



Extron DMP Plus Series Interactive Intelligence PBX System User Guide

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Extron

Extron DMP Plus Series Interactive Intelligence PBX System



DMP Plus Series C V DMP Plus Series C V AT Interactive Intelligence Configuration Guide

Product Information

The DMP Plus Series C V and C V AT models are VoIP devices that can be registered to the Interactive Intelligence PBX system running CIC version 2017 R1 or later. These devices require Firmware Version 1.08.0002 or higher for registration. They have two LAN connections, LAN 1 and LAN 2, that have the same capabilities for connecting, configuring, and controlling the hardware. The recommended LAN for VoIP traffic is LAN 1.

Product Usage Instructions

To register the DMP Plus Series VoIP device to the Interactive Intelligence PBX system, follow these steps:

Step 1: Create a New Line

1. Open the Interaction Administrator platform with administrator credentials.
2. Click on the Lines item on the left-hand side of the screen.
3. Right-click on the right-hand list portion of the screen and select New.
4. Edit the Line Value fields as required and click OK.
5. Edit the Audio fields as shown and check the Allow Multiple Codecs box if more than one codec will be assigned to the DMP Plus Series. Click OK.
6. Set the signaling Transport Protocol as required. The default is UDP.
7. Change Media Timing and Media reINVITE Timing to Normal. Uncheck the Use SIP Session Timer box.
8. Click on Proxy followed by Add. Enter the IP address of the DMP Plus Series and the port number being used (the default port for UDP and TCP is 5060). Click OK.
9. Click on Access followed by the Denied Access radio button. Add the DMP Plus Series as an exception by clicking on the Add button and entering its IP address, followed by OK.

Step 2: Create a New Station

Add a New Station for the DMP Plus Series to complete the registration process.

Revision Log

Date	Version	Notes
Feb 9, 2018	1.0	First Release: Applies to Firmware Version 1.01.0007.004
Mar 7, 2018	1.0.1	Layout and language changes for emphasis
July 26, 2019	1.0.2	Added Appendix C
Feb. 12, 2020	1.1.0	Updated DMP Plus Series
Sep. 1, 2020	1.2.0	Added VoIP Configuration file
Sep. 12, 2022	1.2.2	Updated Appendix
Mar. 27, 2023	1.3.1	Update Network Troubleshooting

Introduction

This document provides essential instructions for registering the VoIP lines of DMP Plus Series, C V and C V AT models to Interactive Intelligence PBX system running CIC version 2017 R1 or later.

DMP Plus Relates to the following products:

- DMP 128 Plus C V / C V AT
- DMP 128 FlexPlus C V AT
- DMP 64 Plus C V / C V AT

Note: Requires Firmware Version 1.08.0002 or higher

Networking Note:

LAN 1 and LAN 2 on the DMP both have the same capabilities for connecting, configuring, and controlling the hardware. If the call manager intends to register the DMP Plus VoIP interface and the control system reside on the same IP subnet, only a single LAN connection is needed.

For best performance, it is recommended that LAN 1 be used to for VoIP traffic.

Configuring Interactive Intelligence for DMP Plus Series VoIP Registration

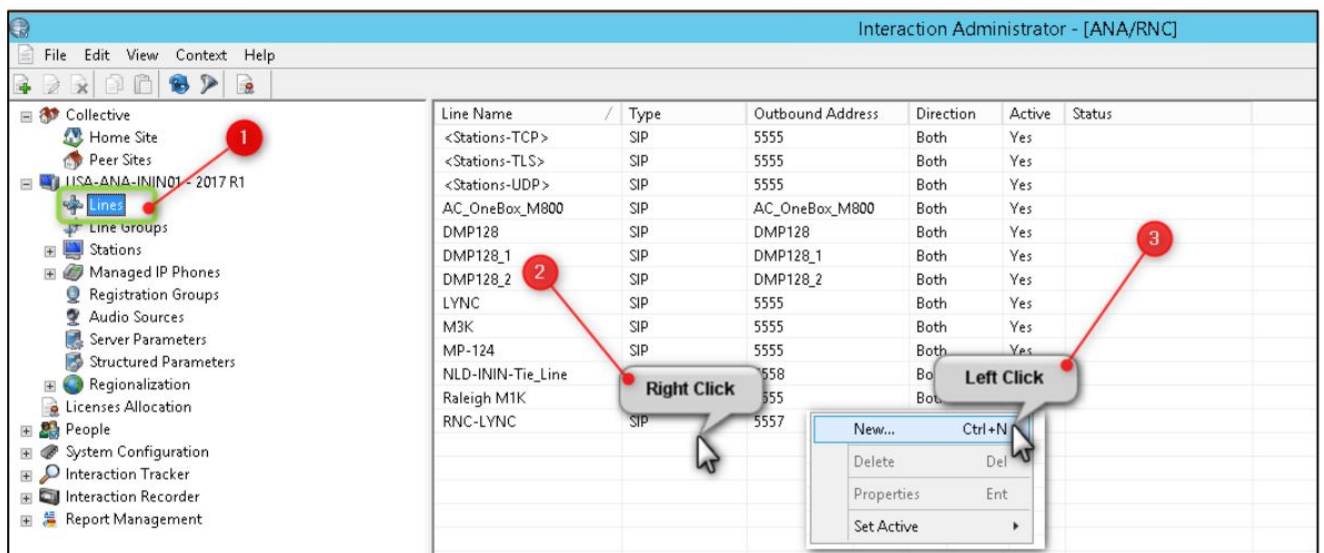
- VoIP functionality within the DMP Plus Series is built around the Session Initiation Protocol (SIP) signaling system, as defined in RFC 3261. The Interactive Intelligence platform must be licensed to allow the addition of generic basic third-party SIP endpoints before any line registration can take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

Note: When TLS is used only Line 1 can be used on the DMP Plus VoIP Interface.

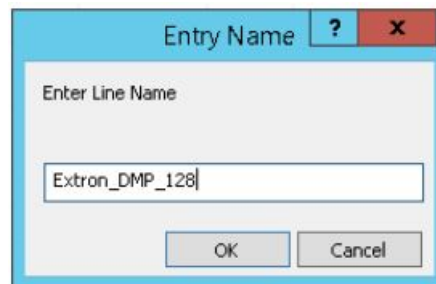
Create a New Line

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

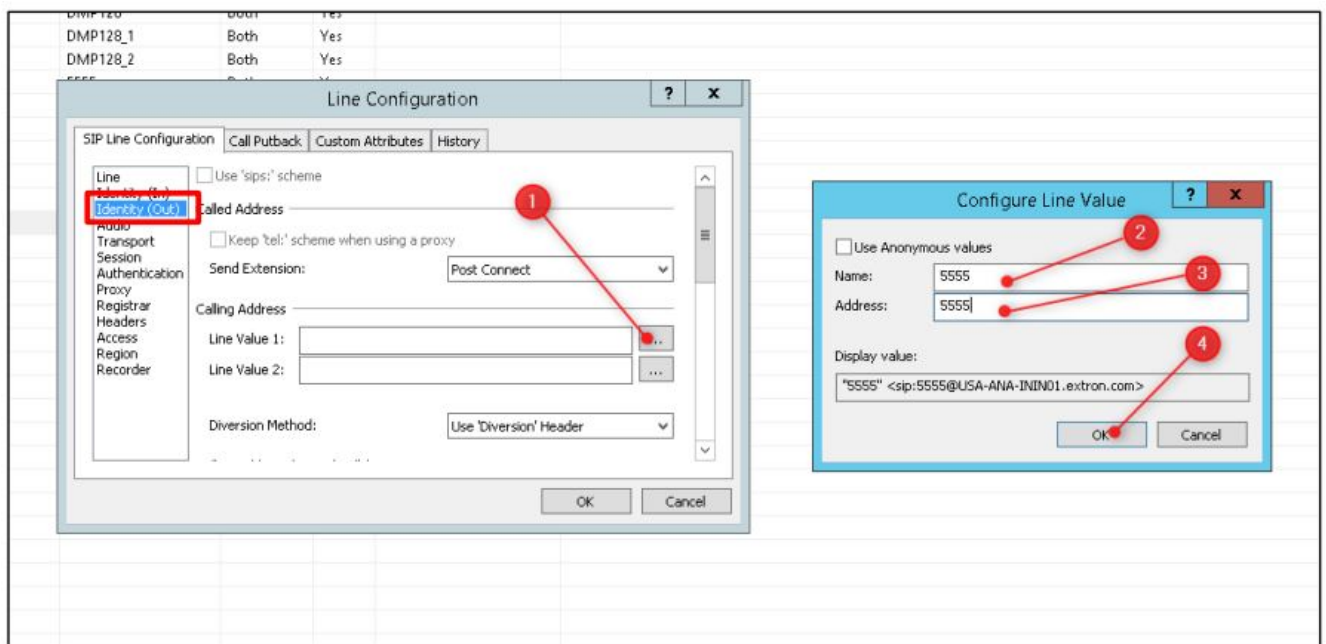
1. Click on the Lines [1] item on the left-hand side of the screen.
2. Right-click on the right-hand list portion of the screen [2].
3. Select New [3].



