SMCDSP-200/SMCDSP-205 Series

VoIP Phone Administration Guide

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* SMC will provide warranty service for one year following discontinuance from the active SMC price list. Under the limited lifetime warranty, internal and external power supplies, fans, and cables are covered by a standard one-year warranty from date of purchase.

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1 Introduction

This administration guide is for the VoIP Phone, SMCDSP-200 & SMCDSP-205. This administration guide explains the keypad instructions, web configuration and command line configuration for the SMC IP Phones. Before using each device, some setup processes are required to make the VoIP Phone work properly. Please refer to the Setup Menu for further information.

1.1 Hardware Overview

The SMC IP Phones have the following interfaces for Networking and telephone interface. SMCDSP-200 supports POE that can connect to POE switch and use the power of POE switch.

Two RJ-45 networking interfaces; these two interfaces support 10/100Mps Fast Ethernet. You can connect one RJ-45 Fast Ethernet port to the ADSL or Switch, and connect the other one to your computer.

1.2 Software Overview

Network Protocol

- SIP v1 (RFC2543), v2(RFC3261)
- IP/TCP/UDP/RTP/RTCP
- IP/ICMP/ARP/RARP/SNTP
- TFTP Client/DHCP Client/ PPPoE Client
- Telnet/HTTP Server
- DNS Client

Codec

- G.711: 64k bit/s (PCM)
- G.723.1: 6.3k / 5.3k bit/s
- G.726: 16k / 24k / 32k / 40k bit/s (ADPCM)
- G.729A: 8k bit/s (CS-ACELP)
- G.729B: adds VAD & CNG to G.729

Vice Quality

- VAD: Voice activity detection
- CNG: Comfortable noise generator
- LEC: Line echo canceller
- Packet Loss Compensation
- Adaptive Jitter Buffer

Call Function

- Call Hold
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- In-Band DTMF
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- PPPoE

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- HTTP 1.1 basic/digest authentication for Web setup
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QoS

- QoS field

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- STUN

Configuration

- Web Browser
- Console/Telnet
- Keypad

Firmware Upgrade

- TFTP
- Console
- HTTP
- FTP

2 Keypad interface for IP Phone demo system

2.1 Keypad description

Key Name	Description	Note
1	“1”	
2	“2”, “a”, “b”, “c”, “A”, “B”, “C”	
3	“3”, “d”, “e”, ”f”, “D”, “E”, “F”	
4	“4”, “g”, “h”, “I”, “G”, “H”, “I”	
5	“5”, “j”, “k”, “l”, “J”, “K”, “L”	
6	“6”, “m”, “n”, “o”, “M”, “N”, “O”	
7	“7”, “p”, “q”, “r”, “s”, “P”, “Q”, “R”, ‘S”	
8	“8”, “t”, “u”, “v”, “T”, “U”, “V”	
9	“9”, “w”, “x”, “y”, “z”, “W”, “X”, “Y”, “Z”	
0	“0”, “space”	

*	"*", "•", ":", "@"	
#	Start dialing process	
4-way Navigation Keys	Press to scroll through lists and menus on the display.	
MENU	Press to access the menu options or cancel your selection and go back to the previous level.	
ENTER	Press to enter a menu or confirm a selection.	
Phone Book	Press to access the personal phonebook directory.	
REDIAL	Press to call the last number dialed.	
DND	Press to block all incoming calls.	
HOLD	Press to put an active call on hold.	
TRANSFER	Press to transfer an active call to another VoIP phone on the system.	
CONFERENCE	Press to activate the three-way conference call.	
FWD	Press to forward all incoming calls to another phone on the system.	
DELETE	Press to erase the number you dialed when making a call.	
M1~M9	Press any of the keys to speed dial the preset contact number.	
REC	Press REC button to record the conversation, please refer to section 2.2 for further details.	
VOICE MSG	Press to listen to voice mail messages.	
SPEAKER	Press to activate the speakerphone to allow handsfree conversations.	
VOLUMN Control Key	Press to increase or decrease the volume of the ringer tone, handset, or the volume of the current call using the speakerphone.	

2.2 Recording

The recording function is for users who want to record their conversations during the calls. The user only needs to press “REC” button to start recording and press again to stop it. The voice file will be saved in the user’s voice mail system. The Message Waiting Indicator on IP phones will be lit to inform the user.

To listen to the recording, make a call to the user’s voice mail system, press “1” to hear the new messages of the recording.

2.3 Keypad Function and setting List

2.3.1 Phone Book

Name	Description
Search	Search Phone Book.
Add entry	Add new phone number to phone book.
Speed dial	Add speed dial phone number to speed dial list.
Erase all	Erase all phone number from Phone Book.

2.3.2 Call history

Name	Description
Incoming calls	Show all incoming call.
Dialed numbers	Show all dialed call.
Erase record	Delete call history. **All:** Delete all call history. **Incoming:** Delete all incoming call. **Dialed:** Delete all dialed out call.

2.3.3 Phone setting

Name	Description	
Call forward	All Forward	**Activation:** To Enabled/Disabled this function. **Number:** Forward to a Speed Dial Number.

	Busy Forward	**Activation:** To Enabled/Disabled this function. **Number:** Forward to a Speed Dial Number.
	No Answer Forward	**Activation:** To Enabled/Disabled this function. **Number:** Forward to a Speed Dial Number.
	Ring Timeout	Set the Ring times to start the no answer forward function, ex: 2 means after 2 rings then forward to the dedicated number.
Do not Disturb	Always	Block all phone calls.
	By Period	Block all phone calls at a certain period of time.
	Period Time	Set the start time and end time to Block Setting.
Alarm setting	Activation	Set the Alarm Enabled or Disabled.
	Alarm time	Set the time for alarming.
Date/Time setting	Date and Time Setting.	
	Date & Time	Set the IP Phone Date and Time.
	Time format	To set the time as 12-hour or 24-hour clock.
	SNTP setting	
	SNTP	Enabled / Disable SNTP.
	Primary SNTP	Set Primary SNTP server IP address.
	Secondary SNTP	Set Secondary SNTP server IP address.
	Time zone	Set Time zone.
	Adjustment Time	Set adjustment time period.
Volume and Gain	Handset volume	Set Handset volume from 0~15 (max.) for you to hear.
	Speaker volume	Set Speaker phone volume from 0~15 (max.) for you to hear.
	Handset gain	Set Handset Gain from 0~15 (max.) for the other site to hear.
	Speaker gain	Set Speaker phone Gain from 0~15 (max.) for the other site to hear.
Ringer	Ringer volume	Ringer volume setting from 0~15 (max.).
	Ringer type	Ringer tone selection from 1~4.
Auto dial	Set Auto Dial time from 1~5 seconds.	

