IP Phone

MTA7308-SLM

User's Manual Ver. 1.2

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1 Introduction

Voice over IP (also known as Internet Phone) is a technology that allows anyone to make a telephone call over the Internet environment. This is an operation manual for the IP Phone. It is intended to help you configure the telephone. Please follow the user guide carefully as troubleshooting the telephone can be very difficult and time consuming.

2 Safety Declaration

- FCC Part 15 Class B
- CE Class B
- VCCI Class B (optional)
- EN60950-1

3 Package Content

The following items are included in the package. Contact your supplier immediately if an item is missing.

Ethernet Cable

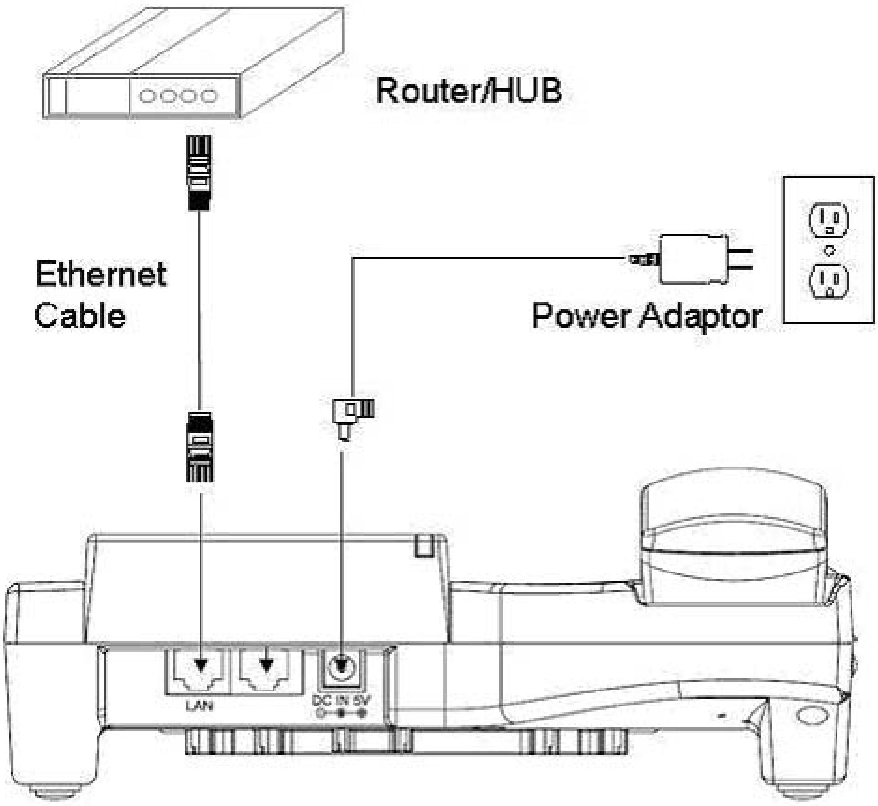
Power Adaptor (DC 5V)

Quick Installation Guide

4 Appearance

A. 96 x 16 Graphics LCD
B. Primary Line/Secondary Line Indicator
C. Transfer
D. Mute
E. Do Not Disturb
F. Voice Mail
G. Speed Dial Keys (M1-M6)
H. Menu
I. OK
J. Cancel
K. Message Waiting Indicator
L. Volume
M. In Use Indicator
N. Menu Navigation Keys
O. Line Keys
P. Redial
Q. Speaker Phone Key

5 Installation



5.1 LCD Menu

Action	LCD view	Other Options	Note
MENU key	Anonymous: Disable	Enable	
√ change	Prov Mode: None	VSP5K, IPPBX-M, IPPBX-D	
√ change	Connection Type: DHCP	Static IP	
* dot	IP Address:	0.0.0.0 when there is no DHCP server	Read Only when DHCP is chosen.
	Subnet Mask:		Read Only when DHCP is chosen
	Default Gateway:		Read Only when DHCP is chosen
	MAC: xxxxxxxxxxxx		Read Only; 16 hexadecimal chars
	Ver:		Read Only
√ OK	Restore: <OK> to confirm		Need confirmation to proceed restore
√ OK	Reboot: <OK> to confirm		Need confirmation to proceed reboot

6 Getting Started

6.1 Web Page Login

IP Phone 7308SLM can be configured via a convenient and user-friendly web interface. The default login User Name is “admin” and Password is also “admin”.

