726 32K, 8: G.726 40K,	You can set the codec you want to the first priority.
Setting	Set Handset Gain	**#131+[00~15]#**	Handset Gain from 0~15	You can set the Handset gain to proper value, default is 10
Setting	Set Handset Volume	**#132+[00~12]#**	Handset Volume from 0~12	You can set the Handset volume to proper value, default is 10
Setting	TFTP Server IP Address	**#135#**	Set Auto configure TFTP Server IP Address	You can set the TFTP Server IP address
Setting	FTP Server IP Address	**#136#**	Set Auto configure FTP Server IP Address	You can set the FTP Server IP address
Setting	Auto configure mode	**#137+[0~2]#**	0: Disable, 1: TFTP mode, 2: FTP mode	You can set the Auto configuration mode, 0: Disable, 1: use TFTP Server, 2: user FTP Server
Setting	IP mode	**#192#**	Set default use IP mode	Only support 1S1P, provide setting change default setting to IP mode
Setting	PSTN mode	**#193#**	Set default use PSTN mode	Only support 1S1P, provide setting change default setting to PSTN mode

3. 5 Phone Book



3.5.1 In Phone Book contains Speed Dial Settings. You can setup the Speed Dial number. If you want to use Speed Dial you just dial the speed dial number (from 0~9) then press “#”.

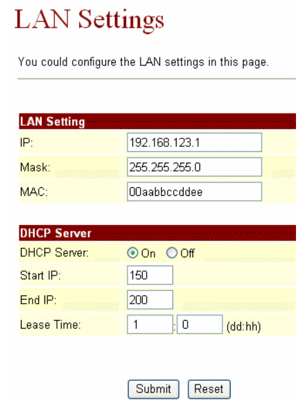
3.5.2 In Speed Dial setting function: you can add/delete Speed Dial number. You can input maximum 10 entries speed dial list.

3. 6 Network Items

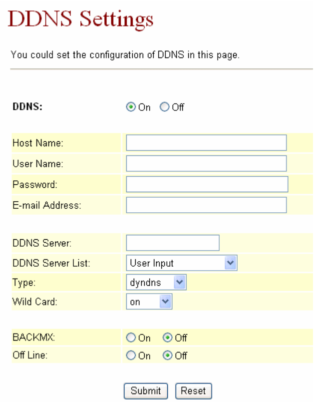
3.6.1 LAN Settings: In this page you can configure the IP Phone LAN port’s setting.

a) The LAN port’s default IP address is 192.168.123.1, Net Mask is 255.255.255.0., and DHCP Server enabled. The start IP address if 150, end IP adress is 200. It is not necessary to change the LAN settings.

b) You can connect your PC to the LAN port, set your PC as DHCP Client mode, then you can get IP address from the TA.



3.6.2 DDNS Setting: You can configure the DDNS setting in this page. You need to have the DDNS account and input the informations properly. You can have a DDNS account with a public IP address then others can call you via the DDNS account. But now most of the VoIP applications are work with a SIP Proxy Server.



3.6.3 VLAN Setting: You can set the VLAN setting in this page. There are two parts in this page. First one is to set the packets related to the TA, and the second parts is if you use the VLAN setting in the NAT Mode.

a) There are two kind of destination packets will come from the TA's WAN port, one kind of packets will go to the TA, the other will go through the LAN port to the PC.

b) VLAN Packets: if you enable the first VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets will be check with the IP Address and

the VID.

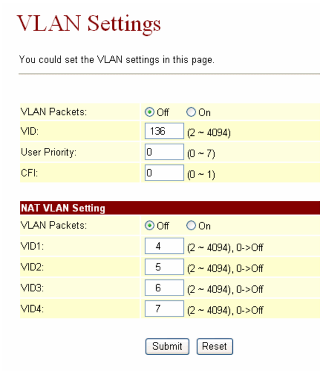
c) VID:You can follow your service provider to set your VID.

d) User Priority: Usually this will be defined by your service provider.

e) CFI: Canonical Format Indicator is always set to zero for Ethernet switches. CFI is used for compatibility reason between Ethernet type network and Token Ring type network. If a frame received at an Ethernet port has a CFI set to 1, then that frame should not be forwarded as it is to an untagged port.

f) When you enable the first VLAN Packets and set the VID, User Priority, and CFI, then all the incoming packets with the TA's IP address and the same VID will be accept by the TA. If the incoming packets with the TA's IP address but the different VID then the packets will be discard by the TA. The Other incoming packets with different IP address will go through the LAN port to the PC.

g) NAT VLAN Setting: When you set your device in NAT mode, the TA can help you to filter the wrong incoming packets. You can separate the other device connectd behind the TA into 4 VLAN group. You can set different VID for these 4 groups. When the incoming packets go through the TA's WAN port then the TA will check the VID, if the packets is not going to the TA(with the TA's IP address and the correct VID), and the VID is not these four VID you set, then the packets will be discard by the TA.



