ESSENTIAL TRADING SYSTEMS CORP.

sales@essentialtel.com

CS-74

Applicable for:
CS-71, CS-71TR, CS-74FM, CS-74FHD/W, CS-701

Administration & Setup Guide

CS-74

CS-74-F

188 Norwich Ave, Colchester, CT 06415

(860) 295-8100

www.essentialtradingsystems.com

THE POSSIBILITIES ARE ENDLESS

Table of Contents

1.0 Product Overview 5
1.1 Product Description 5
1.2 Product features..... 5
2.0 Administration..... 6
2.1 Login 6
2.2 System Settings 7
2.4 System Information Page 9
2.5 Networking Page..... 10
2.6 SIP Configuration Page 11
2.7 DTMF Steering Page 16
2.8 LMR / EIA Tone Settings 17
2.9 Audio Settings Page 18
2.10 Two Tone Paging..... 21
2.11 Management Page..... 22
2.12 Update Firmware 23
2.13 Diagnostics Page..... 24
3.0 Frequently Asked Questions 25
4.0 Appendix..... 26
4.1 Specifications..... 26
4.2 Syslog Messages..... 27
5.0 CS-74 Limited Warranty 29

Revised – September 3rd, 2024, for firmware version 3.1.7, release date – 9/3/2024
AG

Release notes:

09/3/2024	Ver 3.1.7	Tones introduced
4/10/2024	Ver 3.1.0	Fixed and added the following: - Fixed minor SNMP vulnerability - Added Prevent channel mute - Changed the way selected channel works in handset selection mode (CS-74-FCH)
5/23/2023	Ver 3.0.23	GPIO Control (BCD Channel Steering) introduced.
3/8/2023	Ver 3.0.11	<u>Updated OS</u> – This firmware cannot be applied in a form of file upload. Prevent channel mute and EIA hold tone filter introduced.
3/15/2022	Ver 2.1.50	DTMF Steering changed to per channel setting. Minor bug fixes.
4/1/2021	Ver 2.1.34	Adds EIA Tone feature
1/1/2021	Ver 2.1.30	Fixes issue concerning device Tx level when using Secure SIP. Also of note: VAD should not be used when "Secure SIP" is enabled. Fixes issue with log file download.
12/11/2020	Ver 2.1.28	Fixes issue with slow volume adjust (requires Atmel IC firmware upgrade to be effective), Fixes audio bleed through issue, fixes log file download bug, fixes SIP lockup issue associated with invalid/unavailable server, fixes SIP RTP port leak. Added TLS/SRTP feature as "Secure Call" option.
5/9/2020	Ver 2.1.21	Fixes bug related to Privacy mode while operating in Unicast
4/16/2020	Ver 2.1.20	Fixes bug related with randomly setting channels to "off" condition. Would cause device to lock up or present constant "app reloading" condition.
3/16/20	Ver 2.1.17	Introduce new features: DTMF Steering, check boxes for RTP & RTCP keep alives. Fixed bugs: Audio bleed through.
7/19/19	Ver 2.1.0	Introduce new features: CB Mic operation (CS-74), Unicast mode (CS-64), Wall mode, Audible IP address announce & Packet Capture. Implemented security patches & add support for Speex (CS-74). Fixed bugs: "NoPBX" server IP field blank, footswitch & handset selection HF, Tick/audio gap on PTT & listen only operation. Changed colors and Logo on web pages.
9/21/18	Ver 2.0.31	Add "Mic Boost" feature. Bug fixes – SIP disconnect protocol issue when save & reload executed, SIP registration & autoconnect issue after save & reload, SIP Privacy PTT issue – constant Tx with no PTT active.
7/23/18	Ver 2.0.23	Bug Fixes - issue concerning static to dhcp reboot freeze up, issue saving Log File priority changes.
5/7/18	Ver 2.0.20	Bug fixes – Device ID reporting (SIP), Constant RTP output during active call (VAD), RTCP timeout.
4/20/18	Ver 2.0.15	Implemented new "simulcast" feature. Improvements to handsfree audio, normalized levels between hands free, gooseneck & headset operations. Bug fixes – Firmware upgrade, SIP messaging device ID.
3/9/18	Ver 2.0.0	New release for new & improved CS-74/64. This version is not backward compatible. **Applies only to SN-2180xxxx and above. Major hardware changes. New features added: Privacy, full duplex Hands Free.**

12/15/17	Ver 1.1.9	Added: SIP Registration Fail timer field, PTT DTMF * # feature. Fixed Auto Answer check box bug on CS-74
9/10/17	Ver 1.1.8	Added: enable/disable ring tones on CS-74. Made improvements to No PBX mode, added new factory reset function (press all 4 buttons)
5/1/17	Ver 1.1.7	Added features: Call w/o PBX, HTTPS, QoS(DSCP). Fixed bugs with: single channel reload, device type switching. Implemented several changes to web page layout and descriptions.
10/3/16	Ver 1.1.6.1	Added feature: SIP VAD.
8/30/16	Ver 1.1.6	Added features: PTT Mutes Speaker, RTCP. Fixed bugs for : syslog, 20ms packet size, SIP 480 responses, NTP. Minor webpage verbiage changes
6/15/15	Ver 1.1.4	Added features: call without registration, RTP keep alive, single channel reload, SNMP, device type switching. Improved call status reporting.
12/15/15	Ver 1.1.3g	Fix files size limits in OS kernel. Must be loaded on any device with pre-1.1.4 firmware to allow further firmware upgrades.
12/8/15	Ver 1.1.3.2	Fix device feeze up after unknown period of time.
3/18/15	Ver 1.1.3	Improvements to code structure, add microphone limiter circuitry.
11/24/14	Ver 1.1.2	Patch “shellshock bash” vulnerability, improvements to audio quality, improvements to volume adjust controls, add Listen Only feature to channels, remove half duplex mode, fix various other bugs.
9/22/14	Ver 1.1.1	Fixed numerous bugs discovered in 1.1.0
6/11/14	Ver 1.1.0	First official release.

REMAINDER OF THIS PAGE LEFT BLANK INTENTIONALLY

1.0 Product Overview

1.1 Product Description

The CS-74 is a four line/channel SIP endpoint device capable of connecting 4 simultaneous SIP calls.

1.2 Product features

- **Display** – Provides information such as device name, user name and line/channel assignments as defined by administrator and other device status indicators.
- **Volume Control** – Each channel has individual volume control so users can set levels to their liking.
- **Loud Speaker** – Each CS-74 is equipped with a built in loud speaker. Administrators have access to master volume control via web page configuration. Audio for all lines is mixed.
- **LED's** – LED's on the front panel provide status indication for device condition such as: line registered, ring, call connected, and audio activity
- **PTT Buttons** – Buttons on front panel serve not only as PTT (Press to Talk) on the respective line but in certain modes, initiate/drop a call, and activate/deactivate Hands Free operation.
- **F1/F2 Button** – These optional buttons have been added to provide expanded capability of the device. Currently used to enable privacy & simulcast features, with more functions to follow in near future.
- **Handset/Headset** – The CS-74 can be equipped with the option to connect either a mono headset or PTT handset. An RJ-25 (6-wire) jack located on the back of the device can be universally used for either PTT handset or Mono headset with appropriate adapter cable. See Specifications in this manual for pinout.
- **Footswitch** – The CS-74 can be equipped with an optional foot switch for PTT activation. Footswitch will connect to the handset port on back of unit. Pins 1 & 6 are used for contact closure input to activate PTT.
- **Gooseneck Mic** – The CS-74 is typically equipped with a close talking gooseneck microphone. For listen only models this option is removed.
- **Built In Mics** – The CS-74-F has been equipped with microphones built in to the enclosure specifically for use with the Hands Free feature. *Available in version with "F" keys.*

2.0 Administration

2.1 Login

The CS-74 is configured via browser interface. CS-74's ship, default, set to DHCP. Upon connection to appropriate network the device will automatically acquire an IP address. This address will be indicated on the device's display during boot up.

ETC recommends using Chrome or Firefox to ensure best browsing experience. IE9 and above can be used as well.

Once the IP address has been determined, open a browser from a PC that is networked with this device. Type the IP address into the URL bar of the browser and press enter. The CS-74 Login screen is shown in Figure 1.

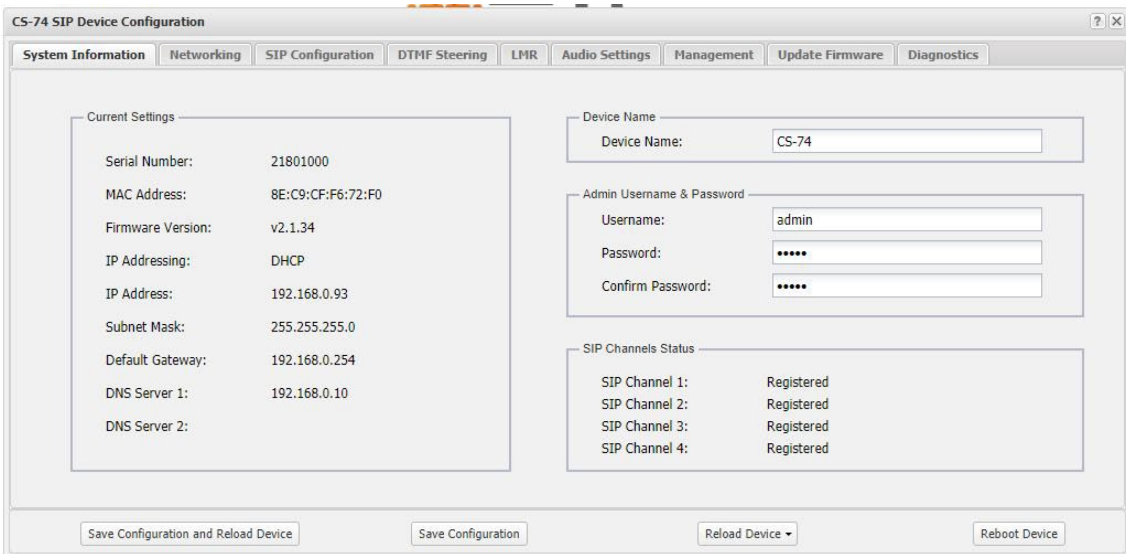
Default Username is: **admin**, default password is: **admin**. Upon logging in, administrative login credentials can be changed to ensure security of system configuration.

