ESSENTIAL TRADING SYSTEMS CORP.

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CS-808/812

Administration & Setup Guide

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THE POSSIBILITIES ARE ENDLESS

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AG

Release notes:

7/23/2024	Ver 1.1.7	Introduced All Call and Stereo Audio Output mode. Minor bug fixes.
	All units manufactured prior to July of 2024 must have the 1.1Edge firmware installed prior to installing 1.1.7	
5/23/2022	Ver 1.0.16	Incoming Audio Indication Delay introduced
4/25/2022	Ver 1.0.12	Headset Only mode introduced
2/1/2022	Ver 1.0.7	Initial release firmware

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1.0 Product Overview

1.1 Product Description

The **CS-812** is a Twelve-channel console capable of simultaneously connecting either SIP, Multicast or Unicast channels. **CS-808** is **identical in operation to the CS-812** but has four fewer Talk Channels. Here after the units will be referred to in generic terms as **CS-8XX**.

1.2 Product features of the CS-8XX units

- **Display** – Provides information such as date and time, network connectivity, PTT status, channel information as defined by administrator and other device status indicators.
- **Volume Control / PTT button** – Each channel has an individual volume control so users can set levels to their liking. Users can PTT on desired channel by pressing and holding down the channel's volume knob.
- **Loudspeaker** – Each CS-8XX is equipped with a built-in loudspeaker. Administrators have access to master volume control via web page configuration. Users can adjust volume for each channel with the volume control knob. Audio for all Channels is mixed in the speaker.
- **A-D Buttons** – These buttons have been added to provide expanded capability of the device. These buttons can be used for numerous different functions such as DTMF steering, EIA Tones, Group call etc.
- **Handset or Headset** – The CS-8XX 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.
- **Foot Switch** – The CS-8XX can be equipped with an optional foot switch for PTT activation. Footswitch will connect to the handset port on the back of unit. Pins 1 & 6 are used for contact closure input to activate PTT.
- **Gooseneck Mic** – The CS-8XX is typically equipped with a close talking gooseneck microphone.

2.0 Administration

2.1 Login

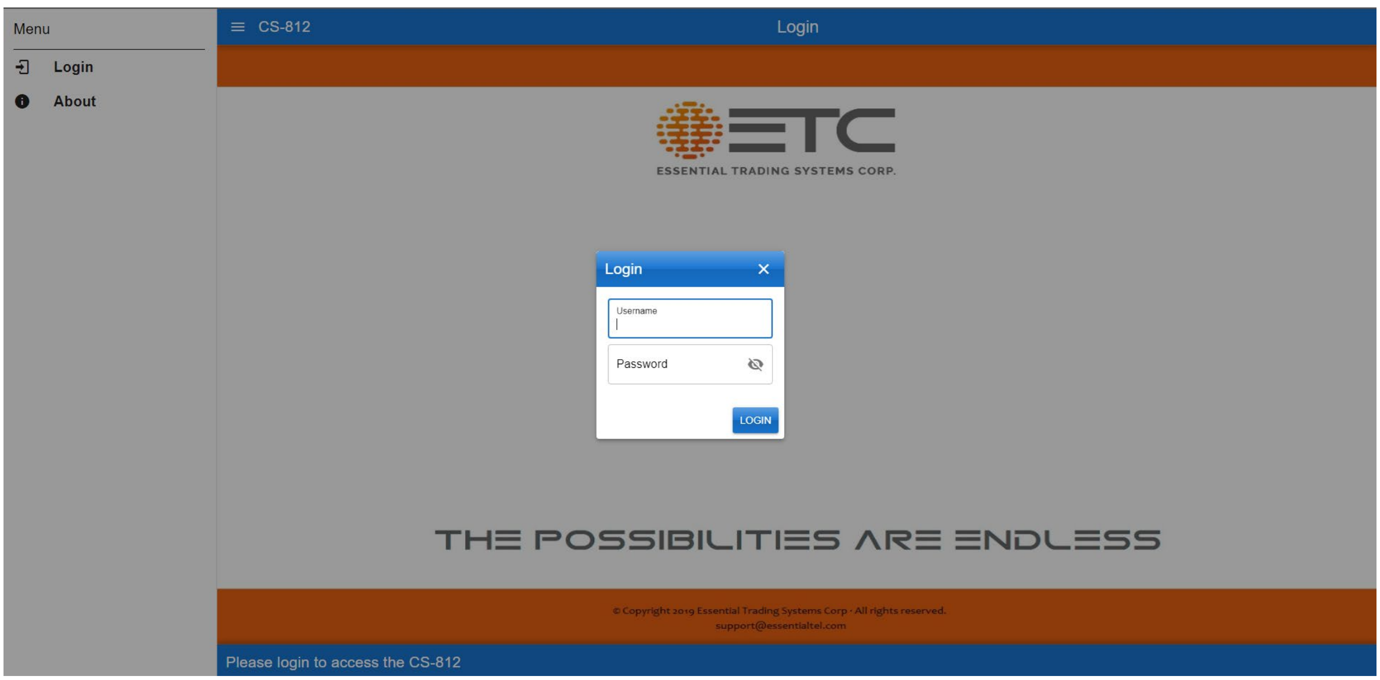
The CS-8XX is configured via web portal. CS-8XX's ship, default, set to DHCP. Upon connection to the appropriate network the device will automatically acquire an IP address. This address will be indicated on the device's display during boot up.

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

Once the IP address has been determined, open a browser, and type the IP address of the CS-8XX into the URL bar of the browser and press enter. The CS-8XX 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 Home page. See Figure 2.

