

ClearOne[®]

MAX[®] IP Conferencing Phone

ADMINISTRATOR'S GUIDE



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MAX IP ADMINISTRATOR'S GUIDE

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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. 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 providing complete microphone and loudspeaker coverage along with ease-of-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 provide simplicity to phone operations.

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

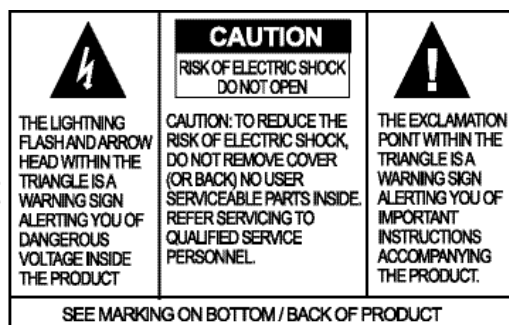
All product returns require a Return Material Authorization (RMA) number. Please contact ClearOne Technical Support before attempting to return your product. Make sure you return all the items that shipped with your product.

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.

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



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 1.1.

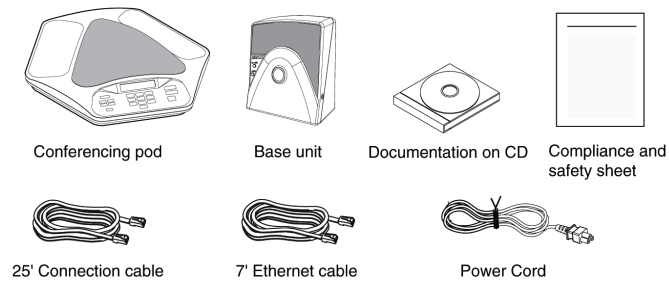


Figure 1.1 - MAX IP Parts

- » **NOTE:** ClearOne is not responsible for product damage incurred during shipping. Claims must be made 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.

REPLACEMENT PARTS & ACCESSORIES

The items shown in Figure 1.2 are included in the MAX IP Expansion Kit.

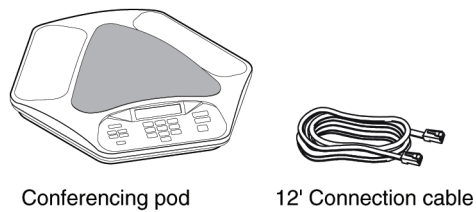


Figure 1.2 - MAX IP Expansion Kit

CHAPTER 2: GETTING STARTED

CONNECTING THE 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 (Figure 2.1).

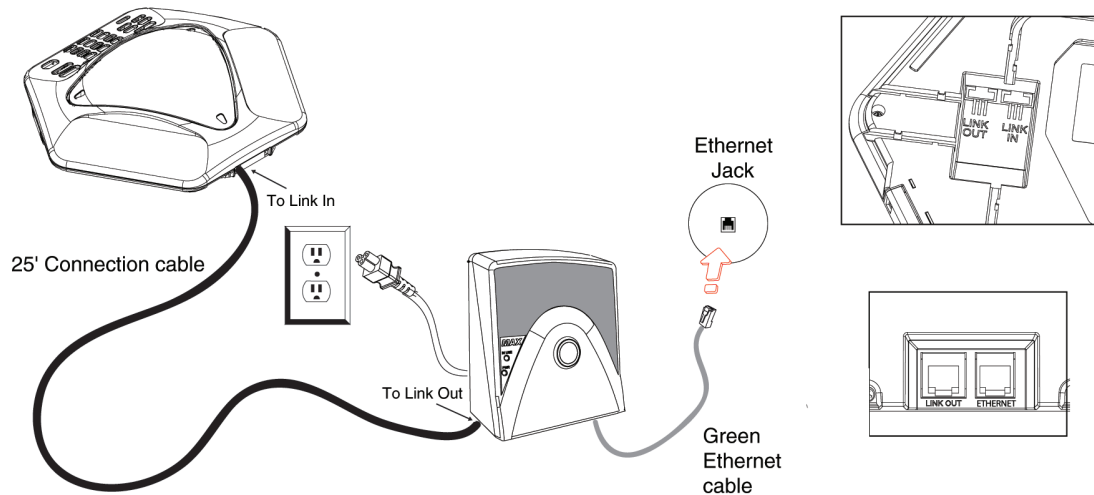


Figure 2.1 - Connecting the MAX IP

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 (Figure 2.2).

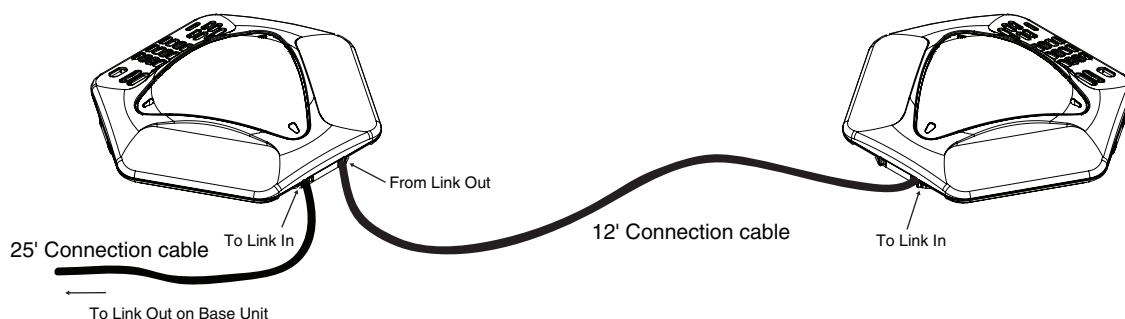


Figure 2.2 - Connecting Additional Units

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

PROVISIONING THE MAX IP PHONE

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

- The first method, called **TFTP Provisioning**, 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.
- The second method, **Web Portal Provisioning**, allows the user to configure the phone via a Web Portal associated with the phone in combination with the phone's keypad. Each phone can be fully provisioned using this method, however, each component of the phone's settings must be manually entered. Consequently, this can be a very time-consuming process depending on the number of phones to be setup.