Figure 1

2.2 System Settings

After logging in, you are brought to the System Settings page. Here, access to all administrative functions of the CS-74 are presented as tabs across the top of the window. See Figure 2.

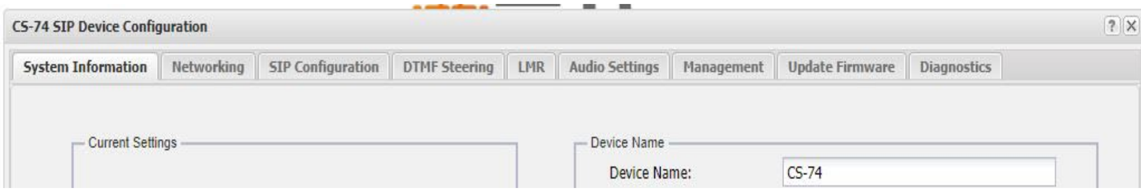
Figure 2



2.3 Menu Options

The menu selections are displayed as tabs across the top of the web page. Each section will be explained in more detail later in the guide. Figure 3 shows the options available.

Figure 3

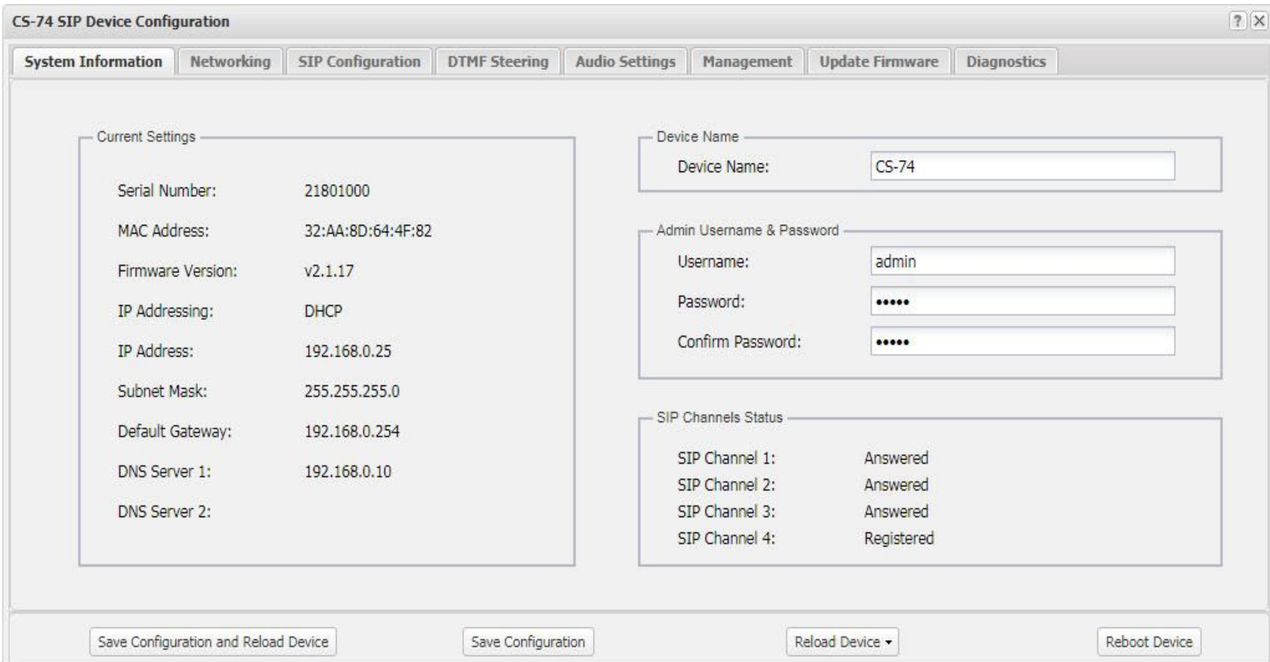


- **System Information** – This page displays device information and administrator definable fields for device name and login credentials.
- **Networking** – This page allows the administrator to configure IP settings for the device and select static or DHCP
- **SIP Configuration** – This page allows the administrator to configure SIP related details for each of the lines/channels.
- **DTMF Steering** – This page allows the administrator to multiple names & DTMF codes on a per channel basis for steering radio systems to use different frequencies/channels for communication between the CS-74 and mobile radios.
- **LMR** – This page allows the administrator to enable and set EIA Tones for each channel
- **Audio Settings** – This page allows the administrator to set master volume & mic gain as well as display brightness/contrast. Other feature settings are accessible from this page.
- **Management** – This page provides the administrator with access to settings for syslog reporting, device type selection & browsing options. New management related features will be added to this page as they are developed.
- **Update Firmware** – This page allows the administrator to upgrade device firmware.
- **Diagnostics** – This page allows the administrator access to diagnostic tools such as activity log & config file download.

2.4 System Information Page

The System Information page displays pertinent information about the device such as IP address, serial number, firmware version etc. Additionally, there are fields the administrator may use to identify the specific device, change the device login credentials for security purposes and view call status. See Figure 4 below.

Figure 4



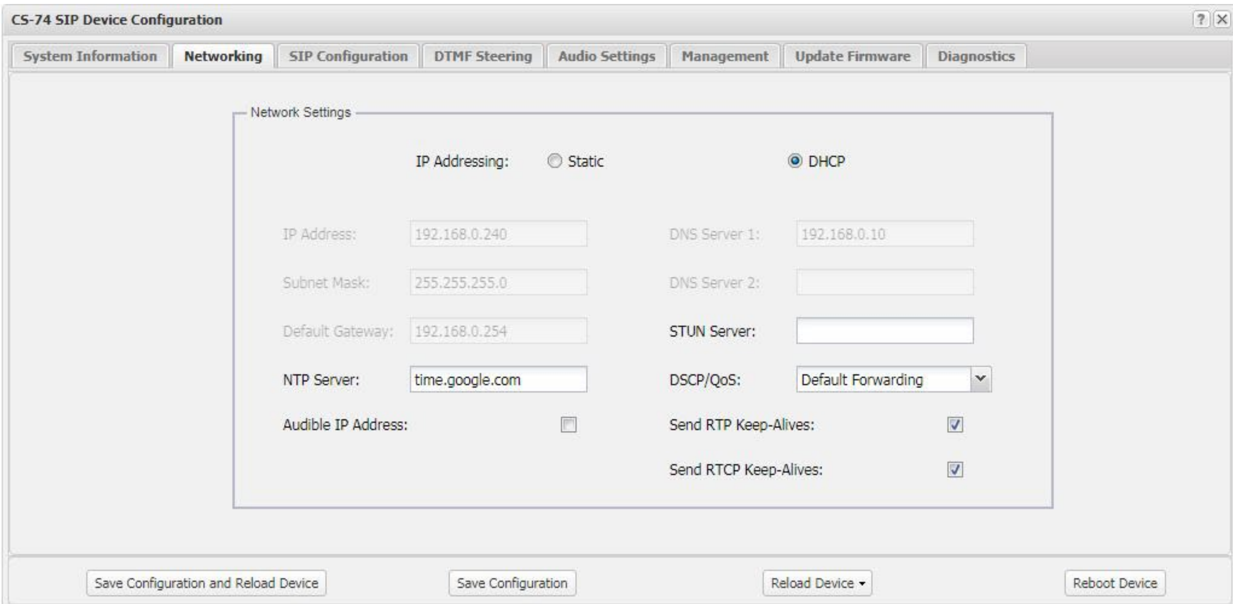
- **Device Name** – Enter any alpha-numeric sequence to uniquely identify the device. Note: information entered here will also indicate on the device display.
- **Username** – Enter new login username. Default is: admin
- **Password** – Enter new login password. Default is: admin
- **Confirm Password** – Renter new password to confirm.
- **SIP Channels Status** – Active status of each channel is displayed

Upon making changes you must click **Save Configuration** if making additional changes on other pages or click **Save Configuration & Reload** to save & activate the changes. These buttons are located at the bottom of the page.

2.5 Networking Page

The Network Settings page allows the administrator to configure the device with a static IP address or configure using DHCP. Device is default DHCP and IP address will be indicated on the display during boot up. See Figure 5 below.

Figure 5



- **IP Address** - Enter static IP address for the gateway. In DHCP mode, device will default to 192.168.0.240 in the event it cannot retrieve an address from a DHCP Server.
- **Subnet Mask** - Enter the Subnet Mask for the gateway.
- **Default Gateway** - Enter the Default Gateway for the gateway.
- **NTP Server** - Enter IP or DNS name of NTP (Network Time Protocol) Server. If left blank device will not poll for time/date.
- **DNS Server 1** - Enter the IP address of the primary DNS server if DNS will be utilized.
- **DNS Server 2** - Enter the IP address of the secondary DNS server if DNS will be utilized.
- **STUN Server** - Enter IP address of STUN Server. Needed for firewall penetration.
- **DSCP/QoS** - Click drop down to select Default Forwarding or Expedite Forwarding. Expedite Forwarding tags RTP packets with DSCP bits 46 (EF). Default Forwarding does not tag RTP packets. CS-74's are shipped factory set to Default Forwarding.
- **Send RTP Keep Alives** - Allows administrator to enable or disable RTP keep alives depending on requirements of SIP server/conference bridge.
- **Send RTCP Keep Alives** - Allows administrator to enable or disable RTCP keep alives depending on application requirements.
- **Audible IP Address** - Clicking this feature will enable the CS-74 to announce the device's IP address over the built in loudspeaker.

