Extron Electronics
INTERFACING, SWITCHING AND CONTROL

DMP 128 Plus C V
DMP 128 Plus C V AT

Interactive Intelligence Configuration Guide

REVISION: 1.0.1
DATE: MARCH 7TH 2018

Revision Log

Date	Version	Notes
Feb 9th 2018	1.0	First Release: Applies to Firmware Version **1.01.0007.004**
Mar 7th 2018	1.0.1	Layout and language changes for emphasis

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1.0 Introduction

This document provides essential instructions for registering DMP 128 Plus C V (AT) VoIP lines to an Interactive Intelligence PBX system running **CIC version 2017 R1** or later.

DMP 128 Plus C V / C V AT must be running firmware Version **1.01.0007-004** or later.

2.0 Configuring Interactive Intelligence for DMP 128 Plus VoIP Registration

- VoIP functionality within the DMP 128 Plus is built around the Session Initiation Protocol (SIP) signaling system, as defined in RFC 3261. The Interactive Intelligence platform must be licensed to allow the addition of generic basic third-party SIP endpoints before any line registration can take place.

- The DMP 128 Plus requires that the **Early Offer** call initialization model be used, referred to as **Normal Media Timing** within Interactive Intelligence systems.

- It is recommended that a **static IP address** is assigned to the network interface used for VoIP traffic on the DMP 128 Plus.

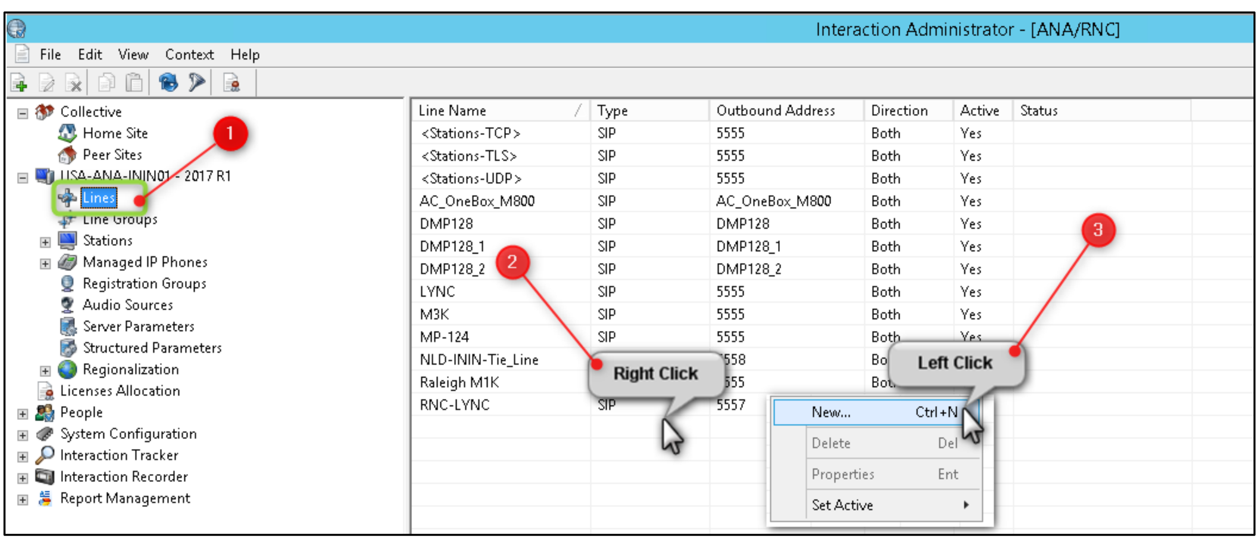
2.1 Create a New Line

Registration of a DMP 128 VoIP device requires the creation of a new line within the **Interaction Administrator** platform. Start the application with administrator credentials.

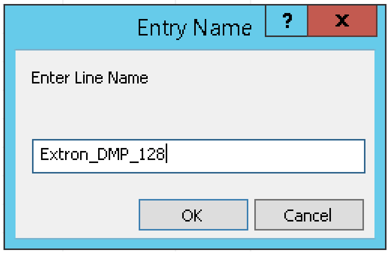
1) Click on the **Lines** [1] item on the left-hand side of the screen.

2) Right-click on the right-hand list portion of the screen [2].

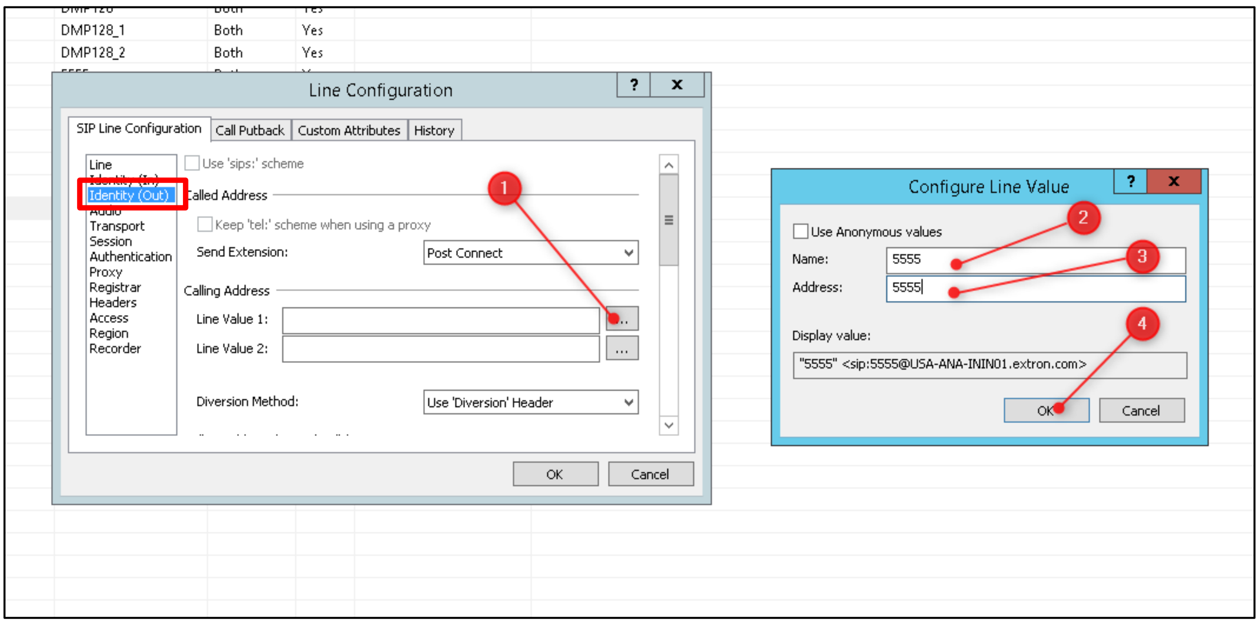
3) Select **New** [3].



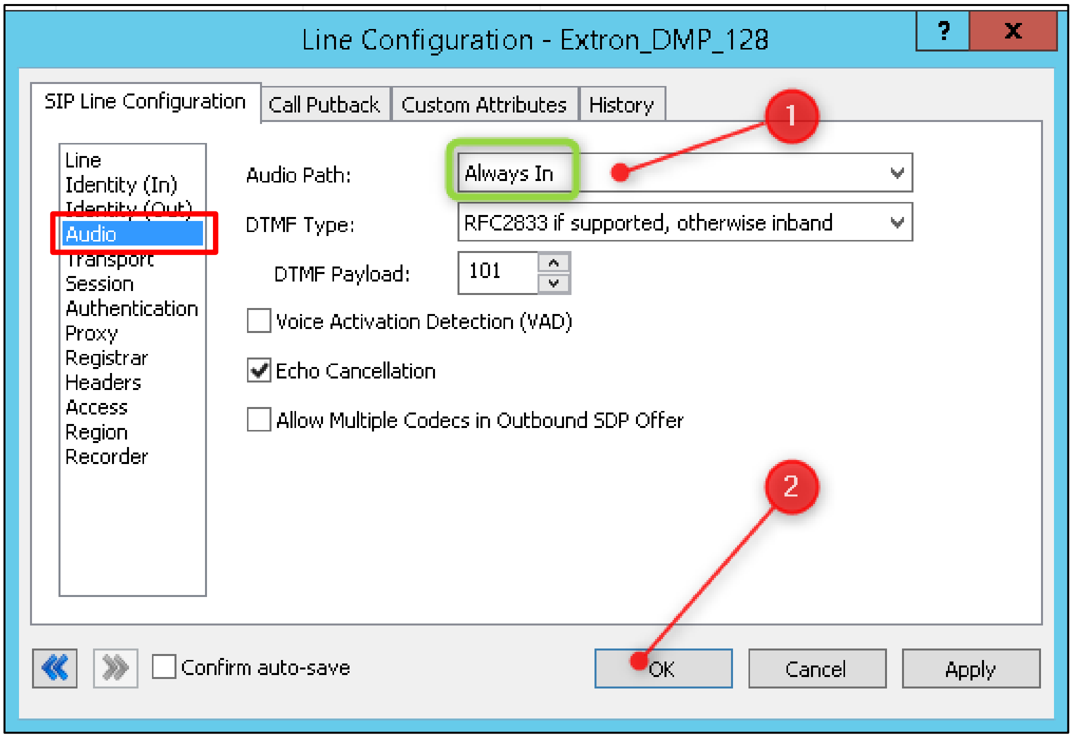
4) Enter a name for the line and click on **OK**.