Keypad Setup is another method for setting up a limited number of options using the phone's keypad. The MAX IP's phone extension and IP Address are a couple examples that can be setup using the phone's keypad which are usually handled via TFTP or Web Portal Provisioning.

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 a large number of phones are to be provisioned, the automatic method using the TFTP server is recommended; otherwise use the **Web Portal** method when only a few phones are to be provisioned.

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 Portal. 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 (see **Keypad Setup, page 28**).

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.

TFTP PROVISIONING

AUTOMATICALLY PROVISIONING THE MAX IP PHONE

If there are multiple units that to be provisioned with the same settings, the quickest and most seamless way of accomplishing this is through TFTP provisioning.

In fact, the 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 on 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

Each phone has its own unique MAC address. When a phone system is provisioned, and if the config file was written with the extension assigned to the respective MAC address for each phone, then nothing needs to be done. See **Keypad Setup, page 28** to manually provision a phone's extension.

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, it searches the server to locate these two files. If the phone locates a newer version of the **config.fil** file than is currently installed on the phone, it is downloaded and installed via TFTP.

The phone will then search for the **ggsip_all** file out on the server. If the phone locates it, it will compare the file date and firmware version to what is stored on the phone. If the date and/or version are more recent than what is stored on the phone, the **ggsip_all** file will be downloaded and installed on the phone. After the new firmware has loaded, the MAX IP reboots and the new firmware becomes the active firmware.

If one or both files cannot be located, it will be necessary to provision the phone via the Web Portal (See **Web Portal Provisioning, page 15**).

PHONE CONFIGURATION

The MAX IP phones are automatically configured through TFTP by searching for the phone configuration files used to provision the MAX IP phones:

1. The phone settings file contains general settings used by all MAX IP phones on a network and **MUST** be named **C1MAXIP.txt**.
2. The phone-specific file contains phone-specific settings and is required to be named "**C1MAXIP_MACAddress.txt**", where **MACAddress** is the actual hardware Ethernet MAC address found on the label on the bottom 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.

If the files are not found during the phone boot process, it will be necessary to configure each phone in the system through the Web Portal (See **Web Portal Provisioning, page 15** or **Keypad Setup, page 28**).

You can edit the files using a general text editor, such as "vi" or "Notepad". A sample **C1MAXIP.txt** file, containing configuration settings.

- » **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>

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

Configuration: General Settings

See **Web Portal Provisioning** screen on **page 15** for the login procedure and/or the **Configuration: General Settings** screen on **page 17** for configuring this setting via the Web Portal.

<**username**> - The username for logging into the Web Portal.

Default value: admin

<**password**> - The password for logging into the Web Portal.

Default value: clearone

<**helpline_num**> - The number dialed when the help line speed dial is called.

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 Portal interface. If a reboot command is issued remotely, it will not take immediate effect if this parameter is set to 0.

Default Value: 0

Allowable Values:

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

1 - Reboot phone immediately

Configuration: User Preferences

See **Configuration: User Preferences** screen on **page 18** for information on how to set these parameters through the Web Portal.

<**alc_enable**> - Automatic Level Control (ALC) Enable.

Default Value: 0

Allowable Range:

0 - Disable

1 - Enable

<**agc_enable**> - Automatic Gain Control (AGC) enable.

Default Value: 0

Allowable Range:

0 - Disable

1 - Enable

<**mute_ringtones**> - Mute Incoming Ringer for an incoming call.

Default Value: 0

Allowable Values:

0 - Disable Mute

1 - Enable Mute

» **NOTE:** If the ringer is disabled, the LED indicators on the pod(s) will still flash on an incoming call.

<**ringtone**> - The default Incoming Ringer Melody when the phone rings.

Default Value: 1

Range: 1-5

<**timezone**> - The Time Zone in which the phone resides.

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 Lumpur, 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)

<**auto-answer**> - Auto Answer Mode.

Default Value: 0 (Dynamic)

Allowable Range:

0 - Dynamic

1 - Always Auto Answer

2- Never Auto Answer

Configuration: Dial Plan

For information on configuring a dial plan, see **Configuration: Dial Plan** screen on **page 19**.

<dialplan> - The filename on the TFTP server containing the Dial Plan file.

Default Value: Undefined

Allowable Values: ASCII text filename limited by the filename length limit on the TFTP server.

Configuration: Network Settings

See **Configuration: Network Settings** details on **page 20** for information on how to set these parameters through the Web Portal.

<SNTP_server_1> - SNTP 1 server IP address.

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.

Default Value: 0.0.0.0

Allowable Values: 0.0.0.0 (Disabled) or Valid IP Address

<qos_precedence> - Quality of Service Precedence.

Values 0-7 sets a static TOS DSCP hex value according to the list below.

A value of 8 allows a valid custom hex setting to be entered.

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.

Default Value: N/A

Allowable Range: 0x00 to 0x3F

<vlan_enable> - The Virtual LAN Enable

Default Value: 0

Allowable Range:

0 - Disable

1 - 4094 Valid LAN ID

<vlan_priority> - Sets the Priority that VLAN tags outbound packets.

Default Value: 0

Allowable Range: 0-7

Configuration: SIP Parameters

See **Configurator: SIP Configuration** details on **pages 21-22** for information on how to set these parameters through the Web Portal.

<localnum> - The unique Local Number of the specific phone being configured.

Default Value: 0

Allowable Characters: 0 - 9

<localphonename> - The unique Name assigned to the specific Local Phone being configured.

Default Value: none

Allowable String Length: 49

<sip_minse_timer> - The SIP minimum session refresh interval in milliseconds.