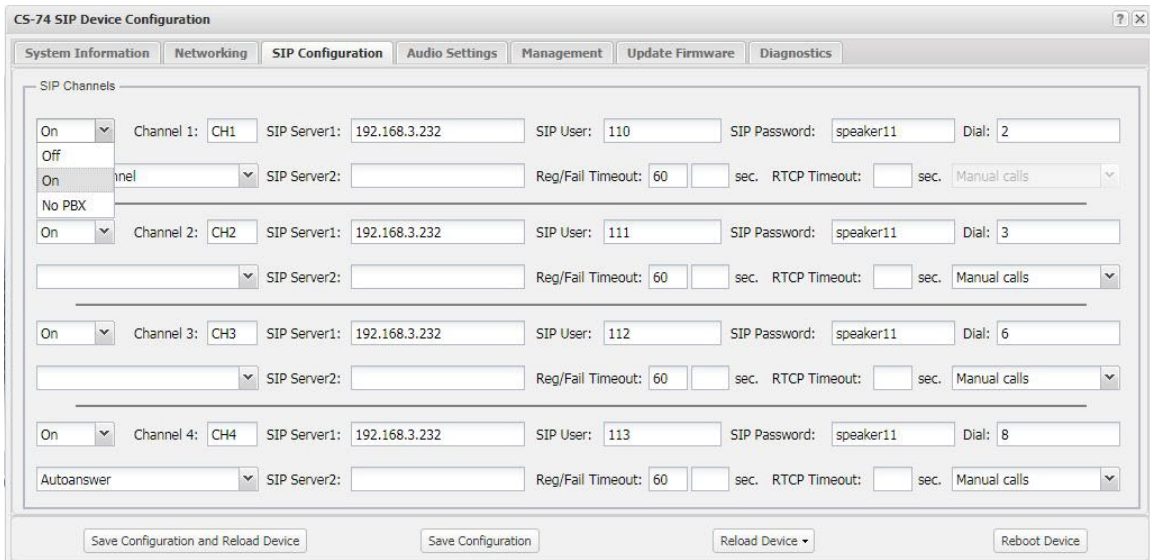
When finished entering IP information click the **Save Configuration & Reload** button at bottom of screen. Once finished with the reload, click the **Reboot** button.

Rebooting is required after making device IP address changes.

2.6 SIP Configuration Page

The SIP Configuration page allows an administrator to configure the respective SIP channels or lines of the device. Lines/channels can be set for different modes of operation, turned on or off, etc., depending on specific application needs. See Figure 6 below.

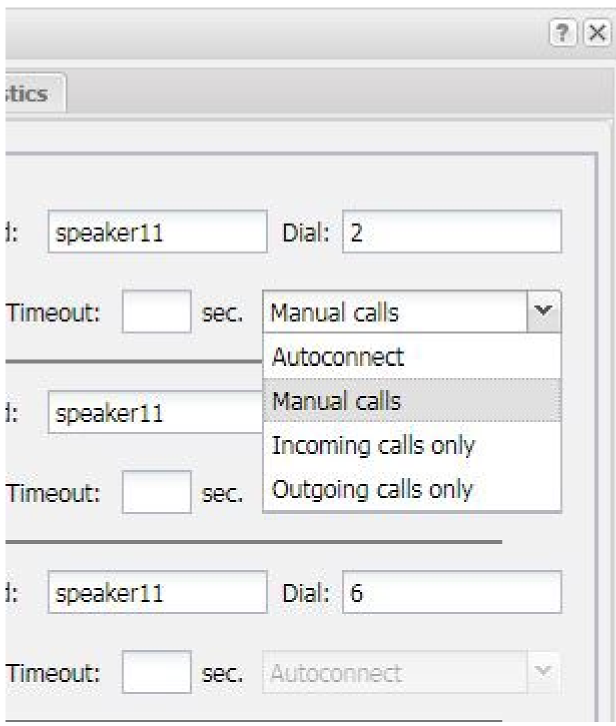
Figure 6



- **'On / Off / No PBX'**– Allows the administrator to turn on / off the respective line or set channel to 'No PBX'. If turned off, 'line information' will not be displayed on the device. No PBX allows device to make calls to other SIP end points without registering with a SIP Server/Proxy.
- **Channels 1-4** – This field allows the administrator to assign a name to the respective channel. This name will be displayed on the device. It is recommended to use upper case letters and limit number of characters to 4 per channel.
- **SIP Server** – This field allows the administrator to input the address of the SIP Server to which the device will register. Can be either DNS name or IP address. Each channel can be configured to register with a different SIP Server if desired.
- **SIP Server 2** – This field allows the administrator to input the IP address of a backup SIP server in the event the primary is not available.
- **SIP User** – This field allows the administrator to assign an extension or SIP identifier for registration.
- **SIP Password** – This field allows the administrator to assign the SIP Password for registration.
- **Reg/Fail Timeout** – These fields allows the administrator to set the time span, in seconds, for re-registration and registration fail of the channel to the SIP server.
- **Number to Dial** – This field allows the administrator to assign an extension the device will dial. Note: This function works in conjunction with specific 'Mode' selected.
- **RTCP Timeout** – This field allows the administrator to set the time span, in seconds, in which the device will drop an active call after RTCP packets are no longer present.
- **Auto Answer** – If checked this option sets the device to automatically answer an incoming call to the respective channel as assigned in the SIP User field. This option only becomes available with specific 'modes.'
- **Listen Only** – Allows administrator to set a specific channel for listen only mode.

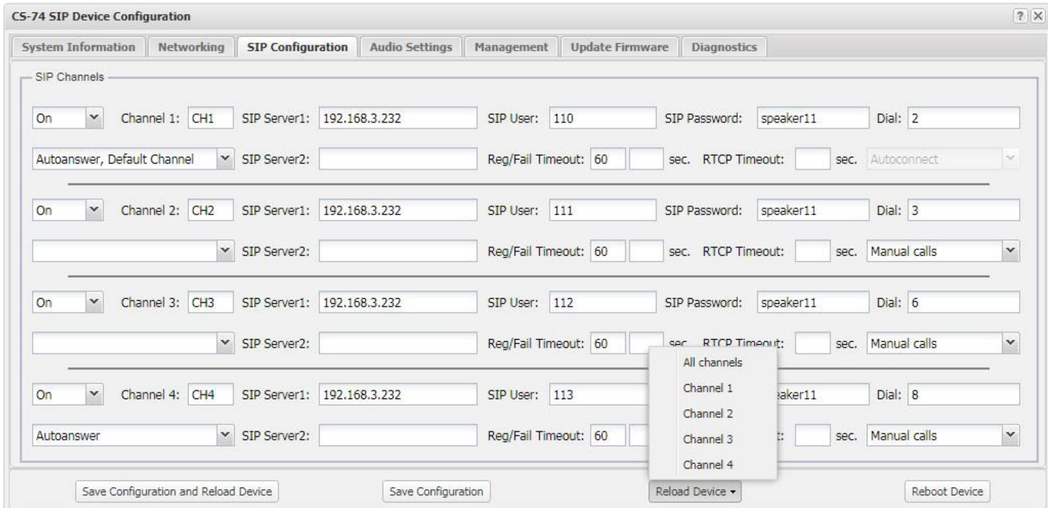
- **SIP Modes –** The drop down menu allows the administrator to select various modes of operation for each line. See Figure 7
    - **Autoconnect –** Selecting this mode instructs the device to automatically connect to the extension/number, as defined in the Number to Dial field immediately upon successful boot up.
    - **Manual –** Selecting this mode instructs the device to wait for user intervention to either accept a call or make a call. Auto answer can be selected with this mode. If selected any incoming SIP calls to the respective extension will be automatically answered by the device. However the user can still initiate an outgoing call from the device. The PTT button for the respective line is used to make/answer a call and double pressing the PTT button quickly will hangup/drop the call.
    - **Incoming Only –** Selecting this mode instructs the device to wait for user intervention to accept an incoming SIP call. In this mode the device will not make outgoing calls. Auto answer is available with this mode.
    - **Outgoing Only –** Selecting this mode instructs the device to wait for user intervention to make an outgoing SIP call. The call is initiated by pressing the PTT button on the respective line. When in this mode the device will not accept incoming SIP calls. Auto answer is disabled in this mode.

Figure 7



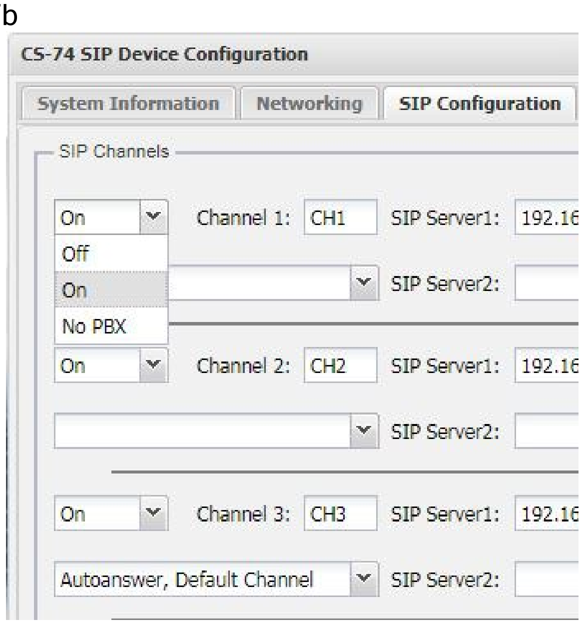
- **Channel Reload** – The CS-74 also has a feature which allows an administrator to change a single channel and load the new configuration for the channel without affecting the connection status of other channels. By clicking the reload device button at the bottom of the screen the administrator is presented with options to reload specific channels or all channels. Once a channel configuration has been changed you must save new configuration then select channel to be reloaded for changes to take effect. See Figure 7a below.