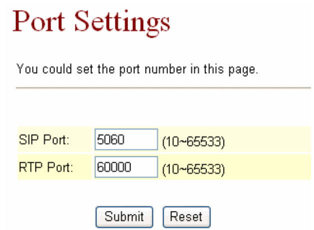
3.7 SIP Settings

3.7.1 In SIP Settings you can setup the Service Domain, Port Settngs, Codec Settings, Codec ID Setting, RTP Setting, RPort Setting and Other Settings. If the VoIP service is provided by ISP, you need to setup the related

informations correctly then you can register to the SIP Proxy Server correctly.

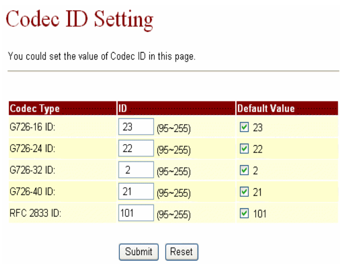
3.7.2 Service Domain Function(see section 2.4)

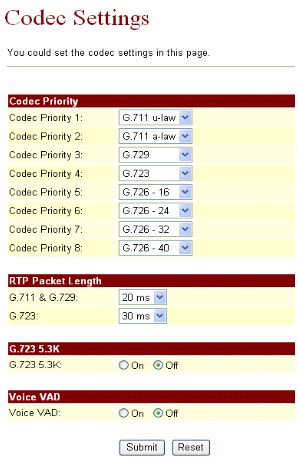
3.7.3 Port Settings: you can setup the SIP and RTP port number in this page. Each ISP provider will have different SIP/RTPport setting, please refer to the ISP to setup the port number correctly.



3.7.4 Codec Settings: you can setup the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP suggestion to setup these items.

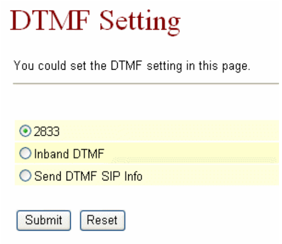
3.7.5 Codec ID Setting: Sometimes 2 VoIP device with different Codec ID will cause the interopability issue. If you are talking with others got some problems, you may ask the other one what kind of Codec ID he use, then you can change your Codec ID.



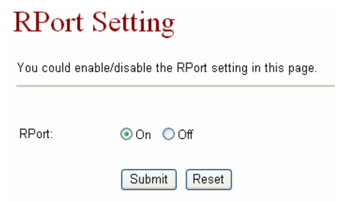


3.7.6 DTMF Setting: you can setup the RFC2833 Out-Band DTMF, Inband DTMF and Send DTMF SIP Info in this page. To change this setting, please following your ISP

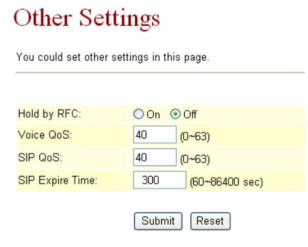
information.



3.7.7 RPort Function: you can setup the RPort Enable/Disable in this page. To change this setting, please following your ISP information.



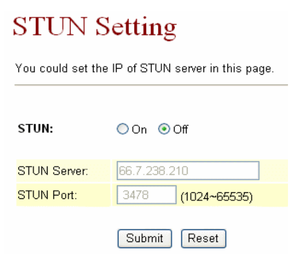
3.7.8 Other Settings: you can setup the Hold by RFC, Voice/SIP QoS and SIP expire time in this page. To change these settings please following your ISP information. The QoS setting is to set the voice packets' priority. If you set the value higher than 0, then the voice packets will get the higher priority to the Internet. But the QoS function still need to cooperate with the others Internet devices.



3.8 NAT Trans

3.8.1 In NAT Trans. you can setup STUN function. These functions can help your TA working properly behind NAT.

3.8.2 STUN Setting: you can setup the STUN Enable/Disable and STUN Server IP address in this page. This function can help your TA working properly behind NAT. To change these settings please following your ISP information.



3.9 Save Change

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