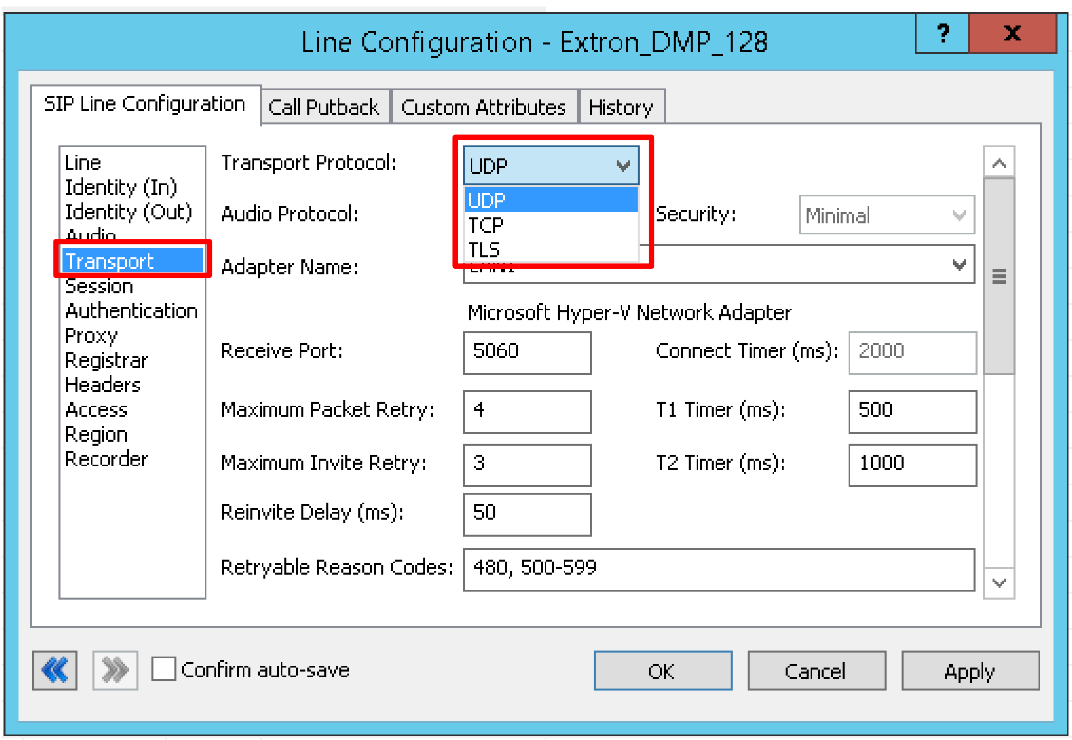
5) Click on **Identity (OUT)** [1] and edit Line Value fields as required [2] and [3]. Click OK [4].



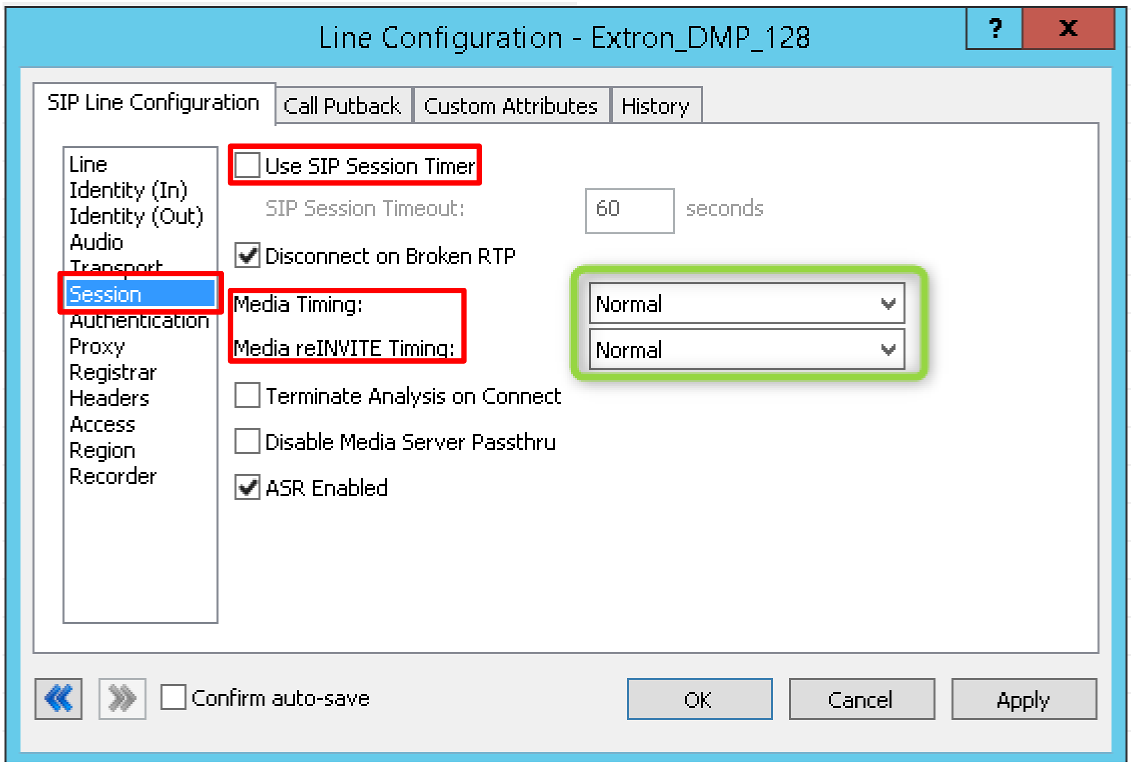
6) Click on **Audio** and edit the fields as shown below [1]. Check the **Allow Multiple Codecs** box if more than one codec will be assigned to the DMP 128 Plus. Click OK [2].



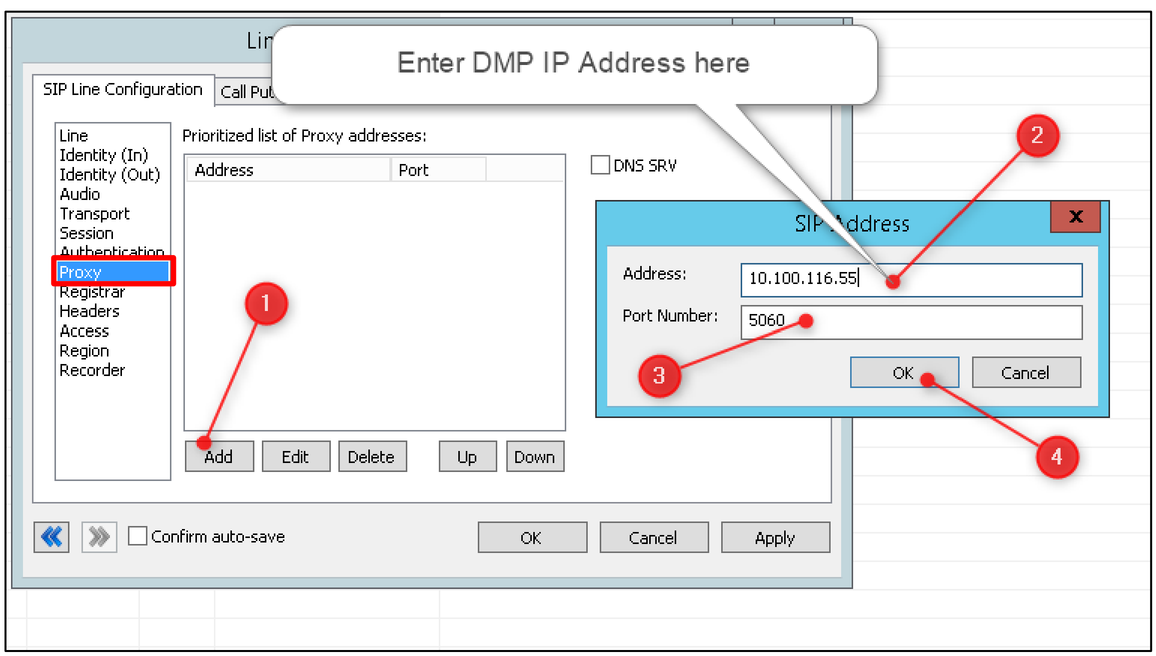
7) Set the signaling **Transport Protocol** as required. The default is UDP.