Figure 7a



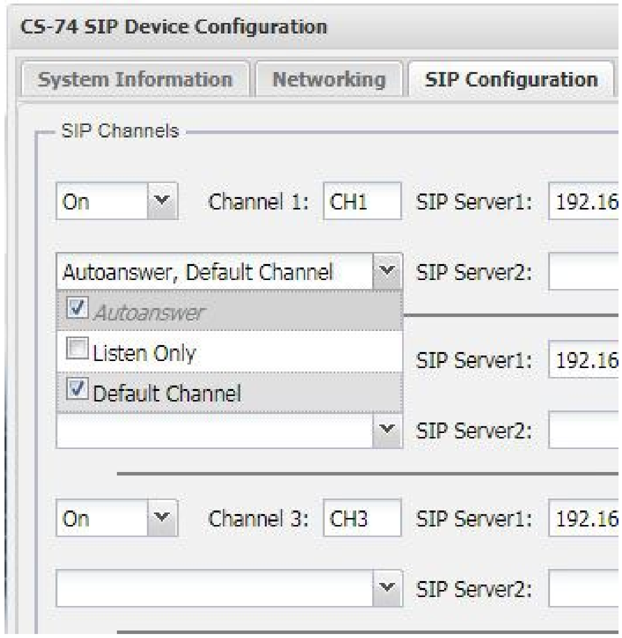
- **Call without PBX** – The CS-74 has the capability to make/take calls without registration or connection to a SIP PBX. This can be enabled on a per channel basis. Figure 7b below shows a drop down to the left of each channel where an administrator can select to turn the channel off, turn it on or select No PBX to set channel for no registration. No PBX allows the CS-74 to make/take direct point to point calls from another device or gateway without a PBX. In this mode the CS-74 will not send registration requests on the respective channel.

Figure 7b



- **Default Channel –** The CS-74 has the ability to set a “Default” pre-selected channel. Figure 7c below shows a drop down where an administrator can select a channel as the default channel. This feature is only accessible when “CB Mic” mode is enabled from the **Audio Setting**s Tab.

Figure 7C



Key Operational Features of Default Channel option

- Enabling this option will automatically set the channel to Autoconnect / Autoanswer. When the device is either rebooted or restarted, the respective channel will register, automatically dial the configured extension then select this channel for instant PTT via an attached CB style Microphone. The selected channel will indicate “SEL” above the channel on the device display. Only one channel can be set as the default and must be unselected before selecting a different channel as the default
- Users can select other channels for PTT by either press & release of a different channel button (use CB Mic) or press and hold a different channel button (use Gooseneck).

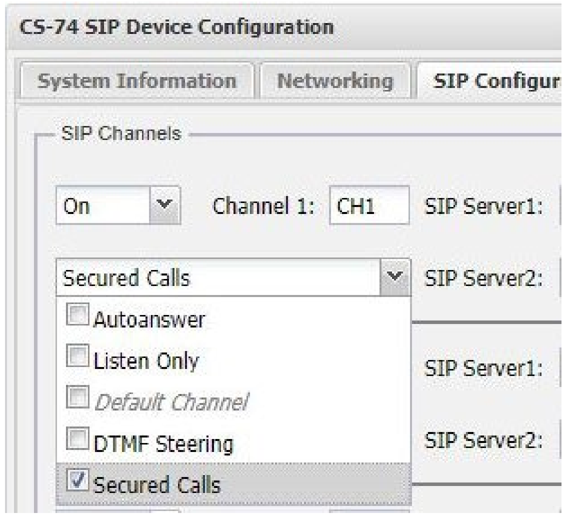
Note: If press & hold is used on a different channel other than default channel, upon release of the console PTT button the device will automatically revert back to the default channel as indicated by “SEL” above the respective channel.

- When a channel is selected (press & release event) by a user for use with the CB Mic, that channel shall stay selected until user 1. Selects a different channel, 2. Does a press and hold on a different channel, 3. Device reboot or App restart.

Note: It is highly recommended but not required that all channels be set to autoconnect when using the “Default Channel” option. If the other channels are set to “Manual Connect” then those channels will become the selected channel as they connect their call. Users must double tap the respective channel PTT button to manually connect a channel other than the default and as a result the last channel connected becomes selected for use with the CB Mic.

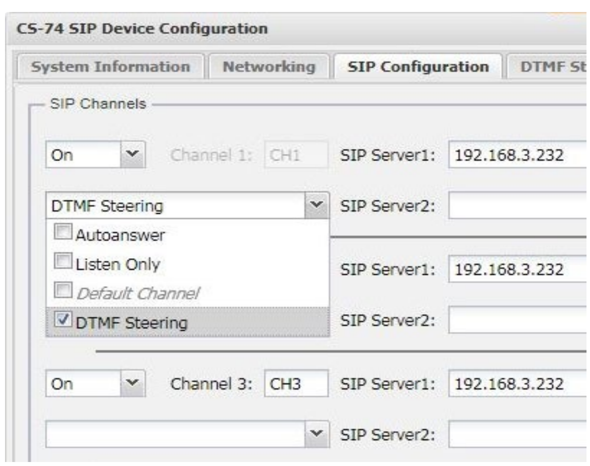
- **Secure SIP Enable** – Enabling this option allows Secure SIP messaging via TLS and Secure RTP for the data stream.

Figure 7D



- **DTMF Steering Enable** – The CS-74 has the ability to output DTMF codes on a per channel basis for the purpose of "steering" radio systems to use different frequencies/channels for communication with mobile radios. Selecting this option will activate the DTMF Steering tab to allow an administrator to input the desired Radio ID's and DTMF for each channel of the CS-74.

Figure 7E

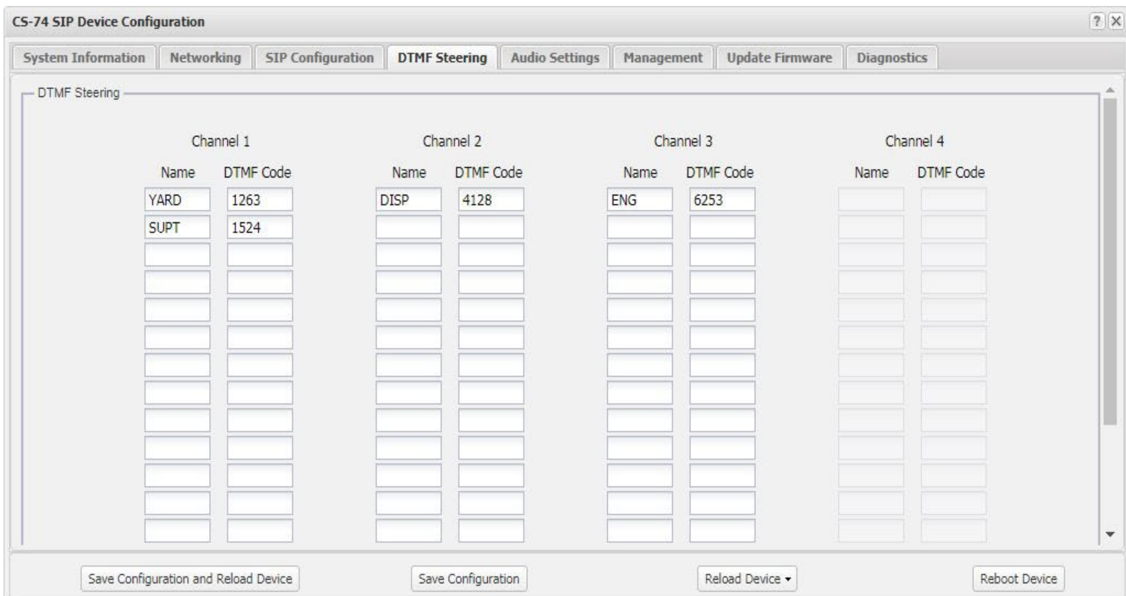


2.7 DTMF Steering Page

The DTMF Steering page allows an administrator to configure multiple radio ID's and DTMF codes for each channel as indicated in Figure 8 below. A max of 4 characters is allowed per Radio ID. Capital letters are recommended as these Radio ID's will be indicated on the CS-74 display for each channel. The first Radio ID in each channel column will be displayed upon save and reload.

Users can select a different Radio Channel by pressing the F1 button (CS-74-F only) on front of the CS-74, will trigger any channels configured for DTMF Steering to flash, press the respective channel button to scroll through Radio IDs until desired ID is found then press F1 to activate. When PTT is pressed on a respective channel, DTMF codes will be sent to the called destination as configured. Depending on application, this also works with DTMF * # option enabled on Audio Settings tab.