Default Value: 1800

Allowable Range: Must be > 90 msec

<sip_proxy_enable> - Enable SIP proxy registration.

Primary Proxy: Index=0, Secondary Proxy: Index=1

Default Value: 0

Allowable Range:

0 - Disable

1 - Enable

<sip_proxy_port> - The default port with which to communicate to the SIP proxy.

Primary Proxy: Index=0, Secondary Proxy: Index=1

Default Value: 5060

Allowable Port Range: 1 - 65535

<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.

Primary Proxy: Index=0, Secondary Proxy: Index=1

Default Value: 0.0.0.0

Allowable String Length: 79

<sip_proxy_type> - The SIP Proxy device Type.

Default Value: 0

Allowable Values:

0 = Cisco Proxy

1 = Generic Proxy

<sip_session_expires_timer> - The SIP session refresh interval in milliseconds.

Default Value: 1800

Allowable Range: Must be > 90 msec

<sip_refer_timer> - The REFER message timer in milliseconds.

Default Value: 2000

Allowable Range: 0 - 4294967295 (0 = Disabled)

<sip_register_timeout> - The SIP Registration Timeout in milliseconds.

Default Value: 3600

Allowable Range: 0 - 4294967295 (0 = Disabled)

<sip_retrans_timer_T1> - The SIP Retransmission Timer T1 Timeout in milliseconds.

Default Value: 500

Allowable Range: 0 - 4294967295 (0 = Disabled)

<sip_retrans_timer_T2> - The SIP Retransmission Timer T2 Timeout in milliseconds.

Default Value: 4000

Allowable Range: 0 - 4294967295 (0 = Disabled)

<sip_retrans_timer_T4> - The SIP Retransmission Timer T4 Timeout in milliseconds.

Default Value: 5000

Allowable Range: 0 - 4294967295 (0 = Disabled)

<use_sipauth> - Use SIP authentication credentials when registering with the SIP proxy.
 Primary Proxy: Index=0, Secondary Proxy: Index=1
 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.
 Primary Proxy: Index=0, Secondary Proxy: Index=1
 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.
 Primary Proxy: Index=0, Secondary Proxy: Index=1
 Default Value: None
 Allowable String Length: 14**<outbound_sip_proxy_enable>** - Enable call routing through Outbound SIP Proxy.
 Primary Proxy: Index=0, Secondary Proxy: Index=1
 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.
 Primary Proxy: Index=0, Secondary Proxy: Index=1
 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.
 Primary Proxy: Index=0, Secondary Proxy: Index=1
 Default Value: 5060
 Allowable Port Range: 1 - 65535

<sip_transport> - The SIP Transport type.
 Primary Proxy: Index=0, Secondary Proxy: Index=1
 Default Value: 0
 Allowable Range:
 0 - UDP
 1 - TCP

<sip_udp_port> - The SIP UDP listen Port.
 Primary Proxy: Index=0, Secondary Proxy: Index=1
 Default Value: 5060
 Allowable port Range: 0 - 65535

<sip_tcp_port> - The SIP TCP listen Port.
 Primary Proxy: Index=0, Secondary Proxy: Index=1
 Default Value: 5060
 Allowable Port Range: 0 - 65535

<tls_enable> - Enable TLS.

Primary Proxy: Index=0, Secondary Proxy: Index=1

Default Value: 0

Allowable Range:

0 - Disable

1 - Enable

<tls_port> - The TLS Port assignment.

Primary Proxy: Index=0, Secondary Proxy: Index=1

Default Value: 5061

Allowable Port Range: 0 - 65535

<tls_private_key> - The TLS Private certification Key. Enter the location of the key file or use the **Browse** button and navigate to the location.

Primary Proxy: Index=0, Secondary Proxy: Index=1

Default Value: None

Allowable String Length: 49

<tls_local_cert> - The TLS Local Certification. Enter the location of the certification file or use the **Browse** button and navigate to the location.

Primary Proxy: Index=0, Secondary Proxy: Index=1

Default Value: None

Allowable String Length: 49

<tls_ca_cert> - The TLS CS Certification. Enter the location of the certification file or use the **Browse** button and navigate to the location.

Primary Proxy: Index=0, Secondary Proxy: Index=1

Default Value: None

Allowable String Length: 49

<dtmf_relay_enable> - The DTMF Relay Enable.

Default Value: 1

Allowable Range:

0 - Disable

1 - Enable In-bound DTMF Relay

<dtmf_relay_type> - The DTMF Relay Type to be enabled.

Default Value: 1

Allowable Range:

0 - Use DTMF over SIP

1 - Use DTMF over RTP

<dtmf_relay_payload> - The DTMF Relay RTP packet Payload.

Default Value: 97

Allowable Range: 96 - 127

Configuration: Audio Parameters

See **Configurator: Audio Settings** on **page 23** for details for information on how to set these parameters through the Web Portal.

<vad_enable> - Enable Voice Activity Detection.

Default Value: 1

Allowable Range:

0 - Disable

1 - Enable VAD

<**vad_noise_match**> - Defines the VAD Noise Matching algorithm.

Default Value: Level

Allowable Range:

None - Disabled

Level

g711a2

<**vad_noise_order**> - The VAD Noise Order.

Default Value: 5

Allowable Range: 0 - 10

<**g711ulaw_priority**> - The G.711 ulaw audio codec Priority.

Default Value: 255

Allowable Range: 1 (Lowest) to 255 (Highest)

<**g711Alaw_priority**> - The G.711 Alaw audio codec Priority.

Default Value: 254

Allowable Range: 1 (Lowest) to 255 (Highest) <**g729ab_priority**> - The G.729 audio codec Priority.

Default Value: 250

Allowable Range: 1 (Lowest) to 255 (Highest)

<**g7231_63_priority**> - The G.723.1 low rate audio codec Priority.