2.3.4 Network

Name	**Description**

WAN Setup	IP Type	**Fixed IP client:** self configure the IP address. **DHCP client:** to get IP address through DHCP. **PPPoE client:** to get IP address through PPPoE.
	Fixed IP setting	**IP Address:** the IP address of the IP phone. **Subnet mask:** configure subnet mask. **Default Gateway:** the gateway address **MAC address:** the MAC address
	PPPoE setting	**User name:** user name of PPPoE **Password:** password of PPPoE
LAN Setup	Bridge	Set LAN as bridging mode
	NAT	Set LAN as NAT mode
DNS	Primary DNS	First DNS address
	Secondary DNS	Second DNS address
VLAN	Activation	Activate or disable the VLAN.
	VID	Set VID from 2 to 4094.
	Priority	Set the priority from 0 to 7.
	CFI	0~1
Status	Show WAN, LAN IP address and MAC address.	

2.3.5 SIP Settings

If you want to use keypad to set the SIP setting, you have to go to item 8 Sys. Authority to input the password, or you can not change the SIP setting.

Name	**Description**	
Service domain	First/Second/Third realm	The realms include following information. You can press “1*#”, “2*#”, and ”3*#” to change among these three SIP realms. **Activation:** to Activate or stop the realm. **User name:** the SIP’s user name. **Display name:** the SIP’s display name. **Register name:** the SIP’s registered name. **Register password:** the SIP’s password. **Proxy server:** the address of SIP proxy.

		Domain server: the address of domain server. **Outbound proxy:** the address of outbound proxy.
Codec	Codec type	The codec type includes G.711 uLaw, G.711 aLaw, G.723, G.729, G.726-16, G.726-24, G.726-32, and G.726-40.
	VAD	Voice Active Detection Enable/Disable.
RTP setting	Outband DTMF	Enable/Disable outband DTMF.
	Duplicate RTP	**No duplicate:** do not resend the voice packages. **One duplicate:** resend voice packages one time. **Two duplicate:** resend voice packages two times.
RPort Setting	RPort Enabled/Disabled	
Hold by RFC	Enable/Disable Holding the calls, according to RFC3261.	
Status	Show the SIP Proxy register status. You can use UP/Down key to check each Realm's status.	

2.3.6 NAT Traversal

If you want to use keypad to set the NAT Traversal settings, you have to go to item 8 Sys. Authority to input the password, or you can not change the NAT Traversal settings.

Name	Description	
STUN setting	STUN	Enable/Disable STUN.
	STUN server	The address of STUN server

2.3.7 Administrator

If you want to use keypad to set the Administrator setting, you have to go to item 8 Sys. Authority to input the password, or you can not change the Administrator setting.

Name	Description	
Upgrade system	This function must work with the SMC IP-PBX.	
	Upgrade Now	Select to direct connect to IP PBX to check if there is any upgrade version. If there is a newer version, the IP phone will upgrade the system automatically.
	Schedule State	Select to see the current status and scheduling time.
	Schedule Stop	Select to stop or start scheduling update.

Default setting	You can restore to the default setting.
Change passwd	Press ENTER and press a new password to replace the original password with the new one.
Version	This will show the system’s firmware version.
Vendor ID	To see the vendor ID of the IP Phone. The vendor ID of SMCDSP-200 is dsp200 and SMCDSP-205 is dsp205.
Watch dog	You can use this to enable Watch Dog function to do the debugging.
Restart	You can use this function to restart your IP Phone.

2.3.8 Sys. Authority

To do the SIP setting, NAT traversal and Administrator from Keypad, you need to input the password first. Default is “9876”.

3 Setup the VoIP Phone via a Web Browser

Default the IP Phone's Bridge is enabled, WAN port is in DHCP Client Mode. LAN port is the same as WAN. Connect the IP Phone to DHCP server, and the server will assign an IP address to the phone. Check the IP address and add the port number ":9999" at the end of the IP address to access the web browser.

Note: When selecting DHCP Client and cannot detecting the IP, the IP Phone will provide a default IP, 192.168.2.25.

Note: It is highly recommended to use Internet Explorer 6.0 for web configurations.

3.1 Login

Please input the username and password into the blank field. The default setting is:

1. For the Administrator, the username is: *admin*; and the password is: *smcadmin*. If you use the account login, you can configure the setting.
2. For a normal user, the username is: *user*; and the password is: *test*. If you use the account login, but you can not configure the SIP setting.

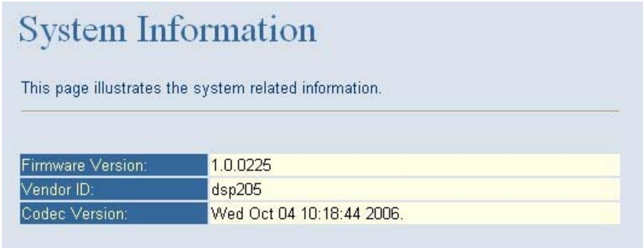
Click **Login** to move into the VOIP PHONE web based management information page.

Any changes in the Web Management interface except for Phone Book require clicking **Submit** in that page, and then go to the Save Change page and click **Save**. The system will restart, and all the settings can work properly.

3.2 System Information for the IP PHONE

When you login to the web page, you can see the current system information of the IP phone, like Firmware Version, Vendor ID and Codec Version in this page.

Also you can see the function lists in the left side. You can use mouse to click the function you want to set up.