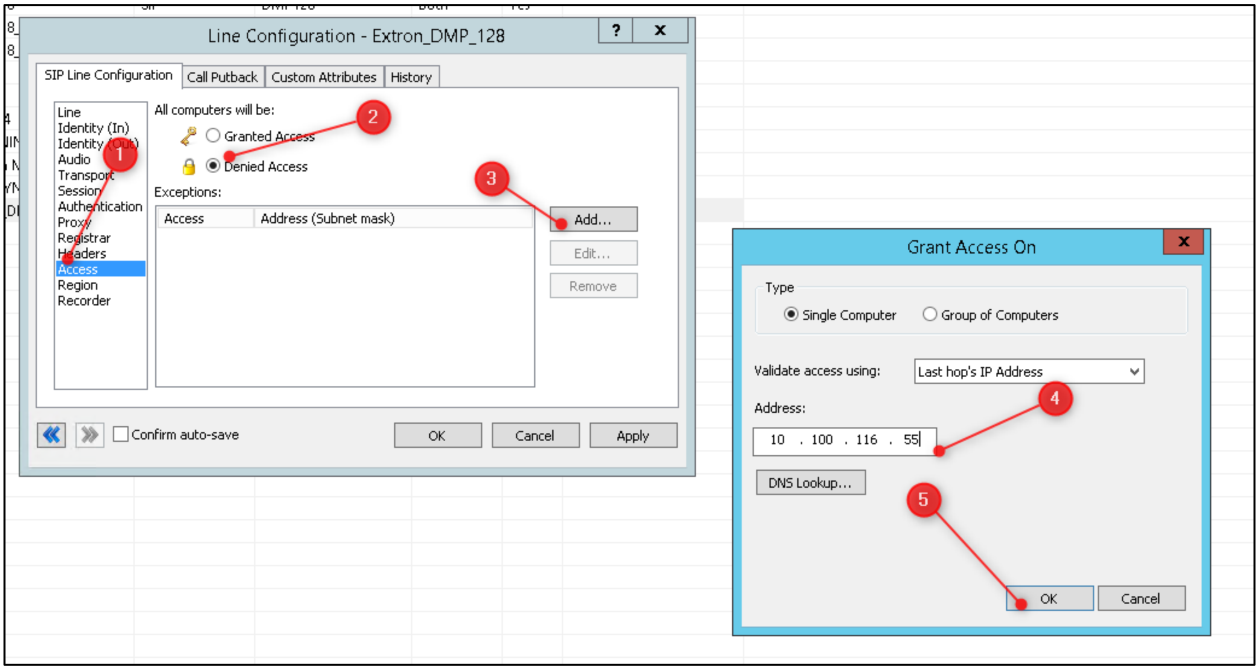
8) Click on **Session** and change **Media Timing** and **Media reINVITE Timing** to **Normal**. Uncheck the **Use SIP Session Timer** box.



9) Click on **Proxy** followed by **Add** [1]. Enter the IP address of the DMP 128 Plus [2] and the port number [3] being used (the default port for UDP and TCP is 5060). Click **OK** [4].



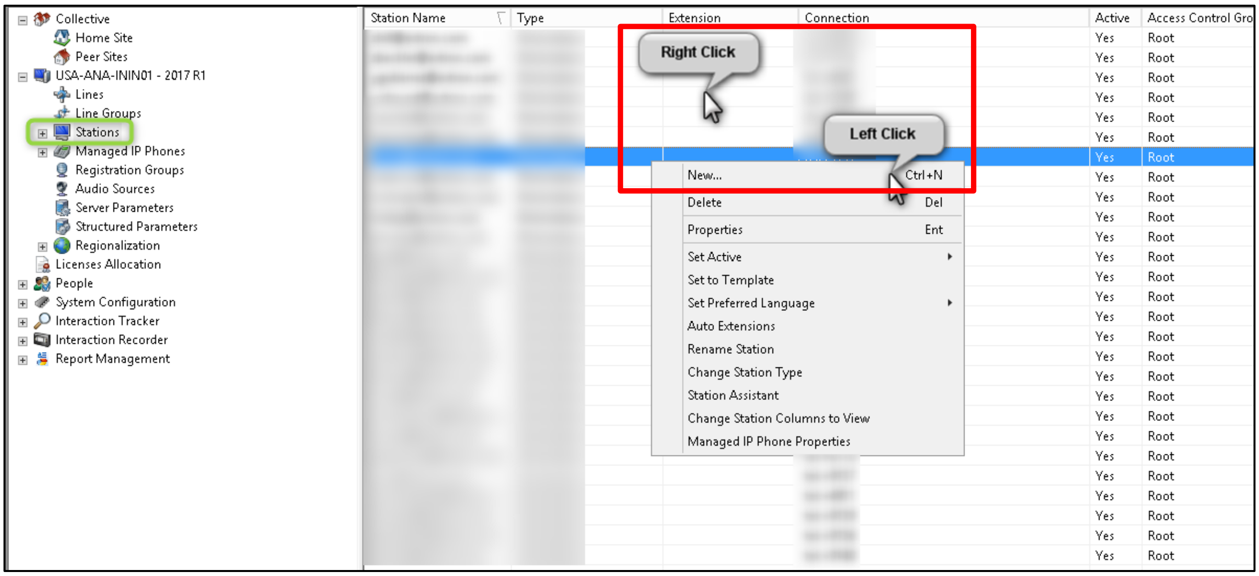
10) Click on **Access** [1] followed by the **Denied Access** [2] radio button. Add the DMP 128 Plus as an exception by clicking on the **Add** [3] button and entering its IP address [4], followed by **OK** [5].



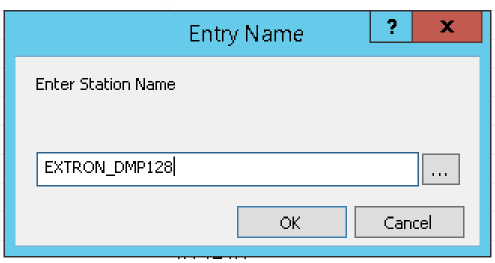
2.2 Create a New Station

Add a New Station for the DMP 128 Plus.

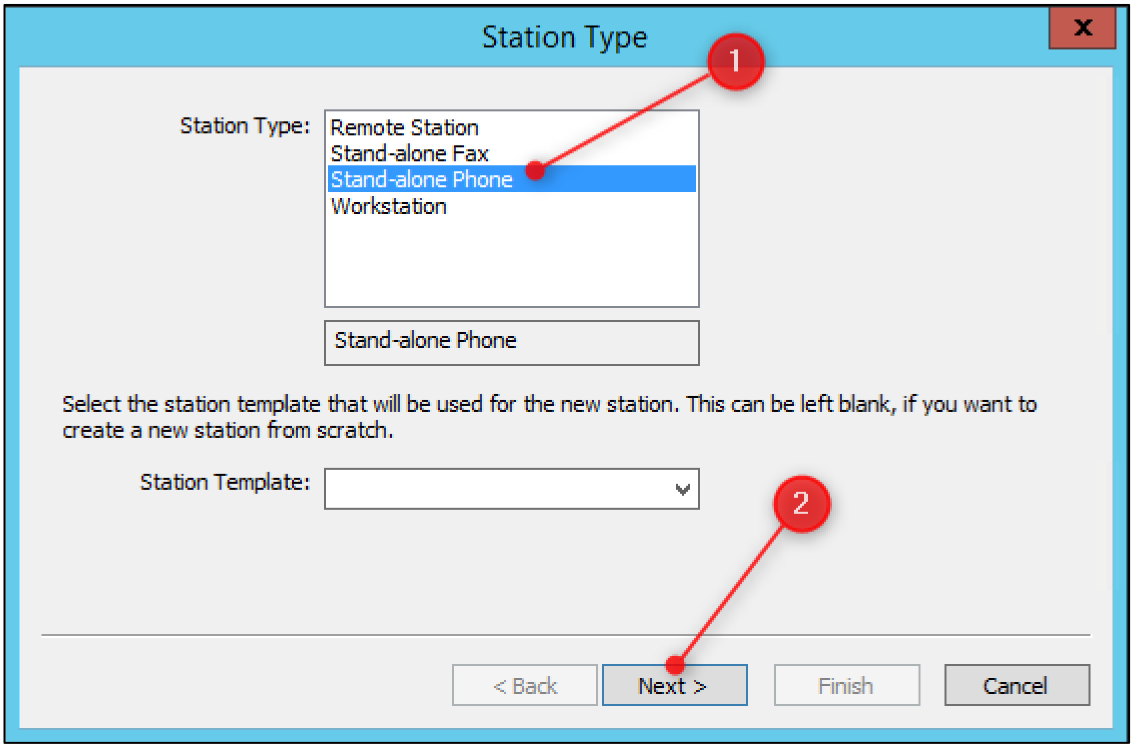
1) Click on the **Stations** container, right-click in the list area, then select **New**.