Figure 8

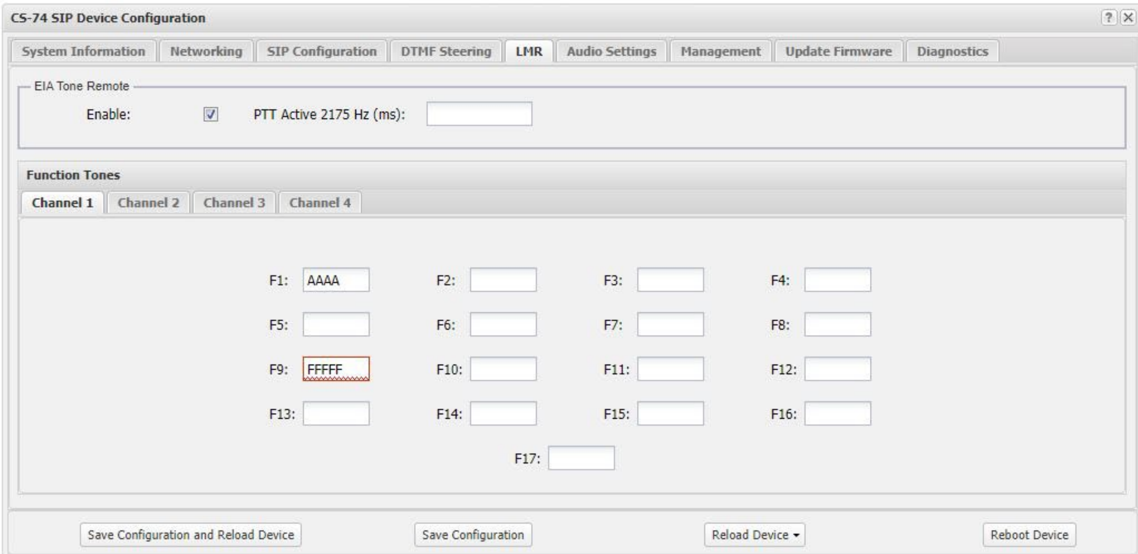


The remainder of this page intentionally left blank.

2.8 LMR / EIA Tone Settings

The LMR Settings page allows the administrator to enable and configure Land Mobile Radio EIA tones for each channel of the CS-74

Figure 9



- **EIA Tone Remote Enable** – Checking this box enables the EIA Tone Remote feature within the CS-74.
- **PTT Active 2175 Hz** – The user definable filed just to the right allows the administrator to set the duration, in milliseconds, of the initial PTT tone. Anytime PTT on a CS-74 is pressed the device will send a 2175Hz HLGT tone at 0dB for the duration specified.
- **Channel 1-4 F1-F17** – The fields next to each Frequency (Fx) allows the administrator to set a 4-character name for each of the EIA tone frequencies. These 4-character names will be displayed on the respective channels of the CS-74. The user will press and release the F1 button then press the respective channel PTT button to scroll through any configured tones then press and release the F1 button again to set. When the user presses the PTT button the device will send the respective Tone to the radio system. During the entire PTT transmission, a 2175Hz Hold Tone is transmitted at -30db.

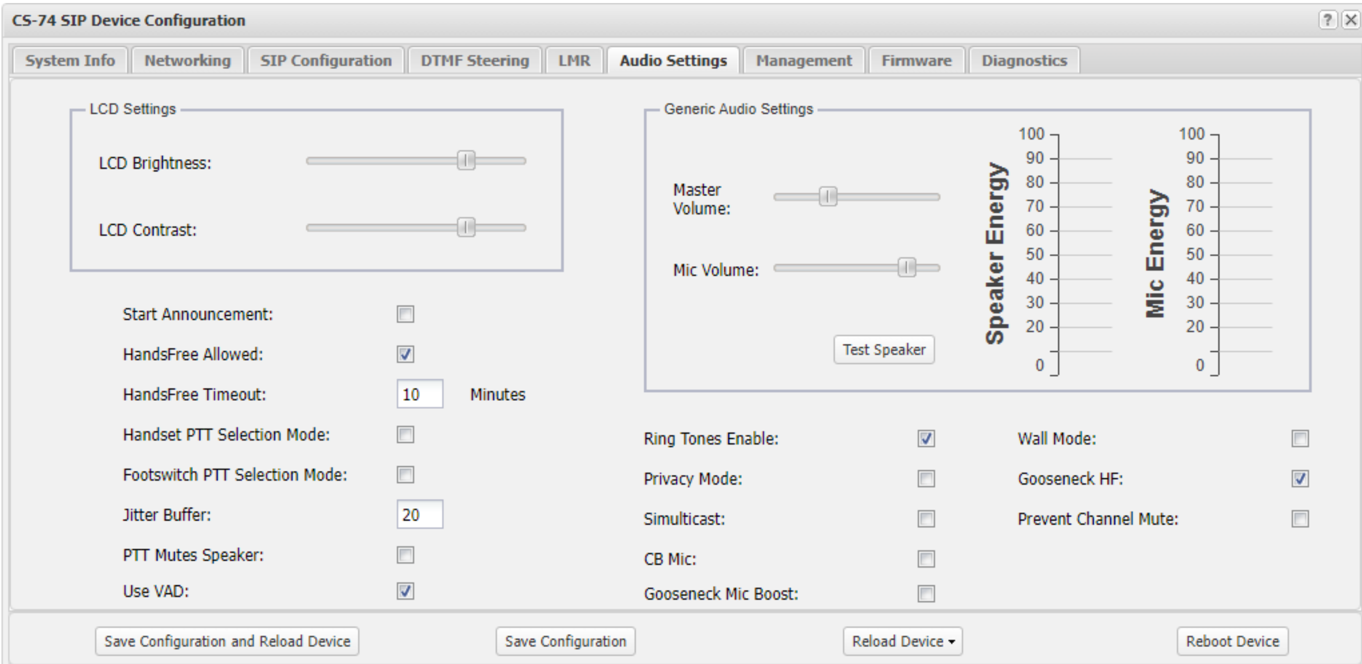
The following table is a quick reference of the respective tones:

Tone	Frequency*	Level & Duration
F1	1950 Hz	0 dBm for 40 msec
F2	1850 Hz	0 dBm for 40 msec
F3	1750 Hz	0 dBm for 40 msec
F4	1650 Hz	0 dBm for 40 msec
F5	1550 Hz	0 dBm for 40 msec
F6	1450 Hz	0 dBm for 40 msec
F7	1350 Hz	0 dBm for 40 msec
F8	1250 Hz	0 dBm for 40 msec
F9	1150 Hz	0 dBm for 40 msec
F10	1050 Hz	0 dBm for 40 msec
F11	950 Hz	0 dBm for 40 msec
F12	850 Hz	0 dBm for 40 msec
F13	750 Hz	0 dBm for 40 msec
F14	650 Hz	0 dBm for 40 msec
F15	550 Hz	0 dBm for 40 msec
F16	2350 Hz	0 dBm for 40 msec
F17	2450 Hz	0 dBm for 40 msec

2.9 Audio Settings Page

The Audio Settings page allows the administrator to adjust specific device properties not covered in other configuration pages such as master volume settings, LCD contrast and other feature settings. See Figure 10.

Figure 10



- **LCD Settings** - Allows the administrator to adjust brightness & contrast of device display. Slide right to increase, left to decrease.
- **Audio Settings** - Allows the administrator to adjust master speaker volume & microphone gain of device. Speaker & microphone level meters are provided as simple diagnostic tools to confirm if device is working properly. Master volume level and Mic Volume are default to set to 0 (unity).
- **Test Speaker** - Pressing this button plays a test message on the device.
- **Start Announcement** - If this box is checked the device will play a message indicating it is being restarted any time the device experiences a reload or reboot.
- **Handsfree Allowed** - Checking this box enables Hands Free functionality. *Available in the CS-74-F model on special order.*
- **Handsfree Timeout** - This setting allows an administrator to set the time, in minutes, the handsfree mode will stay on if a user forgets to turn it off after use. ETC recommends setting this for 10-15 minutes. Device defaults to 10 minutes, max is 60 minutes.
- **Handset PTT Mode** - This option should only be checked if a PTT handset will be connected to the device. When selected the user will select the channel to Tx/Rx on, “SEL” will appear on the display above the selected channel and to talk the user will press the PTT button on the handset.
- **Footswitch PTT Mode** - Checking this box enables use of a foot switch to activate PTT function of a selected channel. In this mode the buttons on front panel are used to select a channel for which the footswitch will activate the PTT.
- **Jitter Buffer** - Enter number for packet size matching with 3rd party SIP systems.
- **PTT Mutes Speaker** - If this box is checked the speaker will be muted when PTT is pressed on any channel. If unchecked the device is in full duplex mode on respective PTT channel.

- **VAD Enable** – If this box is checked the device will stop producing RTP audio packets after 6 seconds of no voice while PTT is pressed. If unchecked device will continuously produce RTP audio packets when PTT is pressed regardless if user is speaking.
- **Ring Tone Enable** – Enables/disables ring tones, default is checked (enabled).
- **PTT DTMF * # -** Enables device to output, "in-band" DTMF * when PTT button is pressed and DTMF # upon release of PTT. This is a global setting for the device. When checked all channels operate the same. Default is unchecked.
- **Gooseneck Mic Boost** – Enabling this feature adds 9dB of gain to the gooseneck mic sensitivity for use in mixed environments with earlier versions of CS-74's (pre-218 serial number) or other 3rd party VoIP products.
- **Wall Mode** – Enabling this feature sets the device to use only microphones built in to a ruggedized enclosure as the device is not outfitted with a gooseneck microphone. *It is not recommended to use this feature on standard units as unusual audio behavior may result*.

Note: The following features are available only on model CS-74-F with the additional function keys.

- **Privacy** – Enabling this feature allows user to select a specific channel to have a private conversation on. **Requires connection of a handset or headset.** User must press the F2 button then select desired channel to converse on by press & release of desired channel PTT button. Incoming audio on unselected channels will be routed to the built in speaker.
- **Gooseneck HF** – Uses the gooseneck microphone for the HandsFree function
- **Prevent Channel Mute** – Prevents channels to be completely muted. Requires an additional configuration entry in the general.conf file>misc section. : minimum_mute="x". "x" is the minimal mute volume.
- **Simul-select or Group Select (AKA "Simulcast")**– Enabling this feature allows the user to transmit ("Broadcast") on 2 or more channels at the same time. User presses and holds F1 button then presses each respective channel button to select channels for 'simulcast' which will be indicated by 'SIMC' over the respective channel indicator. When done selecting channels, user releases F1 button, F1 button LED stays lit indicating mode is active. To "Broadcast" simply presses the PTT button of one of the selected channels to transmit.