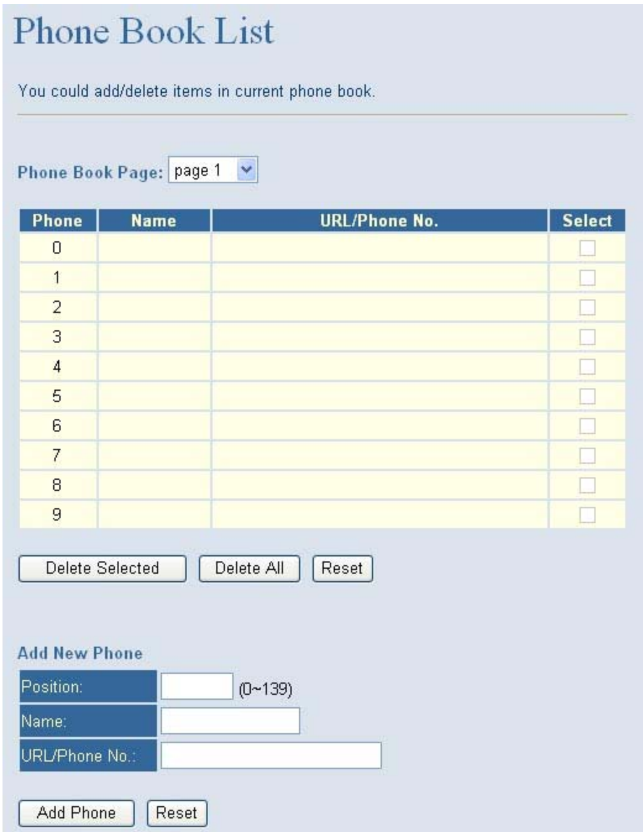


3.3 Phone Book

The Phone Book contains Phone Book and Speed Dial Settings. You can setup the Phone Book and Speed Dial numbers. The Phone Book can store 140 phone numbers and the Speed Dial can store 9 phone numbers. If you want to use Speed Dial you just dial the speed dial number then press “#”.

3.3.1 Phone Book

In the Phone Book function, you can add/delete the phone number in the phone book list. You can add a maximum of 140 entries phone book list.



If you need to add a phone number into the phone book, you need to enter the position, the name, and the phone number (by URL type). When you are finished a new phone list, just click **Add Phone**.

If you want to delete a phone number, you can select the phone number you want to delete then click **Delete Selected**.

If you want to delete all phone numbers, you can click **Delete All**.

3.3.2 Speed Dial List

In Speed Dial List function, you can add/delete a Speed Dial number. You can input a maximum of 9 entries into the speed dial list.

If you need to add a phone number into the Speed Dial list, you need to input the position, the name, and the phone number (by URL type). When you have finished a new phone list, just click **Add Phone**.

If you want to delete a phone number, you can select the phone number you want to delete then click **Delete Selected**.

If you want to delete all phone numbers, you can click **Delete All**.

3.4 Phone Setting

Phone Setting contains Call Forward, SNTP Settings, Volume Settings, Ringer Settings, DND Settings, Dial Plan Settings, Call Waiting Settings, Soft-key Setting functions, Hot Line Settings and Alarm Settings.

3.4.1 Forward Settings

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 clicking the icon.

3.4.1.1All Forward

All incoming calls will forward to the number you select. You can input the name and the phone number in the URL field. If you select this function, then all the incoming call will directly forward to the speed dial number you choose.

3.4.1.2Busy Forward

If you are on the phone, the new incoming call will forward to the number you select. You can input the name and the phone number in the URL field.

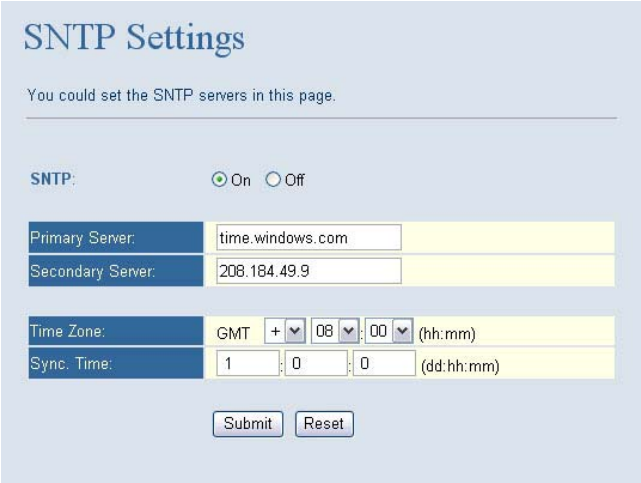
3.4.1.3No Answer Forward

If you can not answer the phone, the incoming call will forward to the number you select. You can input the name and the phone number in the URL field. Also you have to set the Time Out time for the system to start to forward the call to the number you select.

When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

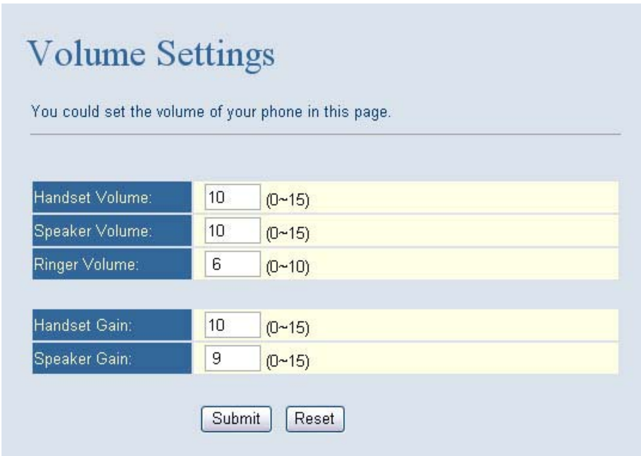
3.4.2 SNTP Settings

You can setup the primary and second SNTP Server IP Address, to get the date/time information. Also you can base it on your location to set the Time Zone, and how long it needs to synchronize again. When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.



3.4.3 Volume Settings

You can setup the Handset Volume, Speaker Volume, Ringer Volume, the Handset Gain, and Speaker Gain.



Handset Volume is to set the volume you hear from the handset.

Speaker Volume is to set the volume you hear from the speaker phone.

Ringer Volume is to set the ringer volume.

Handset Gain is to set the volume send out from the handset.

Speaker Gain is to set the volume send out from the micro phone.

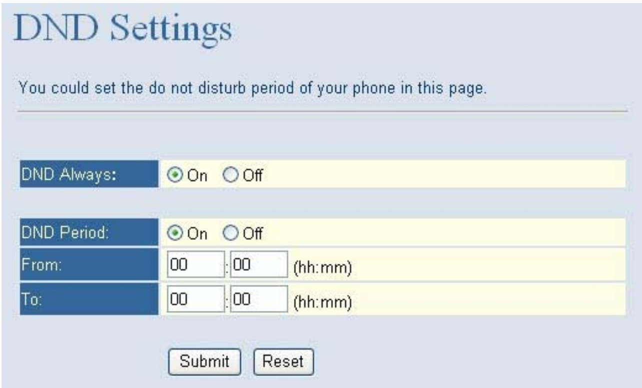
When finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

3.4.4 Ringer Settings

You can select the melody for the incoming calls. When you have finished the setting, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

3.4.5 DND Settings

You can setup the DND Setting to keep the phone silent. You can choose **DND Always** or **DND Period**.