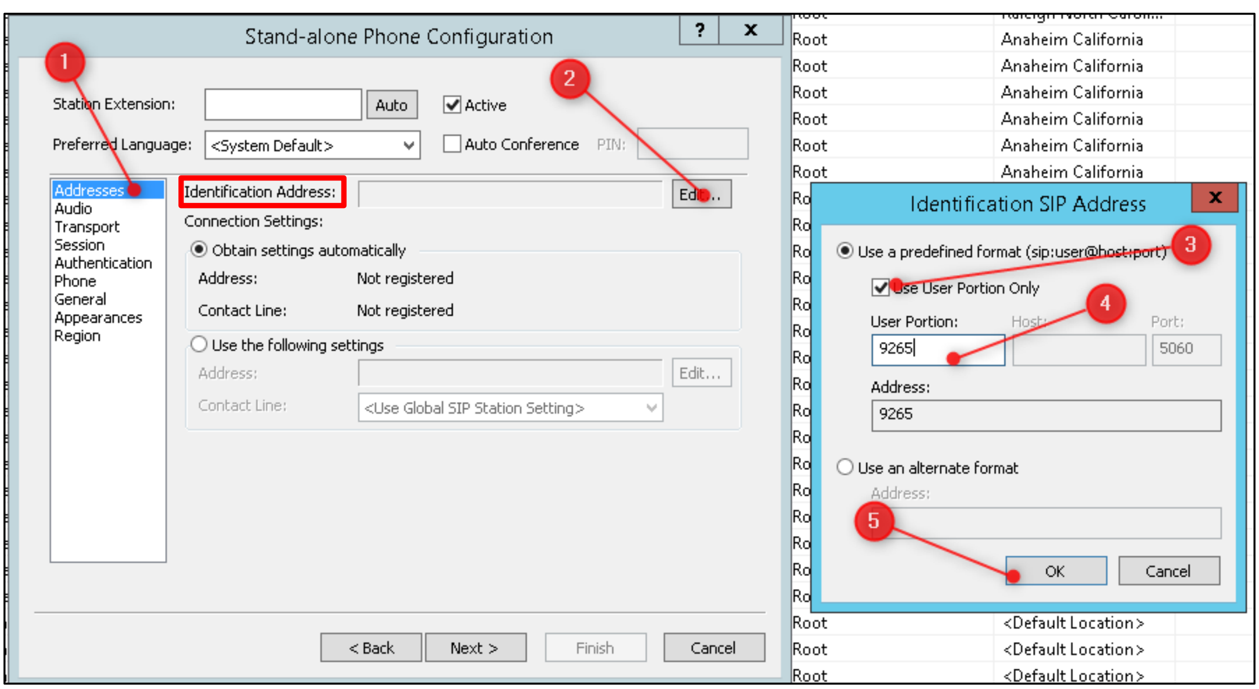
2) Enter a name for the new station and click **OK**.



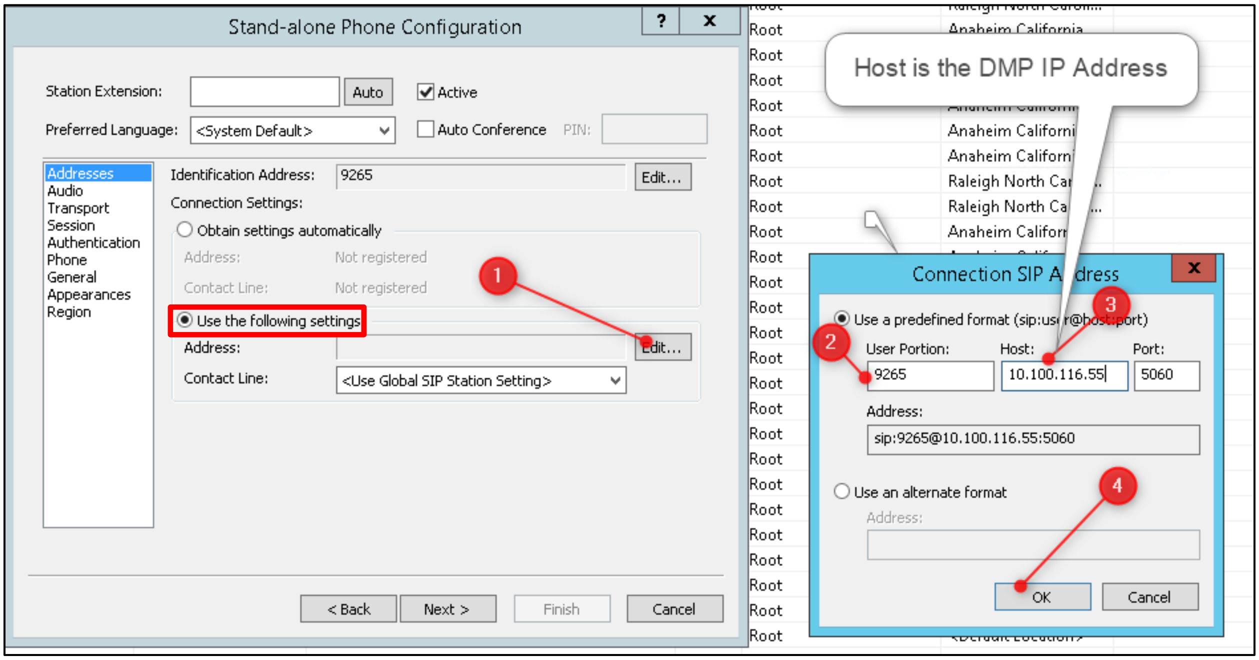
3) Select **Stand-alone Phone** [1] as the Station Type followed by **Next** [2].



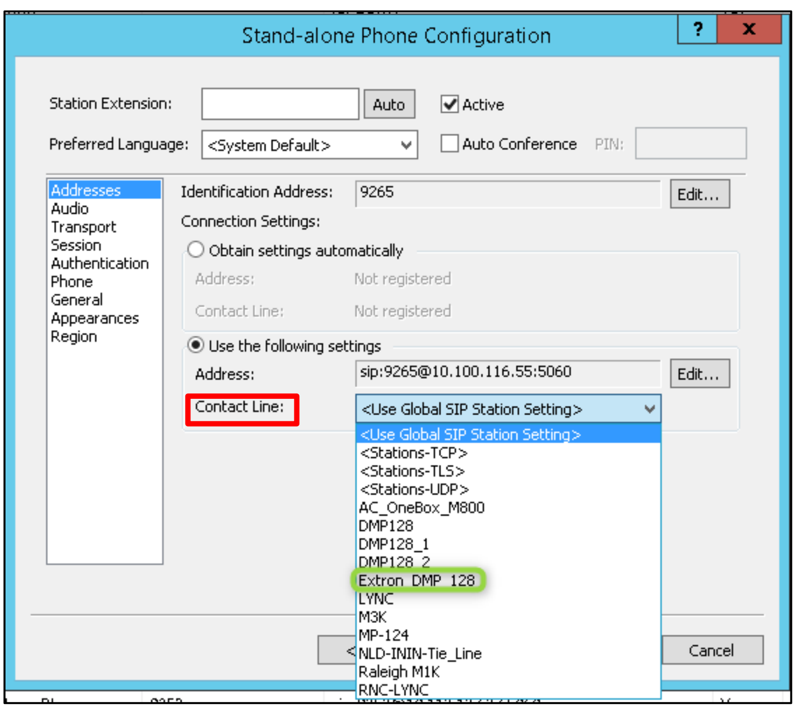
4) Click **Edit** [2] from the **Identification Address** section of the **Addresses** [1] section. Select **Use User Portion Only** [3] and enter the extension number [4] assigned to the DMP 128 Plus, followed by **OK** [5].