Note: If DTMF steering is enabled, Simulcast will be grayed out and not selectable as the F1 key is utilized to enable DTMF Steering selection mode.

The remainder of this page intentionally left blank.

2.9 Audio Settings Page cont.

- **CB Mic** – Enabling this feature sets the device to operate with an external CB style microphone or the gooseneck mic only. Use of this feature requires an additional hardware component, purchased separately from ETC, to interface the CB microphone to the device's Handset/Headset port on the back of the device. See figure 10A for physical description of the CB Mic Adapter Module

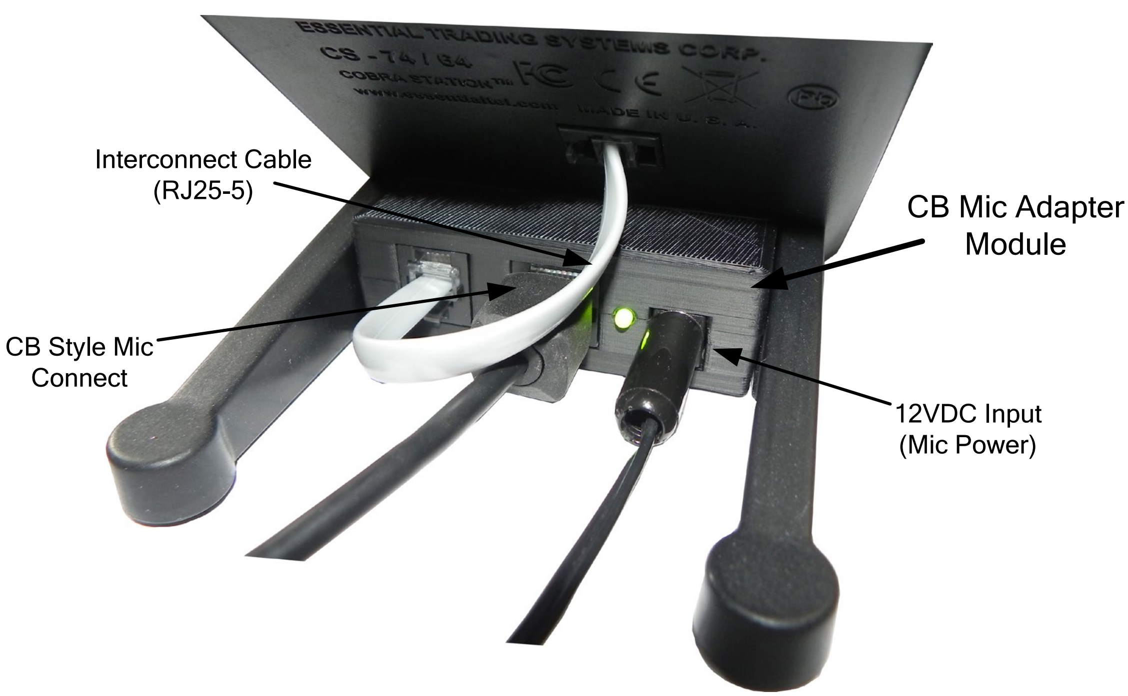
Operational features:

- Requires use of CB Mic Adapter hardware kit (option) and Motorola "CB Style" Microphone. This kit was designed to externally power a CB mic and also trigger the CS-74's Handset/headset mode, as indicated by an "H" in the upper right corner of the device display, to allow "channel select" feature for use with CB mic.

Note: If the device is equipped with the CB Mic Adapter Module (Figure 10A) you must remove the short silver satin interconnect cable, between the Adapter Module and the CS-74, if normal operation is desired, i.e., the "H" must no longer indicate on the display.

- Incoming audio from the network shall be heard via built in speaker. All incoming channel audio is mixed.
- Works in conjunction with "Default Channel" channel option set in the **SIP Settings** tab.
- Selected channel PTT is executed by pressing the PTT button on the side of the CB style mic.
- User can select other channels for CB mic usage by press & release of a different console channel button. SEL will be indicated above the selected channel.
- The gooseneck microphone may be used by press & hold of the respective console channel's PTT button. Upon release of press & hold event, selected channel shall revert to the Default Channel as set on the **SIP Settings** tab and be ready for use with CB mic.
- If "PTT Mutes Speaker" is unchecked, incoming audio from the network shall be heard via the built in loudspeaker during any PTT event on the device.

Figure 10A

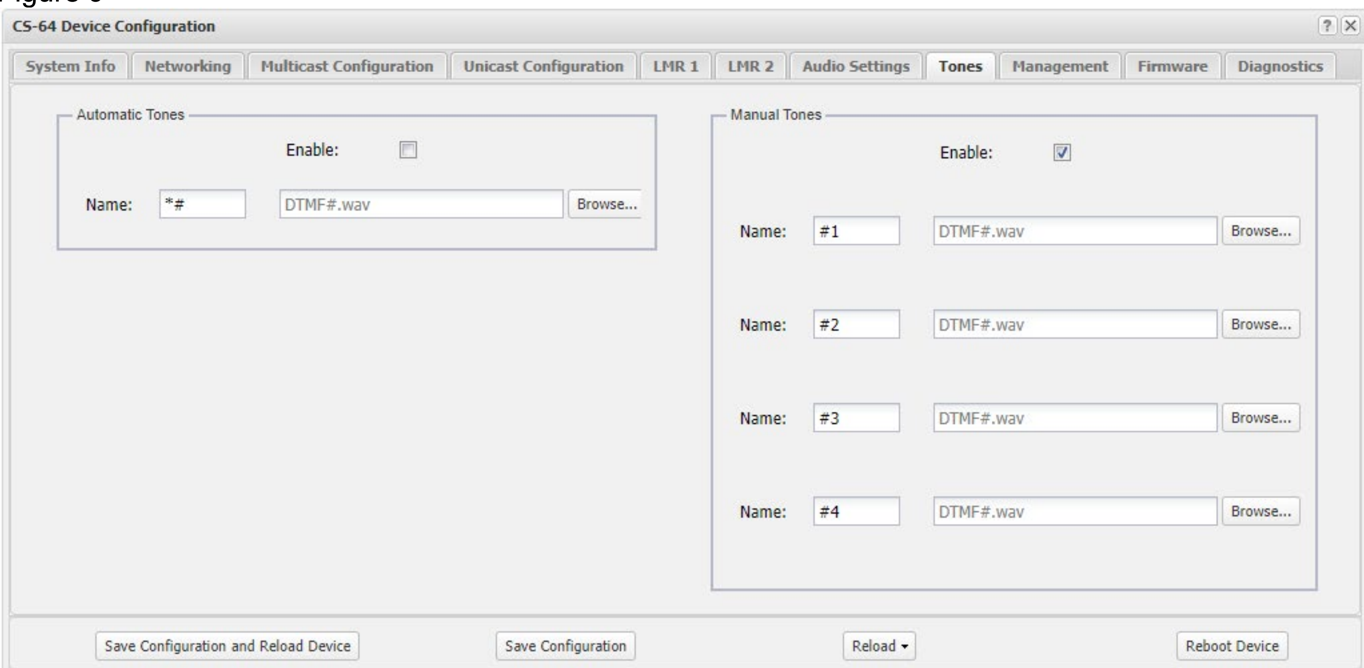


2.10 Two Tone Paging

The Tones tab provides the ability to upload and send out either Automatic or Manual Paging Tones. Automatic tones are sent each time the PTT button is pressed to transmit. Manual tones can be sent by using the F2 button to first select one of the Four Pre-Loaded tones and selecting the channel for the tones to be sent over. Each Tone can be given a unique 4-character Name (Alias).

All tones must be in the .wav format and not exceed 600kb in file size.

Figure 9

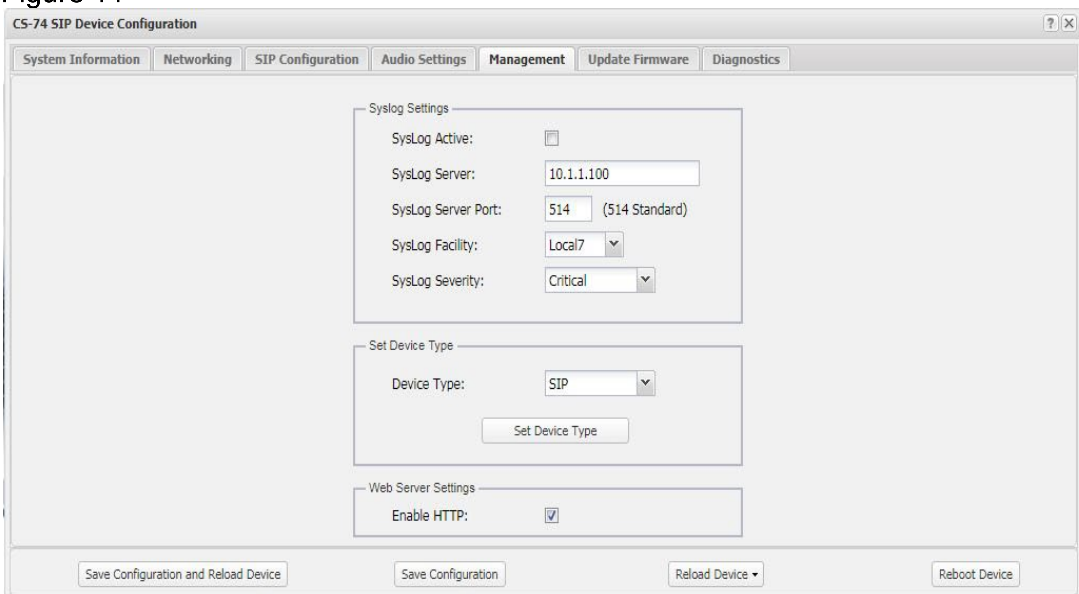


- **Automatic Tones** - If enabled using the Enable Check Box the Administrator is able to upload any desired tone Audio file as the Pre-Set Tone(s). The Tones must meet the restrictions mentioned at the top of the page. The Automatic Pre-Set Tones are sent each time the Talking Channel PTT button is pressed for transmit. As an indication of a Tone(s) being sent the Station will show a blinking amber LED for the duration of the tone(s). Please note the name of the automatic tone is for description purposes only.
- **Manual Tones** - If enabled using the Enable Check Box the Administrator is able to upload up to four individual tone(s) via an Audio file(s). The Tones must meet the restrictions mentioned at the top of the page. Manual Tones can be sent by the Station User at any time by pressing the F2 button. Once the F2 button is pushed the Talk Channels Names (1-4) Change to the Manual Tone Names. Tone 1 is on Talk Channel Button One and so forth for each of the Three remaining Talk Channel Buttons. As an example, To Select “Tone 3” push Talk Channel Button Three which selects the desired Tone(s). Once the Tone Name is selected (in this example Tone 3) the operator must then follow the onscreen instructions and select the Talk Channel which the tone will be sent over. As an indication of a Tone(s) being sent the Station will show a blinking amber LED under the Volume knob of the respective Talk channel for the duration of the tone.