DND Always: All incoming call will be blocked until you disable this feature.

DND Period: Set a time period and the phone will be blocked during the time period. If the "From" time is larger than the "To" time, the Block time will from Day 1 to Day 2.

When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

3.4.6 Dial Plan Setting

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

3.4.6.1Symbol explan:

x or X	0,1,2,3,4,5,6,7,8,9
+	or

Drop Prefix: The default is **No** for adding the prefix to the dial plan in **Replace Rule**. Click **Yes** to delete the prefix of the dial plan in **Replace Rule**.

Replace Rule 1~4: Enter rule for matching.

Example:

Drop Prefix	Replace Rule		Description
No	002	8613+8662	When the dialed number begins with “8613” or “8662”, the number will be added “002” to call out.
No	002	12	When the dialed number begins with “12”, the number will be added “002” to call out.
No	002	5xxx	When the dialed number begins with “5”, and has total of four digits, the number will be add “002” to call out.
Yes	002	003+004+005	When the dialed number begins with “003”, “004” or “005”, the number will be dropped “003”, “004” or “005”, and then add “002” to call out.

Yes	002	55xxxx	When the dialed number begins with “55” and has total of six digits, the number will be dropped “55”, and add “002” to call out.

Auto Dial Time: The default is 5 seconds. Enter a time in sec for auto dial after the time is up.

When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

3.4.7 Call Waiting Settings

If the user doesn’t want to be informed there is a new incoming call, the user can set the function to off. When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

3.4.8 Soft-key Settings

The SMC IP Phone supports soft-key settings for voice messages. You can press SPEAKER, and then VOICE MSG on the IP phone to dial out the entered number in order to get into the voice mail service.

Voice mail key: enter a serial of number for listening to the voice messages.

When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

3.4.9 Hot Line Settings

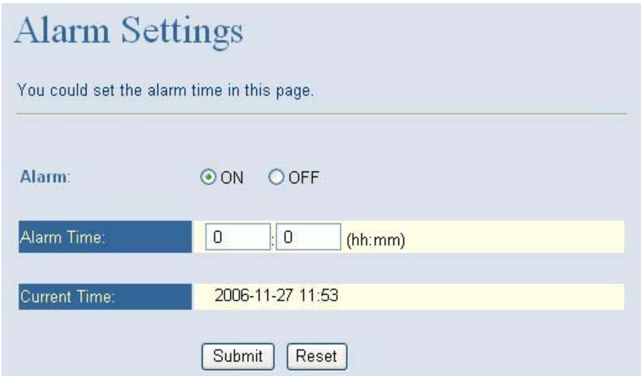
Hot line setting is a special line for the IP Phone. When you enable the function, the phone will directly dial out the number once the phone is picked up.

Hot Line Number: Enter a phone number as a special line.

When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

3.4.10 Alarm Settings

Alarm Settings is to inform a user at a certain time. Click **ON** and then enter the time for ringing.



Alarm Time: enter a time for reminding.

Current Time: display the current time of the IP phone.

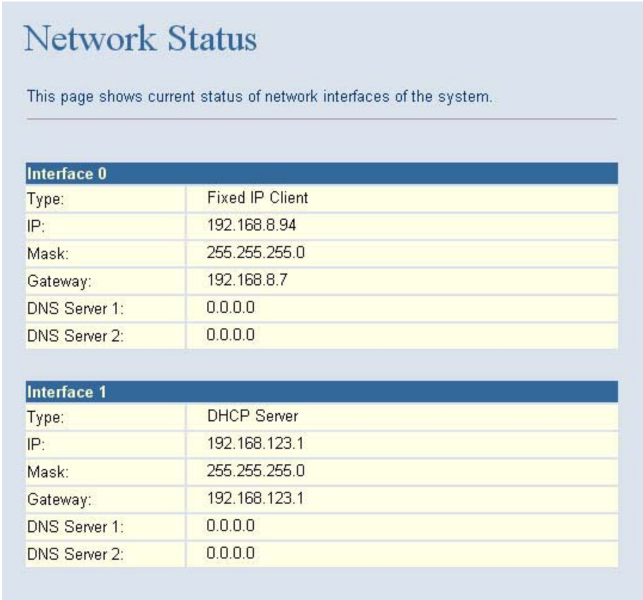
When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

3.5 Network

In Network, you can check the Network status, configure the WAN Settings, LAN Settings, DDNS settings, VLAN Settings, DMZ Settings and Virtual Server.

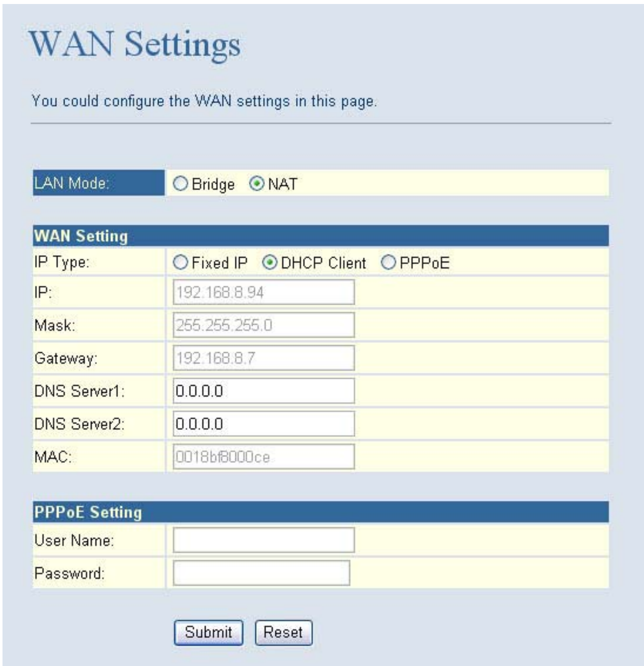
3.5.1 Network Status

You can check the current Network settings in this page.



3.5.2 WAN Settings

In this page, you can configure the IP Phone's WAN port setting. The WAN port is for you to connect to the ADSL Router, or Broadband Router. Also, you can use PPPoE to get the WAN IP address from your ISP.