Figure 2

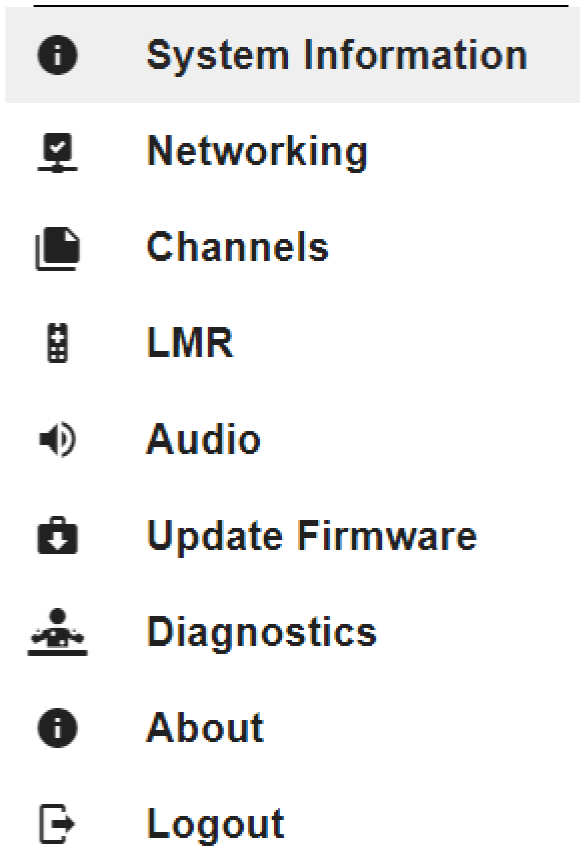
Admin Password Change - The administrator password can be changed from any screen by pressing the icon highlighted on figure 2b below.

Figure 2b

2.3 Menu Options

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

Figure 3

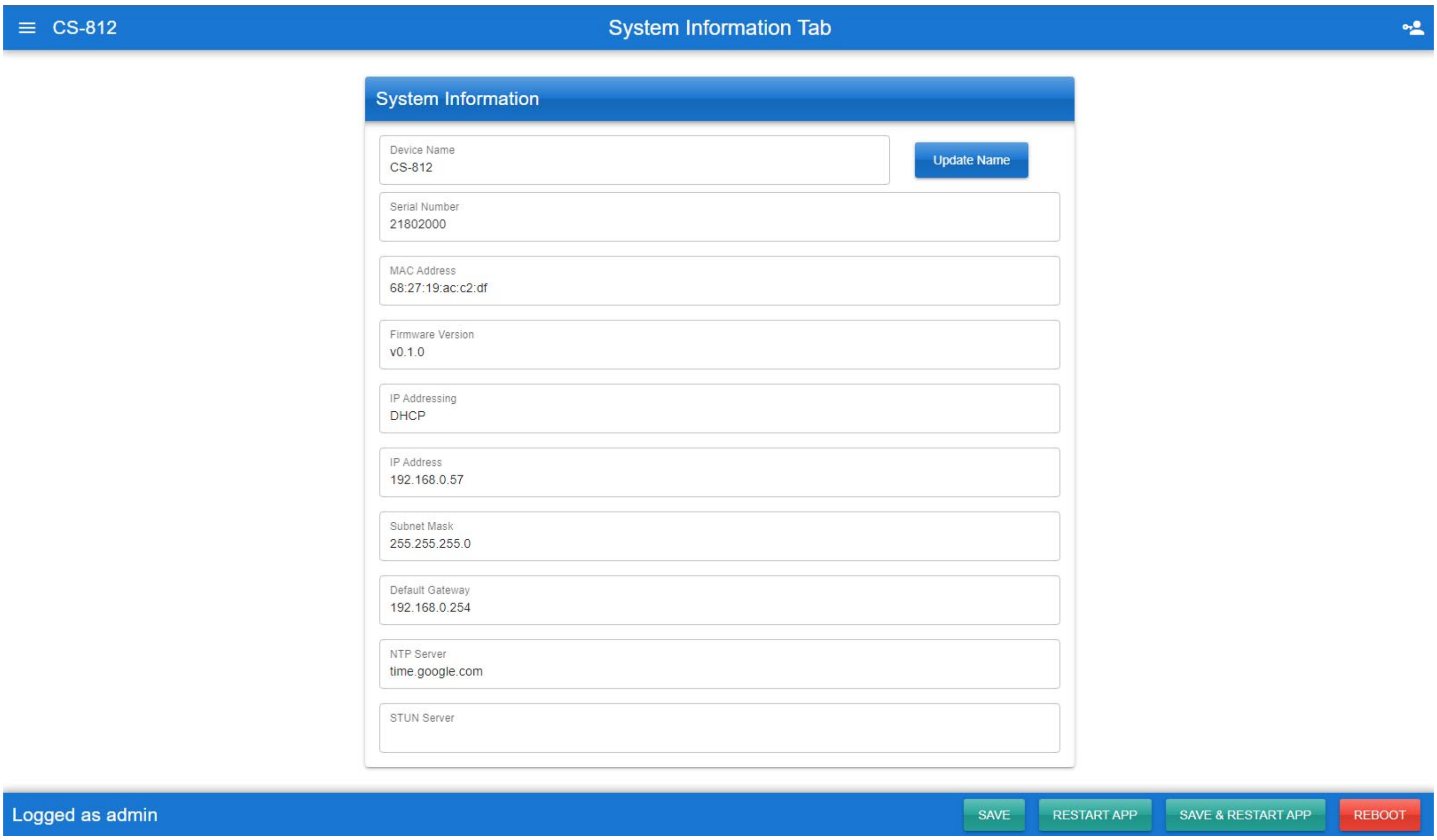


- **System Information** - This page displays device information and administrator definable fields for device name.
- **Networking** - This page allows the administrator to configure network settings for the device.
- **Channels** - This page allows the administrator to configure each of the lines/ Talk 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-8XX and mobile radios.
- **LMR** - This page allows the administrator to enable and set EIA Tones for each channel
- **Audio** - 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.
- **Update Firmware** - This page allows the administrator to upgrade device firmware as well as upload .wav file for the man down function alert.
- **Diagnostics** - This page allows the administrator access to numerous diagnostic tools such as logging and network packet capture.
- **About** - This page displays general information about the device and manufacturer.
- **Logout** - Lets the administrator log off from the web portal

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 is a Device Name field which the administrator may use to identify the specific device. See Figure 4 below.

Figure 4



- **Device Name –** Enter any alpha-numeric sequence to uniquely identify the device.