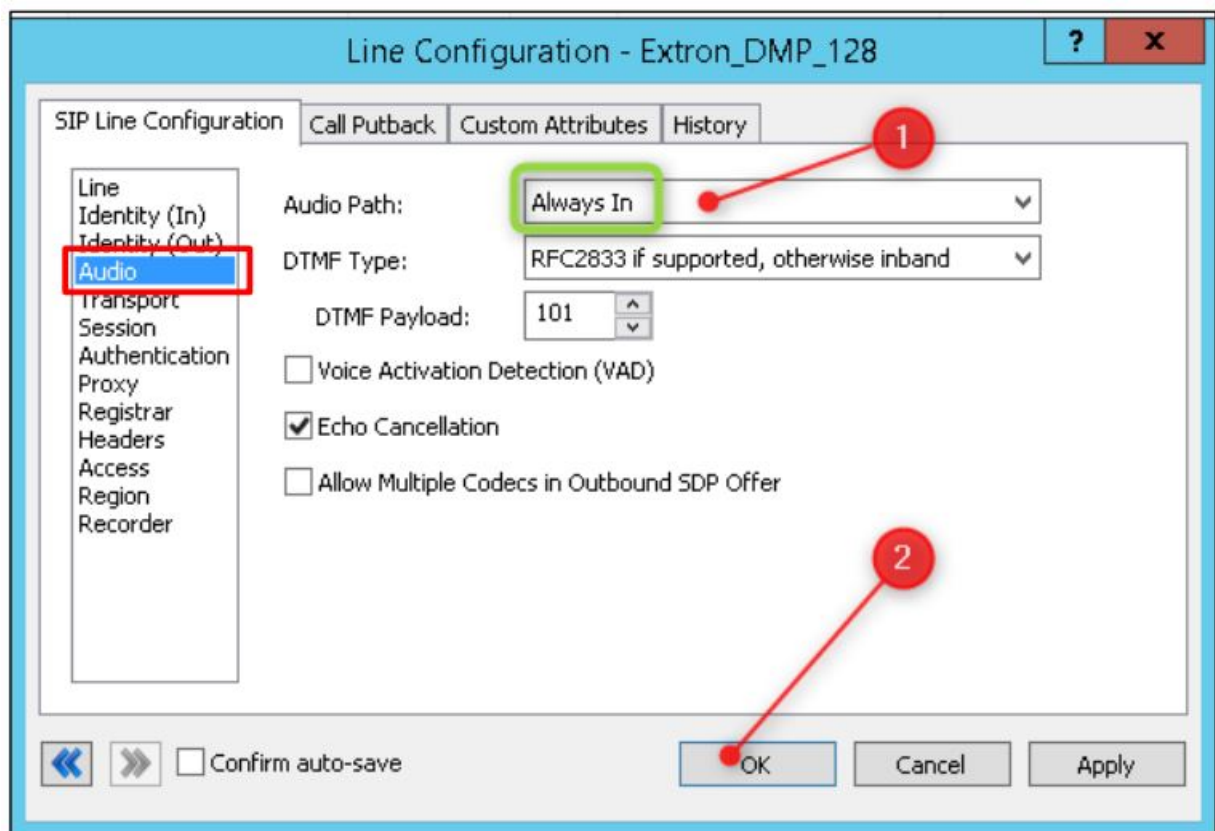
4. Enter a name for the line and click on OK.



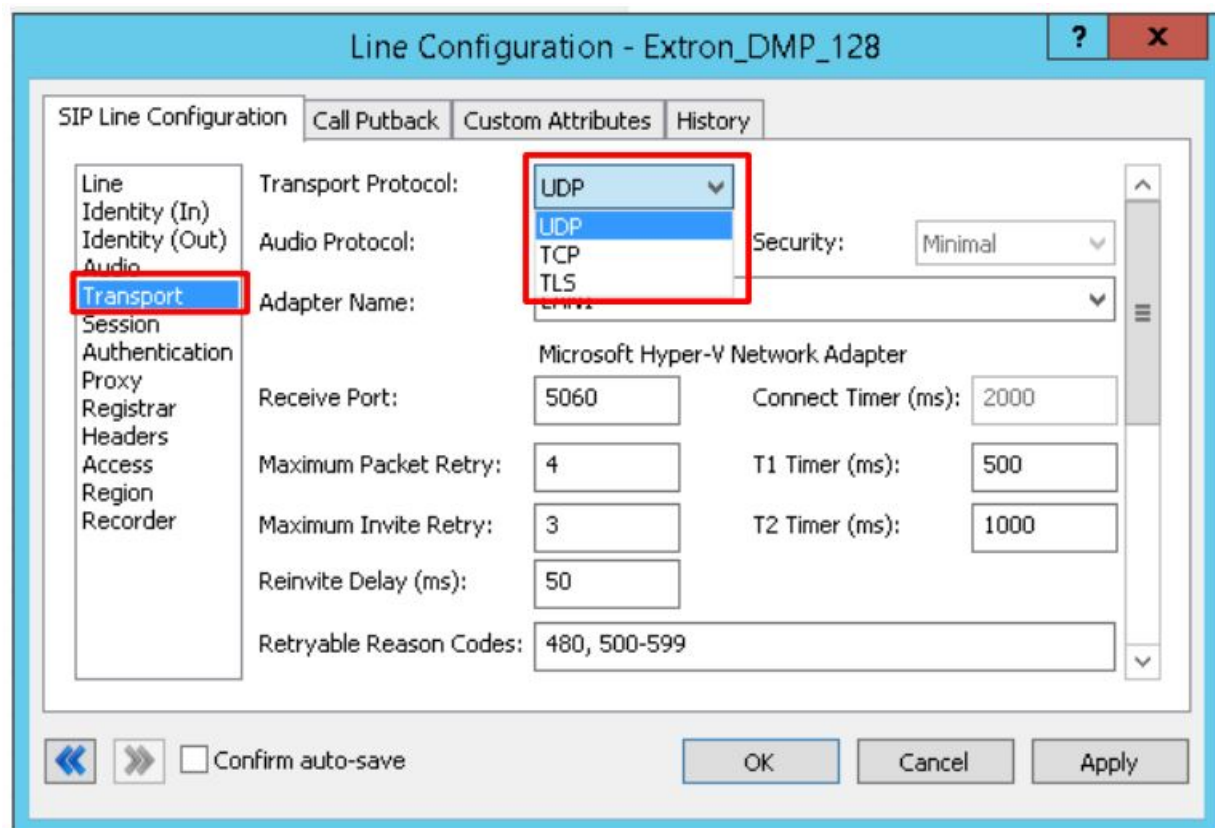
5. Click on Identity (OUT) [1] and edit Line Value fields as required [2] and [3]. Click OK [4].



