MAX™ IP Conferencing Phone

ADMINISTRATOR'S GUIDE

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MAX IP ADMINISTRATOR'S GUIDE

CLEARONE PART NO. 800-158-302. JANUARY 2006 (REV.1.0)

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CHAPTER 1: INTRODUCTION

PRODUCT OVERVIEW

Thank you for purchasing the ClearOne Max™ IP expandable conferencing phone. MAX IP provides premium, full-duplex audio to small conference rooms as a single unit and to larger rooms as an expanded system. Up to four MAX IP units can be linked, expanding not only microphone coverage but loudspeaker coverage and control access as well. This creates even distribution of sound for a more natural communications experience.

Setting up the MAX IP conferencing phone requires only three connections: power, network, and the base unit to the conferencing phone. And, the familiar keypad design ensures that you will be comfortable using the phone, reducing the need for training, and support.

- **EXPANDABLE.** MAX IP is ideal for conference rooms and provides complete microphone and loudspeaker coverage and easy access to controls.
- **SUPERIOR AUDIO.** The clear, full sound of the MAX IP facilitates more natural interaction among participants.
- **ONE-TOUCH CONFERENCING.** Single-button access to 3-way calling.
- **EASY TO USE.** The intuitive, user-friendly controls are simple to operate.

SERVICE AND SUPPORT

If you need additional information on how to set up or operate your MAX IP conferencing phone, please contact us. We welcome and encourage your comments so we can continue to improve our products and better meet your needs.

TECHNICAL SUPPORT

Telephone: 1.800.283.5936(USA) or 1.801.974.3760

Fax: 1.801.977.0087

E-mail: tech.support@clearone.com

Web site: www.clearone.com

SALES AND CUSTOMER SERVICE

Telephone: 1.800.945.7730 (USA) or 1.801.975.7200

Fax: 1.800.933.5107 (USA) or 1.801.977.0087

E-mail: sales@clearone.com

PRODUCT RETURNS

All product returns require a return authorization (RA) number. Please contact ClearOne Technical Support before attempting to return your product. Make sure you return all the items that shipped with your product.

IMPORTANT SAFETY INFORMATION

Read the safety instructions before first use of this product. This conferencing phone is not designed for making emergency telephone calls when the power fails. Make alternative arrangements for access to emergency services.

- Read and understand all instructions and follow all warnings marked on the product.
- Unplug this product from the wall outlet before cleaning. Do not use liquid cleaners or aerosol cleaners. Use a damp cloth for cleaning.
- Do not use this product near water, for example, near a bathtub, washbowl, kitchen sink, or laundry tub, in a wet basement, or near a swimming pool.
- Do not place this product on an unstable cart, stand, or table. The product may fall, causing serious damage to the product.
- Slots and openings in the cabinet and the back or bottom are provided for ventilation, to protect it from overheating, these openings must not be blocked or covered. Never push objects of any kind through cabinet slots as they may touch dangerous voltage points or short out parts that could result in a risk of fire or electric shock.
- This product should never be placed near or over a radiator or heat register. This product should not be placed in a built-in installation unless proper ventilation is provided.
- This product should be operated only from the type of power source indicated on the marking label. If you are not sure of the type of power supply in your location, consult your dealer or local power company.
- Do not overload wall outlets and extension cords as this can result in fire risk or electric shock.
- Never spill liquid of any kind on the product.
- To reduce the risk of electric shock, do not disassemble this product. Opening or removing covers may expose you to dangerous voltages or other risks. Incorrect reassembly can cause electric shock during subsequent use.
- Unplug this product from the wall outlet and refer servicing to qualified service personnel under the following conditions:

  a. When the power supply cord or plug is damaged or frayed.

  b. If liquid has been spilled into the product.

  c. If the product does not operate normally by following the operating instructions.

  d. If the product has been dropped or damaged.

  e. If the product exhibits a distinct change in performance.
- Avoid using a telephone during an electrical storm. There may be a remote risk of electric shock from lightning.
- Do not use this product to report a gas leak in the vicinity of the leak.
- Do not use this product near intensive care medical equipment or by persons with pacemakers.
- This product can interfere with electrical equipment such as answering machines, TV sets, radios, computers and microwave ovens if placed too close.

Save these instructions

UNPACKING

Carefully place the conferencing pod and base unit on a level surface. Ensure you have received all items shown in figure 1.

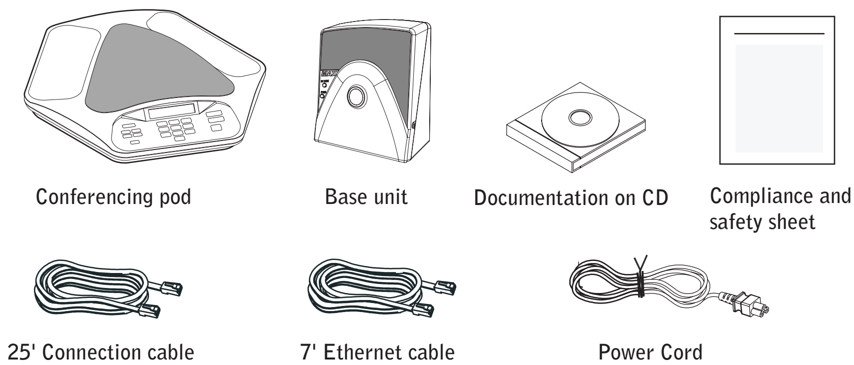

FIGURE 1.1 MAX IP parts

> Note: ClearOne is not responsible for product damage incurred during shipment. You must make claims directly with the carrier. Inspect your shipment carefully for obvious signs of damage. If the shipment appears damaged, retain the original boxes and packing material for inspection by the carrier. Contact your carrier immediately.

The items shown in figure 2 are included in the Max IP Expansion Kit.

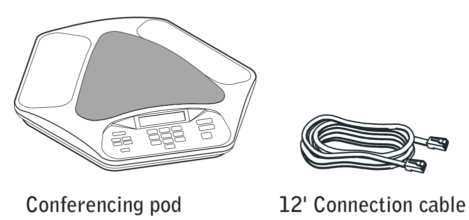

FIGURE 1.2 MAX IP expansion kit

WARNING: TO PREVENT FIRE OR ELECTRICAL SHOCK HAZARD, DO NOT EXPOSE THIS PRODUCT TO RAIN OR MOISTURE.

THE LIGHTNING FLASH AND ARROW HEAD WITHIN THE TRIANGLE IS A WARNING SIGN ALERTING YOU OF DANGEROUS VOLTAGE INSIDE THE PRODUCT	**CAUTION** RISK OF ELECTRIC SHOCK DO NOT OPEN CAUTION: TO REDUCE THE RISK OF ELECTRIC SHOCK, DO NOT REMOVE COVER (OR BACK) NO USER SERVICEABLE PARTS INSIDE. REFER SERVICING TO QUALIFIED SERVICE PERSONNEL.	THE EXCLAMATION POINT WITHIN THE TRIANGLE IS A WARNING SIGN ALERTING YOU OF IMPORTANT INSTRUCTIONS ACCOMPANYING THE PRODUCT.
SEE MARKING ON BOTTOM / BACK OF PRODUCT		

CHAPTER 2: GETTING STARTED

CONNECTING YOUR CONFERENCING PHONE

1. Connect the Connection cable from the Link Out jack on the base unit to the Link In jack on the conferencing pod (see figure 2.1).

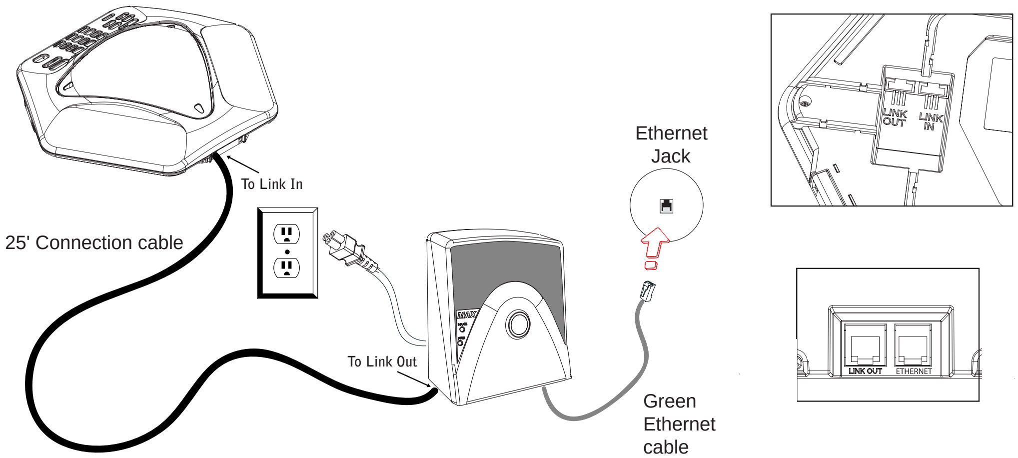

FIGURE 2.1 Connecting the MAX IP

Warning: DO NOT plug a laptop or PC into the Link Out jack on the base unit or conferencing pod--severe electrical damage can occur if this is done.

2. Connect the base unit to the Ethernet jack using the Ethernet cable.
3. Connect the power cord to the base unit and plug it directly into an electrical outlet.

CONNECTING ADDITIONAL MAX IP PHONES

1. Connect the 12' Connection cable to the Link Out jack on the first phone and to the Link In jack on the second phone (see figure 2.2).

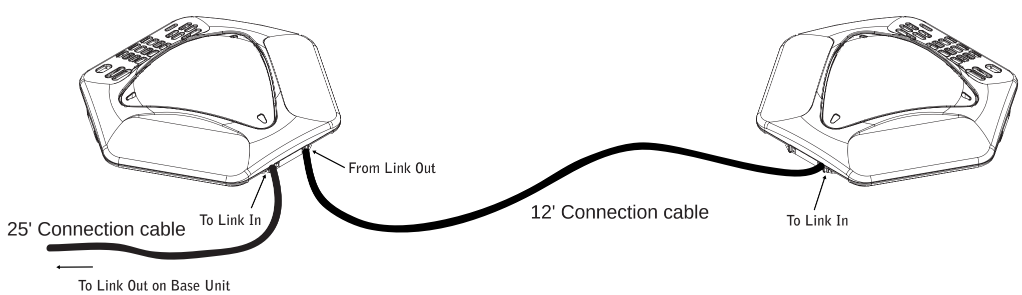

FIGURE 2.2 Connecting additional units

2. Continue linking up to three additional MAX IP phones in the same fashion. A total of four units may be connected.

PROVISIONING YOUR MAX IP PHONE

There are two methods available for configuring your MAX IP phone:

- The first method is manually, through the phone's keypad and a web portal associated with the phone.
- The second method provisions the phone automatically when it is plugged into the network. This method uses a DHCP server to assign the phone minimal IP information so that it can access the network, including an IP address, gateway, subnet mask, and TFTP server address. A TFTP server is then used to automatically upload firmware (when new firmware is available) and provisioning information to the phone, so that it is ready to make a call after it has completed the booting sequence.

By default, the MAX IP is configured for automatic provisioning and assumes the IT administrator has correctly configured the DHCP and TFTP servers on the phone network. When more than a few phones are to be provisioned, it is recommended that you use the automatic method, otherwise the manual method is the best.

CONFIGURING THE IP ADDRESS

Before any other provisioning can be done, you must first configure the host IP address and the subnet mask of the MAX IP phone. These must be known in order to provision the phone through the web interface. These settings are usually obtained automatically from the DHCP server, however, they can also be assigned manually if the host IP address is to be a static IP address.

CONFIGURING THE DHCP SERVER

When configuring a DHCP server for use with the MAX IP phone, the following parameters should be assigned:

- IP address
- Subnet mask
- Gateway IP address
- TFTP server IP address
- DNS server IP address
- Secondary DNS server IP address
- DNS domain

The IP address and subnet masks are defined by DHCP option 1.

The Gateway IP address is defined by DHCP option 3.

The TFTP server is defined first by DHCP option 66. If this is undefined, then the MAX IP examines the siaddr parameter in the DHCP ACK packet. If this is not defined, then the hostname parameter in the DHCP ACK packet is used.