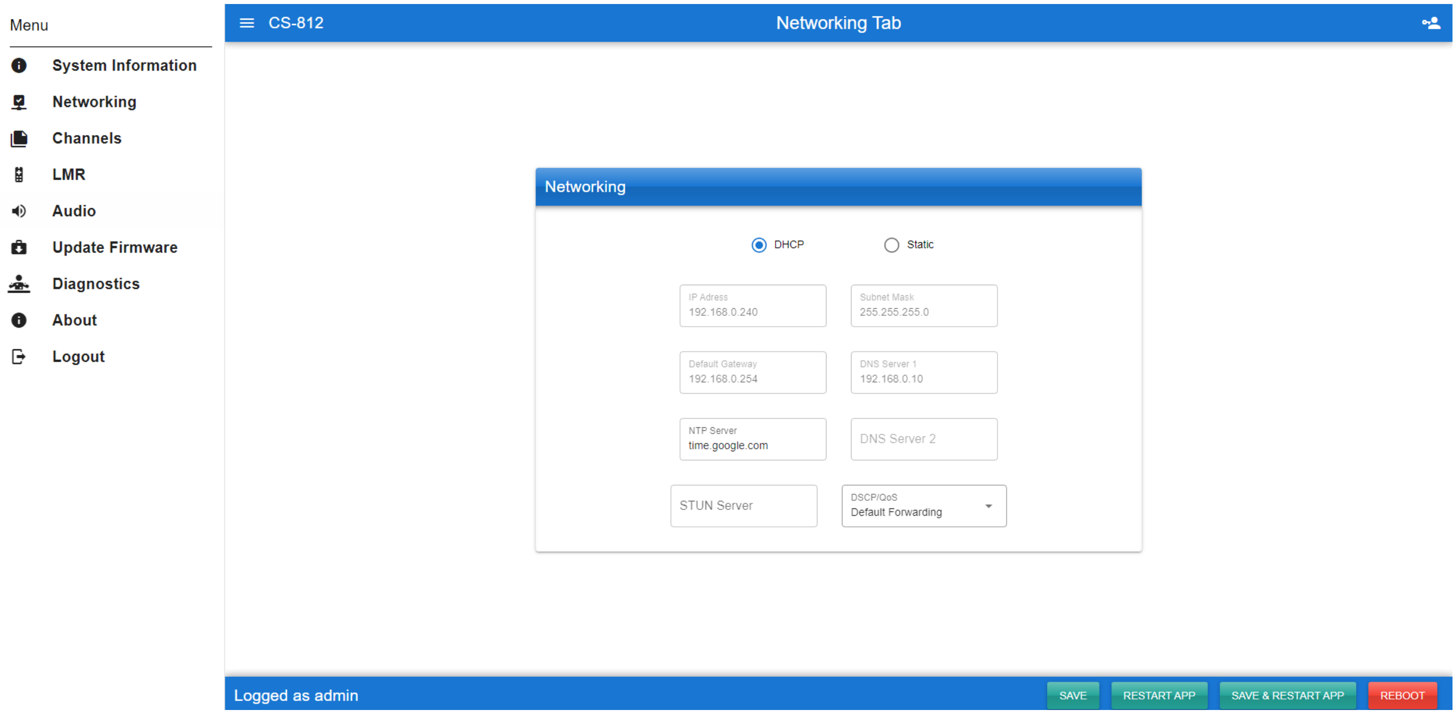
Upon making changes you must click **Save Configuration** if making additional changes on other pages or click **Save Configuration & Reload** to save & activate the changes. These buttons are located at the bottom of the page. See figure 4b.

Figure 4b

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 the 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-812's are shipped factory set to Default Forwarding.
- **Send RTP Keep Alive**- Allows administrator to enable or disable RTP keep Alive depending on requirements of SIP server/conference bridge. - REMOVE
- **Send RTCP Keep Alive** - Allows administrator to enable or disable RTCP keep Alive depending on application requirements.

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

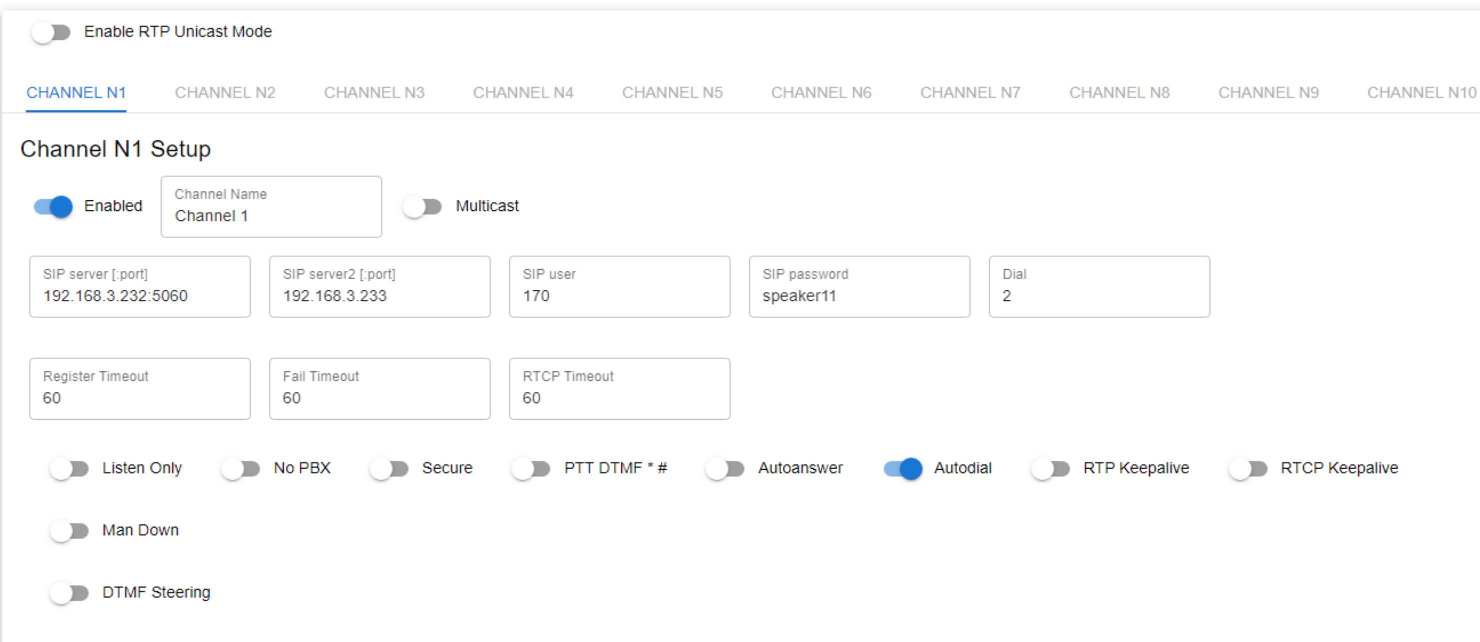
Rebooting is required after making device IP address changes.