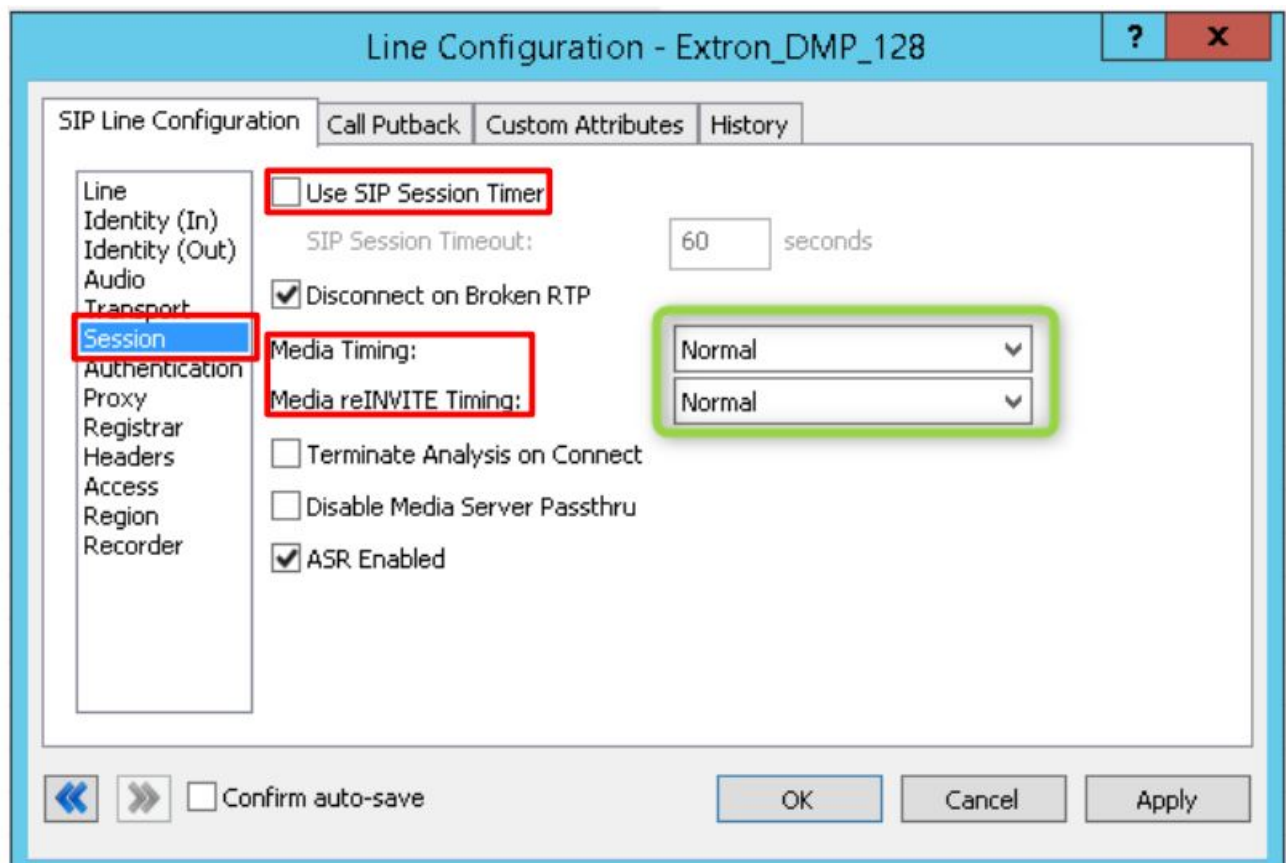
6. Click on Audio and edit the fields as shown below [1]. Check the Allow Multiple Codecs box if more than one codec will be assigned to the DMP Plus Series. Click OK [2].



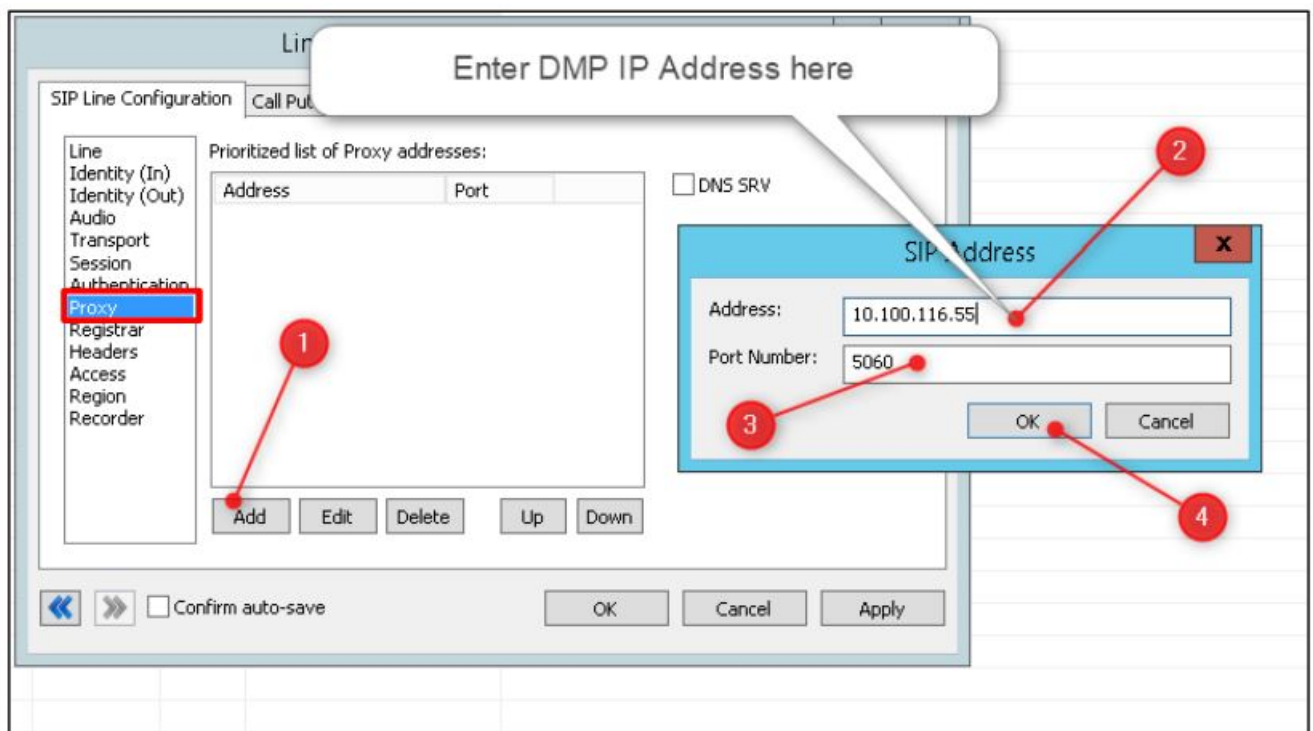
7. Set the signaling Transport Protocol as required. The default is UDP.



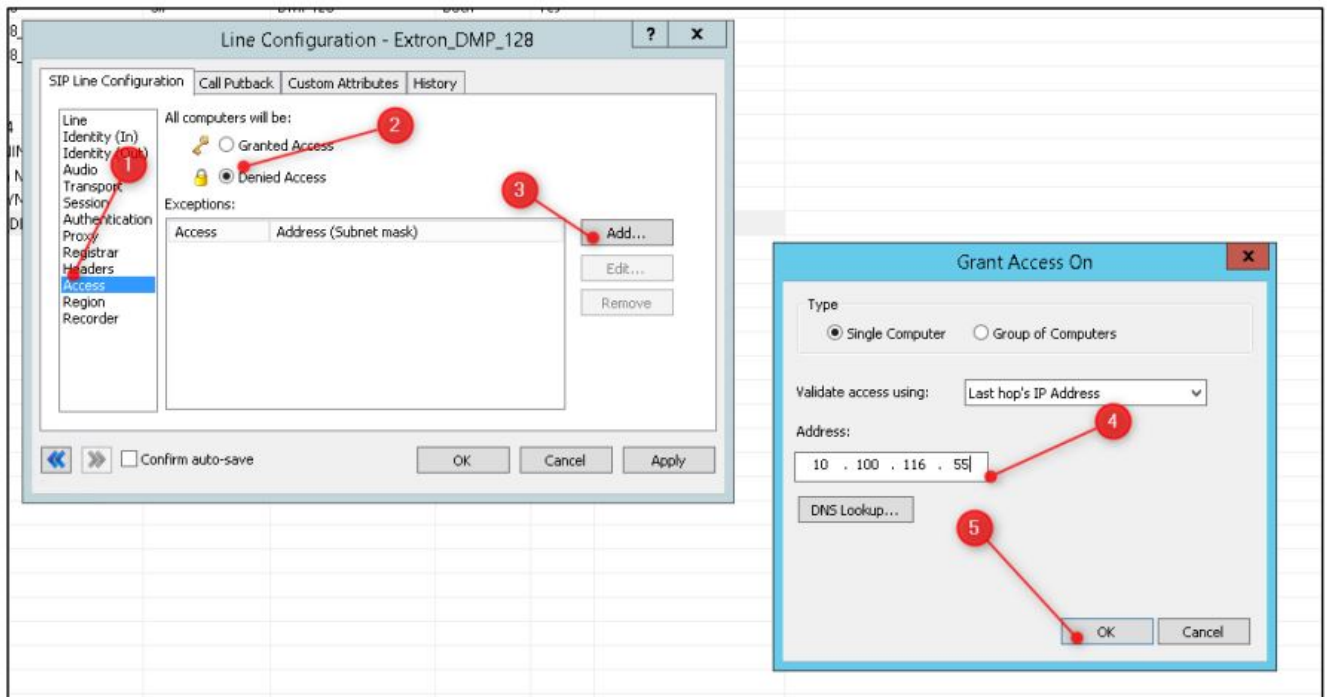
8. Click on Session and change Media Timing and Media reINVITE Timing to Normal.
Uncheck the Use SIP Session Timer box.