The DNS server IP address is defined by DHCP option 6.

The secondary DNS server IP address is defined by DHCP option 6.

The DNS domain is defined by DHCP option 15.

MANUALLY ASSIGNING AN IP ADDRESS

If DHCP is disabled, or you wish to assign a static IP address, then perform the following steps:

1. Press and hold the REDIAL/PROG key until the program icon appears on the LCD screen (see figure 2.3).

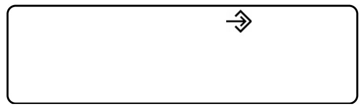

FIGURE 2.3 MAX IP LCD Program icon

> Note: If a key is not pressed within 30 seconds of entering program mode, the MAX IP phone will beep and return to operation mode.

2. Press the 2 key. The default IP address is displayed on the LCD screen. If this address was obtained from DHCP, then you can use it to access the web interface. If you cannot access the web interface using this address, then you need to either configure DHCP as explained above so that the IP address can be obtained automatically, or you must manually enter a static IP address. Continue with step 3 to manually enter a static IP address for the MAX IP phone.
3. Press the 1 key. The current DHCP setting is displayed on the LCD screen. A "1" indicates that DHCP mode is enabled; a "0" indicates that DHCP mode is disabled.
4. If "0" is the current setting, then continue with step 8, otherwise continue with step 5.
5. Press the REDIAL/PROG key. The current DHCP mode "1" flashes on the LCD screen.
6. Press the 0 key. The new DHCP mode "0" flashes on the LCD screen.
7. Press the REDIAL/PROG key. DHCP mode is disabled.
8. Press the 2 key. The current IP address (such as 0.0.0.0) is displayed on the LCD screen.
9. Press the REDIAL/PROG key. The current IP address flashes on the LCD screen.
10. Press the CLEAR key. The current IP address is deleted one character at a time.
11. Using the number keys, enter the static IP address. Use the * key to enter the decimal separators in the IP address.
12. Press the REDIAL/PROG key. The new static IP address is activated in the MAX IP phone.
13. Press the 3 key and repeat steps 9-12 above to set the subnet mask, then continue with step 14.
14. Press the REDIAL/PROG key. The MAX IP phone reboots.

MANUALLY PROVISIONING YOUR MAX IP PHONE

The easiest way to manually provision your MAX IP phone is through the web interface. To access the web interface for your MAX IP phone, perform the following steps:

1. Press and hold the REDIAL/PROG key until the program icon appears on the LCD screen (see figure 2.3).
2. Press the 2 key. The phone's IP address is displayed on the LCD screen.
3. Make note of the IP address and press the CLEAR key. Program mode is exited.
4. Start Internet Explorer (only the Internet Explorer web browser, release 6.0 or later with Java 1.5 or later installed, works with the MAX IP phone).
5. Enter the phone's IP address into the Internet Explorer address field and press the ENTER key on your computer's keyboard. (You can also enter the DNS name or the phone's name on the network in order to get to the web interface login screen.)

6. The web interface login screen appears (see figure 2.4).

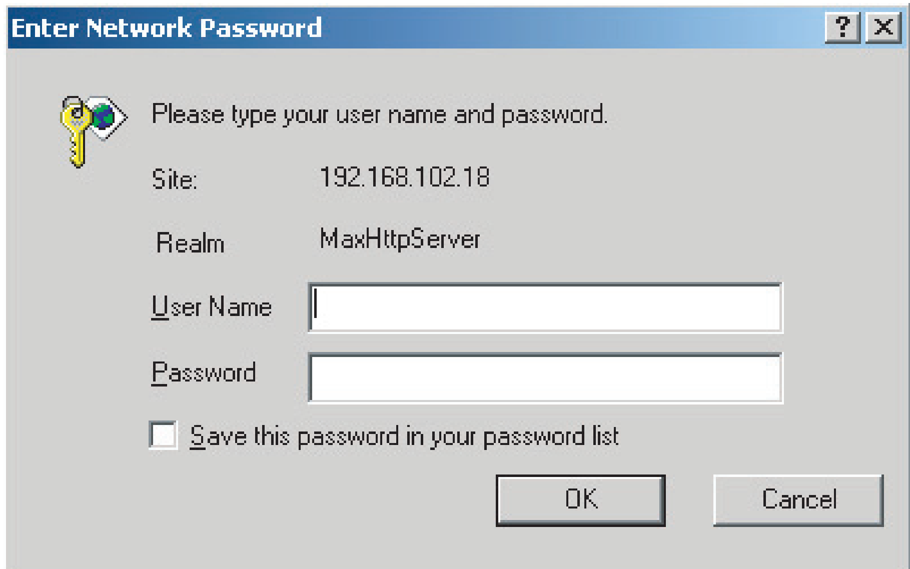

FIGURE 2.4 Login screen

7. Enter the default username admin and the default password clearone and click OK.
8. The web portal appears, displaying the Device Information screen.

WEB PORTAL SCREENS

The following sections display each of the screens included in the MAX IP web portal and describe all of the settings that can be modified through each of the screens.

DEVICE INFORMATION SCREEN

The Device Information screen (see figure 2.5) displays all of the system information for your MAX IP phone. The information fields are self-explanatory, however, note that the last four digits of the MAC address are used as an identifier in the system name. By default, the system name is in the format of MAX1AV- followed by the last four digits of the MAC address. The MAC address is a unique address in hexadecial format that can also be found on the label attached to the back of your MAX IP base unit. This screen (and all other screens) also includes a link to the Registration page in order to register your MAX IP phone with ClearOne. Simply click on the Registration link to go there.

FIGURE 2.5 Device Information screen

CONFIGURATION: GENERAL SETTINGS SCREEN

Use the Configuration: General Settings screen (see figure 2.6) to set up security, provisioning, the MAX IP extension and help line phone numbers, and when a reboot of the phone is allowed.

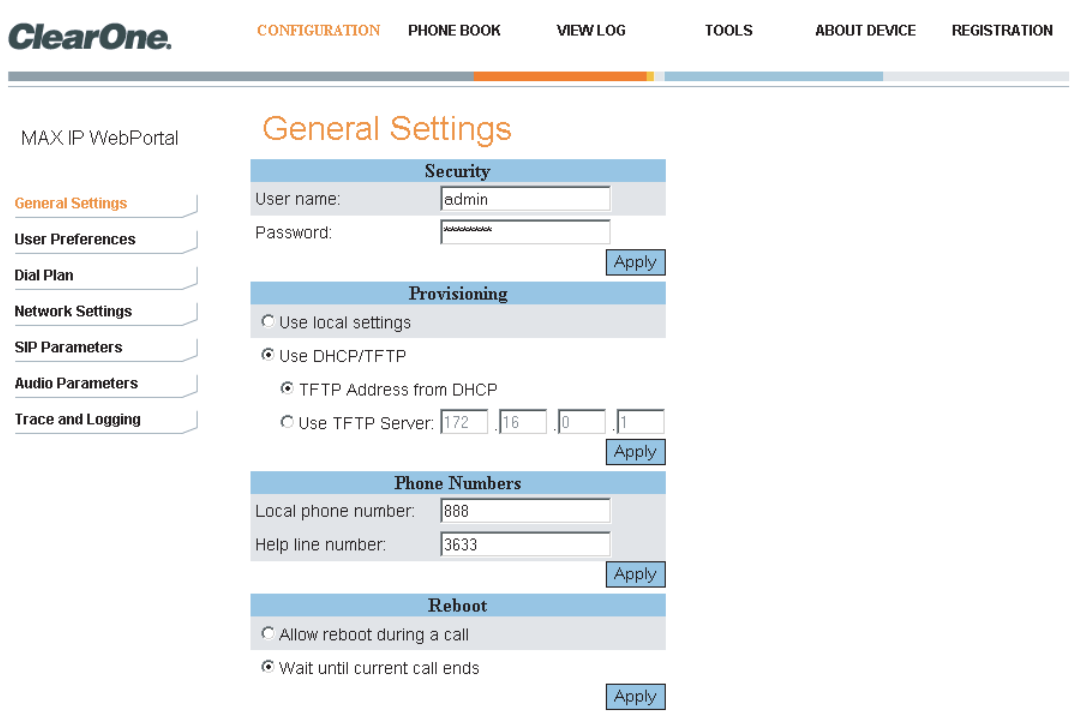

FIGURE 2.6 Configuration: General Settings screen

- Security: To change your username and/or password, enter the new username and/or password in the appropriate fields and then click the Apply button.

- **Provisioning**: Click the **Use local settings** radio button if you wish to use the settings programmed into your MAX IP phone, including IP address, audio settings, and VLAN settings. Use this option if TFTP via DHCP is unavailable or if you wish to manually provision your phone.

  Click the **Use DHCP/TFTP** radio button to set the TFTP IP address. The IP address of the TFTP server can be provided by the DHCP server or you can enter it manually. If the TFTP address is provided by the DHCP server, then click the **TFTP Address from DHCP** radio button; if the TFTP server is set manually, then click the **Use TFTP Server** radio button and enter the IP address of the TFTP server.

  Click the **Apply** button to activate the changes.

- **Phone Numbers**:To change or set the phone numbers for the MAX IP phone and help line, enter the phone number for the MAX IP phone into the **Local phone number** field and the phone number for the help line into the **Help line number** field and then click the **Apply** button.
- **Reboot**: Choose when you wish to allow a reboot of the MAX IP phone. Click the **Allow reboot during a call** radio button to allow a reboot while a call is in progress or click the **Wait until current call ends** radio button to allow a reboot only after a call is completed, then click the **Apply** button.

CONFIGURATION: USER PREFERENCES SCREEN

Use the Configuration: User Preferences (see figure 2.7) screen to enable/disable automatic level control (ALC) and automatic gain control (AGC), to mute/unmute the incoming ringer, to select the incoming ringer melody, to set the time zone, and to determine if you want the time automatically adjusted for Daylight Savings or not.

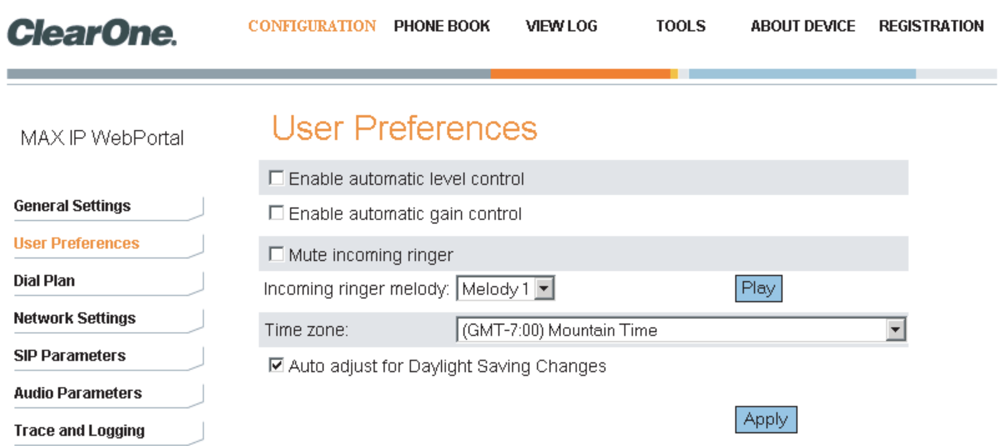

FIGURE 2.7 Configuration: User Preferences screen

- **Enable automatic level control**: Click this check box to enable ALC or uncheck it to disable ALC. ALC (automatic level control) automatically adjusts microphone levels to ensure participants' voices are transmitted at consistent levels regardless of whether people are speaking loudly or softly.
- **Enable automatic gain control**: Click this check box to enable AGC or uncheck it to disable AGC. AGC (automatic gain control) adjusts (softer and louder) input audio to a consistent level.
- **Mute incoming ringer**: Click this check box to mute the incoming ringer (or uncheck it to allow the ringer to ring normally).
- **Incoming ringer melody**: Click the drop-down box to select from the five available melodies. Click the Play button to listen the selected melody on the phone.
- **Time zone**: Click the drop-down box to select from the available time zones. Choose the time zone nearest to your location. The time zone is necessary for logging functions, such as errors, and so forth.