2.11 Management Page

The Management tab currently provides an administrator access to input Syslog Settings for reporting device status, device type selection & browsing settings. Additional features will be added to this page as necessary. See figure 11 below.

Figure 11



- **Syslog Active** – Click the check box to enable Syslog reporting feature, uncheck to disable.
- **Syslog Server** – Input the respective IP address of the Syslog server where status reports will be sent.
- **Syslog Server Port** – Input the respective port the device will be reporting status to.
- **Syslog Facility** – Click the arrow for a drop down menu to select reporting ID. Selecting one of the displayed selections will cause the device to report activity with this 'ID' which can be used for filtering and/or sorting messages from specific devices.
- **Severity Level** - Click the arrow for a drop down menu to select reporting priority.
  - Verbose – All messages are reported
  - Critical – Only messages classified 'critical' will be reported.
- **Device Type** - This drop down allows an administrator to change the device type from a SIP device (CS-74) to a multicast device (CS-64). To change device type, click the drop down, select the device type (SIP or RTP) then click the **Set Device Type** button to activate the change.

WARNING! Please refresh (shift F5) your browser window immediately after changing the device type to avoid configuration data loss.

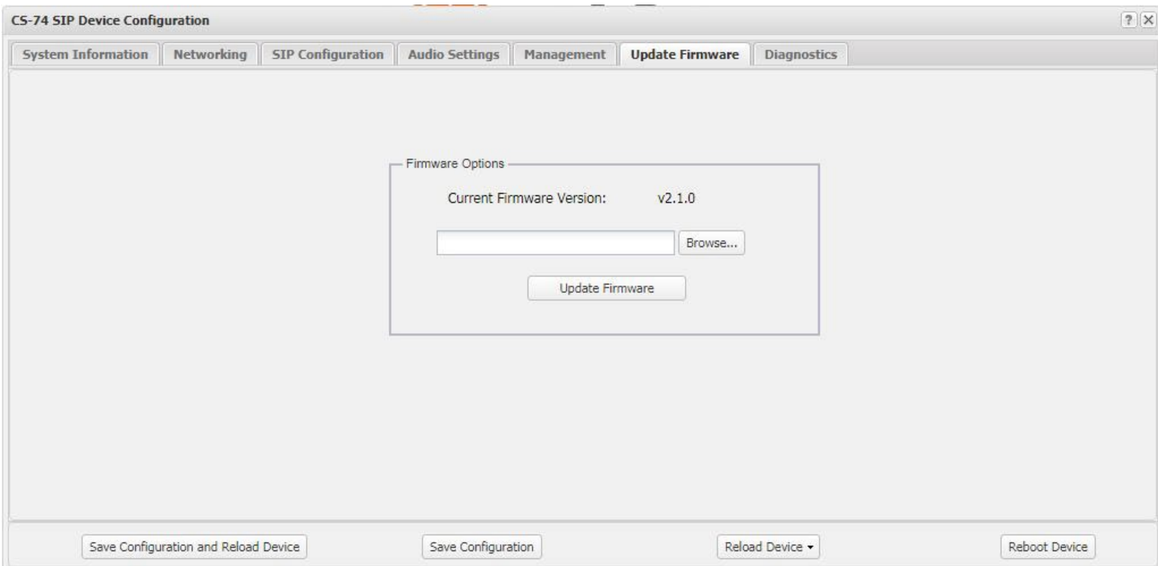
- **Enable HTTP** - Unchecking this option allows only secure (https) browsing to device for configuration. Device is shipped default to 'Enable.'

Note: Syslog message definitions are provided in appendix 4.2 of this guide.

2.12 Update Firmware

The Update Firmware page allows an administrator to easily & quickly update firmware on a device. From time to time ETC will send out firmware releases to fix bugs or add features. Simply click the browse button and navigate to where the firmware file has been saved then click Update Firmware. A pop up window will appear indicating status of firmware update. See Figure 12.

Figure 12



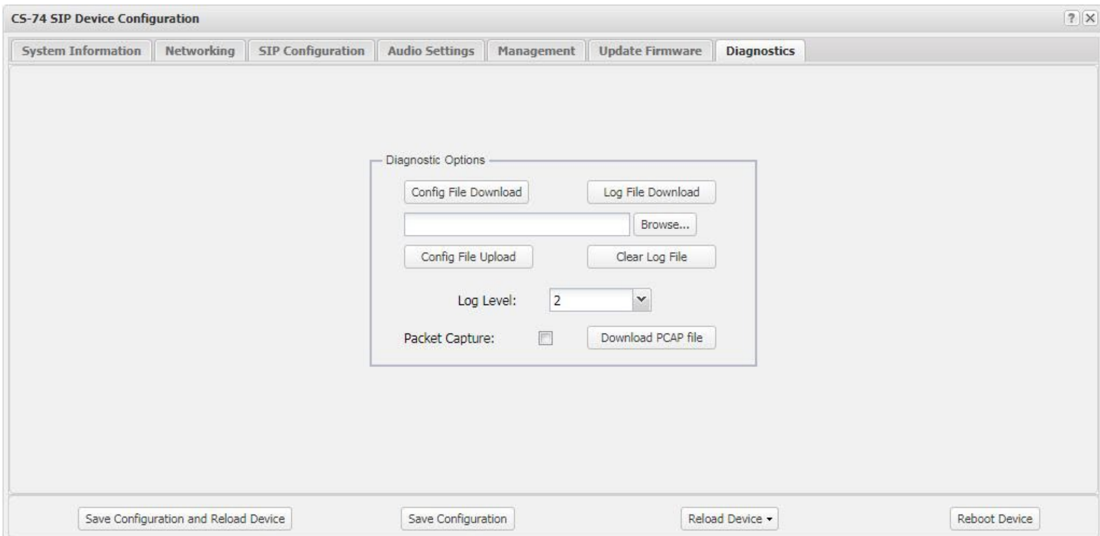
NOTE: AFTER UPDATING FIRMWARE YOU MUST (Shift F5) REFRESH YOUR BROWSER WINDOW FOR NEW FEATURES TO BE DISPLAYED. Most browsers will cache previous pages of the device and therefore a refresh must be performed after a firmware update.

WARNING!! - Firmware Ver 2.0.0 and beyond is not backward compatible 2140xxxx serial number devices. Doing so will render your device unusable and unrecoverable.

2.13 Diagnostics Page

The Diagnostics page has been provided to allow an administrator access to troubleshooting tools such as Activity Log. These tools will be useful in providing ETC information specific to the device to aid in diagnosing problems. See Figure 13.

Figure 13



- **Config file Download** – Allows administrator to download the devices configuration information to a PC or server. File format is .conf and is readable with Notepad or equivalent.
- **Browse** – Allows administrator to find & select CS-74 configuration file for upload.
- **Config File Upload** – Uploads the selected configuration file.
- **Log Level** – Allows administrator to set the logging level to capture event information for troubleshooting purposes. Setting range is 1 (lowest) to 6 (highest). ETC recommends leaving device at level 2-3 unless more in depth logging is requested by ETC Support.
- **Log file Download** – Click this button to download the log info to file on a PC or server. File format is .conf which is readable with Notepad or equivalent.
- **Packet Capture** – Clicking this checkbox will start a packet capture on the device. A pop up will indicate "uncheck box to stop packet capture". When packet capture is stopped a pop up will indicate "Press Download PCAP file button" to download the packet capture to your PC.
- **Download PCAP File** - Press this button to download the PCAP file after packet capture is complete.

3.0 Frequently Asked Questions

Q: What is the IP address of device?

A: When the device boots up the IP, Mask , Gateway and MAC address are presented briefly on the display. You can also press button 1 & 4 at the same time and IP address will be displayed on the device. This also performs a soft reset of the device.

Q: The device is not powering up?

A: If you are using a POE switch, make sure the patch cable is securely seated in the jack of the device. If you are not using a POE switch then you can use an optional PoE injector type power supply.

Q: How do I change the device IP?

A: Once the device has booted up and you have identified its IP address, open a browser and browse to the device's IP address, Login and go to Network settings page to change the network settings.

Q: Channel is not registering? (No amber LED on a Channel)

A: Via device web page, check to make sure the respective channel is enabled and has the correct SIP connection information i.e., SIP server IP, extension, password, etc.

Q: How do I change channel's label?

A: Labels can be changed via the "SIP Configuration" web page. Max of 4 alphanumerical characters. It is recommended to use upper case characters.

Q: User does not hear audio?

A: Is it on a specific channel or no audio from any channel?

1. If a single channel, check if channel is connected (green LED on device). If not connected instruct user on method to connect the channel.
2. Is channel registered? If not investigate reasons for lack of channel registration

Q: IP address of device does not change?