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



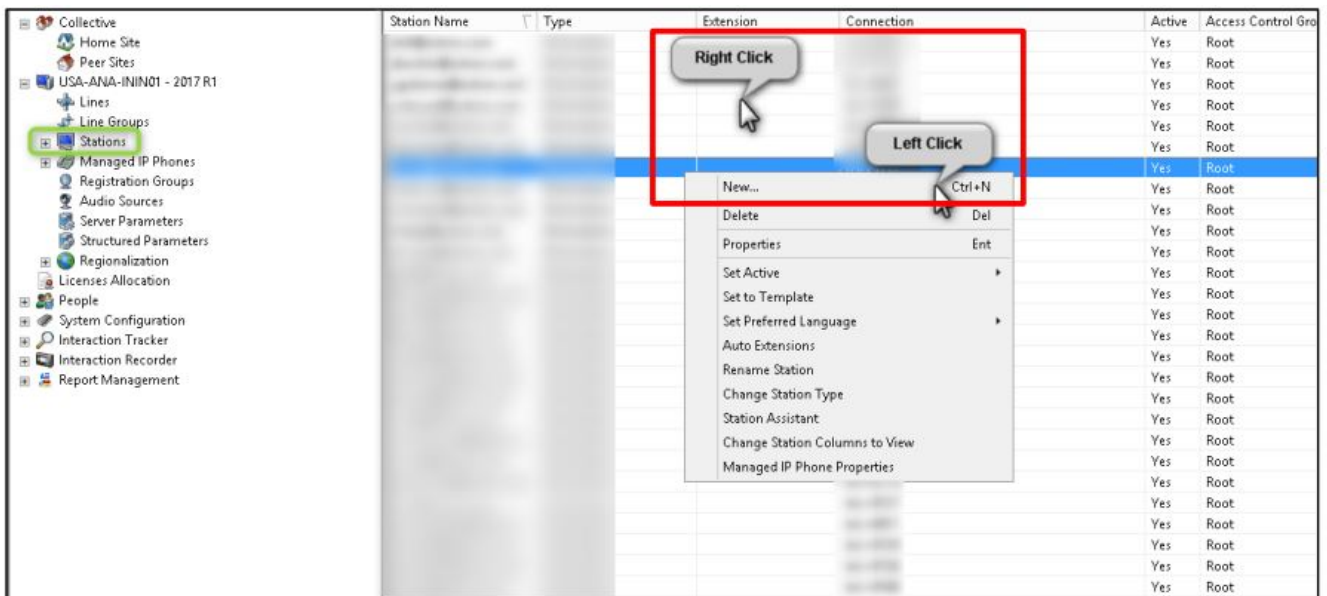
10. Click on Access [1] followed by the Denied Access [2] radio button. Add the DMP Plus Series as an exception by clicking on the Add [3] button and entering its IP address [4], followed by OK [5].



Create a New Station

Add a New Station for the DMP Plus Series.

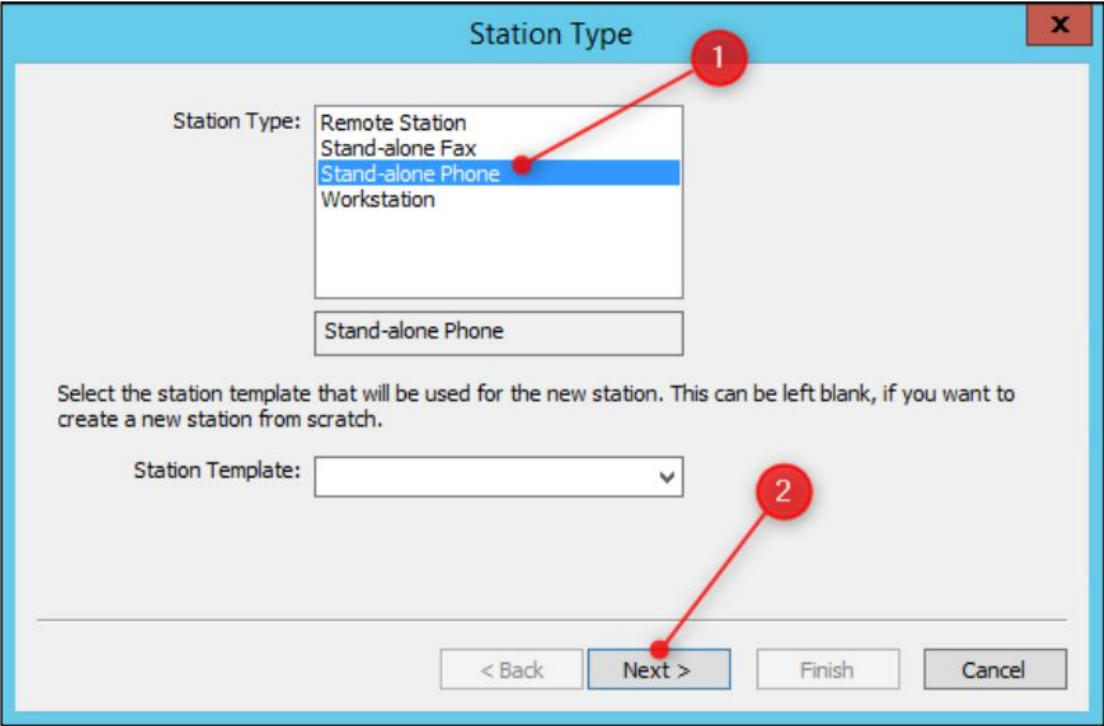
1. Click on the Stations container, right-click in the list area, then select New.



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].



