Extron Electronics
INTERFACING, SWITCHING AND CONTROL

DMP 128 Plus C V
DMP 128 Plus C V AT

Cisco CUCM Configuration Guide

REVISION: **1.2.3**
DATE: **JULY 26, 2019**

Revision Log

Date	Version	Notes
Aug. 4th 2017	1.0	First Release: Applies to Firmware Version **1.01.0004.002**
Sep. 1st 2017	1.1	Corrected error in system import procedure. Applies to Firmware **1.01.0004.002**
Jan. 9th 2018	1.2	Added new features and CUCM configuration updates for **1.01.0007.004**
Mar. 7th 2018	1.2.1	Layout and language changes for emphasis
Dec. 6th 2018	1.2.2	Additional clarification added in Section 2.4
July 26th 2019	1.2.3	Added Appendix C

Table of Contents

1.0 Introduction

This document provides essential instructions for registering DMP 128 Plus C V (AT) VoIP lines to a Cisco Unified Call Manager (CUCM) system (Version 10.0 onwards).

DMP 128 Plus C V / C V AT Firmware Version **1.02.0001-b001** or later.

2.0 Configuring CUCM for DMP 128 Plus CV (AT) 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 CUCM 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.

2.1 Security Profile

A Security Profile is required prior to the registration of any DMP 128 Plus VoIP lines.

1) Access the Cisco Unified CM Management interface.

2) Add a new Phone Security Profile.

3) Select **Third-Party SIP Device (Basic**) as the profile type.

4) Name the security profile using a suitable reference, e.g. *Extron SIP*.

5) Set the **Nonce Validity Time** to **600**.

6) Select the appropriate transport type (the DMP can use either UDP or TCP as required).

7) Set the desired signaling port (default for the DMP is **5060**).

2.2 End User

An End User account is required for each VoIP line registered on the DMP 128 Plus.

1) Specify a User ID for the line (e.g. 1001, DMP1001, DMPConfRoom).

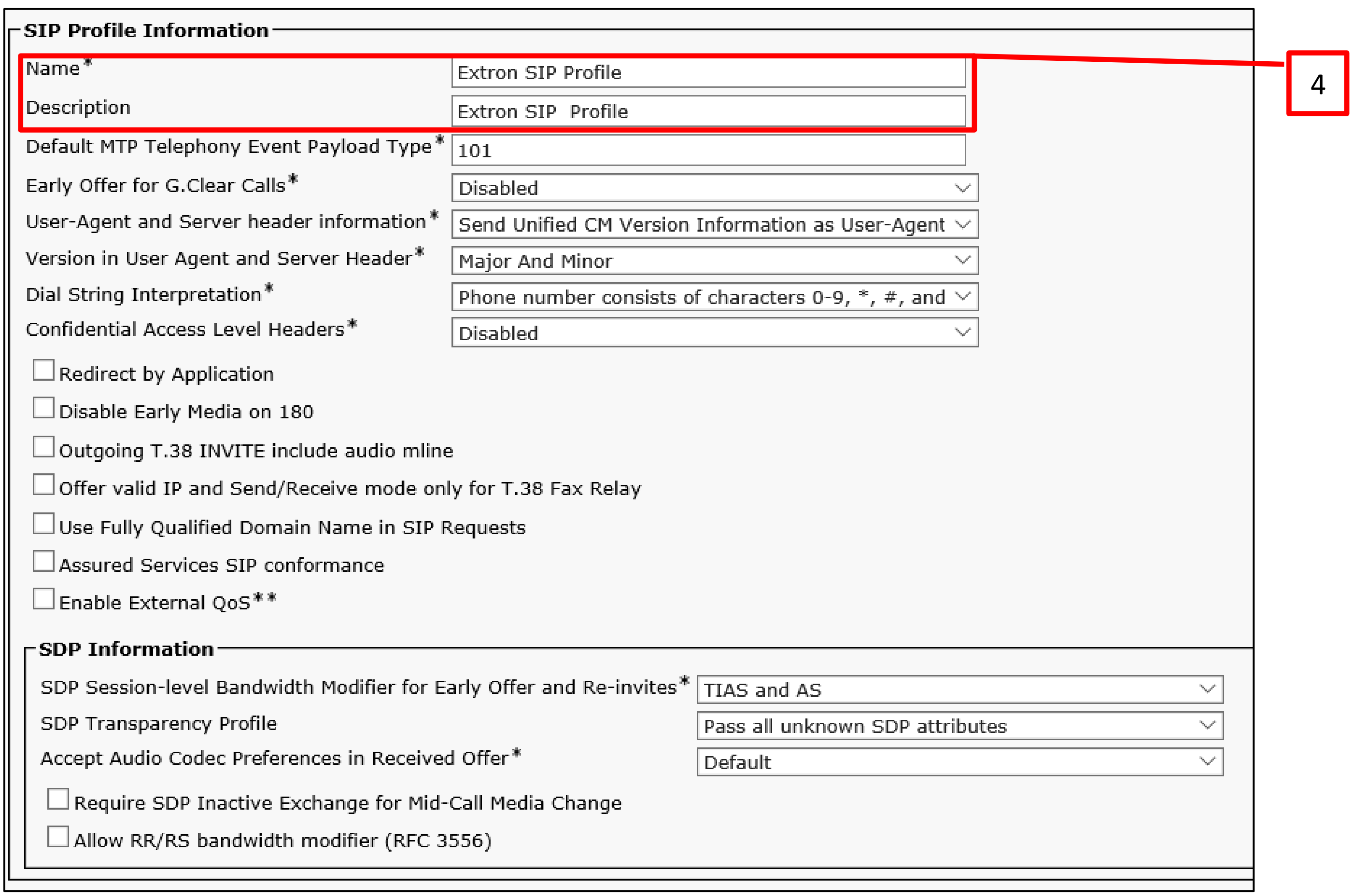
2) Set a password and PIN.