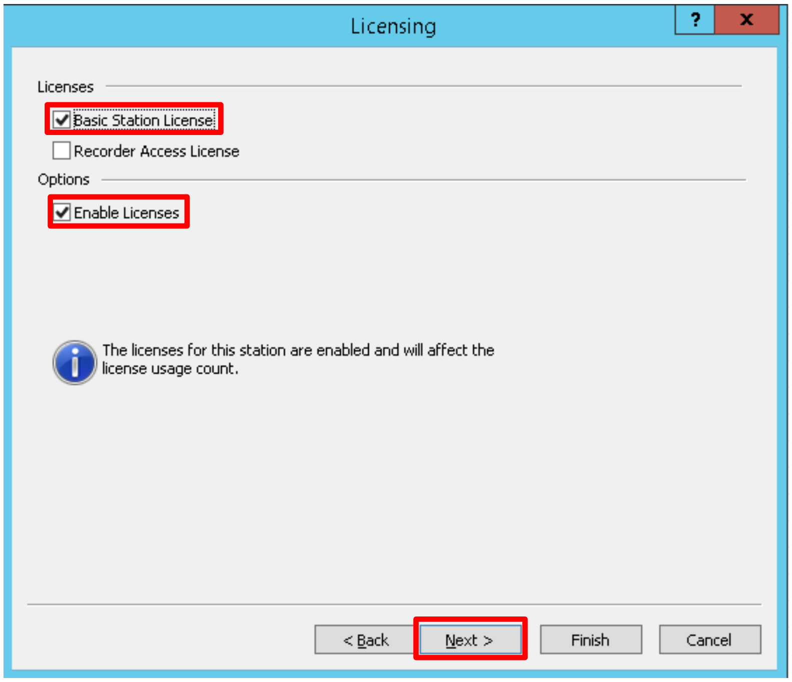
5) Select the **Use the following settings** radio button followed by **Edit** [1]. Enter the extension number assigned to the DMP 128 Plus in the **User Portion** [2] field, followed by the IP address and SIP port number in the **Host** and **Port** fields [3], respectively. Click **OK** [4].



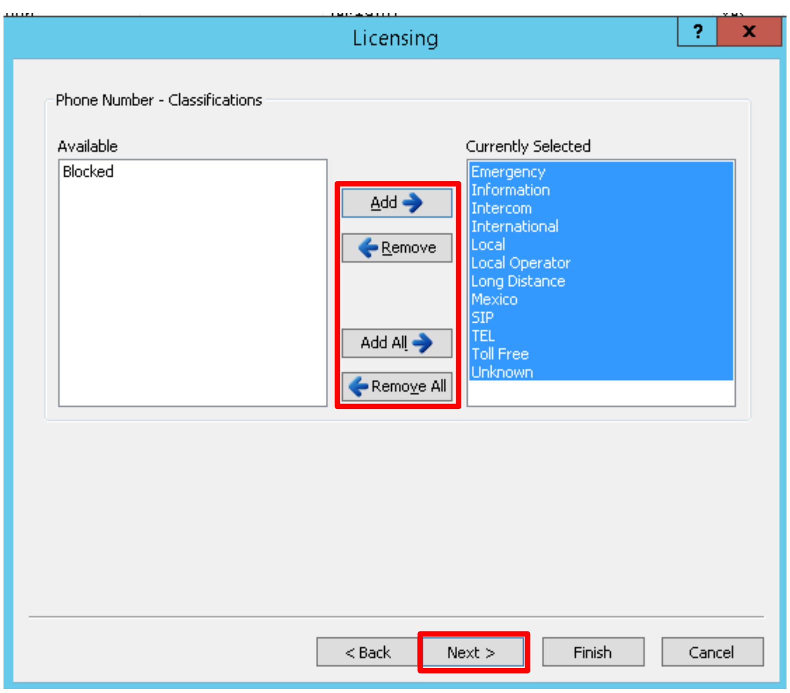
6) From the **Contact Line** drop-down box, select the line created in [**Section 2.1**] followed by **Next**.



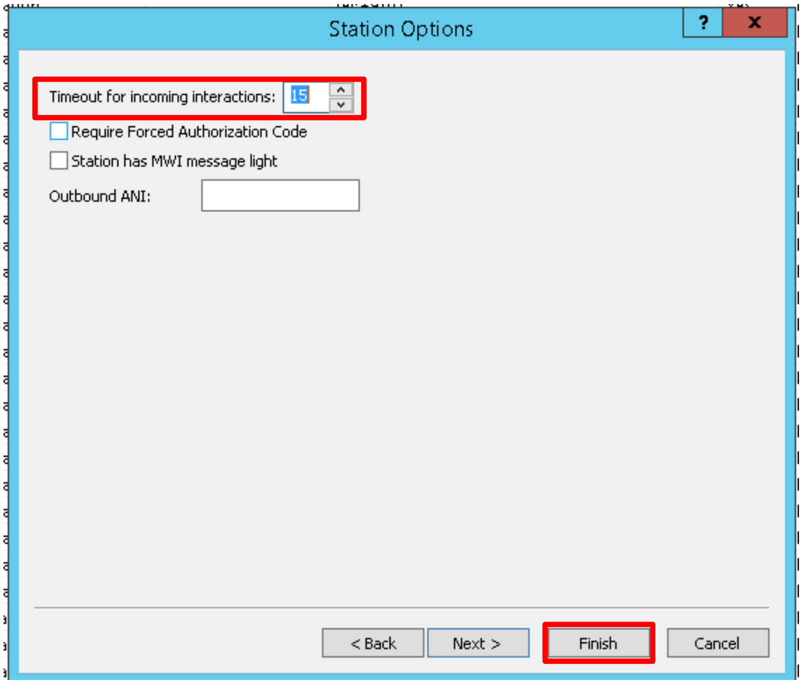
7) Select the **Basic Station License** and **Enable Licenses** checkboxes followed by **Next**.



8) Add any required **Phone Number Classifications** to the **Currently Selected** panel followed by **Next**.



9) Adjust the **Timeout for incoming connections** timer to the desired value and click **Finish.**



3.0 Configuring DMP 128 Plus C V (AT) VoIP Lines

VoIP configuration of the DMP 128 Plus is handled exclusively through a web interface, served from the device itself. The VoIP landing page is accessed through an address of the format -

http://192.168.254.254/www/voip.html

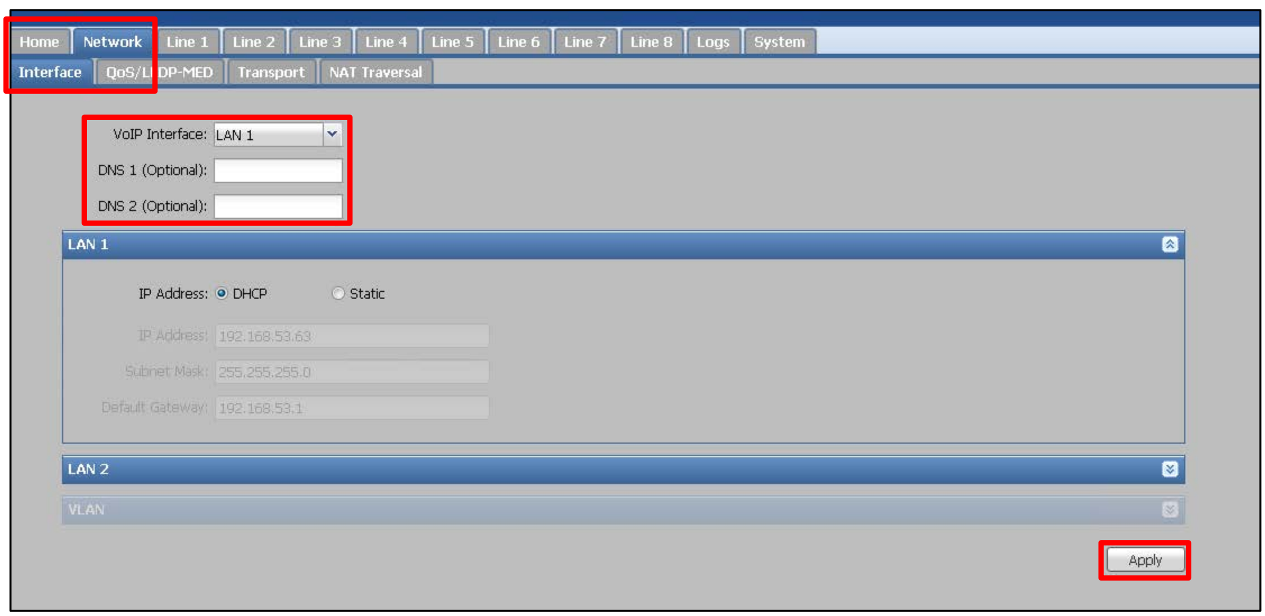
- where 192.168.254.254 in this example is the default IP address of the DMP 128 Plus device.