- **Auto adjust for Daylight Saving Changes**: Click this check box to have the time automatically adjusted for Daylight Savings or uncheck it if Daylight Savings time is not observed in your area.

Click the **Apply** button to activate any changes made to this screen.

CONFIGURATION: DIAL PLAN SCREEN

Use the Configuration: Dial Plan (see figure 2.8) screen to view your current dial plan and to choose how you want the dial plan for your MAX IP phone loaded. You can choose to have it loaded from a file containing all of the settings you desire or you can select the settings you wish for your MAX IP phone manually through this screen.

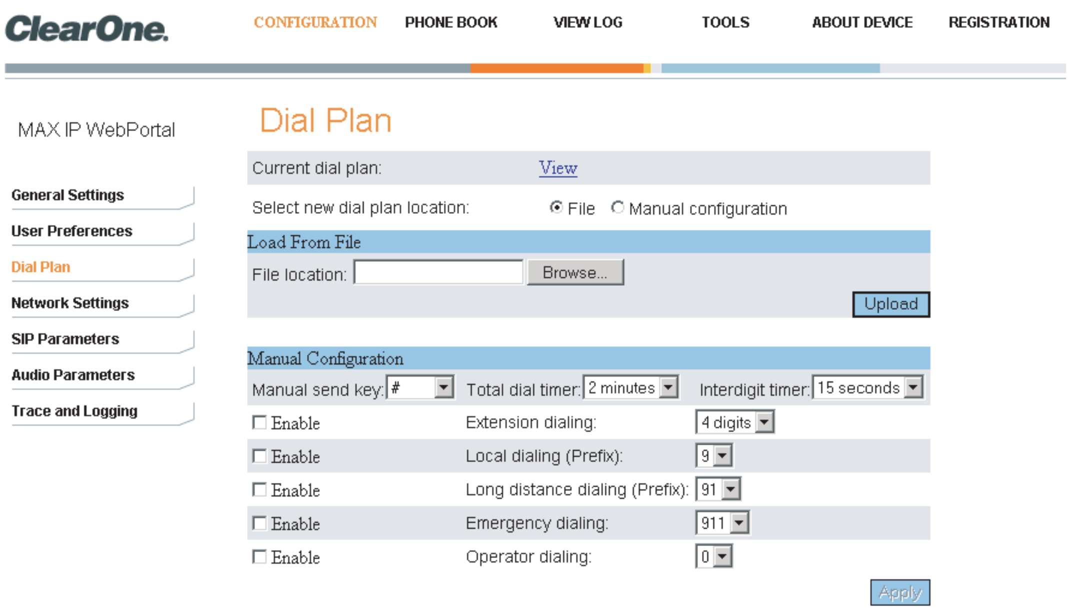

FIGURE 2.8 Configuration: Dial Plan screen

- **Current dial plan**: Click the View hyperlink to see the current dial plan file (see figure 2.9) associated with your MAX IP phone.

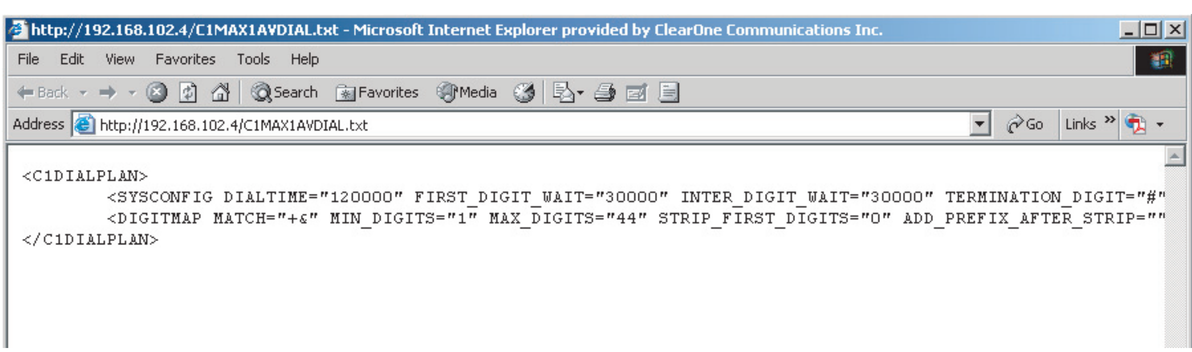

FIGURE 2.9 Current dial plan file

- **Select new dial plan location**: Click the radio button for the method you wish to use to have your dial plan loaded into your MAX IP phone. Click the **File** radio button if you want the dial plan loaded from a file located on the local PC or click the **Manual configuration** radio button if you want to set up the dial plan manually through this screen. Note that when the file method is chosen, the **Upload** button is active and the **Apply** button is inactive; when the manual configuration method is chosen, the **Apply** button is active and the **Upload** button is inactive.
- **Load From File**: If you are setting up this phone to be part of a group of phones with similar settings, then it is recommended that you use the file method (see page 29 for more information on the dial plan configuration file). This method ensures that each phone in the group uses the same dial plan settings. (Although the name of your dial plan file can be any name you wish, it is stored in the MAX IP phone in a file named `C1MAX1AVDIAL.txt`.) Your dial plan file can be located in different directories in order to provide for different settings for different phones, such as different dial plans for domestic calls versus international calls. Click the **Browse** button to navigate to the directory where the dial plan is located and then click the **Upload** button to load the dial plan file into your MAX IP phone.
- **Manual Configuration**: If you do not have a dial plan file set up and stored on your local PC, or if you wish to create a dial plan for only this MAX IP phone, then you should use the manual configuration fields on this screen.

  > **Note:** The manual settings are structured for typical dialing plans in the United States.

  - *Manual send key*: Click the drop-down box and select the key you want to press after entering a dialed number. Values are #, *, or none. If none is selected, then there is no terminating digit and the specified number of digits must be entered before the phone will dial.
  - *Total dial timer*: This is total time allowed to complete digit entry before a re-order is generated. Values are 1, 2, and 3 minutes.
  - *Interdigit timer*: This is the maximum amount of time allowed between digits entered into the phone after the first digit is pressed, before digits are sent automatically. Values are 15, 30, and 45 seconds.
  - *Extension dialing*: Click the drop-down box to select the number of digits in the telephone extension of your organization's extension configuration. Values are 3, 4, and 7 digits. Click the **Enable** check box to enable this feature or uncheck the box to disable it.
  - *Local dialing (Prefix)*: Click the drop-down box to select the prefix required for dialing offsite. Values are 8 and 9. Click the **Enable** check box to enable this feature or uncheck the box to disable it.
  - *Long distance dialing (Prefix)*: Click the drop-down box to select the prefix for dialing offsite long distance. Values are 81 and 91. Click the **Enable** check box to enable this feature or uncheck the box to disable it.
  - *Emergency dialing*: Click the drop-down box to select the number required for emergency dialing. Values are 811 and 911.
  - *Operator dialing*: Click the drop-down box to select the number required to reach the operator. Values are 0.

Click the **Apply** button to activate any changes made to the manual configuration.

CONFIGURATION: NETWORK SETTINGS SCREEN

Use the Configuration: Network Settings screen (see figure 2.10) to set up your MAX IP phone on the network. You can set the hostname, domain name, static IP address, subnet mask, default gateway, primary DNS IP address, secondary DNS IP address, SNTP server 1 IP address, SNTP server 2 IP address, VLAN priority, and VLAN ID from this screen, along with enabling/disabling DHCP and VLAN.

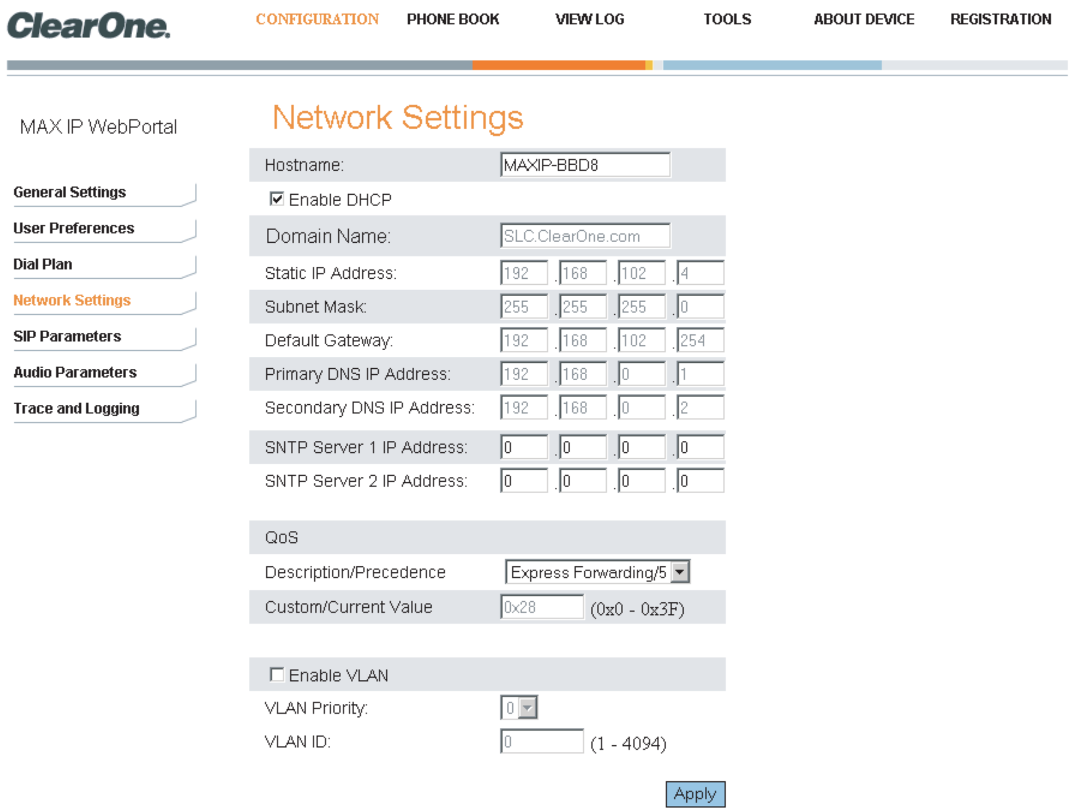

FIGURE 2.10 Configuration: Network Settings screen

- Hostname: This is the name of the host (or the MAX IP's device name on the network) and is the same as the system name found on the Device Information screen.
- Enable DHCP: Click this check box to enable DHCP on the MAX IP phone. When enabled, the settings for domain name, static IP address, subnet mask, default gateway, primary DNS IP address, and secondary DNS IP address are all grayed out as they are obtained automatically from the DHCP server. Uncheck this box to disable DHCP. Note that the above-mentioned settings are now editable as you must enter the appropriate values manually.
- SNTP Server 1 and 2 IP Addresses: The SNTP servers are time servers. Enter the IP address of the desired server and click Apply to go out and get the current time from the corresponding time server. The time returned is in Greenwich Mean Time, adjusted according to the setting of the time zone (see *Configuration: User Preferences Screen* on page 10 for more information on the time zone setting).

- **Enable VLAN, VLAN Priority, and VLAN ID**: VLAN is used to segment a single physical network into several virtual networks. It is used to differentiate between VoIP (voice over IP) data and other data. Once VLAN is enabled by clicking the check box next to **Enable VLAN**, you can set the VLAN priority. VLAN priority is the priority of the MAX IP phone on the VLAN. Click the drop-down box next to **VLAN Priority** and select the desired priority. Values are 0 to 7. The **VLAN ID** is a unique identifier set by the system administrator which can be of any value between 1 and 4094.

  > **Note:** After VLAN is enabled, you will not be able to access the MAX IP phone through the web portal unless your PC has access to the VLAN you just configured. To regain access to the web portal, you will have to disable LAN (see **To Enable/Disable VLAN** on page 32).