3) Leave other values at the default CUCM end user settings.

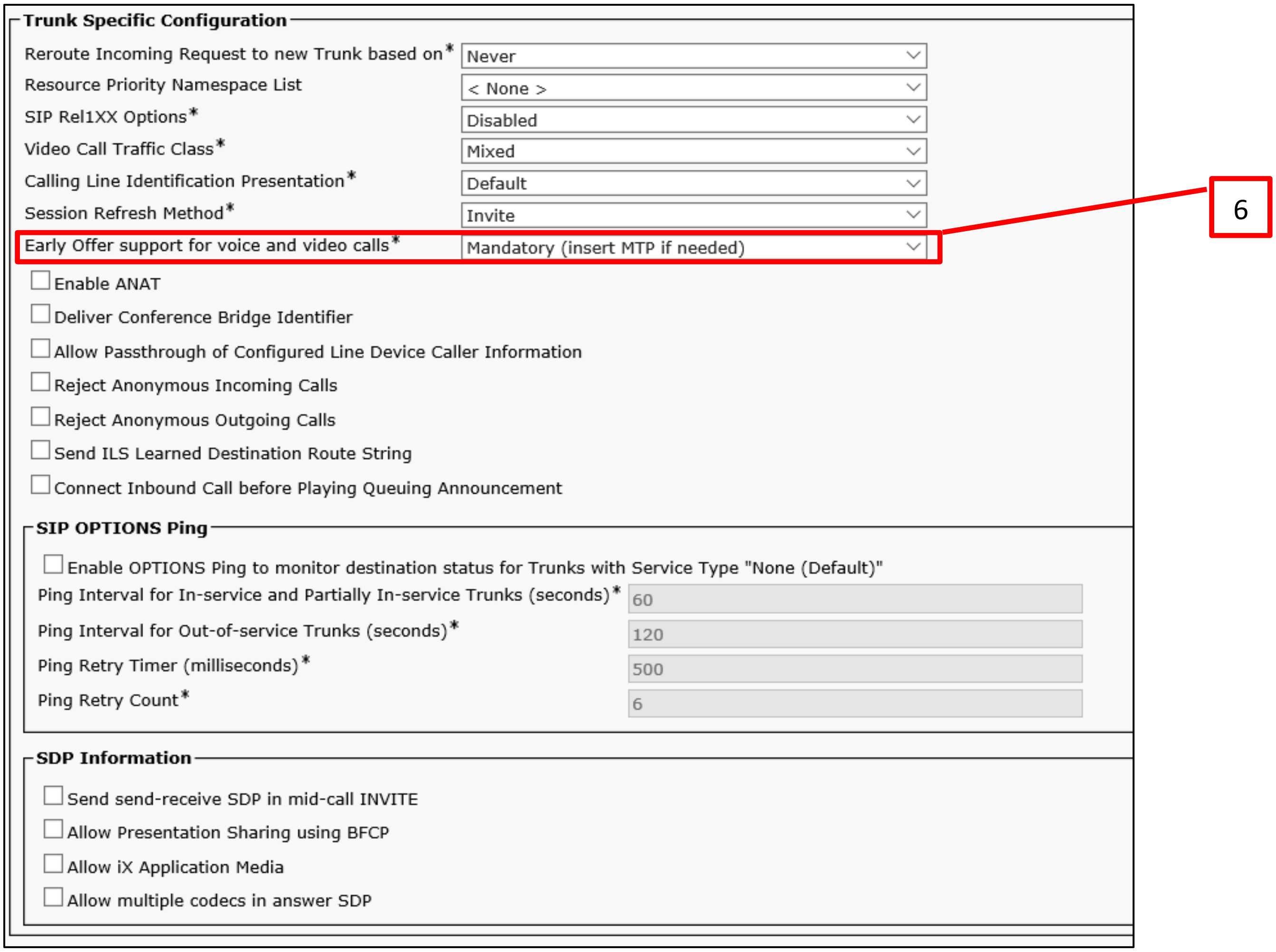
2.3 SIP Profile

A SIP profile is required for the DMP 128 Plus.

1) In the CUCM interface, select **Device** followed by **Device Settings**.
2) Select **SIP Profile**.
3) Locate the **Standard SIP Profile** and make a copy.