Up to 8 lines may be configured. Note that each line intended for use will require a unique extension to be specified as part of the configuration process in Section 2.0.

3.1 Network Interface Configuration

Clicking on the **Network** followed by **Interface** tabs allows changes to be made to the desired network interface on the DMP 128 Plus; either LAN1 or LAN2 may be used for VoIP. VLAN tagging is available on either interface if required. Up to two DNS entries may be manually specified.

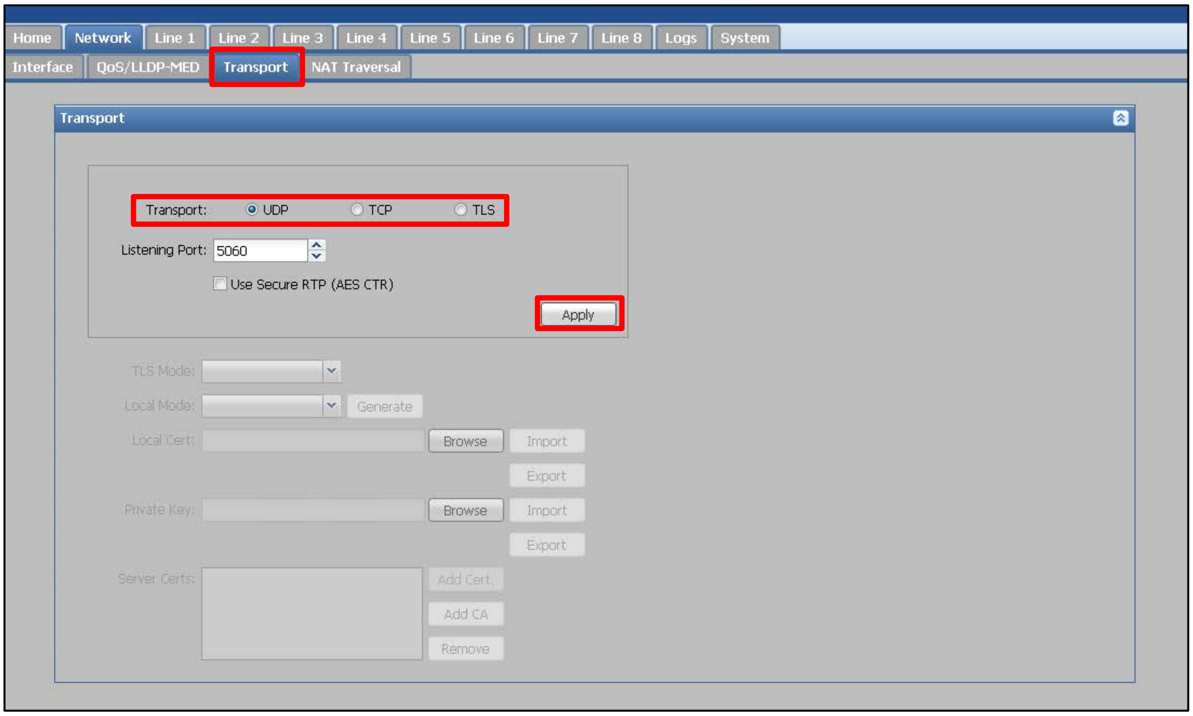
Click **Apply** after making any changes to restart the networking services on the device.



3.2 Transport Configuration

Click on the **Transport** tab to access signaling transport configuration. Set the transport to either UDP or TCP per Section 2.1. The default transport type for the DMP is UDP.

In the event that changes need to be made, click **Apply** to commit any adjustments to the device.



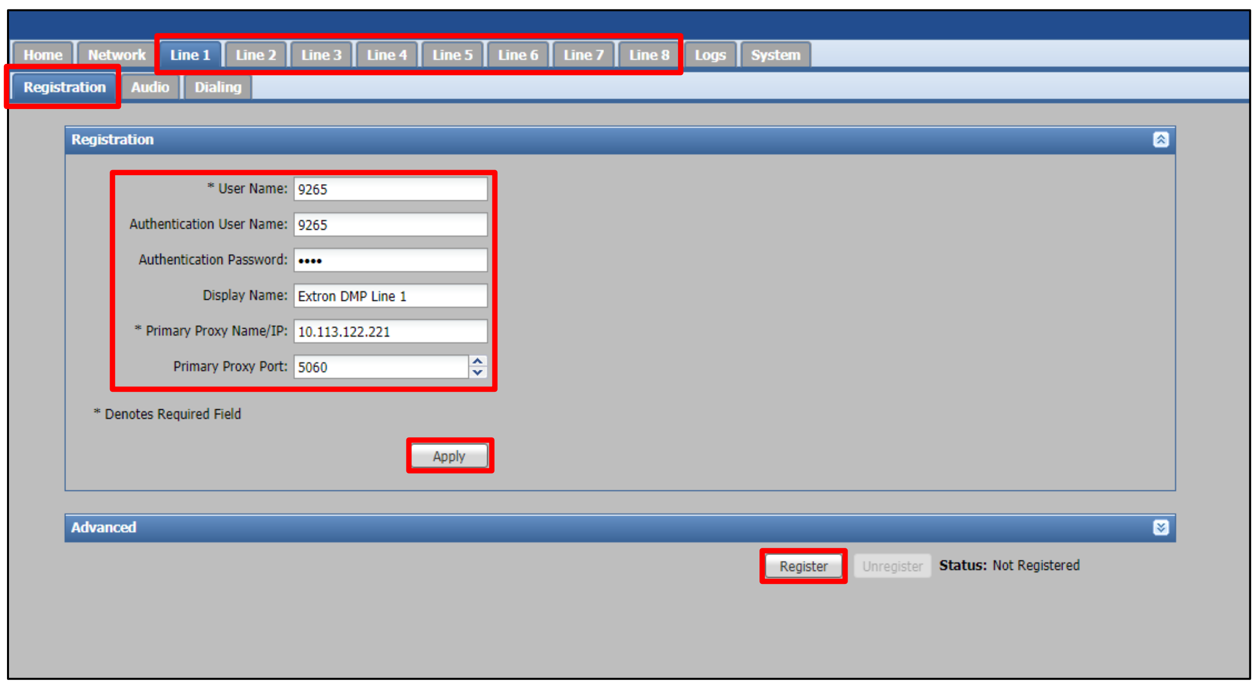
3.3 Line Registration

Click on the first line tab to be configured as part of the system, e.g. **Line 1**.

1) **User Name:** Set this to match the extension number from Section 2.2.
2) **Authentication User Name:** Set this field to match the extension number above.
3) **Authentication Password:** Use the extension number from (1) and (2) as the password.
4) **Display Name:** Optional. Specify an identifier for the line if required.
5) **Primary Proxy Name/IP:** Specify either the IP address or domain name of call server.
6) **Primary Proxy Port:** Specify the port number as required. The default is 5060.

Once the above settings have been entered, click the **Apply** button to save to the device.