The default setting is **Bridge** mode. If you don't need to use the Bridge mode, you can change to **NAT** mode.

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

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.

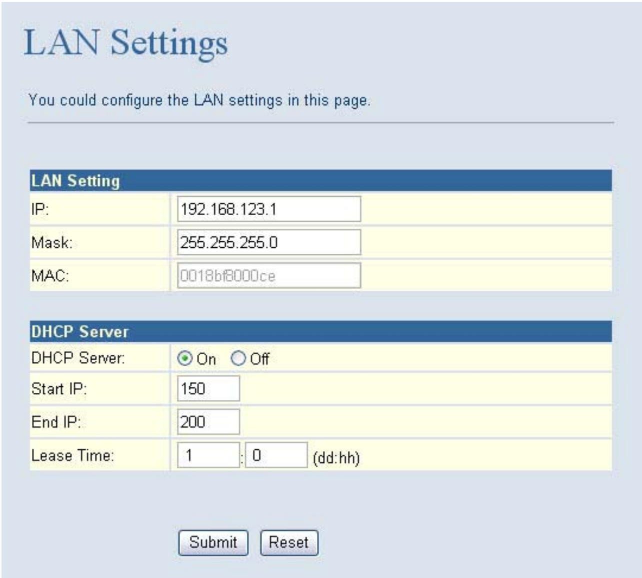
If you change the WAN port's setting to **PPPoE** mode, you have to input a correct username/password to get the IP address from your Internet Service Provider.

When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

Note: When selecting DHCP Client and cannot detecting the IP, the IP Phone will provide a default IP, 192.168.2.25.

3.5.3 LAN Settings

In this page, you can configure the IP Phone LAN port's setting.



The LAN settings' default does not effect as **Bridge** mode is enabled. To configure LAN settings, go to **Network** -> **WAN Settings**, and click **NAT** under **LAN Mode**.

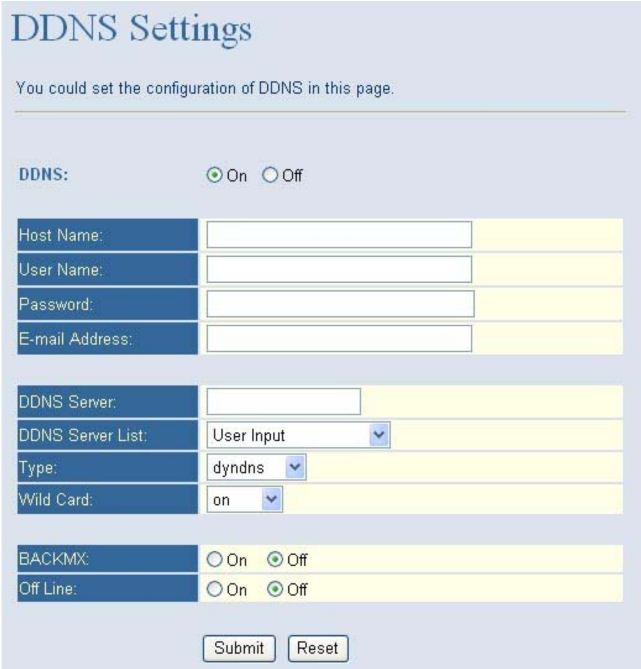
You can connect your PC to the LAN port, set your PC as DHCP Client mode, and then you can get IP address from the Phone.

When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

3.5.4 DDNS Settings

You can configure the DDNS setting in this page. You need to have the DDNS account and input the information properly. You can have a DDNS account with a public IP address then others can call you via the DDNS account.

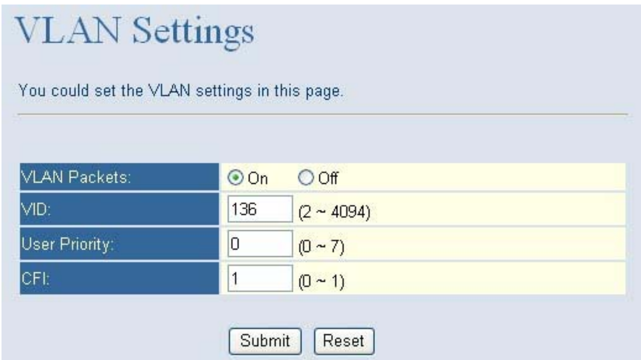
But now most of the VoIP applications work with a SIP Proxy Server. When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.



3.5.5 VLAN Settings

You can set the VLAN setting to set the packets related to the IP Phone.

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



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

VID: You can follow your service provider or Network settings to set your VID.

User Priority: Defines user priority, giving eight (2^3) priority levels. IEEE 802.1P defines the operation for these 3 user priority bits. Usually this will be defined by your service provider.

CFI: Canonical Format Indicator is always set to zero for Ethernet switches. CFI is used for compatibility reasons between Ethernet type networks and Token Ring type networks. 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.

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

When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

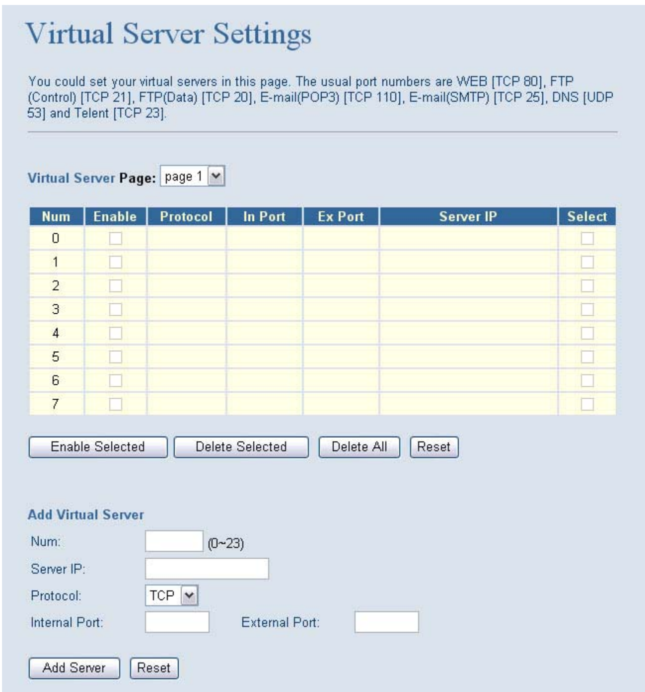
3.5.6 DMZ Setting

You can enable the DMZ Setting of the IP phone. Click **ON**, and enter an IP address of the PC in **DMZ Host IP**.