6.2 Web Home Page

The home page of the IP Phone web interface displays the current status of the phone.

All available web pages are listed on the lower left corner on each web page, where

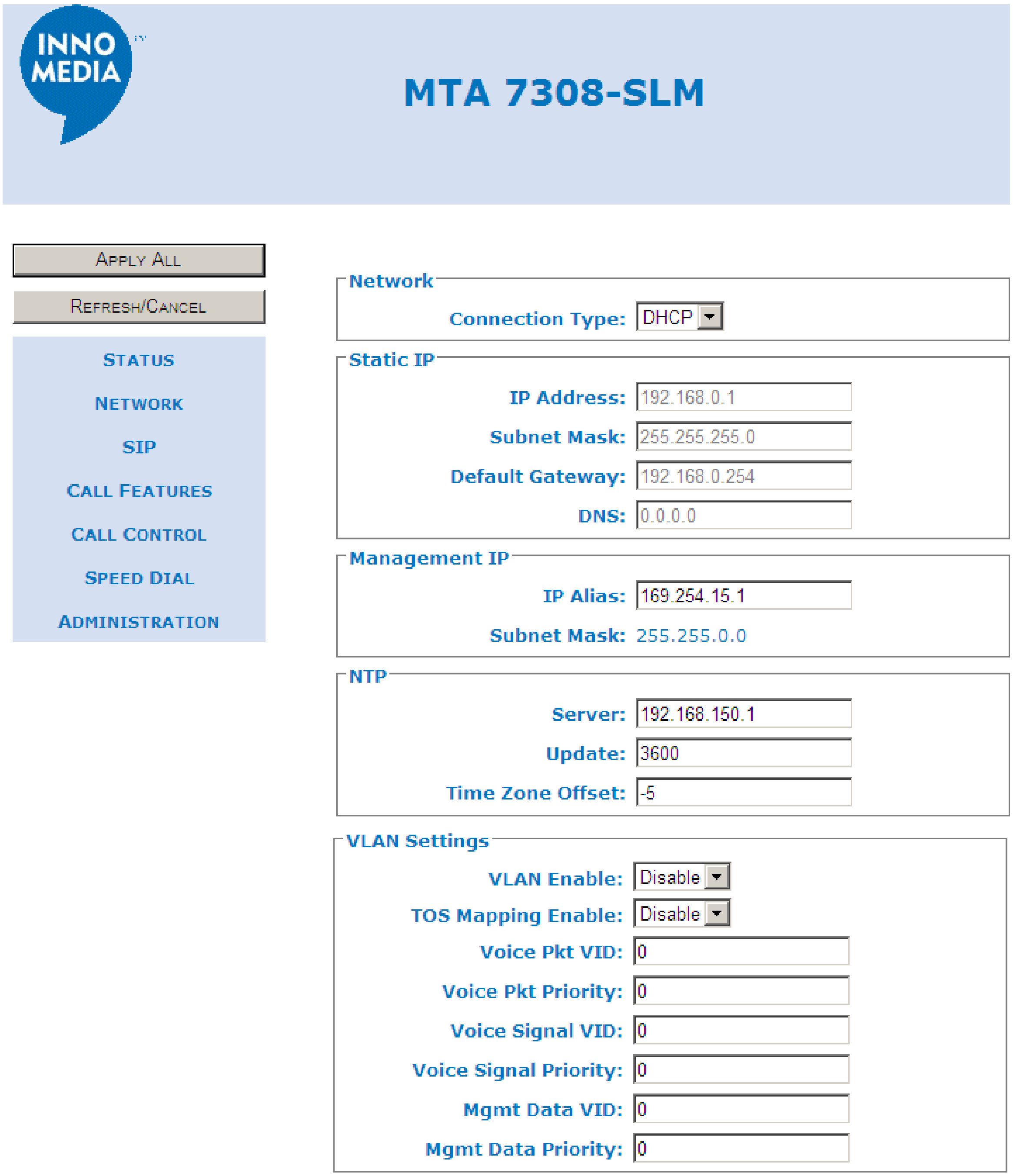
Status	returns to the home page with updated status.
Network	allows user to examine and change network settings.
SIP	allows user to examine and change SIP account and call setup related settings.
Call Features	allows user to examine and change call features supported by the device.
Call Control	allows user to examine and change voice and quality related settings of the device.