4) In the **SIP Profile Configuration** window, assign the copy profile a suitable name, e.g. *Extron SIP Profile*.
5) Specify the following settings:



6) When configuring the Trunk, **Early Offer Support must be set to Mandatory**. An MTP should be used if required:



7) Apply and save the configuration.

2.4 Add a SIP Phone

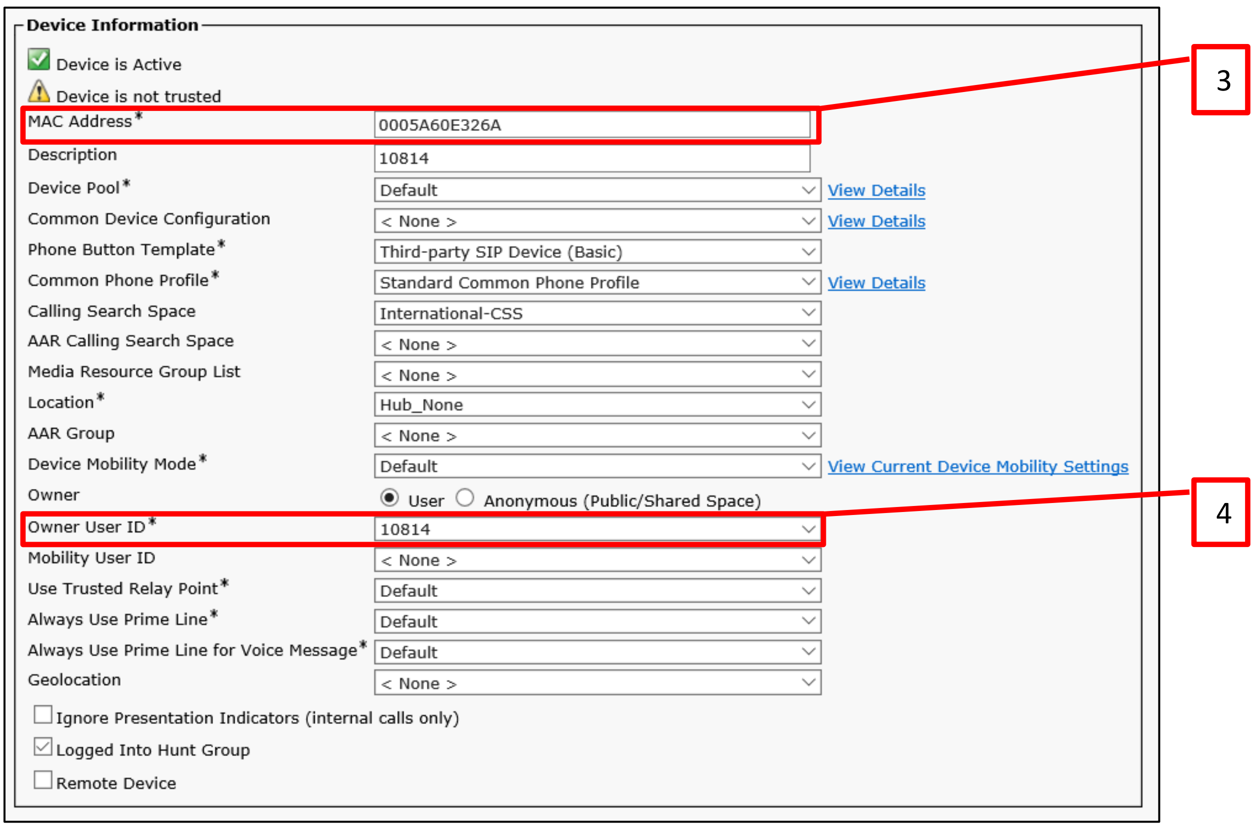
1) Select **Device** followed by **Phone** and **Add New**.