When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

3.5.7 Virtual Server

The SMC IP Phone supports configuring a virtual server function.



If you need to add a virtual server into the Virtual Server list, you need to enter the num, server IP, internal/external port, and select a protocol. When you have finished a new phone list, just click **Add Server**.

If you want to enable a virtual server, you can select the virtual server you want to enable, and then click **Enable Selected**.

If you want to delete a virtual server, you can select the virtual server you want to delete, and then click **Delete Selected**.

If you want to delete all virtual servers, you can click **Delete All**.

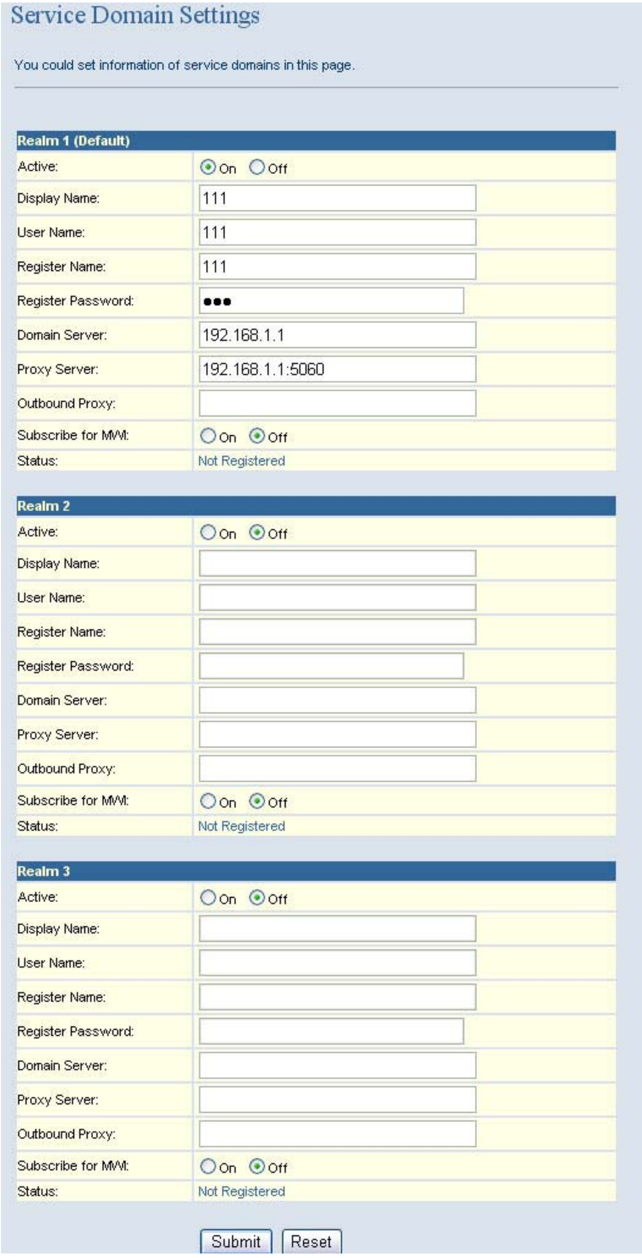
When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

3.6 SIP Settings

In SIP Settings, you can setup the Service Domain, Port Settings, Codec Settings, Codec ID Settings, RTP Setting, RPort Setting and Other Settings. If the VoIP service is provided by an ISP, you need to setup the related information correctly then you can register to the SIP Proxy Server correctly.

3.6.1 Service Domain

In the Service Domain Function, you need to input the account and the related information in this page. Please refer to your ISP provider or Network Administrator. You can register three SIP accounts in the VoIP Phone.



First you need click **ON** in **Active** to enable the Service Domain, and you can input the following items:

Display Name: you can input the name you want to display.

User Name: you need to input the User Name you get from your ISP or Administrator.

Register Name: you need to input the Register Name get from your ISP or Administrator.

Register Password: you need to input the Register Password get from your ISP or Administrator.

Domain Server: you need to input the Domain Server get from your ISP or Administrator.

Note: The default SIP port of the Domain Server is 5060. If you want to change the SIP port, specify the port here.

Example: 192.168.1.100 (Assume Domain server SIP port = 5060)

192.168.1.100:5678 (change the SIP port to 5678)

Proxy Server: you need to input the Proxy Server get from your ISP or Administrator.

Outbound Proxy: you need to input the Outbound Proxy get from your ISP or Administrator. If your ISP does not provide the information, then you can skip this item.

Subscribe for MWI: you can click ON for the IP phone to ask for MWI periodically.

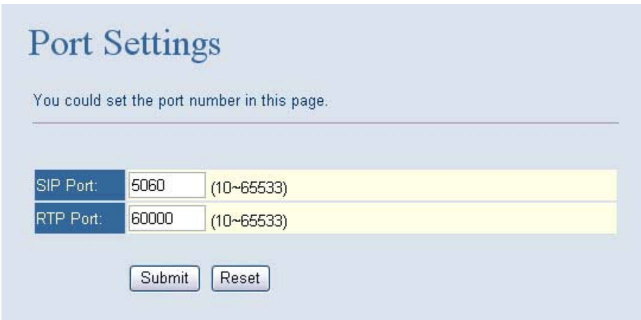
You can see the Register Status in the Status item. If the item shows “Registered”, then your VoIP Phone is registered to the ISP or Network, you can make a phone call directly.

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

When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

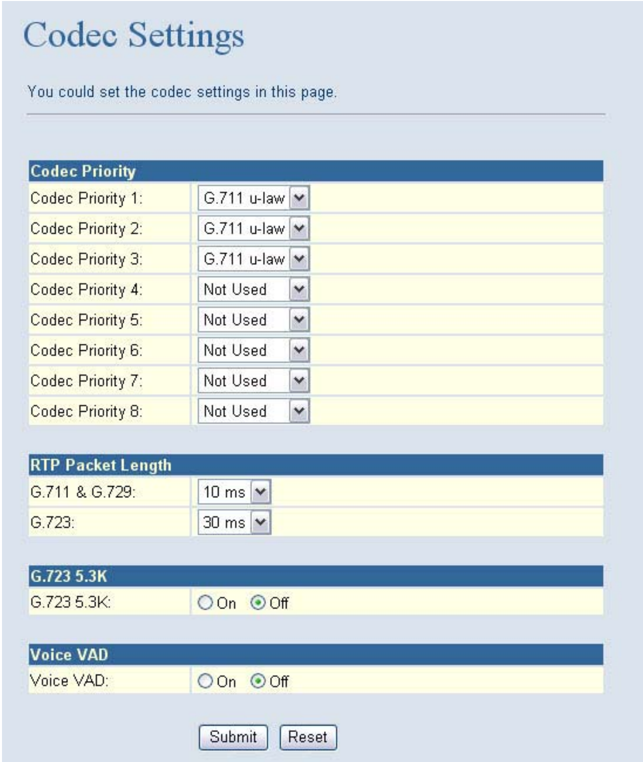
3.6.2 Port Settings

You can setup the SIP and RTP port numbers in this page. Each ISP provider will have different SIP/RTP port settings. When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.