Speed Dial	allows user to examine and change up to six speed dial setting supported by the device.
Administration	allows user to examine and change web management, auto-provisioning, and local TFTP upgrade settings.

You may click **Refresh/Cancel** to check the latest status of the phone.

7 Setting Up

7.1 Network Setting



Connection Type	Select < **Static** > to allow user to input network settings manually; < **DHCP** > to instruct device to request it's network setting from DHCP server at system boot up.
IP Address	Enter the IP address when **Connection Type** is **Static IP**.
Subnet Mask	Enter the subnet mask when **Connection Type** is **Static IP**.
Default Gateway	Enter the IP address of the default gateway when **Connection Type** is **Static IP**.
DNS	Please enter the IP address of DNS server when **Connection Type** is **Static IP**, or obtain the IP address automatically when **Connection Type** is **DHCP**.
Management IP	This is the alternate IP address for local management. The web page is always accessible via this address.
NTP Server	Enter the IP address or domain name of the NTP server that the device can obtain network time. Disabled when left blank.
NTP Update Time	Refresh period in sec. per each network time update.
NTP Time Zone Offset	Local time zone offset to the standard network time.

VLAN Enable	Enable or Disable VLAN settings for the WAN interface of this phone.
TOS Mapping Enable	Enable or disable the mapping of DSCP TOS (Type of Service) priorities to VLAN (802.1p) priorities
Voice Pkt VID	Setting the VLAN ID for voice RTP packets
Voice Pkt Priority	Setting the VLAN priority of voice RTP packets
Voice Signal VID	Setting the VLAN ID for SIP signaling packets
Voice Signal Priority	Setting the VLAN priority of SIP signaling packets
Mgmt Data VID	Setting the VLAN ID for IP Phone management data packets
Mgmt Data priority	Setting the VLAN priority for IP Phone management data packets

Click **Apply All** to activate the changes, or **Refresh/Cancel** to restore the configurations.

7.2 SIP Setting

Server	The host name or IP address of the SIP server or SIP proxy server.
Server Port	The port of the SIP server that is used for SIP signaling.

SIP Port	The local port of the IP phone that is used for SIP signaling.
SIP Display Name	Enter the SIP user name or number that is provided by your service provider.
Username	The ID that is used to authenticate the IP phone when it registers to the SIP Registrar or makes calls through the SIP Proxy server.
Password	The password that is used to authenticate the IP phone when it registers to the SIP Registrar or makes calls through the SIP Proxy server.
Register Expires	The relationship which is established by SIP registration with the Registrar server will be terminated if the time is elapsed.
STUN Enable	Enable or disable the **STUN** function.
STUN Server	Enter the host name or IP address of the STUN server.
NAT Keep Alive	The time period that the IP phone sends out a lightweight packet for keeping the tunnel alive. “0” means to disable this function.
Proxy Domain	If specified, the proxy domain name will be used in place of SIP server address in each SIP command.
URI enable	Select among <None>, <Transport=UDP>, <user=phone>, or both (i.e., <Transport=UDP> & <user=phone>) to be included in SIP command.
Session	Session Timer Enable or Disable.