2) Select **Third Party SIP Device (Basic)** before clicking **Next**.

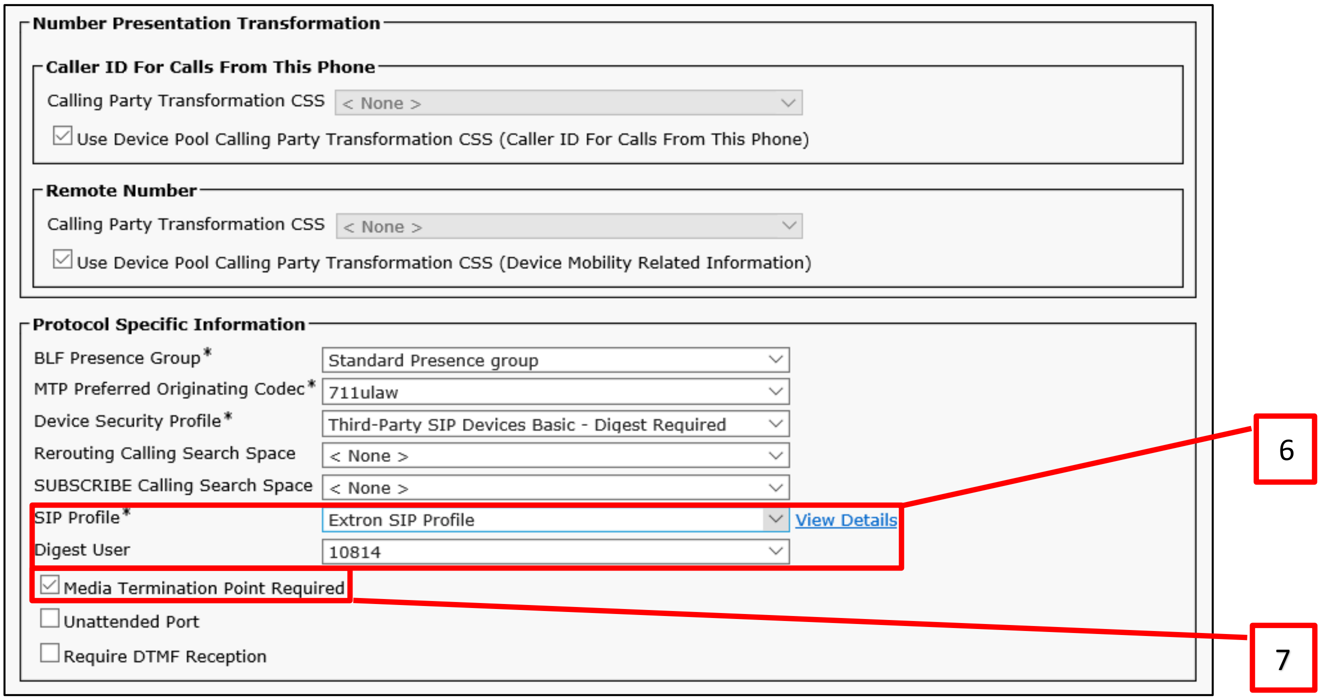
3) Enter the MAC address of the DMP 128 Plus (either LAN1 or LAN2). The MAC address is found on a sticker affixed to the rear panel of the unit.