2.6 Channels

This configuration page allows the administrator to configure the respective multicast/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. Each channel (N1-N12) is configured in a separate channel tab. See Figure 6 below for SIP channel more, Figure 8 for Multicast channel mode and Figure 9 for Unicast channel mode).

SIP Channel Mode

Figure 6

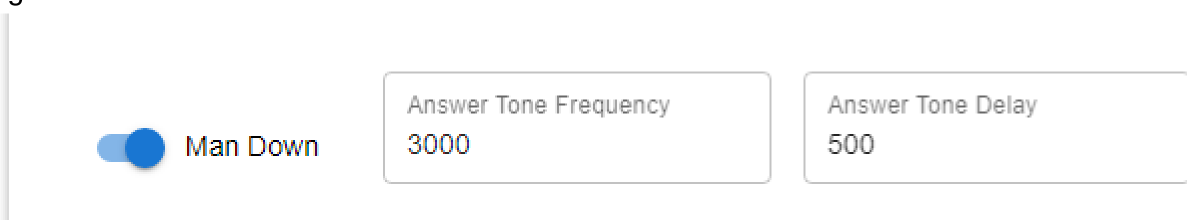


- **Enable RTP Unicast Mode**– Allows the administrator to switch between multicast and unicast channels. Unicast mode is explained further in Unicast Channel mode section on page 14.
- **Enabled -** Allows the administrator to turn on / off the respective channel/line.
- **Channel Name -** This field allows the administrator to assign a name to the respective channel. This name will be displayed on the device. It is recommended that the name does not exceed 10 characters.
- **Multicast -** Allows the administrator to switch the channel for Multicast/Unicast
- **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.
- **Number to Dial -** This field allows the administrator to assign an extension the device will dial.

Channels cont.

- **Reg/Fail Timeout** - These fields allow the administrator to set the time span, in seconds, for re-registration and registration fail of the channel to the SIP server.
  - **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.
  - **Listen Only** – Allows administrator to set a specific channel for listen only mode.
  - **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.'
  - **No PBX** – Allows for a channel to operate without registering to SIP server/proxy
  - **Secure** – Turns on TLS encryption for SIP handshaking and uses SRTP for audio encryption.
  - **PTT DTMF * #** - Enables device to output, "in-band" DTMF * when PTT button is pressed and DTMF # upon release of PTT.
  - **Auto-answer**- Selecting this mode instructs the device to automatically answer all incoming calls on this channel
  - **Autodial** - Selecting this mode instructs the device to automatically connect to the extension/number, as defined in the Number to Dial field immediately upon boot up.
  - **RTP Keep Alive** - Allows administrator to enable or disable RTP keep Alive depending on requirements of SIP server/conference bridge. - REMOVE
  - **RTCP Keep Alive** - Allows administrator to enable or disable RTCP keep Alive depending on application requirements.
  - **Man Down** - Allow administrator to enable/disable the Man Down Alert feature. This feature provides a visual and audible (alert tone file may be uploaded in the Update Firmware section) alert upon receiving DTMF "911" code. Two sets of relays on the back of the device provide contact closure upon receiving the man down code for third party alarming options (ex. loudspeaker, strobe light etc.). Operator may clear the code by pressing the "D" function button. Upon clearing of the alert by the operator an automatic Answer Tone is transmitted. Once Man Down feature is enabled an Answer Tone Frequency and Answer Tone Delay may be configured
    - **Answer Tone Frequency** – Allows the administrator to configure answer tone frequency (300Hz to 3400Hz range)
    - **Answer Tone Delay** - Allows the administrator to configure the delay between clearing of an alarm by the operator and automatic transmission of the Answer Tone (300ms-1500ms).

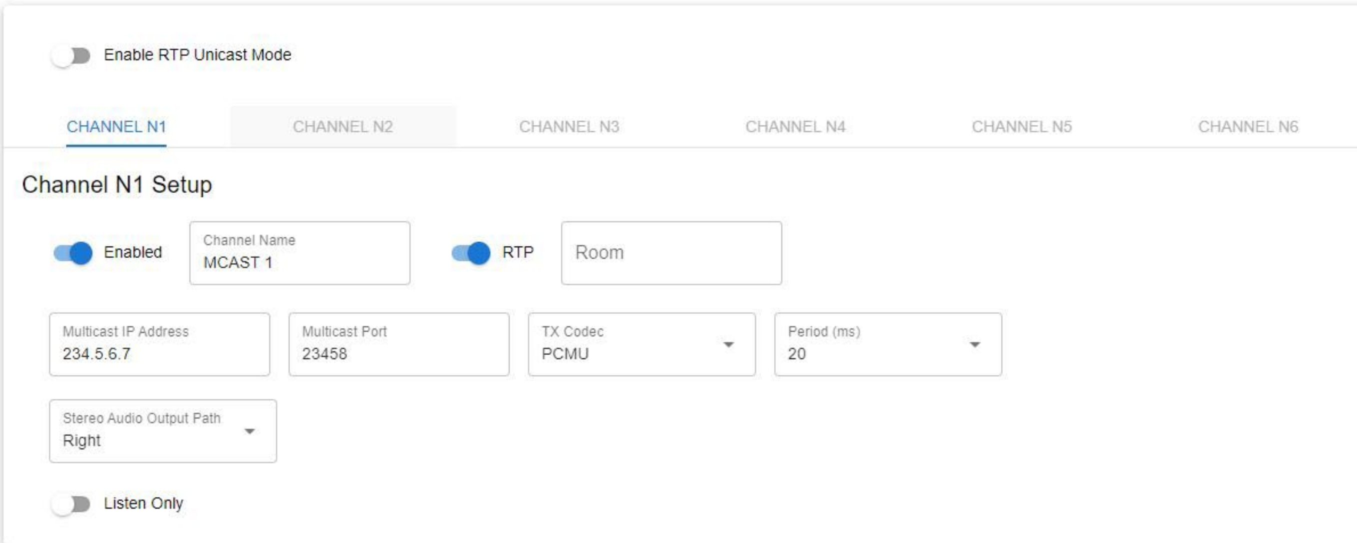
  Figure 7



  - **DTMF Steering Enable** - The CS-8XX can 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-8XX. This feature is further explained on page 15.