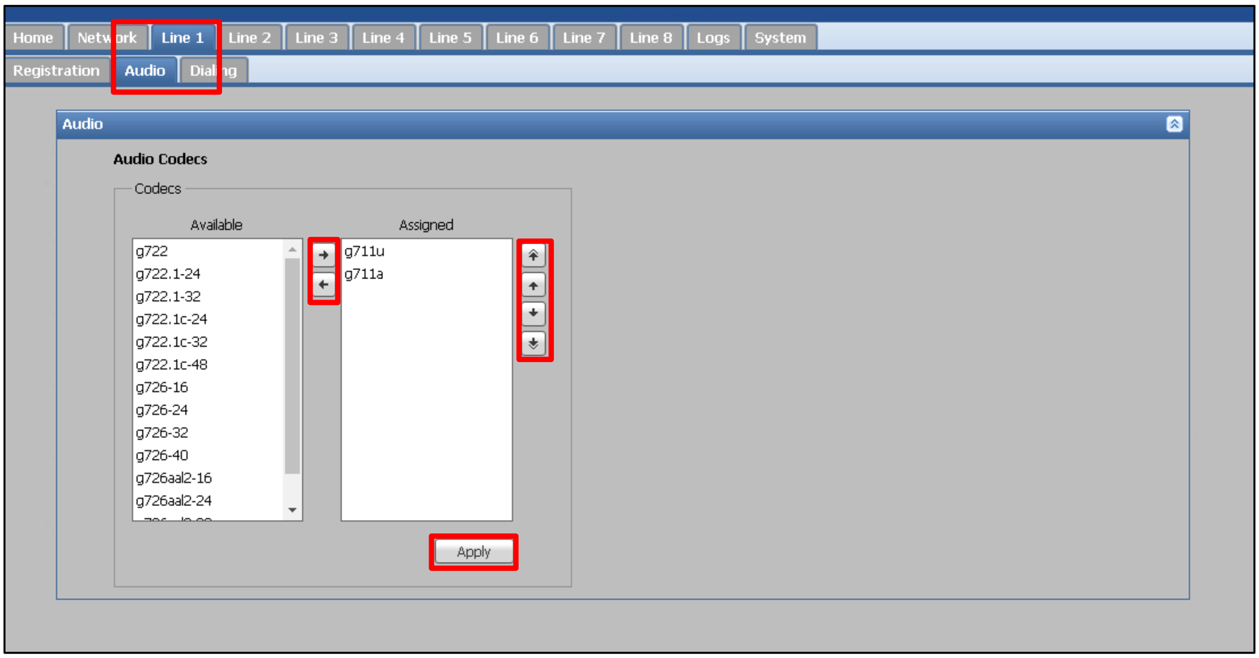
Click the **Register** button to initiate registration to the call server. If successful, the registration status to the right of the Register/Unregister buttons will indicate *Registered – Primary*.



3.4 Codecs

The availability and priority of codecs may be changed from within the **Audio** tab. Codecs will only be available for use within phone calls if they are moved from the **Available** to the **Assigned** column. By default, G.711u and G.711a are assigned to the system. Codec assignment and priority can be set per line.

Click the **Apply** button to commit any changes to the device.

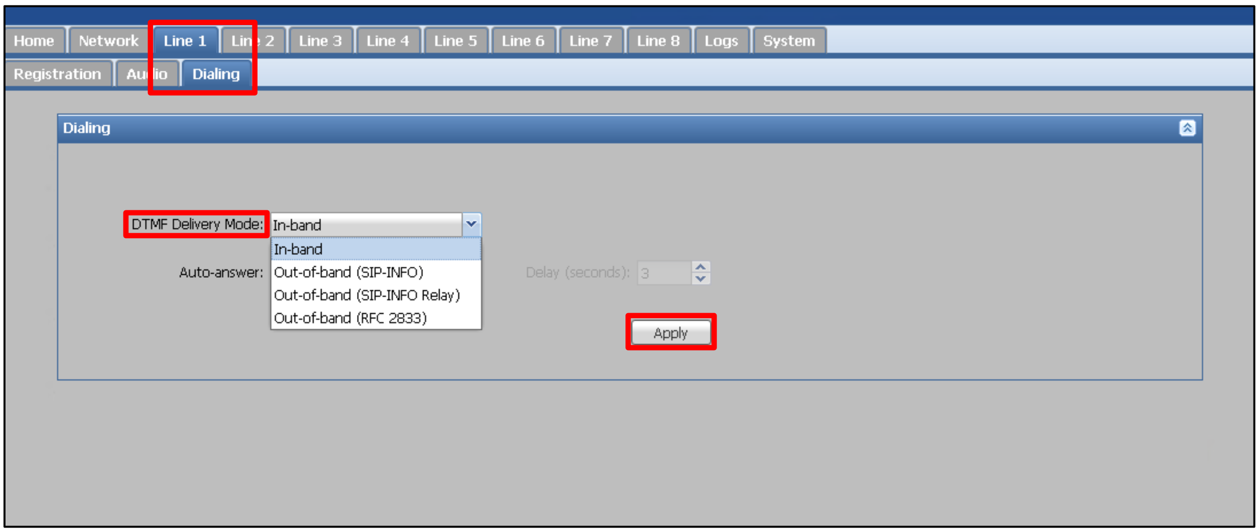


3.5 Dialing

Use the **Dialing** tab to select the desired DTMF signaling method system. The default mode is In-Band. Other available options are as follows:

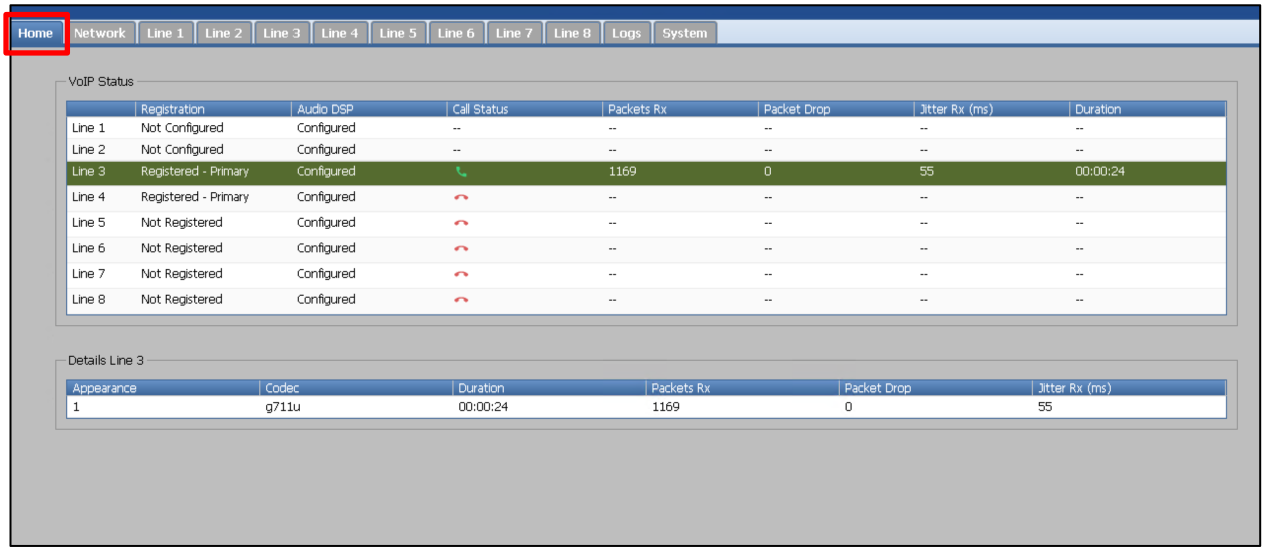
- Out of Band – SIP INFO
- Out of Band – SIP INFO (RELAY)
- Out of Band – RFC 2833

Click **Apply** after selecting the desired DTMF signaling method for the line. This can be set per line.



3.6 System Overview

Once all required lines have been registered to the call server, use the **Home** tab to view a summary of the system, as required. In the example below, one of two registered lines (line 3) is currently in an active call. Appearance-specific (caller-specific) details for active calls can be accessed by clicking on the corresponding Line entry.



	Registration	Audio DSP	Call Status	Packets Rx	Packet Drop	Jitter Rx (ms)	Duration
Line 1	Not Configured	Configured	--	--	--	--	--
Line 2	Not Configured	Configured	--	--	--	--	--
Line 3	Registered - Primary	Configured		1169	0	55	00:00:24
Line 4	Registered - Primary	Configured		--	--	--	--
Line 5	Not Registered	Configured		--	--	--	--
Line 6	Not Registered	Configured		--	--	--	--
Line 7	Not Registered	Configured		--	--	--	--
Line 8	Not Registered	Configured		--	--	--	--

Details Line 3

Appearance	Codec	Duration	Packets Rx	Packet Drop	Jitter Rx (ms)
1	g711u	00:00:24	1169	0	55

3.7 Troubleshooting

In the event of failure to register, review the following:

- Check that the credentials specified as part of the Interactive Intelligence setup are correctly entered into the registration fields for each line.
- Check network interface settings, including DNS fields (particularly if a proxy domain name is being used rather than an IP address).
- Click on the **Logs** tab to inbound and outbound SIP transactions. The absence of inbound transactions indicates a network routing problem. Registration-specific problems may be indicated by corresponding SIP responses such as *403 – Forbidden*.