- **QoS (Quality of Service)**: QoS is implemented on the MAX IP using DSCP (differentiated service code point). The DSCP is a selector for a router's per-hop behaviors. Each group of DSCPs (or class) has the same precedence value, from 0 to 7, with the default precedence value for MAX IP being 5. Select the preferred precedence value from the drop-down menu; advanced users can also enter a custom DSCP value in the text field.

Click the **Apply** button to activate any changes made to the Network Settings screen.

CONFIGURATION: SIP CONFIGURATION SCREEN

Use the Configuration: SIP Configuration screen (see figure 2.11) to configure the SIP (session initiation protocol) settings for your MAX IP phone. SIP is a text-based protocol that is based on HTTP and MIME, which makes it suitable and very flexible for integrated voice-data applications. SIP relies on the session description protocol (SDP) for session description and the real-time transport protocol (RTP) for actual transport.

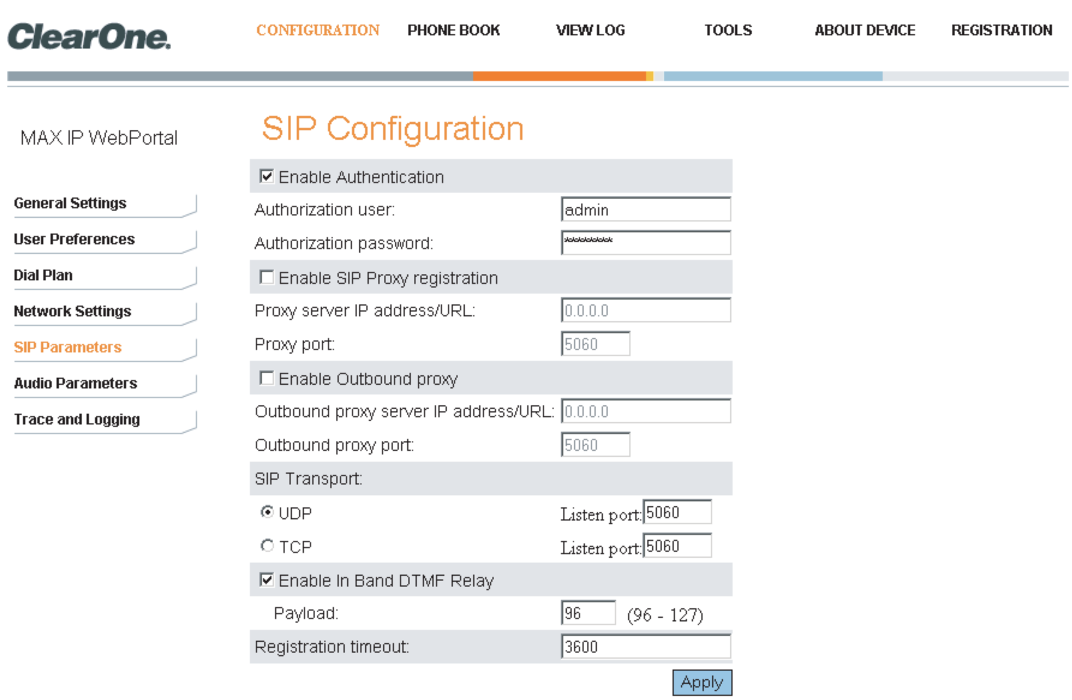

FIGURE 2.11 Configuration: SIP Configuration Settings screen

- Enable Authentication: Click this check box to enable authentication or uncheck it to disable authentication. (Authentication is required if the proxy requires it.) Authentication verifies the username and password as entered in the Authorization user and Authorization password fields. These fields are active only when authentication is enabled and can thus be modified. When authentication is disabled, these fields are inactive.
- Enable SIP Proxy registration: Click this check box to enable SIP proxy registration or uncheck it to disable it. SIP proxy registration is the connection to the SIP proxy server in a SIP-based IP telephony environment that handles call control and serves as the central repository for address translation (name to IP address). When SIP proxy registration is enabled, the Proxy server IP address/URL and Proxy port fields are active. Enter the required IP address/URL of the SIP proxy server and the number of the SIP proxy port. When SIP proxy registration is disabled, these fields are inactive.
- Enable Outbound proxy: Click this check box to enable the outbound proxy server or uncheck it to disable it. The outbound proxy is the IP address to use for outbound calls if the address is different from the registration address. When enabled, the Outbound proxy server IP address/URL and Outbound proxy port fields are active. Enter the required IP address/URL of the outbound proxy server and the number of the outbound proxy port. When disabled, these fields are inactive.
- SIP Transport: Click the radio button next to the type of SIP transport you wish to use and the number of the listen port in the Listen port field for the specified transport. UDP (user datagram protocol) is a protocol within the TCP/IP protocol suite that is used in place of TCP (transmission control protocol) when a reliable delivery is not required. UDP requires less processing of packets and is connectionless, meaning it does not require a handshake to start a session like TCP does. Therefore, it is faster and is often used with VoIP because there is no time to retransmit erroneous or dropped packets. The default port is 5060.
- Enable in Band DTMF Relay: Click the check box to enable in band DTMF (dual tone multi-frequency) relay or uncheck the box to disable it. In band DTMF relay allows dialing information to be sent for gateways that need to receive standard audio. DTMF relay provides a way to transport DTMF digits in an RTP voice stream when the voice codec cannot accurately reproduce the digits, or the sender or receiver DSP (digital signal processor) cannot perform digit sampling. Each DTMF digit is encoded as an RTP-named event and sent as RTP packets over UDP. The packets are encoded with a payload type which is negotiated during connection establishment. When enabled, the Payload field is active. Enter the desired payload: values range from 96 to 127. When Enable in Band DTMF Relay is disabled, this field is inactive.
- Registration timeout: Enter the value (in seconds) your phone is to refresh its registration to the SIP proxy server. The default value is 3600.

Click the Apply button to activate any changes made to the SIP Configuration screen.

CONFIGURATION: AUDIO SETTINGS SCREEN

Use the Configuration: Audio Settings screen (see figure 2.12) to configure the voice activation detection settings and to prioritize your preferred audio codecs.

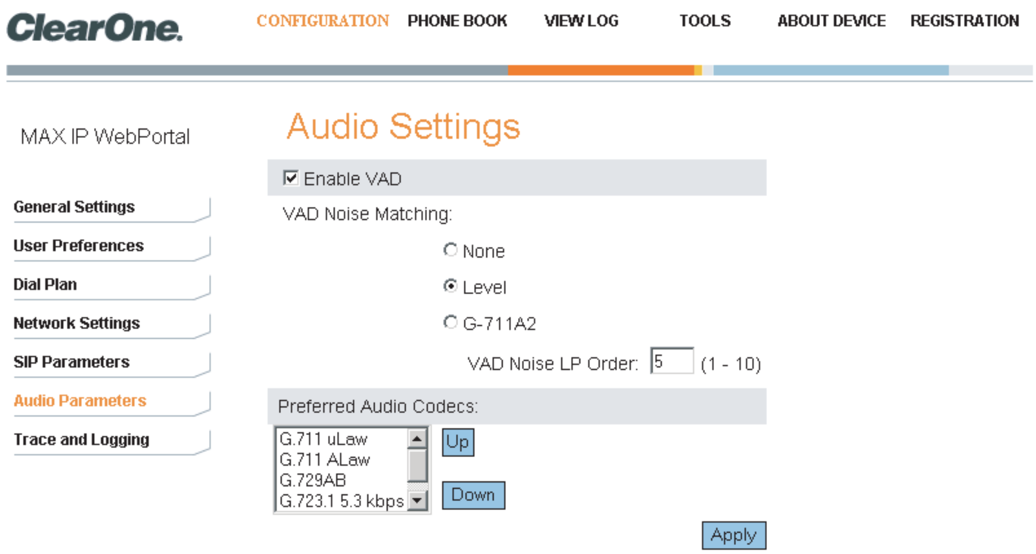

FIGURE 2.12 Configuration: Audio Settings screen

- Enable VAD: Click the check box to enable VAD (voice activity detection) or uncheck the box to disable it (VAD is enabled by default). Voice activity detection is a software application that allows a data network carrying voice traffic over the internet to detect the absence of audio and conserve bandwidth by preventing the transmission of “silent packets” over the network. VAD can also be used to forward idle noise characteristics to a remote IP telephone so that the listener will not think the line has gone dead when the speaker is not actively speaking.When VAD is enabled, VAD Noise Matching is active. When VAD is disabled, the audio sends a constant stream of audio data, even when there is silence.

  If VAD is enabled and the active audio codec is G.723.1 or G.729A/B, silence (SID) packets will be sent when silence is detected according to the description in standards G.723.1 and G.729 Appendix B, respectively.

  If VAD is enabled, G.711 is the active codec, and VAD Noise Matching is set to NONE, no audio is sent and no silence packets are sent when silence is detected.

  If VAD is enabled, G.711 is the active codec, and VAD Noise Matching is set to LEVEL, single-byte CNG packets are sent with the current noise volume level when silence is detected.

  If VAD is enabled, G.711 is the active codec, and VAD Noise Matching is set to G711A2, CNG packets are formatted according to G.711 Appendix II standard and the VAD LP Noise Order corresponds to the M coefficient in the Linear Prediction synthesis filter when silence is detected.

- VAD Noise Matching: Click the radio button for the type of VAD noise matching you prefer. VAD noise matching is the dB level of comfort noise that is transmitted that equals the floor noise in order to ensure that the receiver does not think the phone has hung up when no speaking is happening. You must also enter the VAD Noise LP Order you prefer. VAD noise matching is enabled by default and LP order values range from 1 to 10, with 5 being the default.
- Preferred Audio Codecs: Order the audio codecs in the sequence you prefer, from most preferred to least preferred. Select the codec you wish to move and then Use the Up and Down buttons to move it up/down the list. Repeat this for each codec you wish to re-order.

Click the Apply button to activate any changes made to the Audio Settings screen.

CONFIGURATION: TRACE/LOGGING SETTINGS SCREEN

Use the Configuration: Trace/Logging Settings screen (see figure 2.13) to control logging for general audio processing and SIP subsystems, as well as system logs and trace flags.

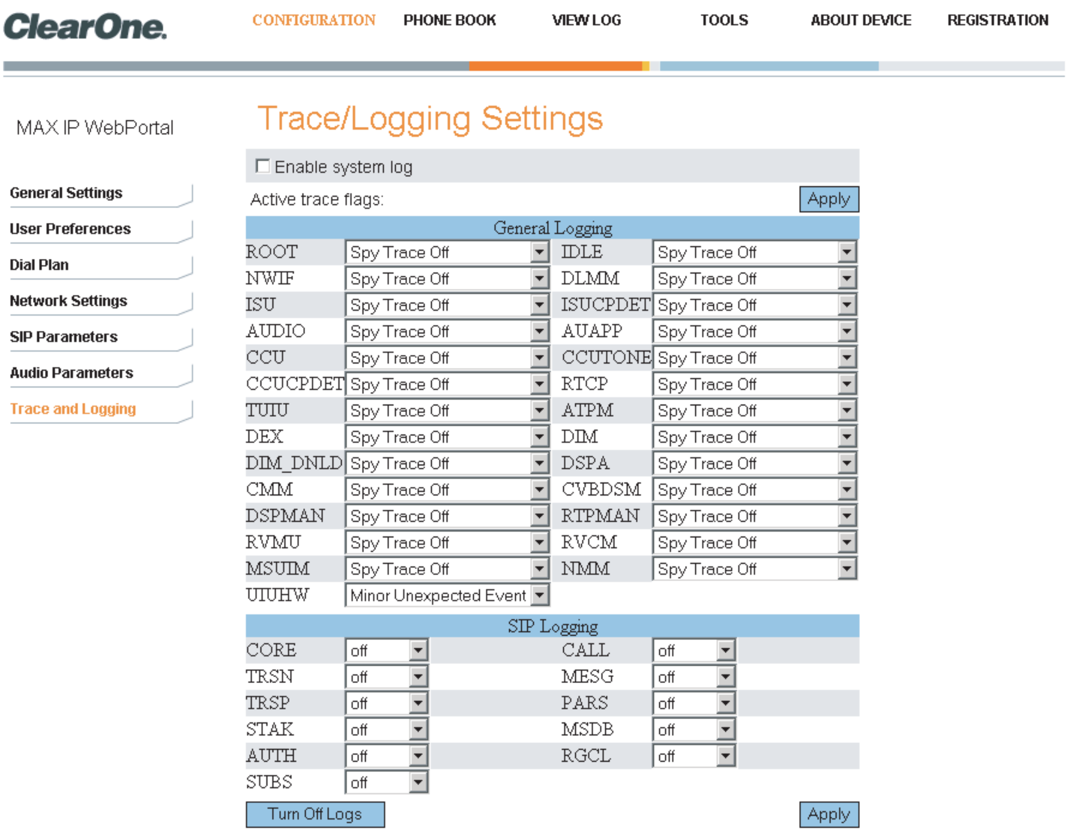

FIGURE 2.13 Configuration: Trace/Logging Settings screen

> Note: This screen is only used to diagnose problems you might be experiencing on your MAX IP phone. Before enabling any of these logs, please call customer service to receive instructions on which logs to enable.