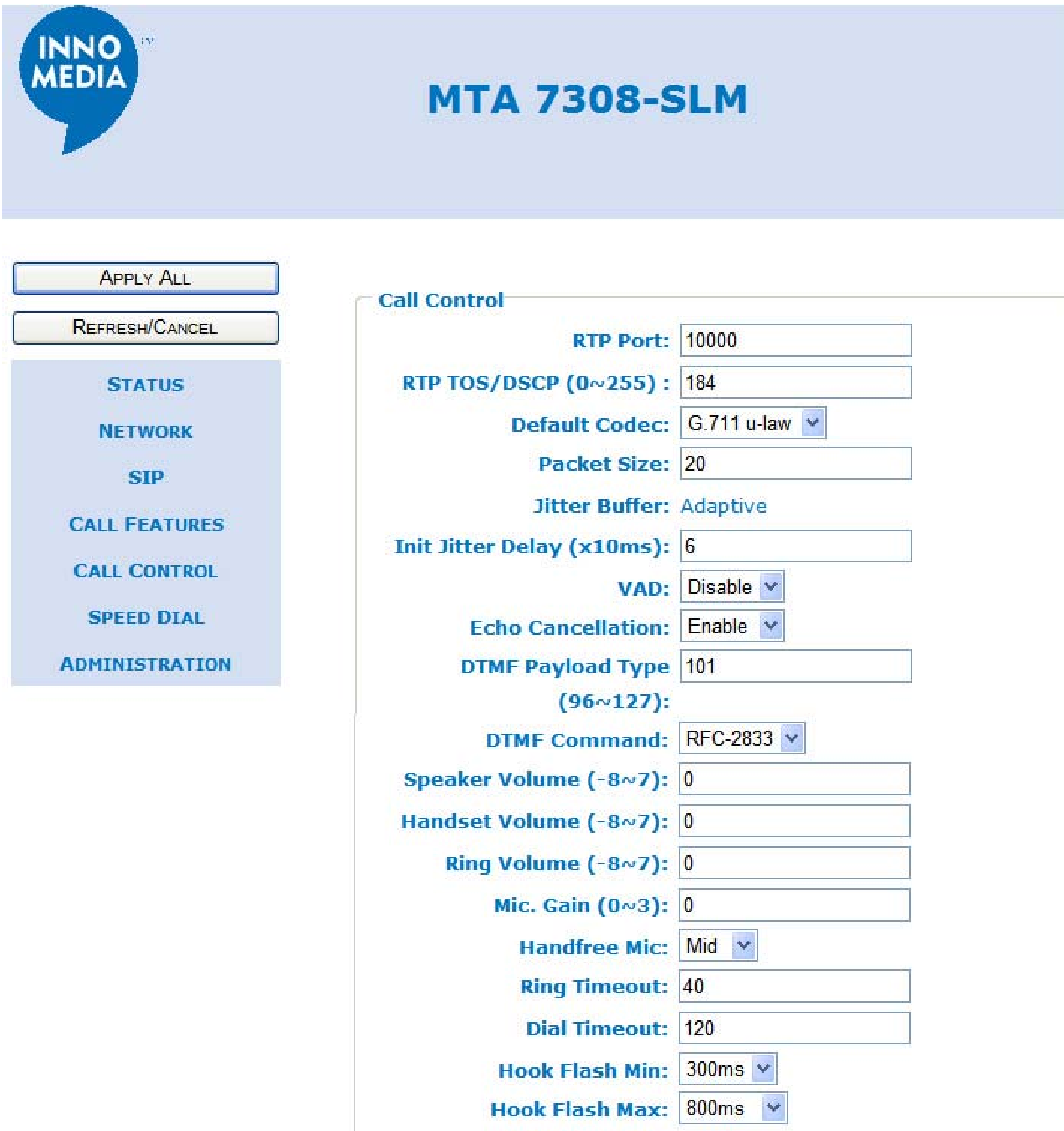
Session Timer	Default time 3600 seconds (0 for inifinity), should larger than min-se
Min. Session Timer	Minimum Session Time: 180 seconds (default, number can not be lower than 90).

Click **Apply All** to activate the changes, or **Refresh/Cancel** to restore the configurations.