Appendix A: RTP Port Range

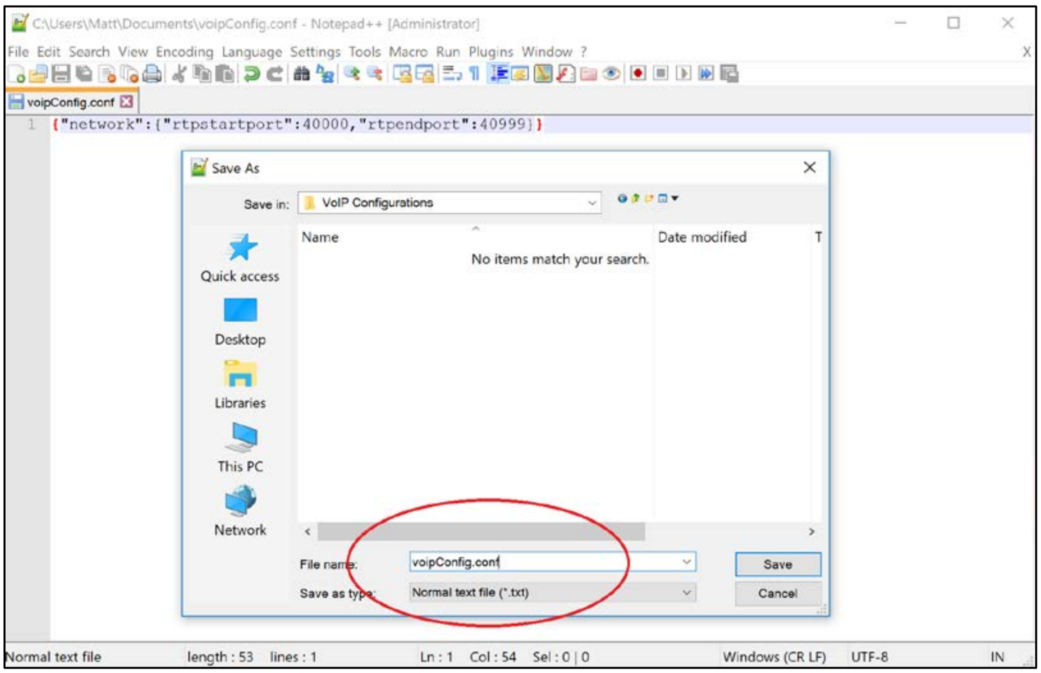
The default port range for VoIP RTP traffic on the DMP is **50000 – 50999**. To change this range, the following steps must be carried out.

1. Create a new blank text file using a suitable basic text editor.

2. Enter the following text into the document (in this example, the port range is being changed to 40000-40999; replace these values with the desired range) -

   ```
   {"network":{"rtpstartport":40000,”rtpendport”:40999}}
   ```

3. Save the file as **voipConfig.conf**.



4. Navigate to the DMP VoIP configuration webpage and click on the **System** tab.

5. Under **Export System Configuration**, click the **Export** button to back up the current VoIP configuration to disk. The file will be saved in the default web browser download directory.

6. Under **Import System Configuration**, click the **Browse** button to locate the **voipConfig.conf** file created in steps 1 to 3.

7. Click the **Import** button to update the DMP with the new RTP Port Range settings. A notification will appear once the settings have applied successfully.

Appendix B: Outgoing Call Termination Mode

Some call server configurations require the use of a CANCEL SIP message to terminate unanswered outgoing calls, rather than a traditional BYE message. This requirement will become apparent if far end devices continue to ring after the DMP has ended the call.

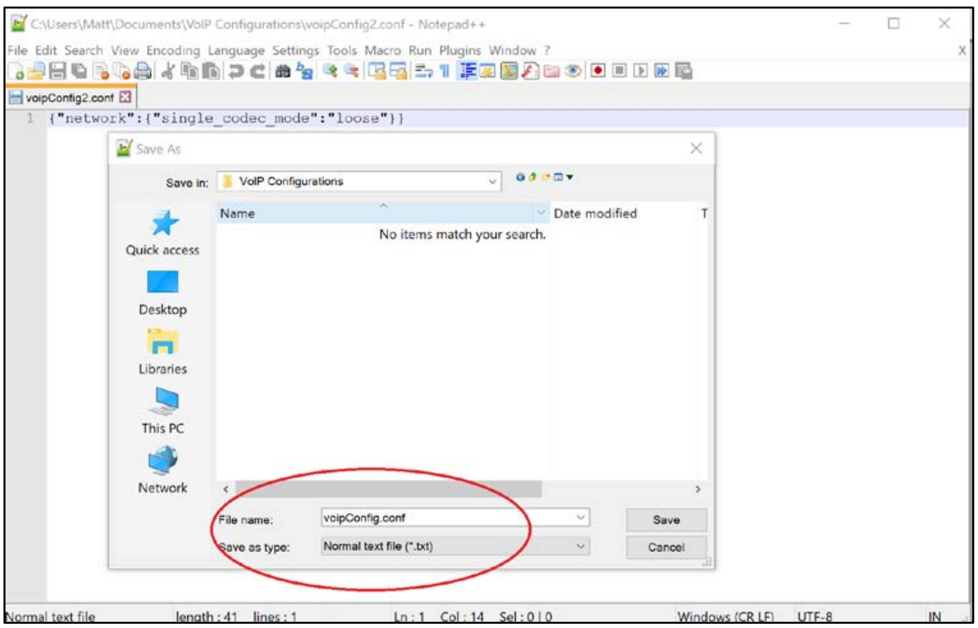
To change the outgoing call terminate mode from BYE to CANCEL (default is BYE), the following steps must be carried out. **Requires FW 1.01.0007-b004 or later.**

1. Create a new blank text file using a suitable basic text editor

2. Enter the following text into the document -

```
{"network":{"sip_outgoing_termination_use_cancel_enable":true}}
```

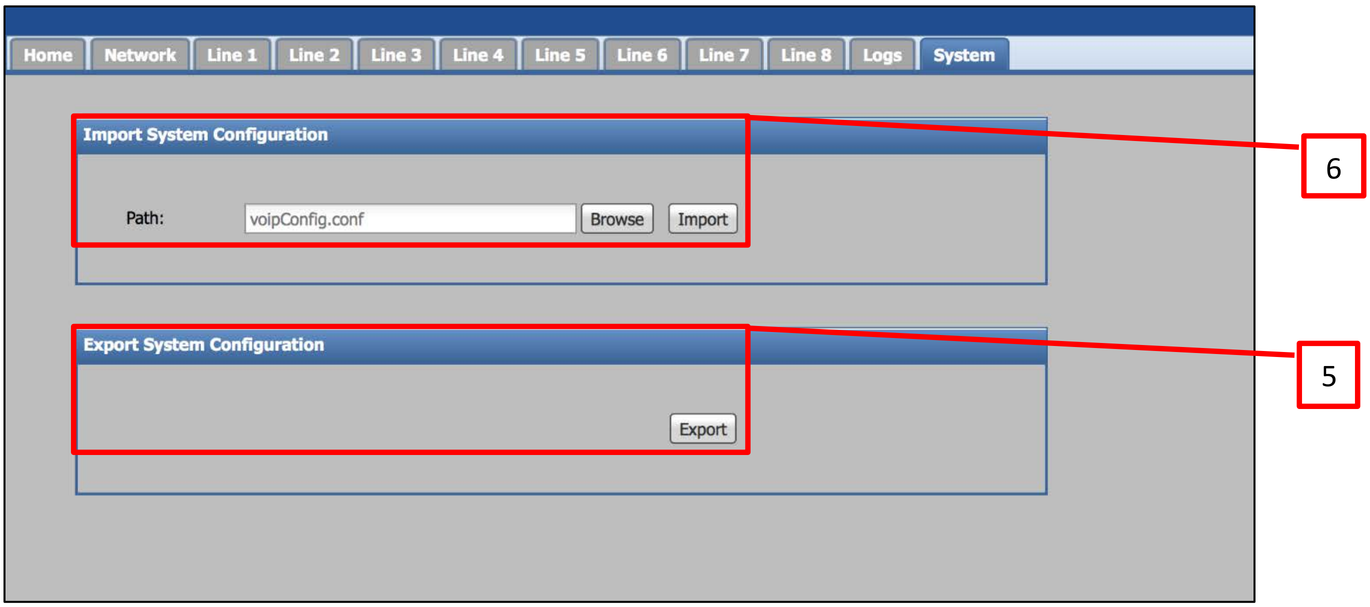
3. Save the file as **voipConfig.conf**.



4. Navigate to the DMP VoIP configuration webpage and click on the **System** tab.

5. Recommended: Under **Export System Configuration**, click the **Export** button in order to back up the current VoIP configuration to disk. The file will be saved in the default web browser download directory.

6. Under **Import System Configuration**, click the **Browse** button to locate the **voipConfig.conf** file created in steps 1 to 3.



Click the **Import** button to update the DMP with the new settings. A notification will appear once the settings have applied successfully.

To return to BYE mode, send the following string using the same method:

```
{"network":{"sip_outgoing_termination_use_cancel_enable":false}}
```