Default Value: 245

Allowable Range: 1 (Lowest) to 255 (Highest)

<**g7231_53_priority**> - The G.723.1 low rate audio codec Priority.

Default Value: 240

Allowable Range: 1 (Lowest) to 255 (Highest)

<**vad_thresh**> - The Audio Threshold level for the system.

Default Value: -20

Allowable Range: -50 (Lowest) to 255 (Highest)

<**adaptive_vad**> - Enable or Disable the Adaptive capabilities of the phone.

Default Value: 0

Allowable Range:

0 = Disable

1 = Enable

Phone Book: Add, Edit and Delete Number

See **Phone Book: Add, Edit and Delete Number** screen on **page 25** for information on how to set this parameter through the Web Portal.

<**speed_dial_0**> through <**speed_dial_9**> - Speed Dial 0 through 9.

Default Value: Undefined

Allowable Range: Valid Phone Number [0-9]

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 below and can be copied and pasted into a text file for use.

```
<C1DIALPLAN>
  <SYSCONFIG FIRST_DIGIT_WAIT="5000" INTER_DIGIT_WAIT="5000" TERMINATION_DIGIT="#"/>
  <DIGITMAP MATCH="911" MIN_DIGITS="3" MAX_DIGITS="3" STRIP_FIRST_DIGITS="0" ADD_
    PREFIX_AFTER_STRIP="" DIAL_STRING="+&@0.0.0.0"/>
  <DIGITMAP MATCH="0" MIN_DIGITS="1" MAX_DIGITS="1" STRIP_FIRST_DIGITS="0" ADD_PRE-
    FIX_AFTER_STRIP="" DIAL_STRING="+&@0.0.0.0"/>
</C1DIALPLAN>
```

- » **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 **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 URL**.

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**.

WEB PORTAL PROVISIONING

- » **NOTE:** Web Portal Provisioning requires Internet Explorer Web browser 6.0 or later with Java 1.5 or later installed.

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 (Figure 2.3).



Figure 2.3 - LCD Program Icon


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 (Figure 2.4). Enter the default username "**admin**" and the default password "**clearone**" (case insensitive), then click the **OK** button.



Figure 2.4 - MAX IP Web Portal Login Screen

The Web Portal appears, displaying the Device Information screen as described in the next section.

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 **About Device: Device Information** screen (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 **MAX1P-** followed by the last four digits of the MAC address. The MAC address is a unique address in hexadecimal format that can also be found on the label attached to the back of your MAX IP base unit.

The screenshot shows the 'Device Information' screen within the ClearOne MAX IP WebPortal. The navigation bar at the top includes 'CONFIGURATION', 'PHONE BOOK', 'VIEW LOG', 'TOOLS', 'ABOUT DEVICE' (which is highlighted), and 'REGISTRATION'. The main content area displays the following information:

System Name:	MAXIP-0CE0
Local phone number:	
Manufacturer:	ClearOne Communications
Copyright Notice:	Manufactured by ClearOne Communications (C) 2010 ClearOne Communications All rights reserved
Protocol Information:	SIP
Base Firmware Version:	4.0.2.0
Pod Firmware Version:	11-03-05
MAC Address:	00:06:24:0E:0C:E0
SIP Proxy Status:	Not configured

Figure 2.5 - Device Information Screen

Configuration: General Settings Screen

Use the Configuration: General Settings screen (Figure 2.6) to set up Security, Provisioning, the Help line number, and when a Reboot of the phone is allowed.

MAX IP WebPortal

CONFIGURATION PHONE BOOK VIEW LOG TOOLS ABOUT DEVICE REGISTRATION

General Settings

User Preferences

Dial Plan

Network Settings

SIP Parameters

Audio Parameters

Trace and Logging

General Settings

Security

User name: admin

Password: ••••••

Apply

Provisioning

☐ Use local settings

☒ Use DHCP/TFTP

☒ TFTP Address from DHCP

☐ Use TFTP Server: 10 .50 .1 .1

Apply

Phone Number

Help line number: 0

Apply

Reboot

☐ Allow reboot during a call

☒ Wait until current call ends

Apply

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 number for the help line, enter 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.

Click the **Apply** button to store any changes made.

Configuration: User Preferences Screen

Use the **Configuration: User Preferences** (Figure 2.7) screen to enable/disable Automatic Level Control (ALC) and Automatic Gain Control (AGC), to Mute Incoming Ringer, to set Incoming Ringer Melody and to set the Auto Answer feature.

ClearOne. CONFIGURATION PHONEBOOK VIEWLOG TOOLS ABOUT DEVICE REGISTRATION

MAX IP WebPortal

User Preferences

☐ Enable automatic level control

☐ Enable automatic gain control

☐ Mute incoming ringer

Incoming ringer melody: Melody 1

Auto Answer: Dynamic

Figure 2.7 - Configuration: User Preferences Screen

- **Enable Automatic Level Control:** Click this checkbox to ON to enable ALC or uncheck it to disable ALC. ALC 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 checkbox to ON to enable AGC or uncheck it to disable AGC. AGC adjusts (softer and louder) input audio to a consistent level.
- **Mute Incoming Ringer:** Click this checkbox to ON 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.
- **Auto Answer:** Click the drop-down box to select the desired auto answer mode. Three options are available:
 1. **Dynamic** (default) allows the phone to auto answer through SIP signalling
 2. **Always** forces the phone to always answer an incoming call
 3. **Never** disables the auto answer feature

Click the **Apply** button to store any changes made.

Configuration: Dial Plan Screen

Use the **Configuration: Dial Plan** (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 load it 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.