- Enable system log: Click this check box to have the system log displayed on the Device Log File screen (see page 19).
- Active trace flags: Click the Apply button to activate trace flags.
- General Logging: These are various audio and call control subsystems for which logs can be generated if enabled.
- SIP Logging: These are various SIP subsystems for which logs can be generated if enabled.
- Turn Off Logs: Click this button to turn off logging.

Click the Apply button to cause logs selected in General Logging and SIP Logging sections to be displayed on the Device Log File screen.

PHONEBOOK: ADD, EDIT AND DELETE NUMBER SCREEN

Use the Phonebook: Add, Edit and Delete Number screen (see figure 2.14) to add, modify, and delete numbers from your phonebook.

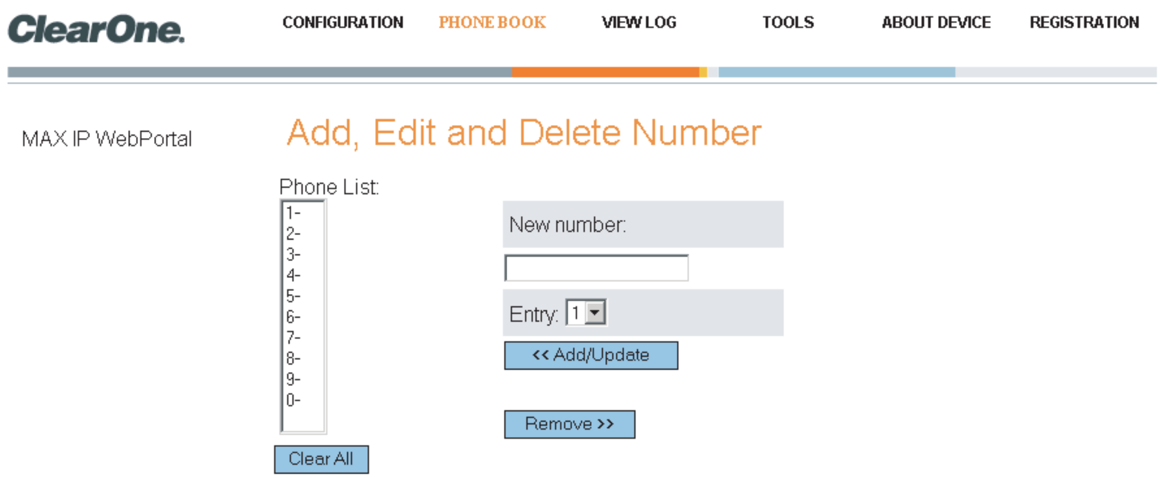

FIGURE 2.14 Phonebook: Add, Edit, and Delete Number screen

- Phone List: This is a list of numbers you have stored in your phonebook. You can have up to 10 numbers stored at a time. To clear all numbers from your list, click the Clear All button.
- New number: Enter the telephone number in the New number field, including area code, that you wish to add to the list. (Enter <P> to program a two-second pause.) Click the Entry drop-down box and select the number you want telephone number assigned to in the phone list. Then click the Add/Update button to add the number to the list. Click the Remove button to remove a number from the list. Note that the entry will also disappear from the phone list but can be added back with the simple addition of a new number stored at that location.

VIEW LOG: DEVICE LOG FILE SCREEN

Use the View Log: Device Log File screen (see figure 2.15) to keep record of device log data. You can then download the log for review by clicking the Download button. The log shows the last 8 KB of log data. Click the Update button to get the most recent 8 KB of data. Click the Clear button to clear the log.

> Note: This screen is only used to diagnose problems you might be experiencing on your MAX IP phone. Please call customer service to receive instructions on how to interpret the logs shown here.

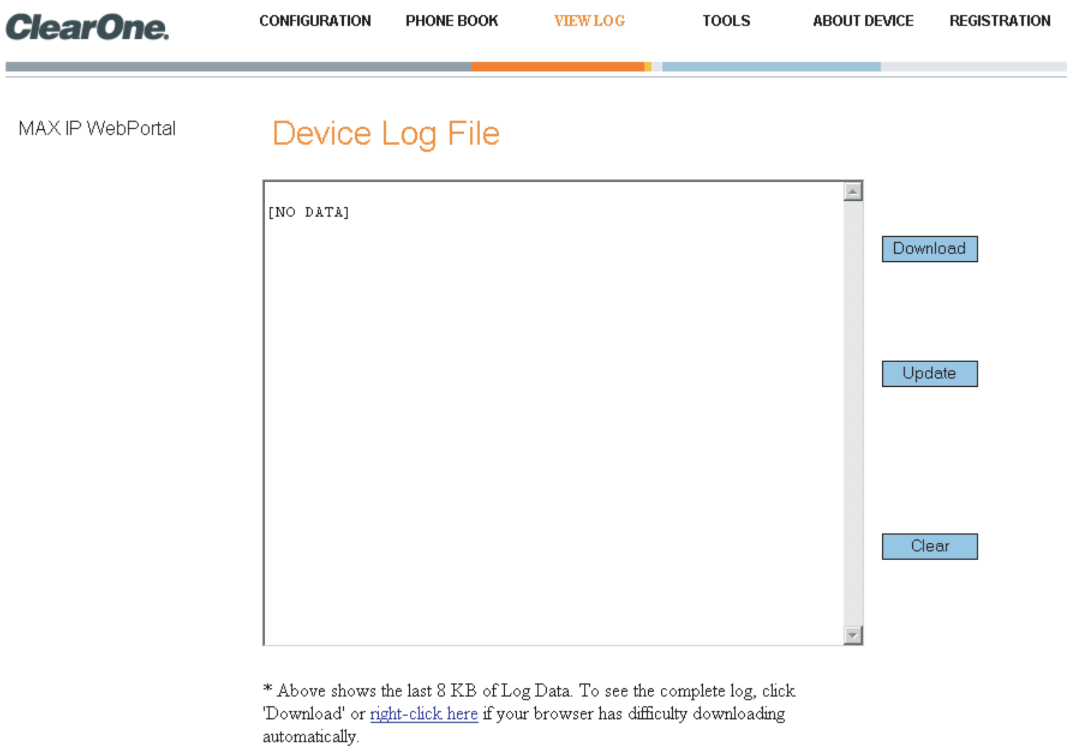

FIGURE 2.15 View Log: Device Log File screen

TOOLS: DIAGNOSTICS - VOIP STATISTICS SCREEN

Use the Tools: Diagnostics - VoIP Statistics screen (see figure 2.16) to check the phone state, update firmware, reboot the phone, and restore default settings.

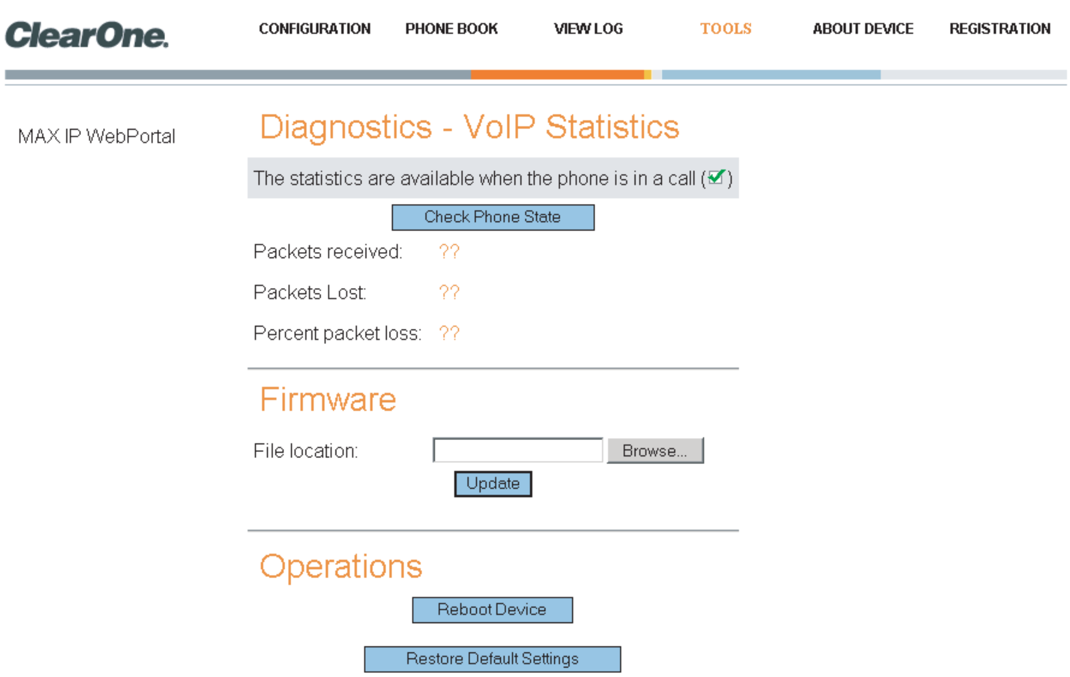

FIGURE 2.16 View Log: Device Log File screen

- Check Phone State: Click this button to view VoIP statistics, including the number of packets received, the number of packets lost, and the percent packet loss. These statistics are displayed in real time and are only available when the phone is in a call. At such time, a green check mark appears in the check box above the Check Phone State button.
- Update Firmware: You will receive the firmware update file (ggsip_all, for example) and, using the web interface only, you will enter that name directly or browse to its location on your computer by using the Browse button, then clicking the Update button. The MAX IP phone's firmware is updated.
- Reboot Device: Click this button to reboot your MAX IP phone.
- Restore Default Settings: Click this button to restore the default settings of your MAX IP phone.

AUTOMATICALLY PROVISIONING YOUR MAX IP PHONE

As mentioned previously, if you have multiple units that you want to provision with the same settings, the quickest and most seamless way of accomplishing this is through automatic provisioning.

In fact, your MAX IP phone is factory-configured to automatically provision by downloading the appropriate configuration files from a TFTP server defined by DHCP on bootup. The settings contained in the configuration file override the default settings stored in the MAX IP phone.

Several configuration files are required to automatically provision the MAX IP phone. These configuration files include:

- Firmware files
- Phone Settings file
- Phone-specific file
- Dial Plan file

FIRMWARE FILES

When new firmware is released, it is delivered with two files: config.fil and ggsip_all. The config.fil file contains date and version information, while the ggsip_all file contains the compressed firmware image. When a MAX IP phone is plugged into the network, the config.fil file is downloaded via TFTP. If the date and version are different from what is stored on the phone, the ggsip_all file is then downloaded. After the new firmware has loaded, the MAX IP reboots and the new firmware becomes active once the reboot completes.

PHONE SETTINGS AND PHONE-SPECIFIC FILES

Two phone configuration files are used to provision the MAX IP phone: the phone settings file and phone-specific file. The phone settings file contains general settings used by all MAX IP phones on a network. This file MUST be named `C1MAXIP.txt`.

The phone-specific file contains phone-specific settings. This file must be named `C1MAXIP_`*`MACAddress`*`.txt`, where the MAC address is the hardware Ethernet MAC address found on the label on the back of the phone's base unit.