7.3 Call Features Setting

Anonymous	< Anonymous > is displayed as caller-ID of the phone and used in SIP command.
SDT Prefix	Second Dial Tone Prefix settings, multiple rules allowed. When any setting matches when dialing, a 2nd dial tone will be heard.
Digitmap	Digitmap string, multiple rules allowed. When any setting matches, the number will be dialed out immediately.
VM Ext Number	Voice Message Extension number. The number to be dialed when Voice_Message key is pressed.

7.4 Call Control Setting



RTP Port	The port is used for voice packets (i.e. RTP packets) transmission.
RTP TOS (0~255)	Enter the value of **Type-of-Service** to modify the priority of RTP

packets.

Default Codec

Decide the default codec that will be the first priority for communication. The actual Codec to be chosen for communication is via negotiation at call setup time.

Packet Size

Select the **Packet Size** in milliseconds (will be truncated to multiple of 10 ms.). It will increase the network traffic load if a shorter packet length is selected; however, it may reduce the voice quality if a longer length is selected.

Jitter Buffer

Adaptive Jitter Buffer (read-only).

VAD

Enable or disable the **VAD** function for silence suppression.

LEC

Enable or disable the function of line echo cancellation.

DTMF Command

The IP phone supports two options, **RFC-2833** and **SIP INFO,** for transporting DTMF tones in a call.

DTMF Payload Type

When **RFC-2833** is selected, please also enter the value of **Payload Type** in the blank. Please contact your service provider for a suitable type.

Speaker Volume

Adjust the speaker volume within the range -8~7. Default is “0”.

Handset Volume

Adjust the handset volume within the range -8~7. Default is “0”.

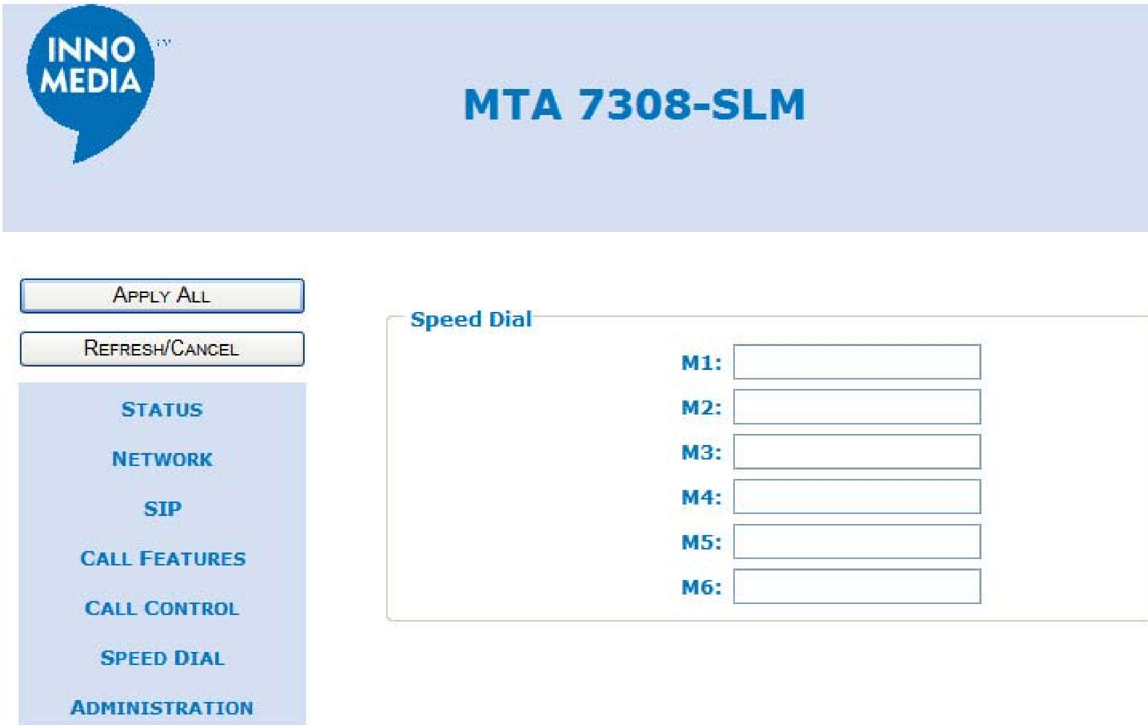
Ring Volume

Adjust the ringer volume within the range -8~7. Default is “0”.