Manual Configuration		
<input checked="" type="checkbox"/> Rule 1	Starts with 911	Auto dials after 3 digits
<input checked="" type="checkbox"/> Rule 2	Starts with 0	Auto dials after 1 digits
<input checked="" type="checkbox"/> Rule 3	Starts with +8	Auto dials after 11 digits
<input type="checkbox"/> Rule 4	Starts with	Auto dials after digits
<input type="checkbox"/> Rule 5	Starts with	Auto dials after digits

Timeout 5 sec (3sec - 30sec)

Figure 2.8 - Configuration: Dial Plan Screen

Load From File Configuration

If the phone is to be part of a group of phones with similar settings, the Load From File method is recommended (see **Dial Plan Configuration File** on page 14 for more information on the dial plan configuration file) to ensure that each phone in the group uses the same dial plan settings. (Although there are no naming rules for the dial plan configuration file, when stored on the MAX IP phone it is always renamed "**C1MAX1AVDIAL.txt**".)

The dial plan file can be located in different directories in order to provide for various settings for each phone, such as individual dial plans for domestic calls versus international calls.

- To load a Dial Plan manually, select the **Load From File** checkbox and enter the path and filename of the file to load. All fields in the Manual Configuration section become inactive.
- Alternatively, select the **Browse** button and navigate to the file using the displayed Open file window.
- When done, click on the **Upload** button to store the Dial Plan file onto the unit.

Manual Configuration

- To manually setup a Dial Plan, select the **Manual Configuration** checkbox. All fields in the Load From File section become inactive. The MAX IP unit is pre-configured prior to shipping with the above settings for Manual Configuration.
- Starting with **Rule 1**, select its checkbox and type in the appropriate information for the rule to follow. Only numeric characters or the "*" character can be entered from this screen. Rule 1 takes precedence over all subsequently assigned rules. **Rule 2** takes precedence over any assigned rules below it, etc.
- Repeat for any of the remaining rules in the list.
- Set the **Timeout** value for the time the phone will take to hang-up if no valid number is entered.
- Select the **Apply** button to store any changes made.
- Select the **View** button to view the current dial plan file (Figure 2.9) associated with your MAX IP phone.

```
<C1DIALPLAN>
<SYS_CONFIG FIRST_DIGIT_WAIT="5000" INTER_DIGIT_WAIT="5000" TERMINATION_DIGIT="#">
<DIGITMAP MATCH="911" MIN_DIGITS="3" MAX_DIGITS="3" STRIP_FIRST_DIGITS="0" ADD_PREFIX_AFTER_STRIP="" DIAL_STRING="+&@0.0.0.0">
<DIGITMAP MATCH="0" MIN_DIGITS="1" MAX_DIGITS="1" STRIP_FIRST_DIGITS="0" ADD_PREFIX_AFTER_STRIP="" DIAL_STRING="+&@0.0.0.0">
</C1DIALPLAN>
```

Figure 2.9 - Example Dial Plan File

Configuration: Network Settings Screen

Use the **Configuration: Network Settings** screen (Figure 2.10) to set up your MAX IP phone on the network. 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.

MAX IP WebPortal

Network Settings

Hostname: MAXIP-0CE0

☒ Enable DHCP

Domain Name: test-network.test.local

Static IP Address: 10.50.1.110

Subnet Mask: 255.0.0.0

Default Gateway: 10.50.1.1

Primary DNS IP Address: 10.50.1.1

Secondary DNS IP Address: 208.67.222.222

SNTP Server 1 IP Address: 132.163.4.101

SNTP Server 2 IP Address: 132.163.4.103

RTP/RTCP QoS Setting

Description/Precedence: Express Forwarding/5

Custom/Current DSCP Value: 0x28 (0x0 - 0x3F)

☐ Enable VLAN

VLAN Priority: 0

VLAN ID: 0 (1 - 4094)

Apply

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 to enter the appropriate values in above-mentioned fields 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 8** for more information on the time zone setting).
- **RTP/RTCP 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.
- **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 30**).

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

Configuration: SIP Configuration Screen

Use the **Configuration: SIP Configuration** screen (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 Protocol (RTP)** for actual transport.

ClearOne. CONFIGURATION PHONE BOOK VIEW LOG TOOLS ABOUT DEVICE REGISTRATION

MAX IP WebPortal

SIP Configuration

Phone number:

Phone name:

Registration timeout:

SIP Proxy Type

☐ Cisco Proxy

☒ Generic Proxy

☐ Select Primary Proxy

☐ Enable SIP Proxy registration

Proxy server IP address/FQDN:

Proxy port:

☒ SIP Credentials

Authorization user:

Authorization password:

☒ Enable Outbound proxy

Outbound proxy server IP address/URL:

Outbound proxy port:

SIP Transport:

☒ UDP Listen port:

☐ TCP Listen port:

☐ Enable TLS

TLS port:

Private Key:

Local Cert:

CA Cert:

☐ Select Secondary Proxy

☐ Enable SIP Proxy registration

Proxy server IP address/FQDN:

Proxy port:

☐ SIP Credentials

Authorization user:

Authorization password:

☐ Enable Outbound proxy

Outbound proxy server IP address/URL:

Outbound proxy port:

SIP Transport:

☒ UDP Listen port:

☐ TCP Listen port:

☐ Enable TLS

TLS port:

Private Key:

Local Cert:

CA Cert:

DTMF Settings

☒ Enable DTMF Relay

☒ Use DTMF over SIP

☐ Use DTMF over RTP

Payload: (96 - 127)