Although all of the phone settings can be defined in one of these two files, the general settings found in the phone settings file will be overridden by the settings in the phone-specific file when the MAX IP phone boots. To leave a setting as is, do not include it in the file; only address those items that are changing.

You can edit the files using a general text editor, such as "vi" or "Notepad." A sample `C1MAXIP.txt` file, containing configuration settings, is shown in figure 2.17.

> Note: Although the phone settings configuration file appears to be well-formed XML, it MUST adhere to the formatting as defined in the example. Parameter settings CANNOT be spread across multiple lines. For example:

Valid:
```
<username> admin </username>
```

Invalid:
```
<username>
admin
</username>
```

```
<C1MAXSIPCONFIG>
   <username> admin </username>                       <!-- web login id -->
   <password> clearone </password>                    <!-- web login password -->
   <ringtone> 1 </ringtone>                           <!-- ringtone index 1 to 5 -->
   <localnum> 1234 </localnum>                        <!-- Local telephone number -->
   <helpline_num> 6356 </helpline_num>                <!-- helpline number -->
   <allow_reboot_in_call> 0  </allow_reboot_in_call> <!-- 1 - allow reboot during a call
                                                           0 - wait till call complete -->
   <mute_ringtone> 0 </mute_ringtone>                 <!-- 1 - mute ringer on incoming call 0 - disable mute -->
   <dialplan> C1MAX1AVDIAL.txt </dialplan>            <!-- File describing MAX Dialplan - TFTPed from server -->
   <timezone> 5 </timezone>                          <!—Timezone – MST -->
   <SNTP_server_1> 0.0.0.0 </SNTP_server_1>           <!-- SNTP 1 IP address -->
   <SNTP_server_2> 0.0.0.0 </SNTP_server_2>           <!-- SNTP 2 IP address -->
   <DSCP_TOS_BITS> 0 </DSCP_TOS_BITS>                 <!-- Diffserv Code Point Type of Service bits 0-7 -->
   <speed_dial_0> </speed_dial_0>                     <!-- Speed dial 0 -->
   <speed_dial_1> </speed_dial_1>                     <!-- Speed dial 1 -->
   <speed_dial_2> </speed_dial_2>                     <!-- Speed dial 2 -->
   <speed_dial_3> </speed_dial_3>                     <!-- Speed dial 3 -->
   <speed_dial_4> </speed_dial_4>                     <!-- Speed dial 4 -->
   <speed_dial_5> </speed_dial_5>                     <!-- Speed dial 5 -->
   <speed_dial_6> </speed_dial_6>                     <!-- Speed dial 6 -->
   <speed_dial_7> </speed_dial_7>                     <!-- Speed dial 7 -->
   <speed_dial_8> </speed_dial_8>                     <!-- Speed dial 8 -->
   <speed_dial_9> </speed_dial_9>                     <!-- Speed dial 9 -->
   <adjust_dst> 1 </adjust_dst>                       <!-- 1 - adjust for daylight savings time 0 - disable -->
   <use_sipauth> 1 </use_sipauth>                     <!-- 1 - enable SIP authentication
                                                           0 - disable sip authentication -->
   <sip_username> test </sip_username>               <!-- sip authentication username -->
   <sip_password> mypwd </sip_password>                <!-- sip authentication password -->
   <sip_proxy_enable> 1 </sip_proxy_enable>           <!-- 1 - enable 0 - disable -->
   <sip_proxy_server> 192.168.0.1 </sip_proxy_server><!-- sip proxy server IP, hostname or FQDN -->
   <sip_proxy_port> 5060 </sip_proxy_port>            <!-- sip proxy server port -->
   <outbound_sip_proxy_enable> 1 </outbound_sip_proxy_enable>   <!-- 1 - enable 0 - disable -->
   <outbound_sip_proxy> 192.168.0.1 </outbound_sip_proxy> <!-- sip outbound proxy IP, hostname or FQDN -->
   <outbound_proxy_port> 5060 </outbound_proxy_port> <!-- sip outbound proxy port -->
   <sip_register_timetout> 3600 </sip_register_timeout> <!-- sip registration timeout in seconds -->
   <sip_transport> 0 </sip_transport>                 <!-- sip transport: 0 - udp or 1 - tcp -->
   <sip_udp_port> 5060 </sip_udp_port>                <!-- sip udp port -->
   <sip_tcp_port> 5060 </sip_tcp_port>                <!-- sip tcp port -->
   <dtmf_relay_enable> 1 </dtmf_relay_enable>         <!-- 1 - enable inband dtmf relay 0 - disable -->
   <dtmf_relay_payload> 96 </dtmf_relay_payload>      <!-- dtmf relay payload - 96 to 127 -->
   <vad_enable> 1 </vad_enable>                       <!-- 1 - enable voice activity detection 0 - disable -->
   <adaptive_vad> 1 </adaptive_vad>                   <!-- 1 - enable adaptive vad 0 - disable -->
   <vad_thresh> -20 </vad_thresh>                     <!-- vad threshold -20 to +10 -->
   <vad_noise_match> none </vad_noise_match>        <!-- vad noise matching none, level, g711a2 -->
   <vad_noise_order> 0 </vad_noise_order>           <!-- vad noise order 0 to 10 -->
   <g711ulaw_priority> 255 </g711ulaw_priority>     <!-- g.711ulaw codec priority 1(lowest) to 255(highest) -->
   <g711Alaw_priority> 254 </g711Alaw_priority>     <!-- g.711Alaw codec priority 1(lowest) to 255(highest) -->
   <g729ab_priority> 250 </g729ab_priority>         <!-- g.729A/B codec priority 1(lowest) to 255(highest) -->
   <g7231_53_priority> 240 <g7231_53_priority>      <!-- g.723.1 lowrate priority 1(lowest) to 255(highest) -->
   <g7231_63_priority> 245 <g7231_63_priority>      <!-- g.723.1 highrate priority 1(lowest) to 255(highest) -->
</C1MAXSIPCONFIG>
```

FIGURE 2.17 Phone settings file example

Configuration Parameters

The following parameters are the parameters available for configuring the phone settings and phone-specific files.

<username>

The username for logging into the web portal. (See *Manually Provisioning Your MAX IP Phone* on page 7 for the login procedure.)

Default value: admin

<password>

The password for logging into the web portal. (See *Manually Provisioning Your MAX IP Phone* on page 7 for the login procedure.)

Default value: clearone

<ringtone>

The default ring tone when the phone rings. (See *Configuration: General Settings Screen* on page 10 for information on how to set this parameter through the web portal.)

Default value: 1

Range: 1-5

<localnum>

The localnum parameter describes the identifier by which the phone will be known. For example, if localnum is set to 1234 and the phone is registered to ClearOneProxy.com, then the phone's SIP URI will be sip:1234@ClearOneProxy.com. (See *Configuration: General Settings Screen* on page 9 for information on how to set this parameter through the web portal.)

Default value: 1111111

Allowable characters: [0-9]

<helpline_num>

The number dialed when the help line speed dial is called. (See *Configuration: General Settings Screen* on page 9 for information on how to set this parameter through the web portal.)

Default value: not set

Allowable characters: [0-9]

<allow_reboot_in_call>

Allow the phone to be rebooted if it's currently in a call. This parameter is useful when remote provisioning through the web interface. If a reboot command is issued remotely, it will not take immediate effect if this parameter is set to 0. (See *Configuration: General Settings Screen* on page 9 for information on how to set this parameter through the web portal.)

Default value: 0

Allowable values:

0 - wait until the call is completed before the phone is rebooted

1 - reboot phone immediately

<mute_ringtone>

Mute the ringer for an incoming call. Note, if the ringer is disabled, the LED indicators on the pod(s) will still flash on an incoming call. (See *Configuration: User Preferences Screen* on page 10 for information on how to set this parameter through the web portal.)

Default value: 0

Allowable values:

0 - disable mute

1 - enable mute

<dialplan>

The filename on the TFTP server containing the dial plan file. For information on configuring a dial plan, see *Configuration: Dial Plan Screen* on page 11.

Default value: undefined

Allowable values: Ascii text filename limited by the filename length limit on the TFTP server.

<timezone>

The time zone in which the phone resides. (See *Configuration: User Preferences Screen* on page 10 for information on how to set this parameter through the web portal.)

Default value: 5

Allowable parameters:

0 - GMT-12:00 (International Date Line West)

1 - GMT-11:00 (Midway Island, Samoa)

2 - GMT-10:00 (Hawaii)

3 - GMT-09:00 (Alaska)

4 - GMT-08:00 (Pacific Time(US & Canada); Tijuana)

5 - GMT-07:00 (Mountain Time(US & Canada); Arizona; Chihuahua; LaPaz; Mazatlan)

6 - GMT-06:00 (Central Time(US & Canada); Central America; Guadalajara; Mexico City; Monterrey; Saskatchewan)

7 - GMT-05:00 (Eastern Time(US & Canada); Bogota; Lima; Quito)

8 - GMT-04:00 (Atlantic Time(Canada); Caracas; La Paz; Santiago)

9 - GMT-03:30 (Newfoundland)

10 - GMT-03:00 (Brasil; Buenos Aires; Greenland)

11 - GMT-02:00 (Mid Atlantic)

12 - GMT-01:00 (Azores; Cape Verde Island)

13 - GMT (London; Edinburgh; Lisbon)

14 - GMT+01:00 (Paris; Stockholm; Madrid; Brussels; Copenhagen; West central Africa)

15 - GMT+02:00 (Athens; Beirut; Istanbul; Cairo; Jerusalem; Helsinki)

16 - GMT+03:00 (Baghdad; Moscow; Kuwait; Nairobi)

17 - GMT+03:30 (Tehran)

18 - GMT+04:00 (Abu Dhabi; Muscat)

19 - GMT+04:30 (Kabul)

20 - GMT+05:00 (Islamabad; Karachi; Tashkent)

21 - GMT+05:30 (New Delhi; KolKata; Mumbai)

22 - GMT+05:45 (Kathmandu)

23 - GMT+06:00 (Dhaka;Sri Jayawardenepura)

24 - GMT+06:30 (Rangoon)

25 - GMT+07:00 (Bangkok; Hanoi; Jakarta)

26 - GMT+08:00 (Beijing; Hong Kong; Kuala Lampur; Singapore; Perth; Taipei)

27 - GMT+09:00 (Tokyo; Osaka; Seoul)

28 - GMT+09:30 (Adelaide; Darwin)

29 - GMT+10:00 (Brisbane; Melbourne; Sidney; Guam; Vladivostok)

30 - GMT+11:00 (Solomon Islands; New Caledonia)

31 - GMT+12:00 (Auckland; Wellington; Fiji; Marshall Islands)

32 - GMT+13:00 (Nuku'alofa)

<adjust_dst>

Adjust for daylight savings time. (See *Configuration: User Preferences Screen* on page 10 for information on how to set this parameter through the web portal.)

Default value: 1

Allowable values:

0 - disable

1 - enable

<SNTP_server_1>

SNTP 1 server IP address. (See *Configuration: Network Settings Screen* on page 13 for information on how to set this parameter through the web portal.)

Default value: 0.0.0.0

Allowable values: 0.0.0.0 (disabled) or valid IP address.

<SNTP_server_2>

SNTP 2 server IP address. (See *Configuration: Network Settings Screen* on page 13 for information on how to set this parameter through the web portal.)

Default value: 0.0.0.0

Allowable values: 0.0.0.0 (disabled) or valid IP address

<speed_dial_0> through **<speed_dial_9>**

Speed dial 0 through 9. (See *Phonebook: Add, Edit and Delete Number Screen* on page 18 for information on how to set this parameter through the web portal.)

Default value: undefined

Allowable range: valid phone number [0-9]

<use_sipauth>

Use SIP authentication when registering with the SIP proxy. (See *Configuraton: SIP Configuration Screen* on page 14 for information on how to set this parameter through the web portal.)

Default value: 0

Allowable range:

0 - disable

1 - enable

<sip_username>

The user name with which the phone will authenticate with the SIP proxy if <use_sipauth> is enabled. (See *Configuraton: SIP Configuration Screen* on page 14 for information on how to set this parameter through the web portal.)