Multicast Channel Mode

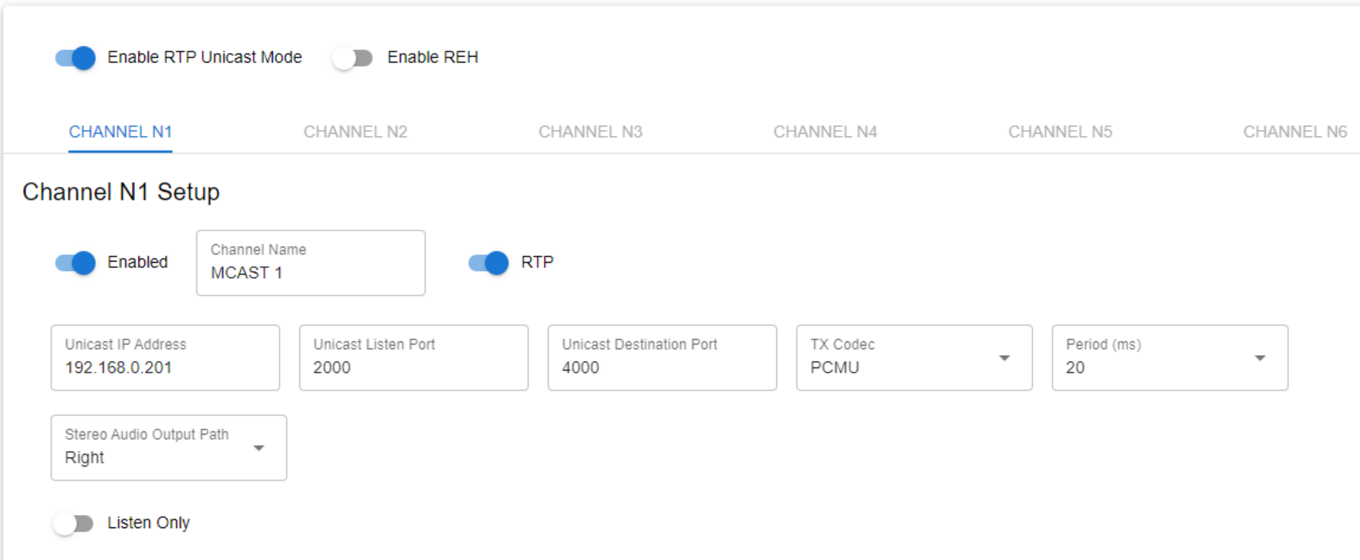
Figure 8



- **Enabled -** Allows the administrator to turn on / off the respective channel/line.
- **Channel Name -** This field allows the administrator to assign a name to the respective channel. This name will be displayed on the device. It is recommended that the name does not exceed 10 characters.
- **Room-** This is the **Cross Mute** (**X-Mute**) Option identifier. Units with the same **Room ID** will not transmit each other's audio out of their respective speakers. As an example [Room **5B**] identifies Room 5B and any station with "5B" in the Room block will be Cross Muted on that Talk Channel.
- **Multicast -** Allows the administrator to switch the channel for SIP
- **Multicast IP Address -** Allows the administrator to enter valid multicast addresses for the respective channels.
- **Multicast Port -** This field allows the administrator to assign ports associated with each channel multicast address. Note: ETC highly recommends using different and even number ports for each of the multicast channels. This ensures compatibility with other 3rd party VoIP applications.
- **TX Codec -** This field allows the administrator to assign the CODEC to be used by each channel. Selections are: PCMU (G.711 uLaw), PCMA (G.711 ALaw) & G.729
- **Period (ms) -** Allows the administrator to choose packet size (10ms, 20ms, 30ms or 100ms).
- **Stereo Audio Output Path -** (STEREO AUDIO OUTPUT MUST BE ENABLED) Administrator is able to assign active channels to either the left side or the right side of external stereo speakers (selected vs. unselected channels).
- **Listen Only -** Allows administrator to set a specific channel for listen only mode.