HandFree Mic.	Adjust the gain of microphone at hand-free speaker-phone mode..
Ring Timeout	Time in seconds for Ring tone to play before IP phone decided that no one answer the call.
Dial Timeout	Time in seconds after last number dialed before IP phone decided that no more numbers the user is intended to dial and started to dialing out.
Hook Flash Min.	The lower threshold of time in milliseconds for phone to decide the hook flash key is pressed.
Hook Flash Max.	The threshold of time in milliseconds for phone to decide the hook flash key is pressed long enough for onhook.

Click **SUBMIT** to activate the changes, or **RESET** to restore the configurations. Please also remember to click **Save** on the left panel to save the changes.

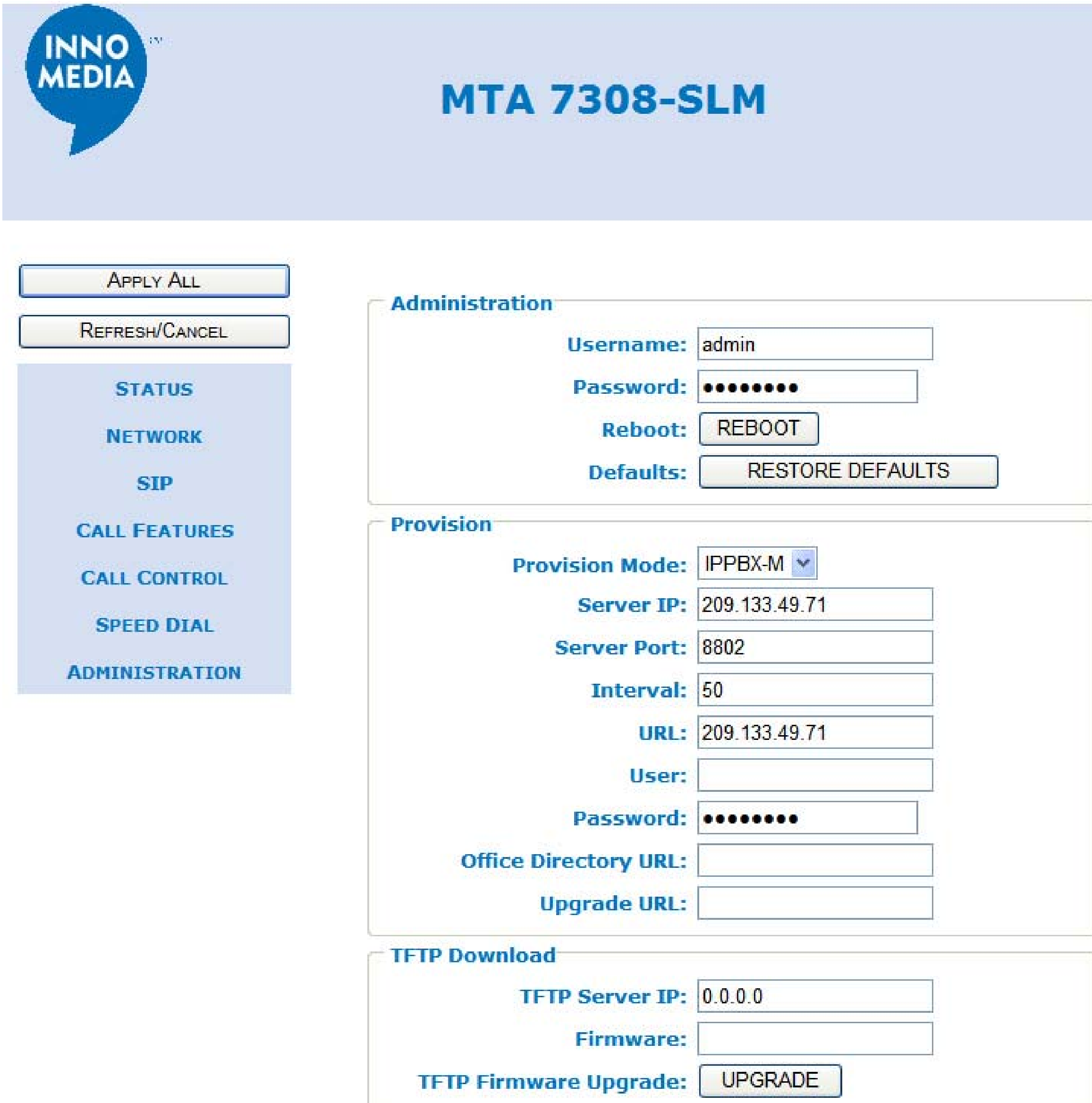
7.5 Speed Dial Setting



Number M1~M6	Enter the phone number or SIP URI per each entry. The phone supports up to 6 phone numbers for Speed Dial. User can do a one touch dial of a preset number by pressing correspondent M1~M6 speed dialing key.

Click **APPLY ALL** to activate the changes, or **REFRESH/Cancel** to restore the configurations.

7.6 Administration, Provision, and TFTP Setting



Username	Modify the username for logging into the phone. By default, the username is “admin”.
Password	Modify the password for logging into the phone. By default, the

	password is "admin".
Reboot	Re-start the phone.
Restore Defaults	Restore the phone with default configuration parameters and re-start.

Provision Mode	Select one of the Auto Provision Modes from the list: None, VSP5K, IP-PBX-M (Multicast), and IP-PBX-D (DHCP).
Server IP	Provision Server IP.
Server Port	Provision Server Port.
Interval	Provision Interval.
URL	Provision URL.
User	Provision User ID.
Password	Provision Password.
Office Directory URL	Office Directory URL (reserved for future upgrade)