Default value: none

Allowable string length: 49

<sip_password>

The password with which the phone will authenticate with the SIP proxy if `<use_sipauth>` is enabled and `<sip_username>` is defined. (See *Configuraton: SIP Configuration Screen* on page 14 for information on how to set this parameter through the web portal.)

Default value: none

Allowable string length: 14

<sip_proxy_enable>

Enable SIP proxy registration. (See *Configuraton: SIP Configuration Screen* on page 14 for information on how to set this parameter through the web portal.)

Default value: 0

Allowable range:

0 - disable

1 - enable

<sip_proxy_server>

SIP proxy server to register with when `<sip_proxy_enable>` is enabled. This parameter can be an IP address, a hostname, or FQDN. (See *Configuraton: SIP Configuration Screen* on page 14 for information on how to set this parameter through the web portal.)

Default value: 0.0.0.0

Allowable string length: 79

<sip_proxy_port>

The default port with which to communicate to the SIP proxy. (See *Configuraton: SIP Configuration Screen* on page 14 for information on how to set this parameter through the web portal.)

Default value: 5060

Allowable port range: 1 - 65535

<outbound_sip_proxy_enable>

Enable call routing through outbound SIP proxy. (See *Configuraton: SIP Configuration Screen* on page 14 for information on how to set this parameter through the web portal.)

Default value: 0

Allowable range:

0 - disable

1 - enable

<outbound_sip_proxy>

Outbound SIP proxy address. This can be a valid IP address, a hostname, or FQDN. (See *Configuraton: SIP Configuration Screen* on page 14 for information on how to set this parameter through the web portal.)

Default value: 0.0.0.0

Allowable string length: 79

<outbound_proxy_port>

The default port with which to communicate to the outbound SIP proxy. (See *Configuraton: SIP Configuration Screen* on page 14 for information on how to set this parameter through the web portal.)

Default value: 5060

Allowable port range: 1 - 65535

<sip_register_timetout>

The SIP registration timeout in milliseconds. (See *Configuraton: SIP Configuration Screen* on page 14 for information on how to set this parameter through the web portal.)

Default value: 3600

Allowable range: 0 - 4294967295 (0 = disabled)

<sip_transport>

The SIP transport type. (See *Configuraton: SIP Configuration Screen* on page 14 for information on how to set this parameter through the web portal.)

Default value: 0

Allowable range:

0 - UDP

1 - TCP

<sip_udp_port>

The SIP UDP listen port. (See *Configuraton: SIP Configuration Screen* on page 14 for information on how to set this parameter through the web portal.)

Default value: 5060

Allowable port range: 0 - 65535

<sip_tcp_port>

The SIP TCP listen port. (See *Configuraton: SIP Configuration Screen* on page 14 for information on how to set this parameter through the web portal.)

Default value: 5060

Allowable port range: 0 - 65535

<dtmf_relay_enable>

The DTMF relay enable. (See *Configuraton: SIP Configuration Screen* on page 14 for information on how to set this parameter through the web portal.)

Default value: 1

Allowable range:

0 - Disable

1 - Enable in-bound DTMF relay

<dtmf_relay_payload>

The DTMF relay RTP packet payload. (See *Configuraton: SIP Configuration Screen* on page 14 for information on how to set this parameter through the web portal.)

Default value: 97

Allowable range: 96 - 127

<vad_enable>

Enable Voice Activity Detection. (See *Configuraton: Audio Settings Screen* on page 16 for information on how to set this parameter through the web portal.)

Default value: 1

Allowable range:

0 - Disable

1 - Enable VAD

<vad_noise_match>

Defines the VAD noise matching algorithm. (See *Configuraton: Audio Settings Screen* on page 16 for information on how to set this parameter through the web portal.)

Default value: level

Allowable range:

none - disabled

level

g711a2

<vad_noise_order>

The VAD noise order. (See *Configuraton: Audio Settings Screen* on page 16 for information on how to set this parameter through the web portal.)

Default value: 5

Allowable range: 0 - 10

<g711ulaw_priority>

The G.711 ulaw audio codec priority. (See *Configuraton: Audio Settings Screen* on page 16 for information on how to select this codec priority through the web portal.)

Default value: 255

Allowable range: 1 (lowest) to 255 (highest)

<g711Alaw_priority>

The G.711 Alaw audio codec priority. (See *Configuraton: Audio Settings Screen* on page 16 for information on how to select this codec priority through the web portal.)

Default value: 254

Allowable range: 1 (lowest) to 255 (highest)

<g729ab_priority>

The G.729A/B audio codec priority. (See *Configuraton: Audio Settings Screen* on page 16 for information on how to select this codec priority through the web portal.)

Default value: 250

Allowable range: 1 (lowest) to 255 (highest)

<g7231_63_priority>

The G.723.1 low rate audio codec priority. (See *Configuraton: Audio Settings Screen* on page 16 for information on how to select this codec priority through the web portal.)

Default value: 245

Allowable range: 1 (lowest) to 255 (highest)

<g7231_53_priority>

The G.723.1 low rate audio codec priority. (See *Configuraton: Audio Settings Screen* on page 16 for information on how to select this codec priority through the web portal.)

Default value: 240

Allowable range: 1 (lowest) to 255 (highest)

<vlan_enable>

The Virtual LAN enable. (See *Configuraton: Network Settings Screen* on page 13 for information on how to set this parameter through the web portal.)

Default value: 0

Allowable range:

0 - Disable

1 - 4094 Valid LAN ID

<vlan_priority>

Sets the priority that VLAN tags outbound packets. (See *Configuraton: Network Settings Screen* on page 13 for information on how to set this parameter through the web portal.)

Default value: 0

Allowable range: 0-7

<agc_enable>

Automatic gain control enable. (See *Configuraton: User Preferences Screen* on page 10 for information on how to set this parameter through the web portal.)

Default value: 0

Allowable range:

0 - Disable

1 - Enable

<alc_enable>

Automatic level control enable. (See *Configuraton: User Preferences Screen* on page 10 for information on how to set this parameter through the web portal.)

Default value: 0

Allowable range:

0 - Disable

1 - Enable

<qos_precedence>

Quality of service precedence. (See *Configuraton: Network Settings Screen* on page 13 for information on how to set this parameter through the web portal.)

Default value: 5

Allowable range:

0 - DSCP = 0x00

1 - DSCP = 0x08

2 - DSCP = 0x10

3 - DSCP = 0x18

4 - DSCP = 0x20

5 - DSCP = 0x28

6 - DSCP = 0x30

7 - DSCP = 0x38

8 - DSCP = CUSTOM

<qos_custom_dscp>

Quality of service custom DSCP. Valid if qos_precedence is set to 8. (See *Configuraton: Network Settings Screen* on page 13 for information on how to set this parameter through the web portal.)

Default value: n/a

Allowable range: 0x00 to 0x3F

DIAL PLAN CONFIGURATION FILE

The dial plan configuration file defines rules for gathering digits when dialing a phone number and also defines the mapping of the gathered digits to a specific target. A sample dial plan is shown in figure 2.18.

```
<C1DIALPLAN>
   <SYSCONFIG DIALTIME="120000" FIRST_DIGIT_WAIT="30000" INTER_DIGIT_WAIT="30000" TERMINATION_DIGIT="#"/>
   <DIGITMAP MATCH="911" MIN_DIGITS="3" MAX_DIGITS="3" STRIP_FIRST_DIGITS="0" ADD_PREFIX_AFTER_STRIP=""
DIAL_STRING="+&@sipgateway.com"/>   <!-- 911 Emergency -->
   <DIGITMAP MATCH="+&" MIN_DIGITS="4" MAX_DIGITS="4" STRIP_FIRST_DIGITS="0" ADD_PREFIX_AFTER_STRIP=""
DIAL_STRING="+&@sipproxy.com"/>      <!-- Enterprise extenions -->
   <DIGITMAP MATCH="9" MIN_DIGITS="8" MAX_DIGITS="43" STRIP_FIRST_DIGITS="1" ADD_PREFIX_AFTER_STRIP=""
DIAL_STRING="+&@sipgateway.com"/>    <!-- Outside dialing -->
   <DIGITMAP MATCH="0" MIN_DIGITS="1" MAX_DIGITS="1" STRIP_FIRST_DIGITS="0" ADD_PREFIX_AFTER_STRIP=""
DIAL_STRING="operator@sipproxy.com"/> <!-- Operator -->

</C1DIALPLAN>
```

FIGURE 2.18 Sample dial plan configuration file

> Note: All tokens associated with `SYSCONFIG` and `DIGITMAP` must appear on separate single lines in the actual configuration file.

Dial Plan Configuration File Tokens

The following tokens are used for configuring the dial plan configuration file.

The **`SYSCONFIG`** token defines timer configuration and termination digit parameters.

The **`DIALTIME`** token defines the total time in milliseconds allowed to enter the dialed digits before the phone will play a re-order tone.

The **`FIRST_DIGIT_WAIT`** token defines the time in milliseconds the phone will wait after going off-hook to enter the first digit before the a re-order tone is played.

The **`INTER_DIGIT_WAIT`** token defines the time in milliseconds the phone will wait after the first digit is entered before a re-order tone is played or the number is dialed.

The **`TERMINATION_DIGIT`** token defines the termination digit to be entered if the maximum number of digits have not yet been entered and the number is to be dialed before the `INTER_DIGIT_WAIT` timer is expired.

The **`DIGITMAP`** token defines the mapping of collected digits to an outbound SIP URI.

The **`MATCH`** token defines the digits which MUST be matched when the user begins entering digits for the `DIGITMAP` rule to take effect.

The **`MIN_DIGITS`** token defines the minimum number of digits which MUST be entered once that match rule has been invoked. This number must be greater than or equal to the number of digits in the `MATCH` string.

The **`MAX_DIGITS`** token defines the maximum number of digits which MAY be entered after the match rule has been invoked. The completion of the number can be achieved when the maximum number of digits have been entered or the `TERMINATION_DIGIT` is pressed. The `MAX_DIGITS` parameter MUST be greater than or equal to the `MIN_DIGITS` parameter.

The **`STRIP_FIRST_DIGITS`** parameter defines the number of digits that will be stripped from the beginning of the complete dial string before it is passed to the underlying stack to be dialed. For example, if the user entered 1234 and `STRIP_FIRST_DIGITS` was set to 2, the string passed to the underlying stack for dialing would be 34.

The **`ADD_PREFIX_AFTER_STRIP`** token defines a set of prefix characters that are to be applied to the beginning of the dial string AFTER the `STRIP_FIRST_DIGITS` rule has been applied. Adding to the previous example, if the `ADD_PREFIX_AFTER_STRIP` were set to "56" and the user entered 1234, the string passed to the underlying stack would be 5634.

The **`DIAL_STRING`** token defines the address which will be dialed when a number satisfying the `MATCH` rule is entered.

The characters "+&" define a wild card, consisting of the digits in the number dialed. In the example shown above, when any four-digit number that was dialed is entered, it is passed to the stack as "`<four-digit number>@sipgateway.com`".

> Note: Although the wild card parameter is defined in the `MATCH` string and in the `DIAL_STRING`, it assumes that the rules applied for `STRIP_FIRST_DIGITS` and `ADD_PREFIX_AFTER_STRIP` still take effect before the entered number replaces the wild card in the `DIAL_STRING`.

CHAPTER 3: USER OPTIONS

PROGRAMMING OPTIONS

To allow for individual preferences and enhance ease of use, the following features can be programmed: DHCP, host IP, subnet mask, default gateway IP address, ringer melody, VLAN on/off, help line number, and AGC/ALC. You can also restore factory defaults.

TO CHANGE DYNAMIC HOST CONFIGURATION PROTOCOL (DHCP)

1. Press and hold the REDIAL/PROG key until the Program icon appears on the LCD screen (see figure 3.1).

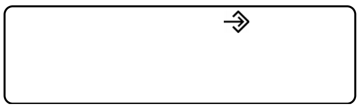

FIGURE 3.1 MAX IP LCD Program icon

2. Press 1 to enter the DHCP menu. There are two options: 1 - enable DHCP; and 0 - disable DHCP.
3. Press REDIAL/PROG. The current DHCP setting flashes.
4. Press keys 1 or 0 to enable or disable DHCP.
5. Press REDIAL/PROG to save the selection; a confirmation tone plays. Press CLEAR to exit programming. The phone reboots.