Figure 2.11 - Configuration: SIP Parameters Screen

- **Phone number:** Enter the phone number associated with the MAX IP unit.
- **Phone name:** Enter the name of the MAX IP unit - usually associated with the room or office it's setup in.
- **Registration timeout:** Enter the value (in seconds) your phone is to refresh its registration to the SIP proxy server. The default value is 3600.
- **SIP Proxy Type:** Select the appropriate radio button to define the specific proxy type for the system. If unsure, select the Generic Proxy option.
- **Select Primary/Secondary Proxy:** If a primary and/or secondary proxy is defined, select the checkbox next to the **Select Primary Proxy** and/or **Select Secondary Proxy** option as needed.
- **Enable SIP Proxy registration:** Click this checkbox 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.
- **SIP Credentials:** Click this checkbox to enable SIP Credentials or uncheck it to disable SIP Credentials. (SIP Credentials are required if the proxy requires it.) The SIP Credentials verifies the username and password as entered in the **Authorization user** and **Authorization password** fields. These fields are active only when the **SIP Credentials** checkbox is enabled and can thus be modified. When the **SIP Credentials** checkbox is disabled these fields are inactive.
- **Enable Outbound Proxy:** Click this checkbox 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 re-transmit erroneous or dropped packets. The default port is 5060.
- **Enable TLS:** In order to utilize **Transport Layer Security (TLS)**, the **Private Key**, **Local Certificate** and **Certificate Authority (CA)** certificates need to be created and available on the server. Check with the local IT Administrator to verify prior to enabling TLS. Click on the **Enable TLS** checkbox to access the TLS settings fields. The TLS port is set by default. Click on the respective **Browse** buttons and navigate to the appropriate location(s) on the server to locate the three certificate files. When located, click on the **Upload** button to enable TLS.
- **DTMF Settings:** Click the **Enable DTMF Relay** checkbox to enable the **DTMF (Dual Tone Multi-Frequency)** relay or uncheck the box to disable it. The 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 DTMF Relay** is disabled, this field is inactive.

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

Configuration: Audio Settings Screen

Use the **Configuration: Audio Settings** screen (Figure 2.12) to configure the **Voice Activity Detection (VAD)** settings and to prioritize your preferred audio codecs.

Figure 2.12 - Configuration: Audio Settings Screen

- **Enable VAD:** Click the check box to enable VAD 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, silence (SID) packets will be sent when silence is detected according to the description in the G.723.1 standard.
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 preferred type of VAD noise matching needed. 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 inadvertently disconnect the phone during periods when conversation is non-existent or the unit does not immediately pick it up. Enter the value for the preferred VAD Noise LP order. 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 preferred sequence, from most to least preferred. Select the codec to move then click on the **Up** or **Down** buttons to move it up/down the list. Repeat this for each codec to be re-ordered.

- **SRTP Settings:** Click the check box to enable the various SRTP setting options.
 - » **NOTE:** The following settings must match the SIP server requirements. To ensure the settings are correct, contact the local IT Administrator to coordinate these settings prior to implementing them.
- **Cipher Mode:** Click on the drop-down window to select the appropriate Cipher Mode setting.
- **MAC Algorithm:** Click on the drop-down window to select the appropriate MAC Algorithm setting.
- **KDR:** Click on the drop-down window to select the appropriate **Key Derivation Rate (KDR)** setting.
- **SRTCP:** Click on the drop-down window to select the appropriate **Secure Real-Time Control Protocol (SRTCP)** setting.

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 (Figure 2.13) to control logging for general audio processing and SIP subsystems, as well as system logs and trace flags.

ClearOne. CONFIGURATION PHONE BOOK VIEW LOG TOOLS ABOUT DEVICE REGISTRATION

MAX IP WebPortal

General Settings
User Preferences
Dial Plan
Network Settings
SIP Parameters
Audio Parameters
Trace and Logging

Trace/Logging Settings

☐ Enable system log

Active trace flags: Apply

SIP Logging

☐ SIP Message logging

General Logging

ROOT	Spy Trace Off	IDLE	Spy Trace Off
NWIF	Spy Trace Off	DLMM	Spy Trace Off
ISU	Spy Trace Off	ISUCPDET	Spy Trace Off
AUDIO	Spy Trace Off	AUAPP	Spy Trace Off
CCU	Spy Trace Off	VIM	Spy Trace Off
SECMAN	Spy Trace Off	RTCP	Spy Trace Off
TUIU	Spy Trace Off	ATPM	Spy Trace Off
DEX	Spy Trace Off	DIM	Spy Trace Off
DIM_DNLD	Spy Trace Off	DSPA	Spy Trace Off
CMM	Spy Trace Off	CVBDSM	Spy Trace Off
DSPMAN	Spy Trace Off	RTPMAN	Spy Trace Off
RVMU	Spy Trace Off	RVCM	Spy Trace Off
MSUIM	Spy Trace Off	NMM	Spy Trace Off
UIUHW	Spy Trace Off	RVCCEXT	Spy Trace Off

Turn Off Logs Apply

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, **page 26**.
- **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 Message Logging:** Allows capture of all inbound and outbound SIP messages for signalling diagnostics.
- **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 (Figure 2.14) to add, modify, and delete numbers from your phonebook. Only numeric characters may be entered into the **New number** field.

ClearOne

CONFIGURATION PHONE BOOK VIEW LOG TOOLS ABOUT DEVICE REGISTRATION

MAX IP WebPortal

Add, Edit and Delete Number

Phone List:

- 1-3577
- 2-
- 3-
- 4-3456789
- 5-
- 6-
- 7-
- 8-
- 9-
- 0-

Clear All

New number:

3456789

Entry: 4

<< Add/Update

Remove >>

Figure 2.14 - Phonebook: Add, Edit and Delete Number Screen

- **Phone List:** This is a list of numbers that have been stored in the **Phone List** - a total of ten numbers can be stored in the Phone list at a time.
 - To view a number, select the specific assignment number from the **Phone List** and the phone number stored at that location will be displayed in the **New number** field to the right for editing if necessary.
 - The number can be reassigned by selecting the it from the **Phone List**. After it is displayed in the **New number** field, select a new assignment number from the **Entry** drop-down list. Click on the **Add/Update** button to commit it to the new assignment number and the **Phone List** will update accordingly.
 - To clear a specific number from the **Phone List**, select the assignment number from the **Phone List**. After it's displayed in the **New number** field, click on the **Remove** button. A confirmation message will be displayed. Click on the **Ok** button to remove it from the list or the **Cancel** button to return to the Phone Book screen.
 - To clear all numbers from the list, click the **Clear All** button. A confirmation message will be displayed. Click on the **Ok** or **Cancel** button to perform the required action.
- **New number:** A new number is easily assigned to the **Phone List**.
 - Enter the telephone number in the **New number** field, including the area code to be added to the **Phone List**.
 - » **NOTE:** To insert a pause for an outside dial tone, enter a "P" prior to where the pause is to occur. Each pause code entered is approximately two (2) seconds in duration. Multiple pauses can be set by entering a "P" for every two second pause required. This is a Web Portal feature and cannot be set via the phone's keypad.
 - Click the **Entry** drop-down box and select the number you want the telephone number assigned to in the **Phone List**.
 - 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 (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.