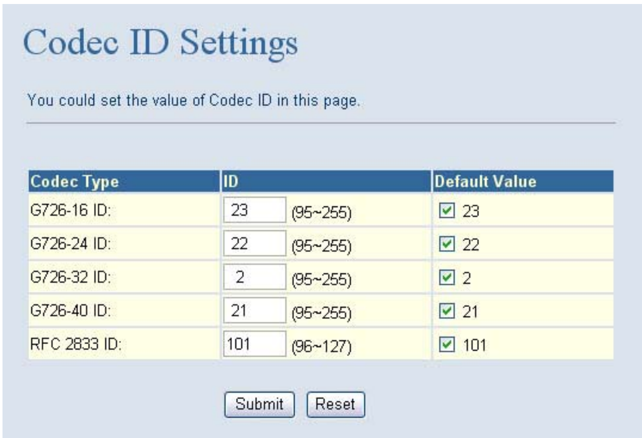
3.6.3 Codec Settings

You can setup the Codec priority, RTP packet length, and VAD function in this page. You need to follow the ISP or Administrator suggestions to setup these items. When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.



3.6.4 Codec ID Settings

You can set the Codec ID to meet the other device's requirement. When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

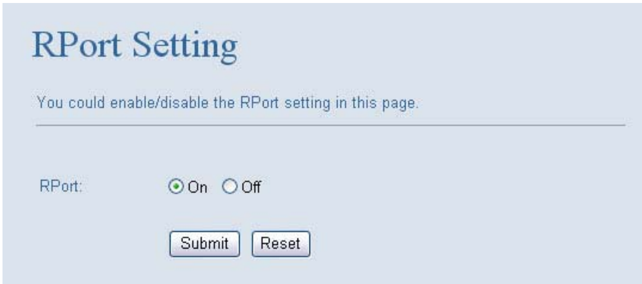


3.6.5 DTMF Settings

You can setup the RFC2833 Out-Band DTMF, Inband DTMF and Send DTMF SIP Info in this page. To change this setting, please follow your ISP or Administrator information. When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

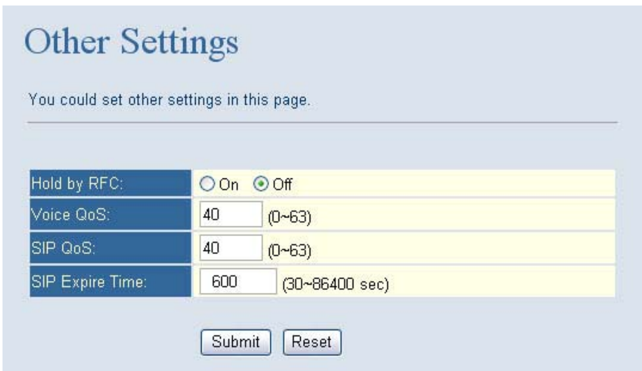
3.6.6 RPort Function

You can setup the RPort Enable/Disable in this page. To change this setting, please follow your ISP information. When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.



3.6.7 Other Settings

You can setup the Hold by RFC, Voice/SIP QoS and SIP Expire Time in this page. To change these settings please follow 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. Note that the QoS function still needs to cooperate with the other Internet devices you connect to. When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.



3.7 NAT Trans

In NAT Trans, you can setup the STUN function. These functions can help your VoIP Phone work properly behind a NAT device.

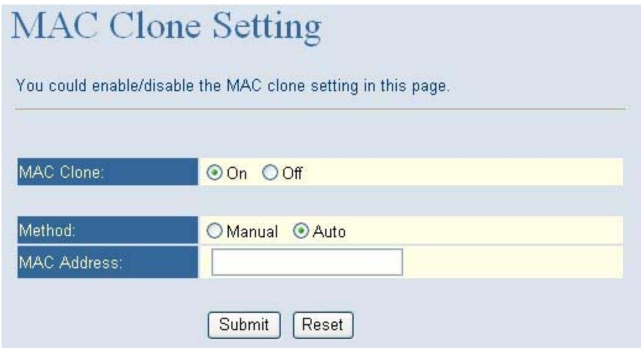
STUN Setting: you can setup the STUN Enable/Disable and STUN Server IP address in this page. This function can help your VoIP Phone working properly behind NAT. To change these settings please following your ISP information. When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

3.8 Others

In Others, you can setup MAC Clone Setting, Tones Settings and Advanced Settings.

3.8.1 MAC Clone Setting

When connecting to ISP via PPPoE, the MAC Clone function can copy the PC's MAC address to IP.



Method: you can manually entered a MAC address or let the IP phone automatically detect the PC's MAC.

MAC Address: When you choose **Manual**, enter a MAC address for the WAN port.

When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

3.8.2 Tones Settings

Tones Settings function let's you configure the tones of various sounds.

When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

	Dial Tone	Ring Back Tone	Busy Tone	Error Tone	Ring Tone	Insert Tone
Cadence On:	☐	☑	☑	☑	☑	☑
Hi-Tone Freq.:	440	480	620	620	480	440
Lo-Tone Freq.:	350	440	480	480	440	350
Hi-Tone Gain:	4522	2261	2261	2261	15360	2261
Lo-Tone Gain:	2261	2261	2261	2261	15360	1130
On Time 1:	0	200	50	30	200	30
Off Time 1:	0	400	50	20	400	20
On Time 2:	0	0	0	0	0	30
Off Time 2:	0	0	0	0	0	400
On Time 3:	0	0	0	0	0	0
Off Time 3:	0	0	0	0	0	0

3.8.3 Advanced Settings

You can configure some more advanced settings such as **ICMP Not Echo**, **Send Anonymous CID**, **Send Flash Event**, and **SIP Encrypt** in this page.