TO CHANGE HOST IP ADDRESS

1. Press and hold the REDIAL/PROG key until the Program icon appears on the LCD screen (see figure 3.1).

> Note: DHCP must be disabled in order to change the host IP address manually.

2. Press 2 to enter the Host IP Address menu. The current host IP address is displayed.
3. Press REDIAL/PROG. The current host IP address flashes.
4. Press and hold CLEAR to erase the current host IP address.
5. Using the number keys, enter the desired host IP address.
6. Press REDIAL/PROG to save the selection; a confirmation tone plays. Press CLEAR to exit programming. The phone reboots.

TO CHANGE SUBNET MASK

1. Press and hold the REDIAL/PROG key until the Program icon appears on the LCD screen (see figure 3.1).

> Note: DHCP must be disabled in order to change the subnet mask manually.

2. Press 3 to enter the Change Subnet Mask menu. The current subnet mask is displayed.
3. Press REDIAL/PROG. The current subnet mask flashes.
4. Press and hold CLEAR to erase the current subnet mask.
5. Using the number keys, enter the desired subnet mask.
6. Press REDIAL/PROG to save the selection; a confirmation tone plays. Press CLEAR to exit programming. The phone reboots.

TO PROGRAM DEFAULT GATEWAY IP ADDRESS

1. Press and hold REDIAL/PROG until the Program icon appears on the LCD screen (see figure 3.1).

> Note: DHCP must be disabled in order to change the subnet mask manually.

2. Press 4 to enter the Default Gateway IP Address menu. The current default gateway IP address is displayed.
3. Press REDIAL/PROG. The current default gateway IP address flashes.
4. Press and hold CLEAR to erase the current default gateway IP address.
5. Using the number keys, enter the desired default gateway IP address.
6. Press REDIAL/PROG to save the selection; a confirmation tone plays. Press CLEAR to exit programming. The phone reboots.

TO CHANGE RINGER MELODY

1. Press and hold the REDIAL/PROG key until the Program icon appears on the LCD screen (see figure 3.1).
2. Press 5 to enter the Ringer Melody menu. There are five available melodies.
3. Press REDIAL/PROG. The current melody selection flashes.
4. Press keys 1–5 to play the corresponding melody. The selected melody plays once.
5. Press REDIAL/PROG to save the selection; a confirmation tone plays. Press CLEAR to exit programming.

TO PROGRAM HELP LINE NUMBER

1. Press and hold the REDIAL/PROG key until the Program icon appears on the LCD screen (see figure 3.1).
2. Press 7 to enter the help line number.
3. Press REDIAL/PROG to save the selection; a confirmation tone plays. Press CLEAR to exit programming.

TO ENABLE/DISABLE VLAN

1. Press and hold the REDIAL/PROG key until the Program icon appears on the LCD screen (see figure 3.1).
2. Press 6 to select VLAN programming. The current VLAN setting appears on the LCD screen (the default setting is to off).
3. Press REDIAL/PROG. The current VLAN setting flashes.
4. Press keys 1 or 0 to enable or disable VLAN.
5. Press REDIAL/PROG to save the selection; a confirmation tone plays. Press CLEAR to exit programming. The phone reboots.

TO PROGRAM THE AUTOMATIC GAIN CONTROL (AGC) AND AUTOMATIC LEVEL CONTROL (ALC) SETTINGS

1. Press and hold the REDIAL/PROG key until the Program icon appears on the LCD screen (see figure 3.1).
2. Press and hold the 8 key. The current AGC setting number will be displayed on the LCD screen.
3. Press REDIAL/PROG. The current setting number will flash.
4. Enter a new setting number using the table shown in figure 3.2.

Setting	Speaker AGC	Microphone ALC
1	On	On
2	On	Off
3	Off	On
4	Off	Off

FIGURE 3.2 MAX IP automatic gain control table

5. Press REDIAL/PROG to save the selection; a confirmation tone plays. Press CLEAR to exit programming.

TO RESTORE FACTORY DEFAULTS

1. Press and hold the REDIAL/PROG key until the Program icon appears on the LCD screen (see figure 3.1).
2. Press and hold the 9 key. The number 8 repeats across the LCD screen.
3. Press REDIAL/PROG to save the selection; a confirmation tone plays. Press CLEAR to exit programming. The phone reboots.

CHAPTER 4: MAINTENANCE

CARING FOR YOUR MAX IP

- Follow all warnings and instructions marked on your MAX IP.
- Unplug base unit and conferencing pod from the wall outlet before cleaning.
- Do not use liquid or aerosol cleaners. Use a damp cloth moistened with water to clean the outside of your conferencing pod or base unit and power supply.

TROUBLESHOOTING

If you are having trouble with your MAX IP, it might be improperly set up or other equipment might be malfunctioning. To begin, check for the following (or consult the chart in figure 4.1):

- The MAX IP base unit is plugged into an electrical outlet with the proper voltage and its power light is on.
- Make sure cables are securely connected.
- The equipment the other party is using is comparable in quality to your MAX IP conferencing phone and is working properly. While the MAX IP works with speakerphones, cell phones, handsets or installed conferencing systems, the quality of the conference will be impacted if the party you are conferencing with has poor quality equipment.

If you hear/see	It could be that	Try this
No dial tone.	Base unit is not connected to Ethernet jack. DHCP failure.	Connect the base unit to the Ethernet jack using the supplied Ethernet cable. Check for IP address.
Static or noise.	The other party's room is too noisy. Packet loss or delay.	Ask the other party to turn off noisy equipment. Contact Network Administrator.
Calls can come in, but you can't dial out.	You have an invalid dial plan. You have not con gured an outbound proxy correctly. You have not pressed the terminating digit at the end of your phone number.	Check your dial plan con guration. Check outbound proxy con guration settings. Press # or "A" after entering your phone number.
Poor audio.	The equipment on the far end is of lesser quality. A G.723.1 codec is selected as the priority codec.	The equipment should be upgraded. Participants can try sitting closer to equipment and eliminate background noises. Move the G.711 or G.729 codecs up in the priority order.
You can dial out, but can't receive incoming calls.	SIP proxy registration is incorrect.	Check registration with SIP proxy.
Error 5 shows up on the display.	Phone-generated error.	Power cycle the phone. If error continues, then set a static IP address. If error continues, contact Technical Support.
Far end is dif cult to hear.	Local AGC is turned off. Far-end ALC is turned off.	Turn local AGC on. Turn far-end ALC on.
Far end is having dif culty hearing you.	Local ALC is turned off. Far-end AGC is turned off.	Turn local ALC on. Turn far-end AGC on.

FIGURE 4.1 MAX IP troubleshooting chart

CHAPTER 5: APPENDIX

ERROR CODES

General Errors (1 - 100)

1 - Memory allocation error
2 - Error reading Flash memory
3 - Error opening Flash memory
4 - Error writing to Flash memory
5 - Task suspended, reboot phone

Networking Errors (101 - 150)

101 - DHCP Error
102 - Device does not have an assigned IP address
103 - VLAN configuration error

Provisioning Errors (151 - 200)

151 - Phone not provisioned
152 - Activate error
153 - Invalid Ring Tone
154 - Invalid local Number
155 - Invalid Time zone
156 - Invalid Noise Match
157 - Invalid VLAN priority
158 - Error setting VLAN priority
159 - Invalid encryption key length
160 - Invalid local phone number
161 - Invalid SIP proxy Port number
162 - Invalid Conference Number
163 - Invalid Tech Support Number
164 - Invalid Speed Dial 0
165 - Invalid Speed Dial 1
166 - Invalid Speed Dial 2
167 - Invalid Speed Dial 3
168 - Invalid Speed Dial 4
169 - Invalid Speed Dial 5
170 - Invalid Speed Dial 6
171 - Invalid Speed Dial 7
172 - Invalid Speed Dial 8
173 - Invalid Speed Dial 9
174 - Invalid SIP Authorized User
175 - Invalid SIP Password
176 - Invalid SIP registration Timeout
177 - Invalid SIP transport
178 - Invalid SIP UDP port
179 - Invalid SIP TCP Port
180 - Invalid Reboot In Call
181 - Invalid Mute Ringtone
182 - Invalid DTMF payload
183 - Invalid VAD Threshold
184 - Invalid VAC Noise Order
185 - Invalid ULAW priority
186 - Invalid ALAW priority
187 - Invalid 729 priority

Provisioning Errors (151 - 200) (continued)

188 - Invalid G-723.1 5.3 kbps priority
189 - Invalid G-723.1 6.3 kbps priority
190 - Invalid SNTP address
191 - Invalid QoS procedure
192 - Invalid QoS custom value

Dial Plan Errors (201 - 250)

201 - Invalid SYSCONFIG parameter
202 - Invalid or missing string in DIGITMAP line
203 - Invalid or missing MIN_DIGITS in DIGITMAP line
204 - Invalid or missing MAX_DIGITS in DIGITMAP line
205 - Invalid or missing STRIP_FIRST_DIGITS in DIGITMAP line
206 - Error creating Address
207 - Invalid or missing DIAL_STRING in DIGITMAP line
208 - Error creating destination
209 - Error creating Hunt group
210 - Error updating dial plan database
211 - Downloading the dialplan failed
212 - Invalid dial plan line

SPECIFICATIONS

DIMENSIONS (W x D x H)
Phone section: 10.5" x 10.5" x 3"
(26.7 cm x 26.7 cm x 7.6 cm)
Base unit: 4.25" x 5.5" x 2.5"
(10.8 cm x 14 cm x 6.4 cm)

WEIGHT
Phone section: 2.7 lb (1.2 kg)
Base unit: .6 lb (0.27 kg)
Shipping: 10 lb (4.5 kg)

ENVIRONMENTAL
Operating Temperature: 0–50° C (32–122° F)
Storage temperature: 5–70° C
(41–158° F)
Operating Humidity: 15 to 80%
Storage humidity: 10 to 90%

POWER
Base unit:
Auto-adjusting power module;
100–240VAC; 50/60 Hz

NETWORK
10/100 Ethernet
RJ-45

KEYPAD
Alphanumeric standard keypad

LOUDSPEAKER
Volume: 90 dBspl A weighted @ 1 ft
Bandwidth: 200Hz - 3.3kHz

RECORD OUTPUT
Connector: 2.5 mm mono audio jack
Impedance: <1000 ohm
Bandwidth: 200Hz–3.3kHz
Dynamic Range: 60dB
THD <.01%

ECHO CANCELLATION
Tail Time: 60 mS x 3

NOISE CANCELLATION
Dynamic noise cancellation

CERTIFICATIONS
FCC Part 15 Class A
FCC Part 68
UL Certified

MODELS
MAX IP*
MAX IP Expansion Kit*

*Call your sales representative for part numbers

COMPLIANCE

FCC PART 15/ICES-003 COMPLIANCE

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC rules and Industry Canada ICES-003. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses, and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference to radio communications. Operation of this equipment in a residential area is likely to cause harmful interference, in which case the user will be required to correct the interference at his/her own expense.

Operation is subject to the following two conditions: (1) This device may not cause interference, and (2) This device must accept any interference including interference that may cause undesired operation of the device.

Changes or modifications not expressly approved by ClearOne Communications could void the user's authority to operate the equipment.

CE EUROPEAN COMPLIANCE

This equipment has been approved in accordance with Council Directive 1999/5/EC "Radio Equipment and Telecommunications Equipment"

See the enclosed Declaration of Conformity (DOC) that is enclosed with the equipment for full details. Compliance of the equipment with the Directive is attested by the application of the CE mark on the equipment.

WARRANTY

ClearOne Communications, Inc. (Manufacturer) warrants that this product is free of defects in both materials and workmanship. For warranty information and coverage, refer to the ClearOne website at www.clearone.com.

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