Upgrade URL	Upgrade Image URL.

The above provision parameters should be preloaded or manually entered in VSP5K modes. They will be automatically downloaded from Auto-Provisioning Server while in IP-PBX-M (multicast) and IP-PBX-D (DHCP) modes.

TFTP Server IP	Enter the IP address or host name of the TFTP server for local image upgrade.
Firmware	Modify the name of firmware to be fetched from TFTP server.
TFTP Firmware Upgrade	Upgrade Firmware from local TFTP server.

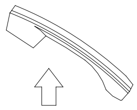
To make the function of Firmware Upgrade work properly in TFTP mode, user shall install a TFTP server on PC and put the files of firmware and configuration into the root directory. The PC with the TFTP server is now the Firmware Upgrade server. The phone will request for downloading the new image of firmware, writing it into the Flash memory and booting up to the updated firmware.

Firmware Upgrade can be invoked manually anytime when both TFTP server and filename are appropriately set.

Click **APPLY ALL** to activate the changes, or **REFRESH/Cancel** to restore the configurations.

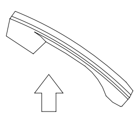
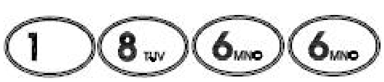
8 Operating the Phone

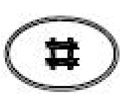
8.1 Dialing IP Address

- Lift handset  or press the SPEAKER button
- Dial IP address
- For example, dialing 192.168.0.1
- 

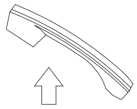
- Press  and SPEAKER button or wait until the timer expires to dial

8.2 Dialing SIP Number

- Lift handset  or press the SPEAKER button
- Dial SIP Number
- For example, dialing 1866
- 

- Press  and SPEAKER button or wait until the timer expires to dial

8.3 Speed Dialing

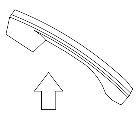
- Lift handset  or press the SPEAKER button

- Dial Speed Dial number with the button M1 ~ M6.