ICMP Not Echo: setup the ICMP echo Enable/Disable. This function can disable echo when someone pings this device, it can avoid a hacker trying to attack the device.

Send Anonymous CID: click **Yes** for the IP phone not to send the phones caller ID out.

Send Flash Event: click **DTMF EVENT** or **SIP INFO** for the flash event.

SIP Encrypt: provide four kinds of SIP encryption format, INFINET, AVS, WALKERSUN1 and WALKERSUN2.

Click one of them to enable SIP encryption for the format.

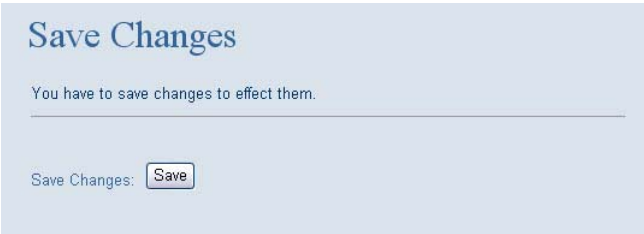
When you have finished the settings, click **Submit**. Go to the Save Change page and click **Save** to reflect the changes.

3.9 System Auth

In System Authority, you can change your login name and password.

3.10 Save Changes

In Save Change, you can save the changes you have done. If you want to use new settings in the VoIP Phone, you have to click **Save**. After you click **Save**, the VoIP Phone will automatically restart and the new settings will take effect.



3.11 Update

In Update, you can update the VoIP Phone's firmware to the new one or perform a factory reset to reset the VoIP Phone back to default settings.

3.11.1 Update Firmware

In New Firmware function you can update new firmware via HTTP. You can upgrade the firmware from your local PC or TFTP.

Update from Local PC:

Click **Browse** at the right the File Location to select a firmware or you can type the correct path and the filename in File Location blank.

Select the correct file you want to download to the VoIP Phone, and then click **Update**.

Update from TFTP:

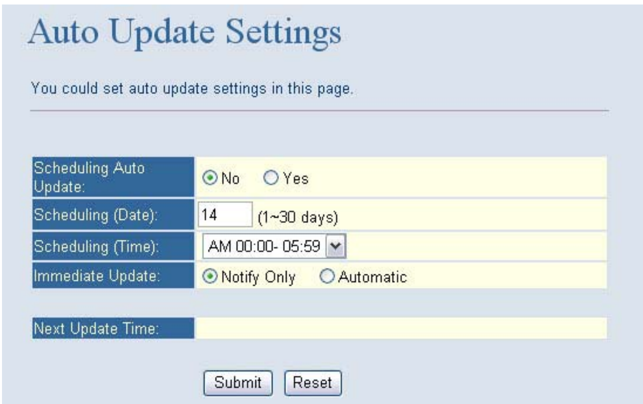
Enter the address of the TFTP server, and then click **Update**.

Note: Do not change the firmware file name, otherwise the system will reject it.

Note: For TFTP server must contain updatelist.dat which reveals the intended update filename.

3.11.2 Auto Update Settings

SMC IP Phones provide an automatic update function. Once enabled, the phone will check the updates at the time in order to have the latest version of firmware. Note that the function must be in DHCP Client mode.



Note: Please check if the IP PBX supports the function.

Scheduling Auto Update: click **Yes** to enable the function.

Scheduling (Date): enter a duration in days for the phone to check the firmware.

Scheduling (Time): select a period of time for updates.

Immediate Update: When the IP phone detects there is newer version of firmware, you can select either to send out a notification only or directly update to the newer version.

3.11.3 Restore Default Settings

In Default Settings, you can restore the VoIP Phone to factory default in this page. You can just click **Restore**, and the VoIP Phone will restore to default and automatically restart again.

3.12 Reboot

The Reboot function is to restart the VoIP Phone. If you want to restart the VoIP Phone, you can just click Reboot, and then the VoIP Phone will restart automatically.

4 Automatic Client Configurations with SMC IP PBX

Auto Client Configuration (ACC) function can be used to download the original configurations stored in the SMC IP-PBX. This is very useful for the administrator who needs to setup large amounts of VoIP phones. The administrator can set a new user account in the IP PBX web page. Once the VoIP phone is connected to the IP PBX, it automatically downloads a predefined configuration setting from SMC IP-PBX. ACC function is enabled in the IP Phone default settings. The vendor ID of SMCDSP-200 is dsp200 and SMCDSP-205 is dsp205.

5 Appendix: Specifications

Network Protocol

- SIP v1 (RFC2543), v2 (RFC3261)
- IP/TCP/UDP/RTP/RTCP
- IP/ICMP/ARP/RARP/SNTP
- FTP/ DNS/ TFTP/ DHCP/ PPPoE Client
- HTTP /NAT/ DHCP server

Tone

- Ring Tone
- Ring Back Tone
- Dial Tone
- Busy Tone
- Programming Tone

Phone Functions

- Volume Adjustment
- Speed Dial
- Phone Book
- Flash
- Speaker Phone
- Call History
- Caller ID
- Voice Message / Incoming Call Indicator

Codec

- G.711: 64k bit/s (PCM)
- G.723: 6.3k / 5.3k bit/s
- G.726: 16k / 24k/ 32k/ 40k bit/s (ADPCM)
- G.729A: 8k bit/s (CS-ACELP)
- G.729B: adds VAD & CNG to G.729

IP Assignment

- Static IP
- DHCP
- PPPoE

SIP Server

- Registrar Server (three SIP account)

Voice Quality

- VAD: Voice activity detection
- CNG: Comfortable noise generator
- LEC: Line echo canceller
- Packet Loss Compensation
- Adaptive Jitter Buffer Security

Security

- HTTP 1.1 basic /digest authentication for web setup
- MD5 for SIP authentication (RFC 2069/RFC 2617)

Qos

- IEEE 802.1q VLAN
- ToS field

Call Function

- Call Holding
- Call Waiting
- Call Forwarding
- Caller Transferring
- Call Blocking
- Call Redial
- Three-way conference

NAT Transversal

- STUN
- Outbound Proxy

Configuration

- Web browser
- TFTP
- Keypad

DTMF Function

- In-band DTMF
- Out-of-band DTMF
- SIP information

Firmware Upgrade

- TFTP
- HTTP