Unicast Channel Mode

Figure 9



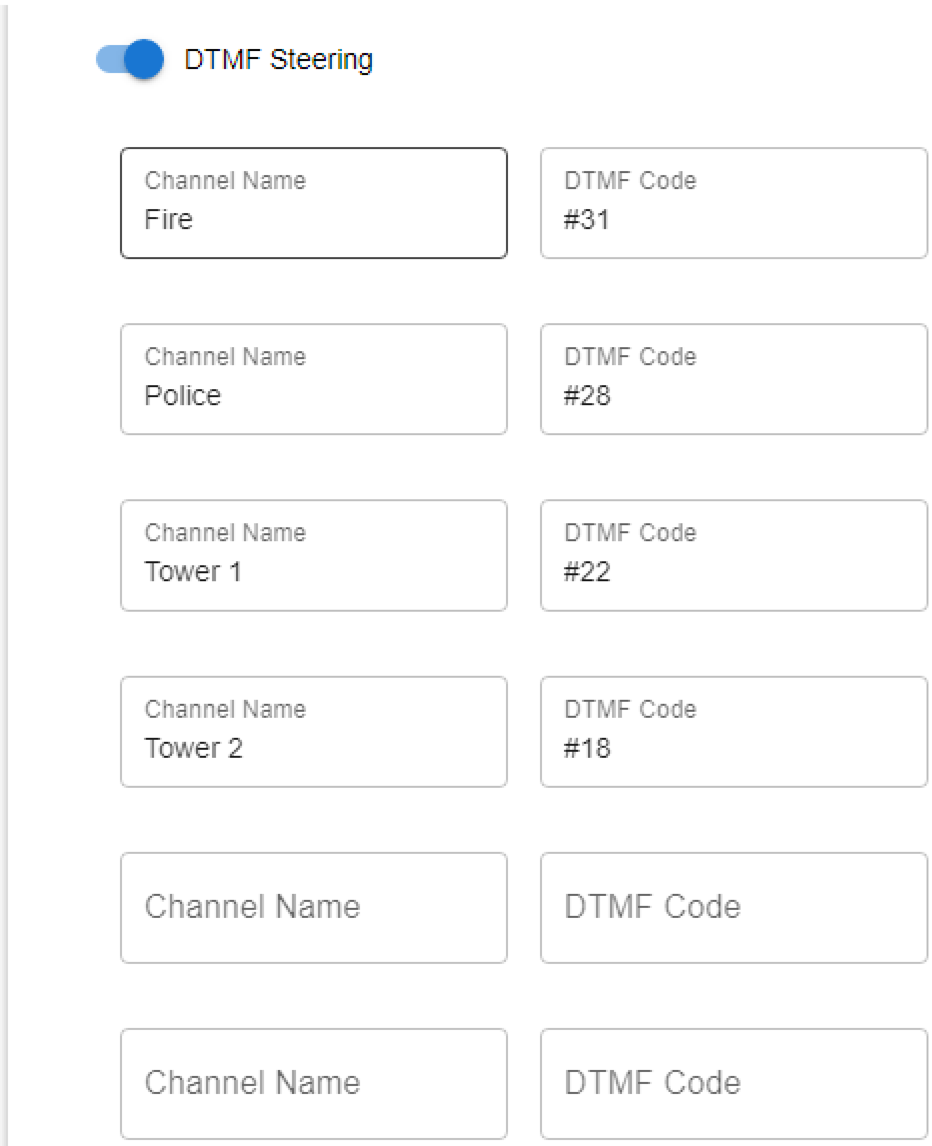
- **Enabled -** Allows the administrator to turn on / off the respective channel/line.
- **Channel Name -** This field allows the administrator to assign a name to the respective channel. This name will be displayed on the device. It is recommended that the name does not exceed 10 characters.
- **Multicast -** Allows the administrator to switch the channel for SIP
- **Unicast IP Address -** Allows the administrator to enter destination IP addresses for the respective channel. Channels can be configured to use the same IP address if needed.
- **Unicast Destination Port -** Allows the administrator to enter the destination port of the channel on which unicast traffic will be sent.
- **Unicast Listen Port -** Allows the administrator to enter the listen port of the channel on which unicast traffic will be received.
- **TX Codec -** This field allows the administrator to assign the CODEC to be used by each channel. Selections are: PCMU (G.711 uLaw), PCMA (G.711 ALaw) & G.729
- **Period (ms) -** Allows the administrator to choose packet size (10ms, 20ms, 30ms or 100ms)
- **Stereo Audio Output Path -** (STEREO AUDIO OUTPUT MUST BE ENABLED) Administrator is able to assign active channels to either the left side or the right side of external stereo speakers (selected vs. unselected channels).
- **Listen Only -** Allows administrator to set a specific channel for listen only mode.

DTMF STEERING

DTMF Steering – The DTMF Steering section allows an administrator to configure multiple radio ID's and DTMF codes for each channel as indicated in Figure 8 below. These Radio ID's will be indicated on the CS-8XX 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 "A" function button on front of the CS-8XX which will trigger any channels configured for DTMF Steering to flash, rotate the respective channel volume knob to scroll through Radio IDs until desired ID is found then press "A" function button 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.

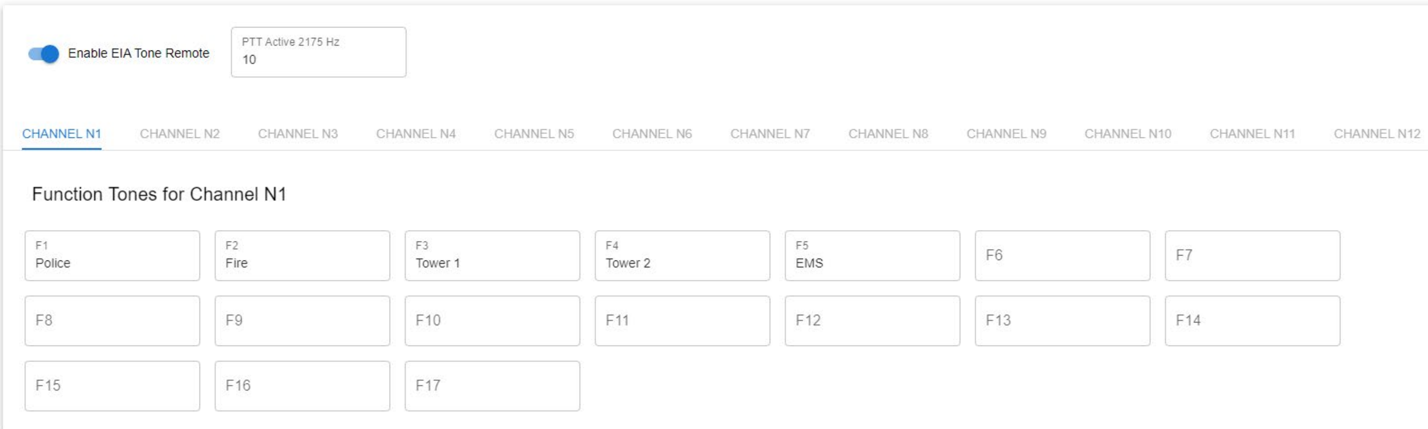
Figure 10



2.7 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-8XX

Figure 11



- **EIA Tone Remote Enable** – Checking this box enables the EIA Tone Remote feature within the CS-8XX.
- **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-8XX is pressed the device will send a 2175Hz HLGT tone at 0dB for the duration specified.
- **Function Tones F1-F17** – These fields allows the administrator to set a name for each of the EIA tone frequencies. These names will be displayed on the respective channels of the CS-812. The user will press and release the "A" function button then press the respective channel PTT button to scroll through any configured tones then press and release the "A" function 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.