Station Type

Station Type: Remote Station
Stand-alone Fax
Stand-alone Phone
Workstation

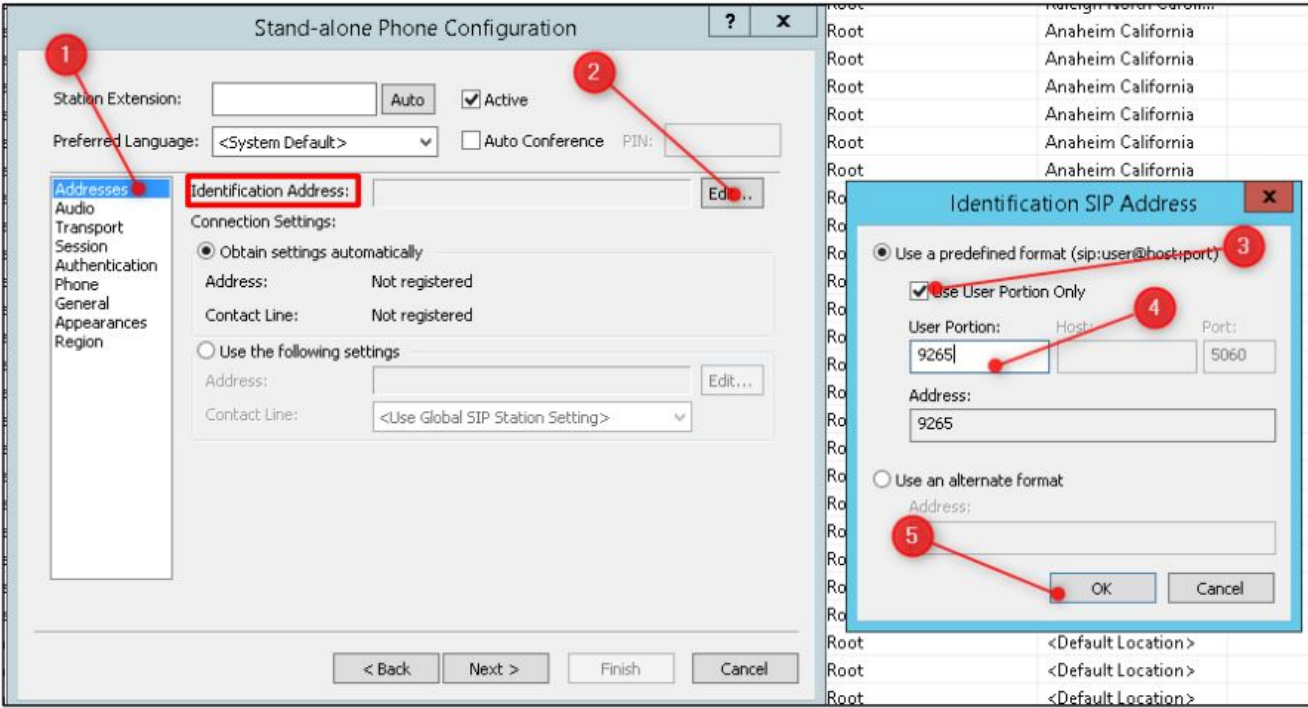
Stand-alone Phone

Select the station template that will be used for the new station. This can be left blank, if you want to create a new station from scratch.

Station Template:

< Back **Next >** Finish Cancel

- 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 Plus Series, followed by OK [5].



Stand-alone Phone Configuration

Station Extension: Auto ☒ Active

Preferred Language: <System Default> ☐ Auto Conference PIN:

Addresses Identification Address: Edit...

Connection Settings:

☒ Obtain settings automatically

Address: Not registered

Contact Line: Not registered

☐ Use the following settings

Address: Edit...

Contact Line: <Use Global SIP Station Setting>

< Back Next > Finish Cancel

Identification SIP Address

☒ Use a predefined format (sip:user@host:port)

☒ Use User Portion Only

User Portion: Host: Port:

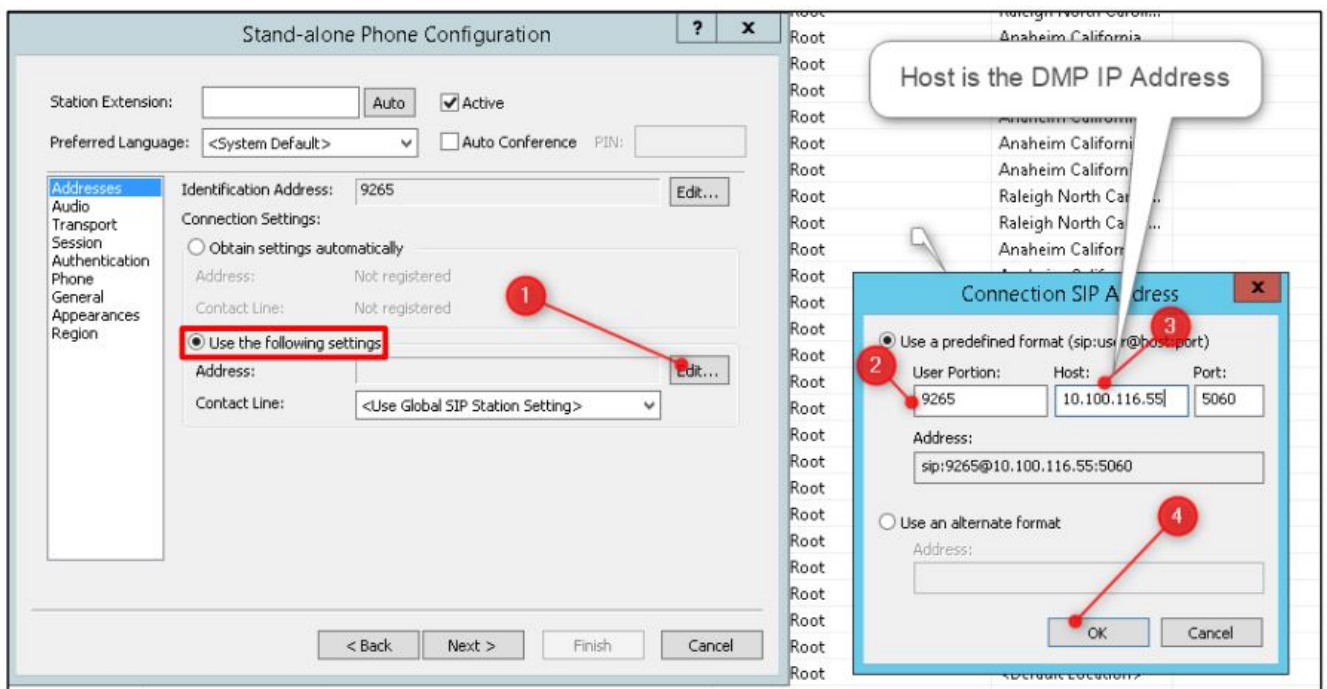
Address:

☐ Use an alternate format

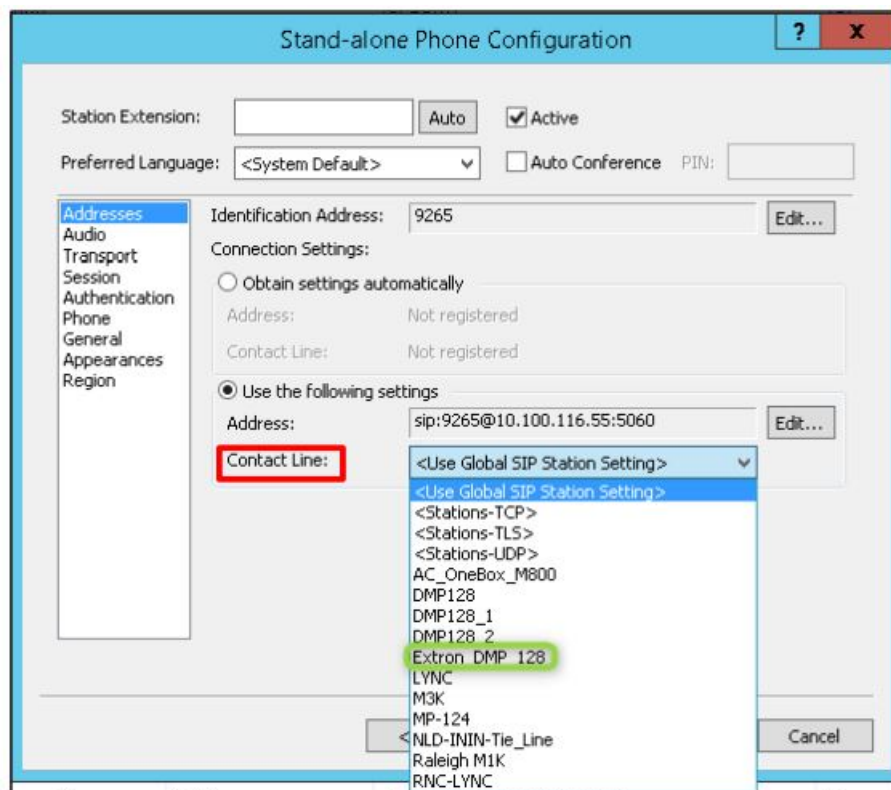
Address:

OK Cancel

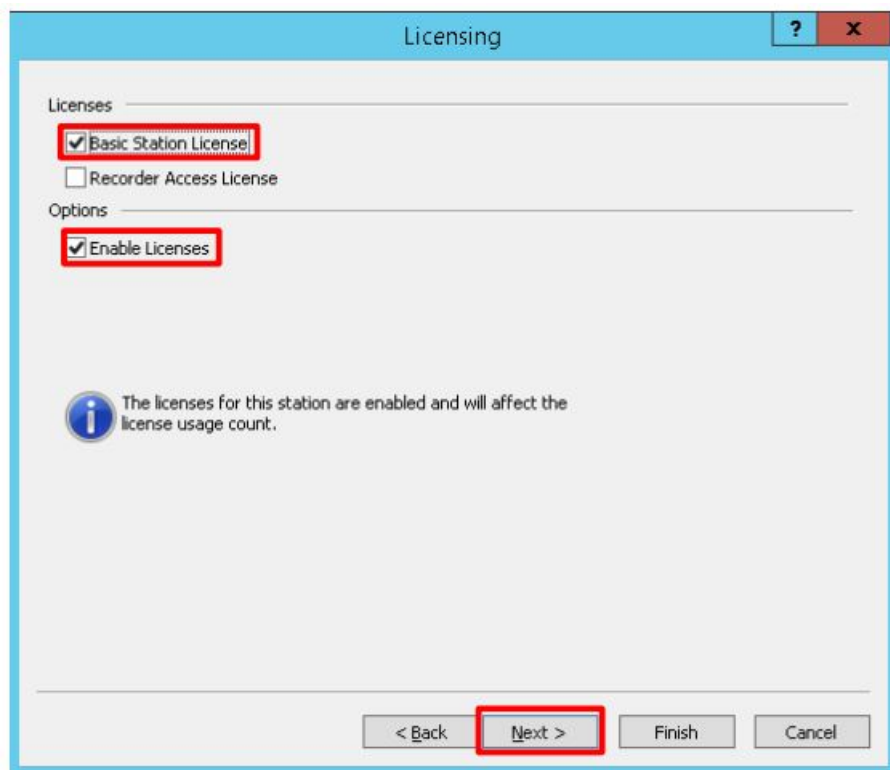
- Select the Use the following settings radio button followed by Edit [1]. Enter the extension number assigned to the DMP Plus Series in the User Portion [2] field, followed by the IP address and SIP port number in the Host and Port fields [3], respectively. Click OK [4].



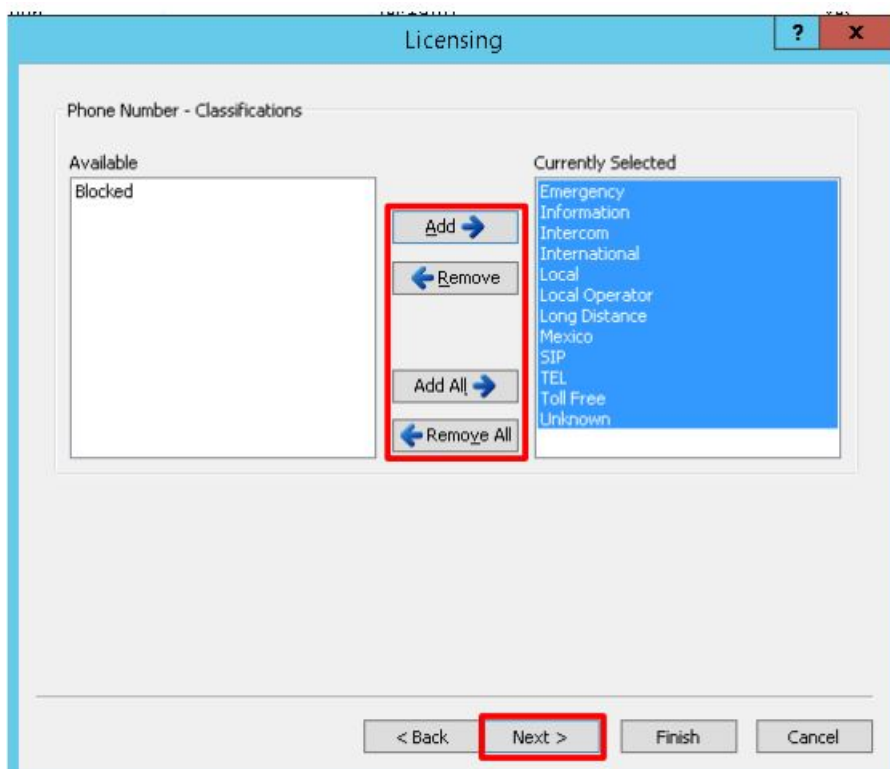
6. From the Contact Line drop-down box, select the line created in Section 2.1 followed by Next.



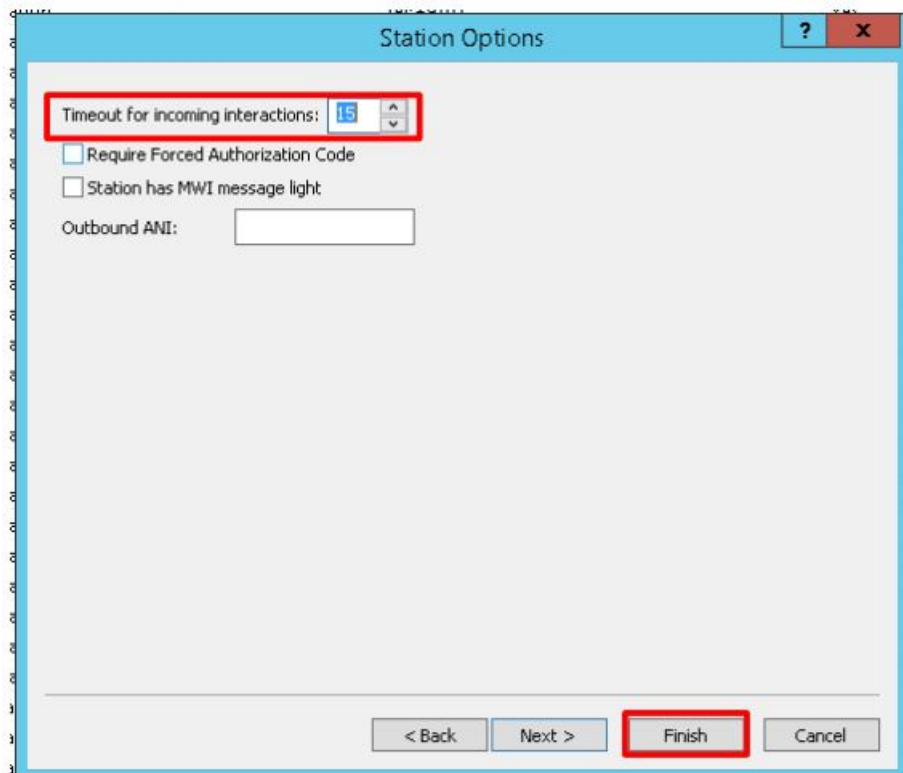
7. Select the Basic Station License and Enable Licenses checkboxes followed by Next.



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.



The image shows a 'Station Options' configuration window. At the top, there is a title bar with a question mark and a close button. Below the title bar, the 'Timeout for incoming interactions' is set to 15 seconds, highlighted with a red box. Below this, there are two checkboxes: 'Require Forced Authorization Code' and 'Station has MWI message light', both of which are unchecked. Below the checkboxes is a text field for 'Outbound ANI'. At the bottom of the window, there are four buttons: '< Back', 'Next >', 'Finish' (highlighted with a red box), and 'Cancel'.

Configuring DMP Plus Series C V (AT) VoIP Lines

VoIP configuration of the DMP Plus Series 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>