- Press # and SPEAKER button or wait until the timer expires to dial

8.4 Answering a Phone Call

When phone rings:

- Lift handset  or press the SPEAKER button to begin conversation

Note: The CANCEL key can be used to reject a call.

8.5 Muting

To mute the phone microphone, speaker, or headset microphone:

- Press **MUTE** key.

- Press **MUTE** key again to resume conversation

8.6 Putting A Call on Hold

To put a call on hold:

- Press the SEC. LINE button will hold the PRI LINE, press the PRI LINE button will hold the SEC. LINE.

8.7 Call Waiting

While having a conversation, another call comes in, you can hear call waiting tone:

- Press the SEC. LINE button to hold the PRI. LINE call and to answer come in call.

8.8 Transferring Calls

You can perform two types of transfers:

- Attended—You call the person to whom you are transferring the call and speak to them before transferring the call.
- Blind—You transfer the call without speaking to the other party to which you are transferring the call.

During an active call

- Press the SEC. LINE button to hold current call (PRI. LINE)
- Dial the third party phone number:

  To transfer the call without waiting for the other person to answer, press Transfer after the call begins to ring and hang up. If you hang up before the second call rings, the transfer fails and the first call disconnected.

  To transfer after speaking privately to the other person, press Transfer at any time during the conversation.

8.8.1 Handfree Call transfer

Press the SPEAKER button to make a handfree call

9 Troubleshooting

9.1 Hardware Installation

Q: I connect the socket of the phone to outlet with a power adaptor, but nothing is displayed on the LCD.

A: It seems the phone cannot be offered with power or gets the hardware problem. Please check the following items.

a. Check if the socket of the phone is unstable or not assembled properly.

b. Check if you use the right power adapter.

c. Check if you have connected the device to the outlet properly.

d. Check if the outlet can offer electric power properly. The power adapter which is attached to the phone supports the power range from 100V to 240V.

If the above methods do not solve your problem, please contact the retailer where you purchased the phone or your service provider.

9.2 Network Connection

Q: I connect my PC to the same HUB or switch and install the Web Prob on the PC. I try to configure my phone through the Web Prob, but I can't.

A: Please ensure that you have read the instruction in *Chapter 6 Getting Started*, and follow the steps to set up your device. If it is not helpful to you, please try the following methods.

a. Enter command line (e.g. the application provided by Microsoft ® Windows™) to examine the communication between the phone and PC by the command "PING".

b. Check the physical connection between the HUB and the phone (It assumes that you can access the Internet service properly). The LED of the port (on switch HUB) where you connect the phone must be lit up or blink. If it is not lit up or blink, please check if one end of the Ethernet cable is plugged into the port on HUB properly and the other end is plugged into the LAN port of your PC.

c. Replace the Ethernet cable with another one and try again.

d. Contact the retailer where you purchased the device or the service provider.

9.3 Configuration and Operation

Q: I pick up the handset of the phone, but I cannot hear the Dial tone.

A: Please check whether the handset is connected to the phone set properly by a cable. You can also reboot the phone, wait for a few minutes and try again. If the problem still exists, please contact the technical support.

Q: My IP phone cannot register with the SIP server.

A: Please check the following items.

a. Check the network connection of the phone. You may enter command "PING" in the command line interface to examine the connection.

b. Check the physical network connection of the phone.

c. Ensure that you have configured the IP phone with right SIP number, ID, password, and IP address of the SIP server or Registrar.

d. Reboot the phone, wait for a few minutes and try again.

e. If the problem still exists, please contact the technical support.

Q: I cannot use my IP phone to make/answer calls to/from other IP phones.

A: Please check the following items.

a. Check if your IP phone has registered with the correct SIP server.

b. Check if the IP phone which makes/answers calls to/from you has also registered with the same SIP server.

c. Pick up the handset of the phone and ensure that you can hear Dial tone.

d. Reboot the phone, ensure that the phone can register to the SIP server and try again.

e. If the problem still exists, pleas contact the technical support.