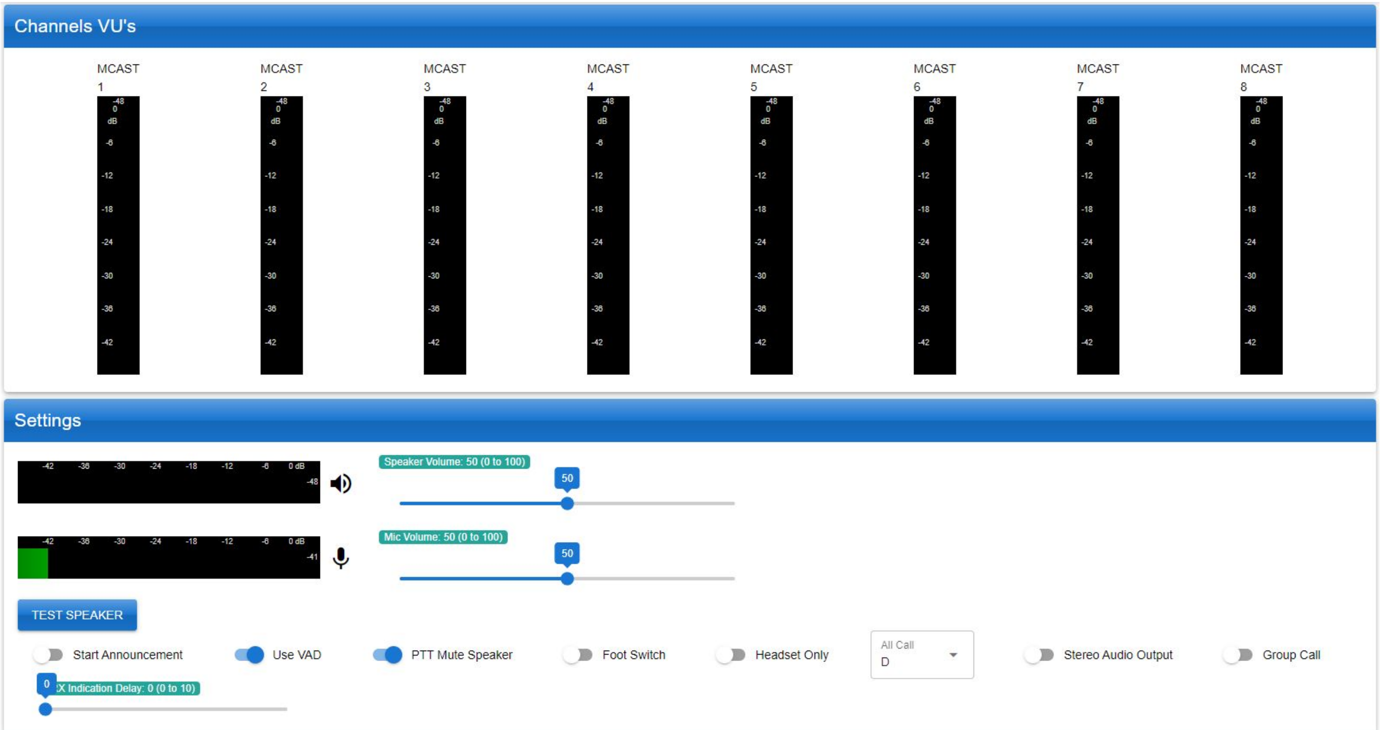
The following table is a quick reference of the respective tones:

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 and other feature settings.

Figure 12



- **Channel VU's -** Allows the administrator to look at channel activity of all 12 or 8 channels.
- **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 the device is working properly. Master volume level and Mic Volume are default to set to 80.
- **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.
- **Use VAD** - 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, the device will continuously produce RTP audio packets when PTT is pressed even if the user is still speaking.
- **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.
- **Foot Switch** - Checking this box enables the use of a foot switch to activate PTT function on a selected channel. In this mode the buttons on the front panel are used to select a channel for which the foot switch will activate the PTT.
- **Headset Only** - Checking this box enables the use of a headset without a PTT button. All PTT operation is done with the PTT knobs.
- **All-Call or All-Stop** - Configuring an All-Call or All-Stop function button requires the user to choose a function button (A-D) so one button which can simultaneously Transmit on all active channels. In the All-Call Selection Box use the Drop-down menu button to select Function Button "A" through "D". The selected Function Button is now set for All-Call or All-Stop.

- **Stereo Audio Output** – (ONLY APPLICABLE TO UNITS WITH STEREO OUT PORT) Checking this box enables the use of the stereo out port. Administrator as well as the user can assign active channels to either the left side or the right side of external stereo speakers (selected vs. unselected channels). Pressing the B&C function buttons simultaneously opens Channel Assignment Mode which allows the user to change channel assignments to either left or right by rotating each channel PTT knob. Simultaneously pressing functions buttons B&C again will save the assignments and exit Channel Assignment Mode.
- **Group Call –** Enabling this feature allows the user to transmit ("Broadcast") on 2 or more channels at the same time. User presses the "A" function button then presses each respective channel button to select channels for 'simulcast' which will be indicated by a check mark next to each selected channel name. When selecting channels, the user simply presses the volume knob/PTT button of one of the selected channels to transmit on all selected channels. Pressing "A" function button again will exit the Group Select mode.
- **RX Indication Delay (seconds) –** Extends incoming audio indication by the number of configured seconds.

2.10 Handset/Headset Operation

The CS-8XX console has the ability to connect either an ETC handset (TH-3) or a Mono headset through a RJ-25 jack on the back of the device. Once a handset is connected the unit will show a “H” icon in the status bar to indicate the unit is now in Handset or Headset mode.

To transmit on a specific channel, the operator needs to first select a channel by pressing the respective Volume Control/PTT button (aka VCB). This channel will have a visual indication in the form of a checkbox.

Users can now simply press the PTT button on the ETC Handset or the Headset Belt Pack PTT button to transmit on the selected channel. The VCB can also be used to transmit on any channel including the selected on using the gooseneck microphone.

PLEASE NOTE:

Incoming audio of the selected channel will go to the handset only for privacy of the conversation. All other channels will still be transmitted out of the built-in speaker.