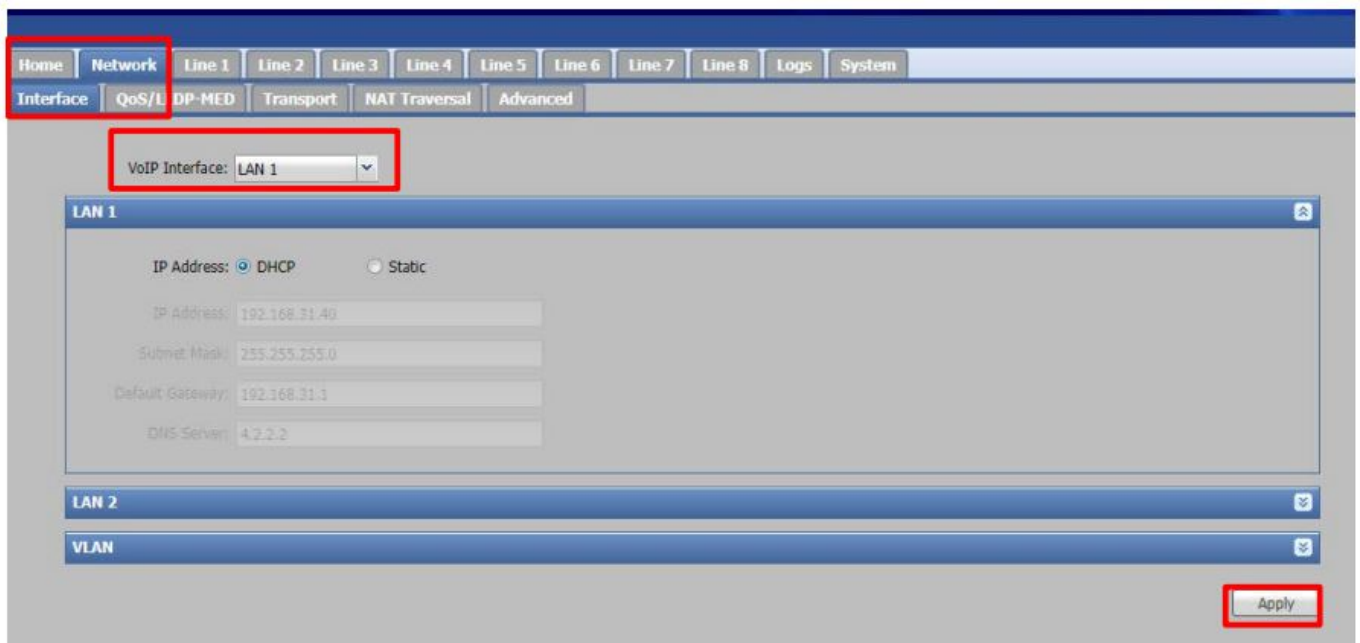
where 192.168.254.254 in this example is the default IP address of the DMP Plus Series device.

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.

Network Interface Configuration

Clicking on the Network followed by Interface tabs allows changes to be made to the desired network interface on the DMP Plus Series; 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.



The image shows the 'Network Interface Configuration' web interface. At the top, there is a navigation bar with tabs: 'Home', 'Network' (highlighted with a red box), 'Line 1', 'Line 2', 'Line 3', 'Line 4', 'Line 5', 'Line 6', 'Line 7', 'Line 8', 'Logs', and 'System'. Below the navigation bar, there is a sub-navigation bar with tabs: 'Interface' (highlighted with a red box), 'QoS/L', 'DP-MED', 'Transport', 'NAT Traversal', and 'Advanced'. Below the sub-navigation bar, there is a dropdown menu for 'VoIP Interface' set to 'LAN 1' (highlighted with a red box). Below the dropdown menu, there is a section for 'LAN 1' configuration. The 'IP Address' is set to 'DHCP' (radio button selected). Below this, there are text fields for 'IP Address' (192.168.31.40), 'Subnet Mask' (255.255.255.0), 'Default Gateway' (192.168.31.1), and 'DNS Server' (4.2.2.2). Below the 'LAN 1' section, there are sections for 'LAN 2' and 'VLAN', each with a dropdown menu. At the bottom right of the interface, there is an 'Apply' button (highlighted with a red box).

Note: When using multiple network connections, it is recommended to use LAN 1 for VoIP. Configuration and control are available on both network connections.

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 Plus Series is UDP.

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

Note: When TLS is used only Line 1 can be used on the DMP Plus VoIP Interface.

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.

Home Network **Line 1** Line 2 Line 3 Line 4 Line 5 Line 6 Line 7 Line 8 Logs System

Registration Audio Dialing

Registration

* User Name: 9265

Authentication User Name: 9265

Authentication Password: ****

Display Name: Extron DMP Line 1

* Primary Proxy Name/IP: 10.113.122.221

Primary Proxy Port: 5060

* Denotes Required Field

Apply

Advanced

Register Unregister Status: Not Registered

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.

Home Network **Line 1** Line 2 Line 3 Line 4 Line 5 Line 6 Line 7 Line 8 Logs System

Registration **Audio** Dialing

Audio

Audio Codes

Codecs

Available		Assigned
g722	→	g711u
g722.1-24	←	g711a
g722.1-32		
g722.1c-24		
g722.1c-32		
g722.1c-48		
g726-16		
g726-24		
g726-32		
g726-40		
g726aal2-16		
g726aal2-24		

Apply

Dialing

Use the Dialing tab to select the desired DTMF signaling method. The default DMP Plus Series 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

Recommend DTMF delivery method is Out of Band – RFC 2833

Click Apply after selecting Out of Band – RFC 2833 DTMF signaling method for the line. This can be set per line.

DTMF Delivery Mode: In-band
Auto-answer: Out-of-band (RFC 2833)
Delay (seconds): 3
Apply

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.

VoIP Status

	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

Troubleshooting

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

- When using multiple network connections, it is recommended to use LAN 1 for VoIP.
 - Configuration and control are available on both network connections.
 - If VoIP and Control system reside on the same IP subnet, only a single network is required.
- Check that the DMP Plus can ping the VoIP server by using proxy ping. This can verify network access and verify DNS name resolution.
 - Using telnet (port 23) or SSH (22023) ping the address of the VoIP server.
 - Command: (esc)<VoIP server Address>ping(cr)
 - **Example:** (esc)192.168.10.10ping(cr)

- 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).
- When TLS is used only Line 1 can be used on the DMP Plus VoIP Interface.
- 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.

Configuration File – Attached to PDF

If needed, the configuration file “voipConfig.conf” is attached to the PDF

- To access the file select “Attachments”  from the left side bar – see figure A1
- Then save the attachment, before uploading to DMP Plus – see Figure A2 below

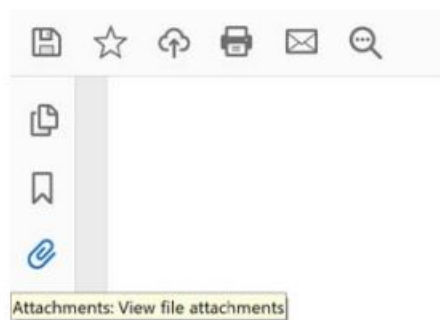


Figure A1 Save Attachment

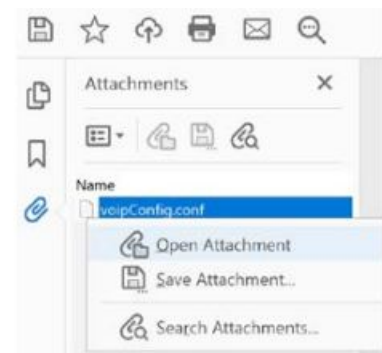


Figure A2 Show Attachments

Appendix A: RTP Port Range

The default port range for VoIP RTP traffic on the DMP Plus Series is 50000 – 50999. To change this range, the following steps must be carried out. There are two methods that can be used to change adjust the RTP port Range

Note: Requires Firmware 1.08.0002 or later.

Method 1 – Internal Webpage

1. From internal webpage Select Network then Advanced Tab
2. Adjust the Start and End port for RDP

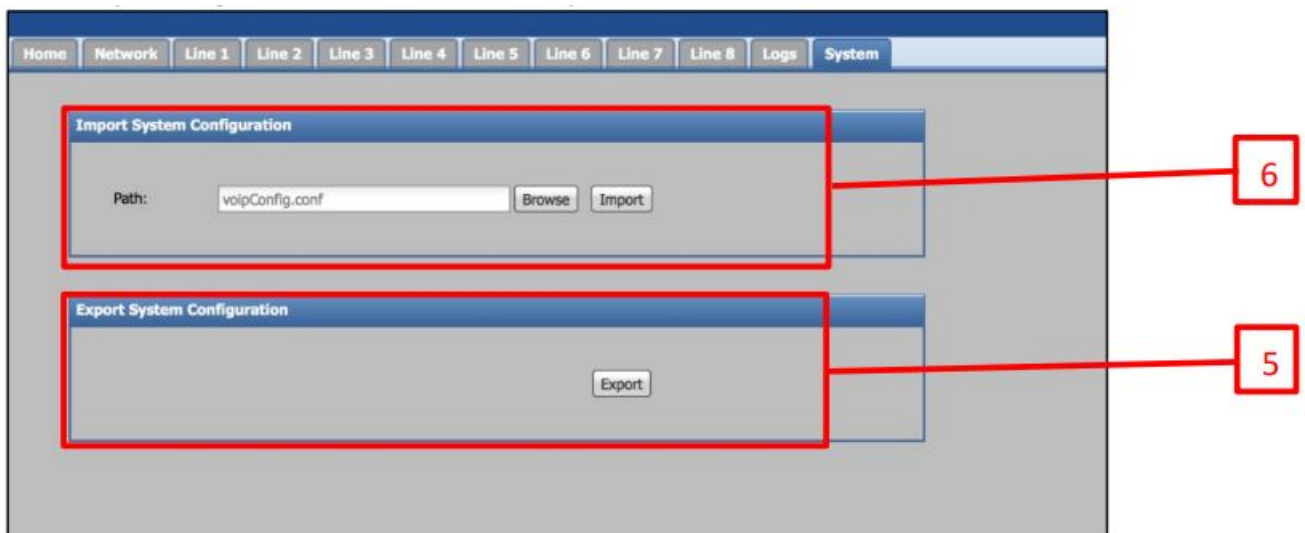
The screenshot shows the VoIP configuration web interface. The 'Network' tab is selected, and the 'Advanced' sub-tab is highlighted. The 'RTP Port Range' section is highlighted with a red box, showing 'Start Port' set to 50000 and 'End Port' set to 50999. Other sections include 'Registration' with 'Automatic Line Re-Registration' and 'Outgoing Calls' with 'Enable Codec Compatibility Mode'.