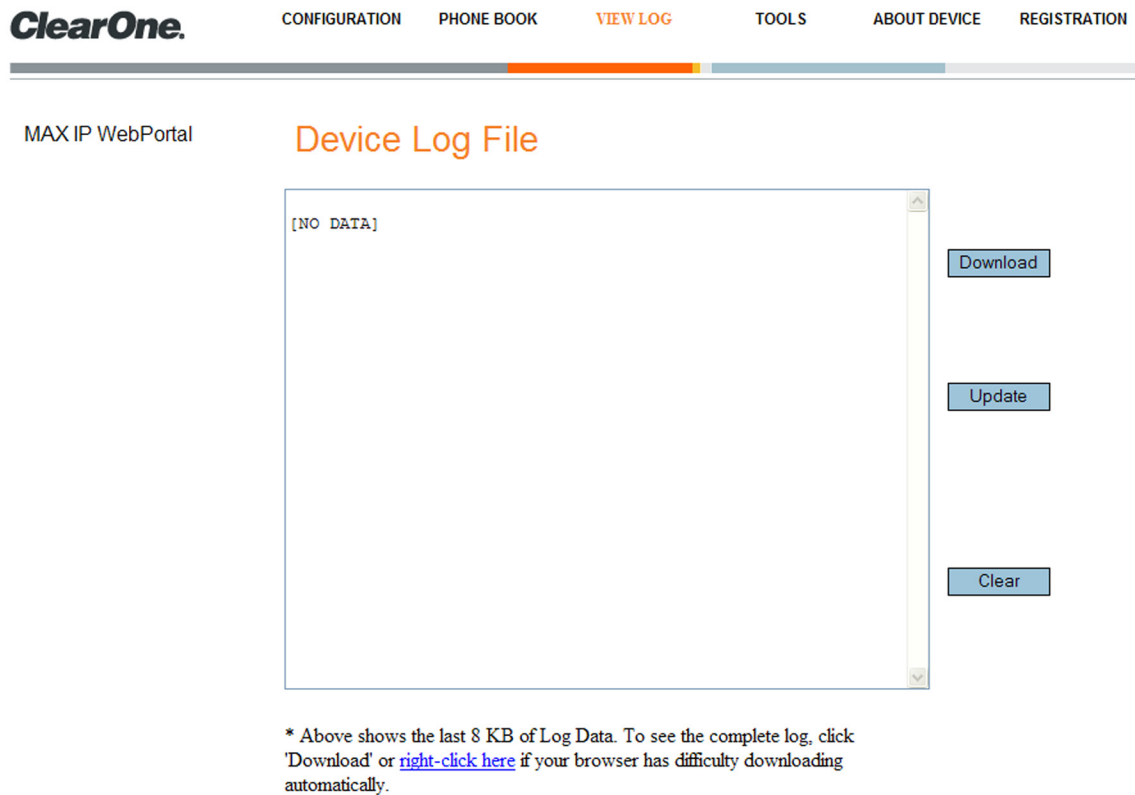


Figure 2.15 - View Log: Device Log File Screen

Tools: Diagnostics - VoIP Statistics Screen

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

ClearOne. CONFIGURATION PHONE BOOK VIEW LOG **TOOLS** ABOUT DEVICE REGISTRATION

MAX IP WebPortal

Diagnostics - VoIP Statistics

The statistics are available when the phone is in a call (☐)

Packets received:

Packets Lost:

Percent packet loss:

Firmware

File location:

Operations

Figure 2.16 - Tools: Diagnostics VoIP Statistics 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 Portal 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.

KEYPAD SETUP

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 (Figure 2.17).

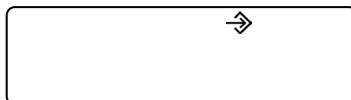



Figure 2.17 - MAX IP Program Icon

» **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 Portal interface. If you cannot access the Web Portal 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 will reboot.

OPTION FOR MANUALLY ASSIGNING AN EXTENSION

If the phone is auto-provisioned via TFTP and if the extensions were not assigned in the configuration file, then an error code will display on the phone's screen.

To configure a phone's extension using the phone's keypad:

1. Press the **REDIAL/PROG** key until the I/O icon is displayed on the phone's screen.
2. Press the **7** key on the phone's keypad.
3. Press the **REDIAL/PROG** key again.
4. Enter the new extension number.
5. Press the **REDIAL/PROG** key again to store the new number and return to the main phone display.
6. Press the **CLEAR** key to finish.

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.

Change Dynamic Host Configuration Protocol (DHCP)

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

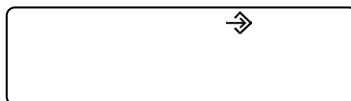



Figure 3.1 - MAX IP Program Icon

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 will play.
6. Press **CLEAR**  to exit programming. The phone will reboot.

Change Host IP Address

1. Press and hold the **REDIAL/PROG** key until the Program icon appears on the LCD screen (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.
7. Press the **CLEAR** button to exit programming. The phone will reboot.

Change Subnet Mask

1. Press and hold the **REDIAL/PROG** key until the Program icon appears on the LCD screen (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.
7. Press **CLEAR** to exit programming. The phone will reboot.