A: The device requires a reboot after changing the IP address. A 'Save & Reload' does not activate IP address changes.

Q: User reports unable to transmit?

A: Is user pressing PTT button for respective channel?

A: On the 'Audio Settings' web page is there activity on the 'Mic Energy' meter, if yes the physical microphone is working.

A: Check Mic volume on device 'Audio Settings' webpage, should be at 50% or higher depending on the user.

A: Is channel connected? Indicated by green LED above respective channel PTT button.

Q: Audio received on device is choppy/garbled

A: Please check if the codec chosen by the SIP server is compatible with the CS-74 device. The list of compatible codecs is in Specifications section of the CS-74 Admin Guide.

A: Please check with the network administrator to ensure a proper QOS policy is in place.

Q: Reports of single user transmitting louder/quieter than other users

A: Once you have identified the IP address of that user's device, open a browser and browse to the device's IP address, Login and go to Audio settings to adjust the master microphone and speaker levels as needed.

4.0 Appendix

4.1 Specifications

Channels
- (4) SIP Lines

Call Types
- SIP, hoot conferencing
- SIP Private line (ARD/MRD)

Signaling
- SIP
- EIA Tones
- DTMF

Interfaces
- 12" Gooseneck Microphone
- Handset w/PTT
- Plantronics Mono Headset (HW251N)
- Footswitch (PTT only)
- NIC, (1) RJ45, 10Mb Ethernet,

Network Requirements
- 100 Base T, (full duplex)
- IEE 802.3af (PoE) compliant
- Built in Ethernet Hub
- Protocols - SIP, UDP, NTP, DHCP, TCP, HTTP, HTTPS, Syslog, DNS, SFTP, RTCP

Dimensions
- Width - 4" / 102 mm
- Depth – 5.5" / 140 mm
- Height – 5.5" / 140 mm
- Weight – 1.1 lbs / 525 g
- 12"/305 mm - Gooseneck microphone

Media
- Bandwidth - supports codecs: G.711 80 kbps, G.729 8kbps and Speex.
- SIP, UDP
- Linux OS,
- Audio – 300Hz – 3kHz, 1 Watt RMS, EIA Tones, DTMF

Management
- Browser based, Internet Explorer, Google Chrome
- Supports HTTP & HTTPS (self signed certificate)
- Upgradeable application firmware via file upload
- Syslog output

Power
- 48 VDC, 1/2 A , external power supply via injector
- 48 VDC, IEEE 802.3af, Alt A & B, Power over Ethernet compliant (PoE).

Thermal
- 3 Watts
- 10 BTU/hr
- Cooling – Ambient air

Other
- Handset/Headset Pinout (RJ-25)
  1 – PTT+
  2 – EAR-
  3 – MIC+
  4 – MIC-
  5 – EAR+
  6 – PTT-

Optional Accessories
- PTT Belt pack, PN -2318
- Mono Headset, PN - HW251N (Plantronics)
- Foot switch, PN - FP-115
- Handset, PN - TH-3
- Dongle headset/footswitch combo, PN–CDA-FPHW251
- Dongle, Headset no PTT, PN – QD-RJ25

4.2 Syslog Messages

Below is a list of generic Syslog messages the CS-64 may produce which can be used with a customer provided Syslog Server. The messages have been classified into 2 categories; Critical & Verbose.

Verbose - "Device type is "
Verbose - "RTPD started"
Verbose - "No channels configured ! Please check the cfg file"
Verbose - "Old_Dev mode is "
Verbose - "Connected to MIC"
Verbose - "RTPD TX part done"
Verbose - "Connected to Speaker"
Verbose - "RTPD RX part done"
Verbose - "Mixer thread started"
Verbose - "Player thread started"
Verbose - "Can't install SIGUSR2 signal !"
Verbose - "Can't install SIGPIPE signal !"
Verbose - "External Mic inserted"
Verbose - "External Mic is present"
Verbose - "External Mic removed"
Verbose - "External Mic absent"
Verbose - "Start playing playfile"
Verbose - "Error opening playfile"
Verbose - "Stop playing playfile"
Verbose - "Error reading playfile"
Verbose - "fill_samples_buf: wrong len "
Critical - "spk: can't open device "
Critical - "spk: can't allocate hardware configuration structure"
Critical - "spk: hardware configuration structure cannot be assigned to device"
Critical - "spk: access method cannot be configured : " << snd_strerror(err);
Critical - "spk: can't get access method"
Critical - "spk: access method set failed : "
Critical - "spk: can't configure format : "
Critical - "spk: can't get format : "
Critical - "spk: format set failed : "
Critical - "spk: can't set sample rate : "
Critical - "spk: can't get sample rate : "
Critical - "spk: sample rate set failed : "
Critical - "spk: can't set channels : "
Critical - "spk: can't get channels : "
Critical - "spk: channels set failed : "
Critical - "spk: can't set buffer size : "
Critical - "spk: can't get buffer size : "
Critical - "spk: buffer size set failed : "
Critical - "spk: can't set period size : "
Critical - "spk: can't get period size : "
Critical - "spk: period size set failed : "
Critical - "spk: can't configure hw_params : "
Critical - "spk: buffer overrrun cannot be recovered, snd_pcm_prepare fail: "
Critical - "spk: ESTRPIPE"
Critical - "spk: suspend cannot be recovered, snd_pcm_prepare fail: "
Critical - "spk: EBADFD"
Critical - "spk: unknown error: "
Critical - "spk: Invalid poll descriptors count"
Critical - "spk: Unable to obtain poll descriptors for write: "
Critical - "spk: Write error: "
Critical - "spk: Wait for poll failed"

Critical - "g729ab_decode: Invalid parameter !"
Verbose - "\nStat: total streams: "
Verbose - "Stat: streamX: [ChX] ssrc: 0x"
Critical - "*** rw_and_poll_loop failed, restarting ..."
Critical - "mic: can't open device "
Critical - "mic: can't allocate hardware configuration structure : "
Critical - "mic: hardware configuration structure cannot be assigned to device : "
Critical - "mic: access method cannot be configured : "
Critical - "mic: can't get access method : "
Critical - "mic: access method set failed : "
Critical - "mic: can't configure format : "
Critical - "mic: can't get format : "
Critical - "mic: format set failed : "
Critical - "mic: can't set sample rate : "
Critical - "mic: can't get sample rate : "
Critical - "mic: sample rate set failed : "
Critical - "mic: can't set channels : "
Critical - "mic: can't get channels : "
Critical - "mic: channels set failed : "
Critical - "mic: can't set buffer size : "
Critical - "mic: can't get buffer size : "
Critical - "mic: buffer size set failed : "
Critical - "mic: can't set period size : "
Critical - "mic: can't get period size : "
Critical - "mic: period size set failed : "
Critical - "mic: can't configure hw_params : "
Critical - "mic: buffer overrun"
Critical - "mic: buffer overrrun cannot be recovered, snd_pcm_prepare fail: "
Critical - "mic: ESTRPIPE"
Critical - "mic: suspend cannot be recovered, snd_pcm_prepare fail: "
Critical - "mic: EBADFD"
Critical - "mic: unknown error: "
Critical/Verbose - "alsa_read: "
Verbose - "channel_disabled: Invalid channel: "
Verbose"Disabling MIC"
Critical"enable_channel: Invalid channel: "
Verbose - "Enabling MIC on channel"
Critical - "random32: failed"
Critical – “rebooting”
Critical – “booted”

5.0 CS-74 Limited Warranty

ETC warrants that your ETC hardware product shall be free from defects in material and workmanship for One Year, beginning from the date of purchase. Except where prohibited by applicable law, this warranty is nontransferable and is limited to the original purchaser. This warranty gives you specific legal rights, and you may also have other rights that vary under local laws

ETC's entire liability and your exclusive remedy for any breach of warranty shall be, at ETC's option, (1) to repair or replace the hardware, or (2) to refund the price paid, provided that the hardware is returned to the point of purchase or such other place as ETC may direct with a copy of the sales receipt or dated itemized receipt. Shipping and handling charges may apply except where prohibited by applicable law. ETC may, at its option, use new or refurbished or used parts in good working condition to repair or replace any hardware product. Any replacement hardware product will be warranted for the remainder of the original warranty period or thirty (30) days, whichever is longer or for any additional period of time that may be applicable in your jurisdiction. This warranty does not cover problems or damage resulting from (1) accident, abuse, misapplication, or any unauthorized repair, modification or disassembly; (2) improper operation or maintenance, usage not in accordance with product instructions or connection to improper voltage supply; or (3) use of consumables, such as replacement batteries, not supplied by ETC except where such restriction is prohibited by applicable law.

Before submitting a warranty claim, we recommend you contact ETC support at support@essentialtel.com for technical assistance. Valid warranty claims are generally processed through the point of purchase during the first thirty (30) days after purchase; however, this period of time may vary depending on where you purchased your product – please check with ETC for details. Warranty claims or other product related questions should be addressed directly to ETC. The addresses and customer service contact information for ETC can be found in the documentation accompanying your product and on the web at www.essentialtel.com.

ETC SHALL NOT BE LIABLE FOR ANY SPECIAL, INDIRECT, INCIDENTAL OR CONSEQUENTIAL DAMAGES WHATSOEVER, INCLUDING BUT NOT LIMITED TO LOSS OF PROFITS, REVENUE OR DATA (WHETHER DIRECT OR INDIRECT) OR COMMERCIAL LOSS FOR BREACH OF ANY EXPRESS OR IMPLIED WARRANTY ON YOUR PRODUCT EVEN IF ETC HAS BEEN ADVISED OF THE POSSIBILITY OF SUCH DAMAGES. Some jurisdictions do not allow the exclusion or limitation of special, indirect, incidental or consequential damages, so the above limitation or exclusion may not apply to you.