SMCDSP-200/SMCDSP-205 Series

VoIP Phone Administration Guide

Manual abstract:

@@@@@@SMC will endeavor to repair or replace any product returned under warranty within 30 days of receipt of the product. The standard limited warranty can be upgraded to a Limited Lifetime warranty by registering new products within 30 days of purchase from SMC or its Authorized Reseller. Registration can be accomplished via the enclosed product registration card or online via the SMC web site. Failure to register will not affect the standard limited warranty. The Limited Lifetime warranty covers a product during the Life of that Product, which is defined as the period of time during which the product is an 'Active' SMC product. A product is considered to be 'Active' while it is listed on the current SMC price list. As new technologies emerge, older technologies become obsolete and SMC will, at its discretion, replace an older product in its product line with one that incorporates these newer technologies. At that point, the obsolete product is discontinued and is no longer an 'Active' SMC product. All products that are replaced become the property of SMC. Replacement products may be either new or reconditioned.*

Any replaced or repaired product carries either a 30-day limited warranty or the remainder of the initial warranty, whichever is longer. SMC is not responsible for any custom software or firmware, configuration information, or memory data of Customer contained in, stored on, or integrated with any products returned to SMC pursuant to any warranty. Products returned to SMC should have any customer-installed accessory or add-on components, such as expansion modules, removed prior to returning the product for replacement. SMC is not responsible for these items if they are returned with the product. Customers must contact SMC for a Return Material Authorization number prior to returning any product to SMC.

Proof of purchase may be required. Any product returned to SMC without a valid Return Material Authorization (RMA) number clearly marked on the outside of the package will be returned to customer at customer's expense. For warranty claims within North America, please call our toll-free customer support number at (800) 762-4968. Customers are responsible for all shipping charges from their facility to SMC. SMC is responsible for return shipping charges from SMC to customer.

*WARRANTIES EXCLUSIVE: IF AN SMC PRODUCT DOES NOT OPERATE AS WARRANTED ABOVE, CUSTOMER'S SOLE REMEDY SHALL BE REPAIR OR REPLACEMENT OF THE PRODUCT IN QUESTION, AT SMC'S OPTION. THE FOREGOING WARRANTIES AND REMEDIES ARE EXCLUSIVE AND ARE IN LIEU OF ALL OTHER WARRANTIES OR CONDITIONS, EXPRESS OR IMPLIED, EITHER IN FACT OR BY ii 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A OPERATION OF LAW, STATUTORY OR OTHERWISE, INCLUDING WARRANTIES OR CONDITIONS OF MERCHANTABILITY AND FITNESS FOR A PARTICULAR PURPOSE. SMC NEITHER ASSUMES NOR AUTHORIZES ANY OTHER PERSON TO ASSUME FOR IT ANY OTHER LIABILITY IN CONNECTION WITH THE SALE, INSTALLATION, MAINTENANCE OR USE OF ITS PRODUCTS. SMC SHALL NOT BE LIABLE UNDER THIS WARRANTY IF ITS TESTING AND EXAMINATION DISCLOSE THE ALLEGED DEFECT IN THE PRODUCT DOES NOT EXIST OR WAS CAUSED BY CUSTOMER'S OR ANY THIRD PERSON'S MISUSE, NEGLECT, IMPROPER INSTALLATION OR TESTING, UNAUTHORIZED ATTEMPTS TO REPAIR, OR ANY OTHER CAUSE BEYOND THE RANGE OF THE INTENDED USE, OR BY ACCIDENT, FIRE, LIGHTNING, OR OTHER HAZARD. LIMITATION OF LIABILITY: IN NO EVENT, WHETHER BASED IN CONTRACT OR TORT (INCLUDING NEGLIGENCE), SHALL SMC BE LIABLE FOR INCIDENTAL, CONSEQUENTIAL, INDIRECT, SPECIAL, OR PUNITIVE DAMAGES OF ANY KIND, OR FOR LOSS OF REVENUE, LOSS OF BUSINESS, OR OTHER FINANCIAL LOSS ARISING OUT OF OR IN CONNECTION WITH THE SALE, INSTALLATION, MAINTENANCE, USE, PERFORMANCE, FAILURE, OR INTERRUPTION OF ITS PRODUCTS, EVEN IF SMC OR ITS AUTHORIZED RESELLER HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES. SOME STATES DO NOT ALLOW THE EXCLUSION OF IMPLIED WARRANTIES OR THE LIMITATION OF INCIDENTAL OR CONSEQUENTIAL DAMAGES FOR CONSUMER PRODUCTS, SO THE ABOVE LIMITATIONS AND EXCLUSIONS MAY NOT APPLY TO YOU. THIS WARRANTY GIVES YOU SPECIFIC LEGAL RIGHTS, WHICH MAY VARY FROM STATE TO STATE. NOTHING IN THIS WARRANTY SHALL BE TAKEN TO AFFECT YOUR STATUTORY RIGHTS. * 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. SMC Networks, Inc. 38 Tesla Irvine, CA 92618 iii 1/10/2007 VoIP Phone Administration Guide Table of Contents MAG-07010 Rev. A 1 Introduction.......