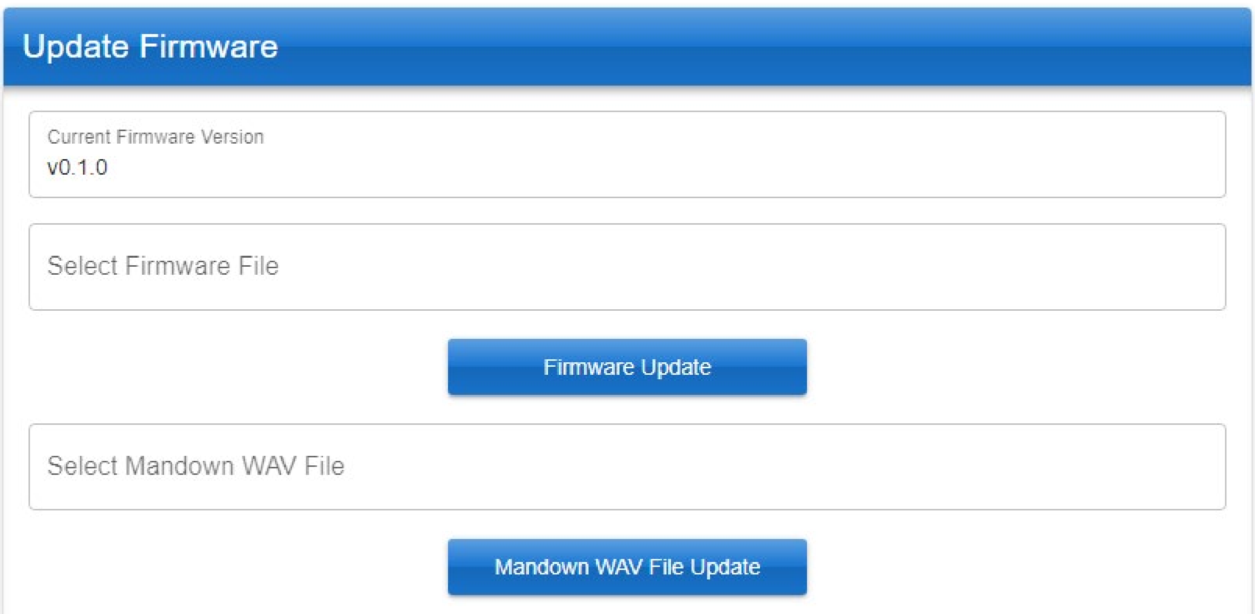
Shall the user want to receive all incoming audio on the built-in speaker the “C” function button may be pressed to disable the privacy function (selected channel incoming audio being routed to the Handset or Mono Headset only). H->S symbol will be displayed on the status bar to indicate all audio is being routed to the built-in speaker.

The operator can press the “C” function button to reengage/toggle standard ETC Handset or Mono Headset operation with privacy mode.

2.11 Update Firmware

The Update Firmware page allows an administrator to update firmware easily & quickly 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 13



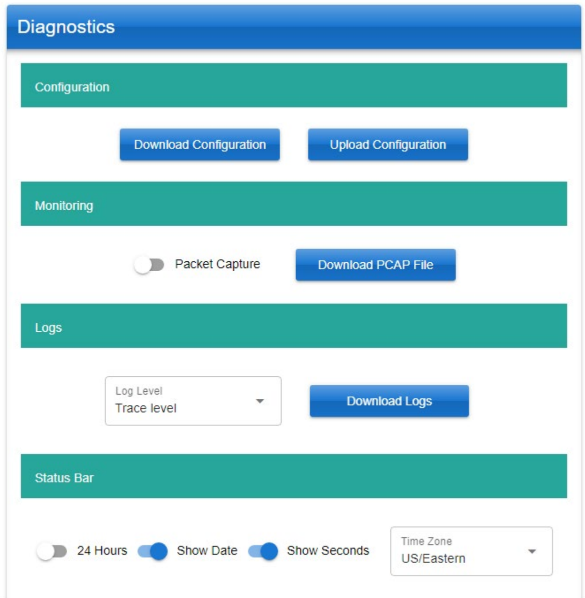
NOTE: AFTER UPDATING FIRMWARE, YOU MUST (Shift F5) REFRESH YOUR BROWSER WINDOW FOR NEW FEATURES TO BE DISPLAYED. Most browsers will cache previous pages of the device and therefore a refresh must be performed after a firmware update.

Select Man down WAV File – Allows the administrator to upload a .wav file to be used as the Man Down Audible Alert Tone.

2.12 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 14



- **Download Configuration** - Allows administrator to download the devices configuration information to a PC or server. File format is .conf and is readable with Notepad or equivalent.
- **Upload Configuration** - Uploads the selected configuration file
- **Browse** - Allows administrator to find & select CS-8XX configuration file for upload.
- **Config File Upload** - Uploads the selected configuration file.
- **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.
- **Log Level** - Allows administrator to set the logging level to capture event information for troubleshooting purposes. Setting range is 1 (Error) to 5 (Trace). ETC recommends leaving devices at level 2-3 unless more in depth logging is requested by ETC Support.
- **Download Logs** - 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.
- **Status Bar Settings** - Allows the administrator to adjust numerous options of the status bar
    - **24 Hours** - Switches Time shown on the CS-8XX between a 12-hour and 24-hour format.
    - **Show Date** - Allows for the date to be shown/not shown
    - **Show Seconds** - Allows for seconds to be shown/not shown
    - **Time Zone** - Allows for setting the proper time zone

3.0 Frequently Asked Questions

Q: What is the IP address of the device?

A: When the device boots up the IP, Mask , Gateway, and MAC address are presented briefly on the display. You can also press buttons 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: How do I change the device IP?

A: Once the device has booted up and you have identified its IP address, open up 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?

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 the channel's label?

A: Labels can be changed on each channel tab.

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 the user pressing PTT button for respective channel?

A: On the 'Audio Settings' web page is there any 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? Check status of the channel

Q: Audio received on device is choppy/garbled

A: Please check if the codec chosen by the SIP server is compatible with the CS-812 device. The list of compatible codecs is in the Specifications section of the CS-812 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
- (12) or (8) SIP/Multicast/Unicast Lines

Call Types
- SIP, hoot conferencing
- SIP Private line (ARD/MRD)
- Multicast IP broadcast
- Unicast

Signaling
- SIP
- EIA Tones
- DTMF

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

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

Dimensions
- Width - 7" /
- Depth – 11.5"
- Height – 9.5"
- Weight – 7.7 lbs.
- 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
- 5 VDC, 2 A , external power supply

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-115P
- Handset, PN - TH-3
- Dongle headset/footswitch combo, PN–CDA-FPHW251
- Dongle, Headset no PTT, PN – QD-RJ25

5.0 CS-8XX 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 it is 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 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 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.

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