Method 2 – Configuration file

- a. Create a new blank text file using a suitable basic text editor.
 - i. Example “voipConfig.conf” is attached to this PDF, see Section 3.9
- b. Enter the following text into the document (in this example, the port range is being changed to 50000 – 50999; replace these values with the desired range) -{"network":{"rtpstartport":50000,"rtpendport":50999}}
- c. Save the file as voipConfig.conf.



- d. Navigate to the VoIP configuration webpage and click on the System tab.
- e. 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.
- f. Under Import System Configuration, click the Browse button to locate the voipConfig.conf file created in steps 1 to 3.



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

Appendix B: Automatic Line Re-Registration

Some call managers and networks go into maintenance windows which do not allow VoIP endpoints to register or maintain their registration. To help resolve this issue the Automatic Line Re-Registration function can be configured to re-register a line if line registration is unexpectedly lost. This function causes the VoIP interface to re-attempt a line re-registration if the first automatic re-registration attempt fails.

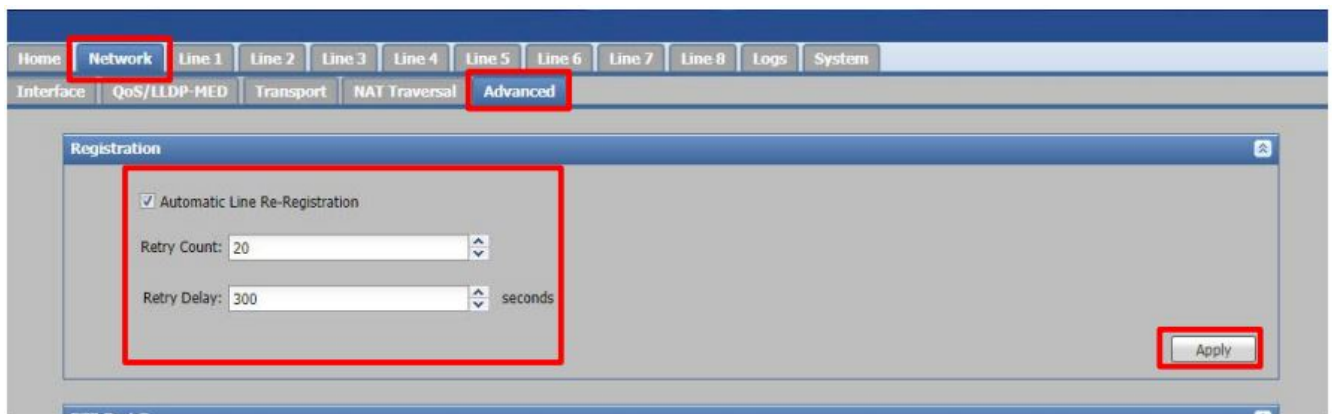
In order to use this feature, the line must first be registered to the call manager.

Note: When enabled, this function will attempt re-registration once the SIP timer has expired. By default the SIP timer is set to 3600 seconds (60 mins). By default, the Automatic Line Re-Registration feature is disabled, with the "registration_fail_retry_count" set to zero (0).

To set up Automatic Line Re-Registration, the following steps must be carried out.
Requires Firmware 1.08.0002 or later.

Method 1 – Internal Webpage

1. From internal webpage Select Network then Advanced Tab
2. To Enable the Automatic Line Re-Registration select the check box
3. Enter Retry Count (0 – 99)
 - **a.** This is the number of attempts a Line will make to re-register
 - **i.** Example below is set to twenty (20) reconnections attempts
 - **ii.** If this is set to zero (0), the feature is disabled
4. Enter Retry Delay (120 – 3600 seconds)
 - **a.** Amount time between registration attempts in seconds
 - **i.** Example above is set to 300 seconds (5 mins) between reconnections attempts
5. Once Set hit Apply



Method 2 – Configuration file

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

- **a.** Example “voipConfig.conf” is attached to this PDF, see Section 3.9

2. Enter the following text into the document –{“network”:

{“registration_fail_retry_count”:5,“registration_fail_retry_delay”:300}}

- **a.** registration_fail_retry_count”:5

This is the number of attempts a Line will make to re-register

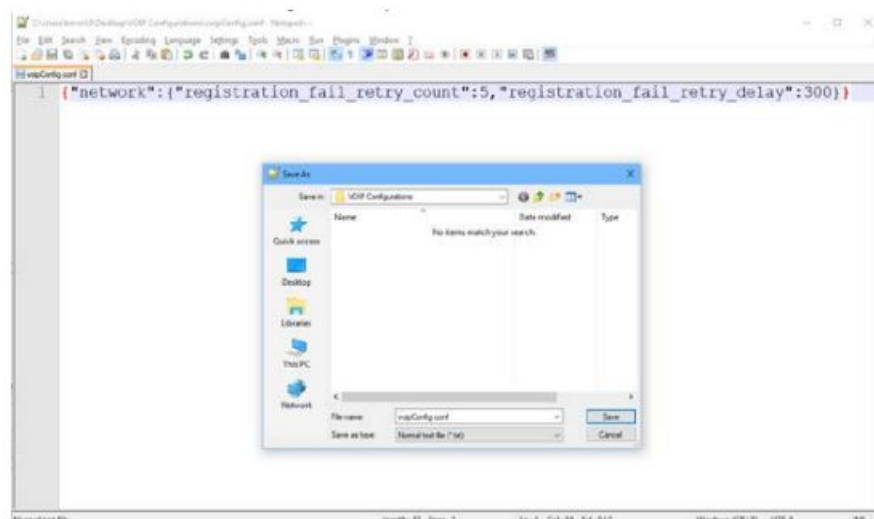
- **i.** Example above is set to five (5) reconnections attempts
- **ii.** If this is set to zero (0), the feature is disabled
- **iii.** Valid Range of values: 0 – 99

- **b.** registration_fail_retry_delay”:300

Amount time between registration attempts in seconds

- **i.** Example above is set to 300 seconds (5 mins) between reconnections attempts
- **ii.** Valid Range of values: 120 – 3600

3. Save the file as voipConfig.conf.



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

5. 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.

The screenshot shows the 'System' configuration page. The top navigation bar includes 'Home', 'Network', 'Line 1' through 'Line 8', 'Logs', and 'System'. The 'System' tab is active. Below the navigation bar, there are two main sections: 'Import System Configuration' and 'Export System Configuration'. The 'Import' section has a 'Path' field with the value 'voipConfig.conf', a 'Browse' button, and an 'Import' button. The 'Export' section has an 'Export' button. Red boxes highlight the 'Import' and 'Export' sections, with callout numbers 6 and 5 respectively.

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

To disable Auto-Reregistration mode, send the following string using the same method:

```
{“network”: {“registration_fail_retry_count”:0,“registration_fail_retry_delay”:200}}
```

Documents / Resources

Extron DMP Plus Series C.V DMP Plus Series C.V AT Interactive Intelligence Configuration Guide Version 1.0.1 2010	Extron DMP Plus Series Interactive Intelligence PBX System [pdf] User Guide DMP Plus Series, DMP Plus Series Interactive Intelligence PBX System, Interactive Intelligence PBX System, Intelligence PBX System, PBX System
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