....uot;, "n", "o", "M", "N", "O" "7", "p", "q", "r", "s", "P", "Q", "R", "S" "8", "t", "u", "v", "T", "U", "V" "9", "w", "x", "y", "z", "W", "X", "Y", "Z" "0", "space" 2 Description Note 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev.
A * # 4-way "*", "·", ":", "@" Start dialing process Press to scroll through lists and menus on the Navigation Keys display. MENU Press to access the menu options or cancel your selection and go back to the previous level. ENTER Phone Book Press to enter a menu or confirm a selection. Press to access the personal phonebook directory. REDIAL DND HOLD TRANSFER Press to call the last number dialed.

Press to block all incoming calls. Press to put an active call on hold. Press to transfer an active call to another VoIP phone on the system. CONFERENCE FWD Press to activate the three-way conference call. 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 SPEAKER Press to listen to voice mail messages.

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 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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 Search Add entry Speed dial Erase all Description Search Phone Book. Add new phone number to phone book. Add speed dial phone number to speed dial list.

Erase all phone number from Phone Book. 2.3.2 Call history Name Incoming calls Dialed numbers Erase record Description Show all incoming call. Show all dialed call.

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 Call forward Description All Forward Activation: To Enabled/Disabled this function. Number: Forward to a Speed Dial Number. 3 1/10/2007 Busy Forward VoIP Phone Administration Guide MAG-07010 Rev. A 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 By Period Period Time Block all phone calls. Block all phone calls at a certain period of time. Set the start time and end time to Block Setting. Set the Alarm Enabled or Disabled. Set the time for alarming. Alarm setting Activation Alarm time Date/Time setting Date and Time Setting.

Date & Time Time format SNTP setting SNTP Primary SNTP Secondary SNTP Time zone Adjustment Time Enabled / Disable SNTP. Set Primary SNTP server IP address. Set Secondary SNTP server IP address. Set Time zone. Set adjustment time period.

Set Handset volume from 0~15 (max.) for you to hear. Set Speaker phone volume from 0~15 (max.) for you to hear. Set Handset Gain from 0~15 (max.) for the other site to hear. Set Speaker phone Gain from 0~15 (max.) for the other site to hear. Set the IP Phone Date and Time. To set the time as 12-hour or 24-hour clock. Volume and Gain Handset volume Speaker volume Handset gain Speaker gain Ringer Ringer volume Ringer type Ringer volume setting from 0~15 (max.). Ringer tone selection from 1~4. Auto dial Set Auto Dial time from 1~5 seconds. 2.

3.4 Network Name Description 4 1/10/2007 WAN Setup IP Type VoIP Phone Administration Guide MAG-07010 Rev. A 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 NAT Set LAN as bridging mode Set LAN as NAT mode First DNS address Second DNS address Activate or disable the VLAN. Set VID from 2 to 4094. Set the priority from 0 to 7.

0~1 DNS Primary DNS Secondary DNS VLAN Activation VID Priority CFI 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 Service domain Description 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. 5 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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 RTP setting Outband DTMF Duplicate RTP Voice Active Detection Enable/Disable. Enable/Disable outband DTMF. No duplicate: do not resend the voice packages. One duplicate: resend voice packages one time. Two duplicate: resend voice packages two times.

RPort Setting Hold by RFC Status RPort Enabled/Disabled Enable/Disable Holding the calls, according to RFC3261. 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 STUN setting Description STUN STUN server Enable/Disable STUN. 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 Upgrade system Description 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 Schedule Stop Select to see the current status and scheduling time. @@@@To see the vendor ID of the IP Phone. @@You can use this function to restart your IP Phone. 2.3.

8 Sys. @@@@@@LAN port is the same as WAN. @@@@@@@@The default setting is: 1. @@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. @@@@@@Also you can see the function lists in the left side. @@@@You can setup the Phone Book and Speed Dial numbers. @@@@@@A number (by URL type). When you are finished a new phone list, just click Add Phone.

@@@@@@When you have finished a new phone list, just click Add Phone. @@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. 10 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev.
A 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. 11 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev.
A 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. 12 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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.
13 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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 No Replace Rule 002 8613+8662 Description 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. 14 1/10/2007 Yes 002 VoIP Phone Administration Guide 55xxxx MAG-07010 Rev. A 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. 15 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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. 16 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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. 17 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A When you have finished the settings, click Submit. Go to the Save Change page and click Save to reflect the changes. 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. 18 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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 19 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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. 20 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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. 21 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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) 22 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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. 23 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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.

24 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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. 25 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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. @@@@When you have finished the settings, click Submit. Go to the Save Change page and click Save to reflect the changes. 26 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. @@When you have finished the settings, click Submit. @@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. 27 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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. 28 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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. 29 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev. A 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. 30 1/10/2007 VoIP Phone Administration Guide MAG-07010 Rev.

A 5 Appendix: Specifications Network Protocol Tone Ring Tone Ring Back Tone Dial Tone Busy Tone Programming Tone Qos IEEE 802.1q VLAN ToS field 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 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) Phone Functions Volume Adjustment Speed Dial Phone Book Flash Speaker Phone Call History Caller ID Voice Message / Incoming Call Indicator Call Function Call Holding Call Waiting Call Forwarding Caller Transferring Call Blocking Call Redial Three-way conference NAT Transversal STUN Outbound Proxy 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 Configuration Web browser TFTP Keypad IP Assignment Static IP DHCP PPPoE DTMF Function In-band DTMF Out-of-band DTMF SIP information SIP Server Registrar Server (three SIP account) Firmware Upgrade TFTP HTTP Voice Quality VAD: Voice activity detection 31 .