Program Default Gateway IP Address

1. Press and hold **REDIAL/PROG** until the Program icon appears on the LCD screen (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.
7. Press **CLEAR** to exit programming. The phone will reboot.

Change Ringer Melody

1. Press and hold the **REDIAL/PROG** key until the Program icon appears on the LCD screen (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.
6. Press **CLEAR** to exit programming.

Enable/Disable VLAN

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

Program the Extension Number

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

Programming AGC and ALC Settings

1. Press and hold the **REDIAL/PROG** key until the Program icon appears on the LCD screen (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 Program Icon

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

Restore Factory Defaults

1. Press and hold the **REDIAL/PROG** key until the Program icon appears on the LCD screen (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.
4. Press **CLEAR** to exit programming. The phone will reboot.

Rebooting the MAX IP Phone

Rebooting the MAX IP phone is sometimes useful when making changes to the SIP settings or troubleshooting. Most often, however, the phone will automatically reboot after making and saving phone setting changes via the Web Portal interface.

1. Press and hold the **REDIAL/PROG** key until the Program icon appears on the LCD screen (Figure 3.1).
2. Press the **0** key - "rEboot" will appear on the display.
3. Press **REDIAL/PROG** to begin the reboot.

CHAPTER 4: MAINTENANCE

CARING FOR THE 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 configured an outbound proxy correctly. You have not pressed the terminating digit at the end of your phone number.	Check your dial plan configuration. Check outbound proxy configuration 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 difficult 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 difficulty 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

ERROR CODES

General Errors (1 - 100)		Networking Errors (101 - 150)	
1	Memory allocation error	101	DHCP Error
2	Error reading Flash memory	102	Device does not have an assigned IP address
3	Error opening Flash memory	103	VLAN configuration error
4	Error writing to Flash memory		
5	Task suspended, reboot phone	Provisioning Errors (151 - 200)	
6	tExcTask task suspended	151	Phone not provisioned
7	tLogTask task suspended	152	Activate error
8	tNetTask task suspended	153	Invalid Ring Tone
9	tDhpcStateTask task suspended	154	Invalid local Number
10	tDhpcReadTask task suspended	155	Invalid Time zone
11	tShell task suspended	156	Invalid Noise Match
12	dhcpcEvent task suspended	157	Invalid VLAN priority
13	tMVX_XSPY task suspended	158	Error setting VLAN priority
14	tMVX_IDLE task suspended	158	Invalid encryption key length
15	tMVX_MEAS task suspended	160	Invalid local phone number
16	tMVX_ROOT task suspended	161	Invalid SIP proxy Port number
17	tMVX_DLMM task suspended	162	Invalid Conference Number
18	tMVX_ISU task suspended	163	Invalid Tech Support Number
19	tMVX_AU task suspended	164	Invalid Speed Dial 0
20	tMVX_AUAP task suspended	165	Invalid Speed Dial 1
21	tMVX_CCU task suspended	166	Invalid Speed Dial 2
22	tMVX_RTCP task suspended	167	Invalid Speed Dial 3
23	tMVX_TUIU task suspended	168	Invalid Speed Dial 4
24	tMVX_ATPM task suspended	169	Invalid Speed Dial 5
25	tMVX_DEX task suspended	170	Invalid Speed Dial 6
26	tMVX_CMM task suspended	171	Invalid Speed Dial 7
27	tMVX_RVMU task suspended	172	Invalid Speed Dial 8
28	tMVX_RVCM task suspended	173	Invalid Speed Dial 9
29	TaskChecker task suspended	174	Invalid SIP Authorized User
30	Timer Mgr task suspended	175	Invalid SIP Password
31	VTDatagram task suspended	176	Invalid SIP registration Timeout
32	DynamicServer task suspended	177	Invalid SIP transport
33	VTSocketServer task suspended	178	Invalid SIP UDP port
34	RingTest task suspended	179	Invalid SIP TCP Port
35	MaxDialing task suspended	180	Invalid Reboot In Call
36	RingSM task suspended	181	Invalid Mute Ringtone
37	RV IPPhone SIP task suspended	182	Invalid DTMF payload
38	PodSerial task suspended	183	Invalid VAD Threshold
39	WatchDog task suspended	184	Invalid VAC Noise Order
40	tMVX_STAT task suspended	185	Invalid ULAW priority
41	tMVX_NMM task suspended	186	Invalid ALAW priority
42	VTSocket task suspended	187	Invalid 729 priority
43	tNVIF1 task suspended	188	Invalid G-723.1 5.3 kbps priority
44	tNVIF2 task suspended	189	Invalid G-723.1 6.3 kbps priority
45	(NULL) - unnamed task suspended	190	Invalid SNTP address
46	tNWIF1 task suspended	191	Invalid QoS procedure
47	tNWIF2 task suspended	192	Invalid QoS custom value
48	tTftpTask task suspended		
49	ConfTimer task suspended		
75	A task failed to start		

ERROR CODES continued...

Dial Plan Errors (201 - 300)			
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 Dial Plan failed		
212	Invalid Dial Plan Line		
300	SIP registration failure		

CHAPTER 5: APPENDIX

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:
0.6 lb (0.27 kg)

Shipping:
10 lb (4.5 kg)

Environmental

Operating Temperature:
0–48° C (32–118° 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 dB SPL A-Weighted @ 1 meter

Bandwidth:
200Hz - 3.3kHz

Record Output:

Connector:
2.5 mm Mono Audio Jack

Impedance:
< 1kΩ

Bandwidth:
200Hz – 3.3kHz

Dynamic Range:
60dB THD < 0.01%

Echo Cancellation:

Tail Time: 60 ms x 3

Noise Cancellation:

Dynamic Noise Cancellation

Certifications:

FCC Part 15 Class A
FCC Part 68
UL Certified

Model

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.



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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