4) The Owner User ID must be set to match the User ID specified in Section 2.2.

5) The remaining parameters must be set as follows:



6) The **Protocol Specific Information** must be set as follows, where **SIP Profile** is the name of the profile defined in Section 2.3, and **Digest User** is the user ID defined in Section 2.2.



7) Ensure that Media Termination Point Required is checked

8) Save and Apply the configuration.

2.5 Add a Directory Number

1) Click on the link named **Add a New DN**.

2) Specify a directory number.

3) Apply and Save the configuration.

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

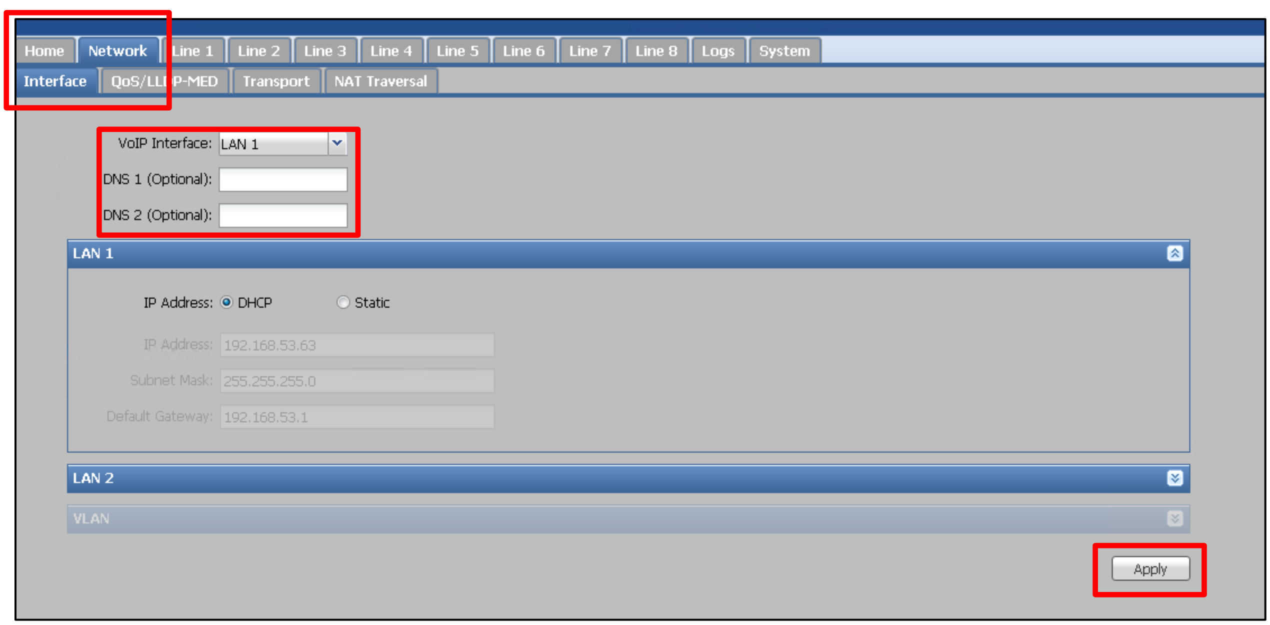
- where 192.168.254.254 in this example is the default IP address of the DMP 128 Plus device.

Up to 8 lines may be configured. Note that each line intended for use will require a unique Directory Number to be specified as part of the CUCM configuration process.

3.1 Network Interface Configuration

Click on the **Network** tab followed by **Interface** tab to set up 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.

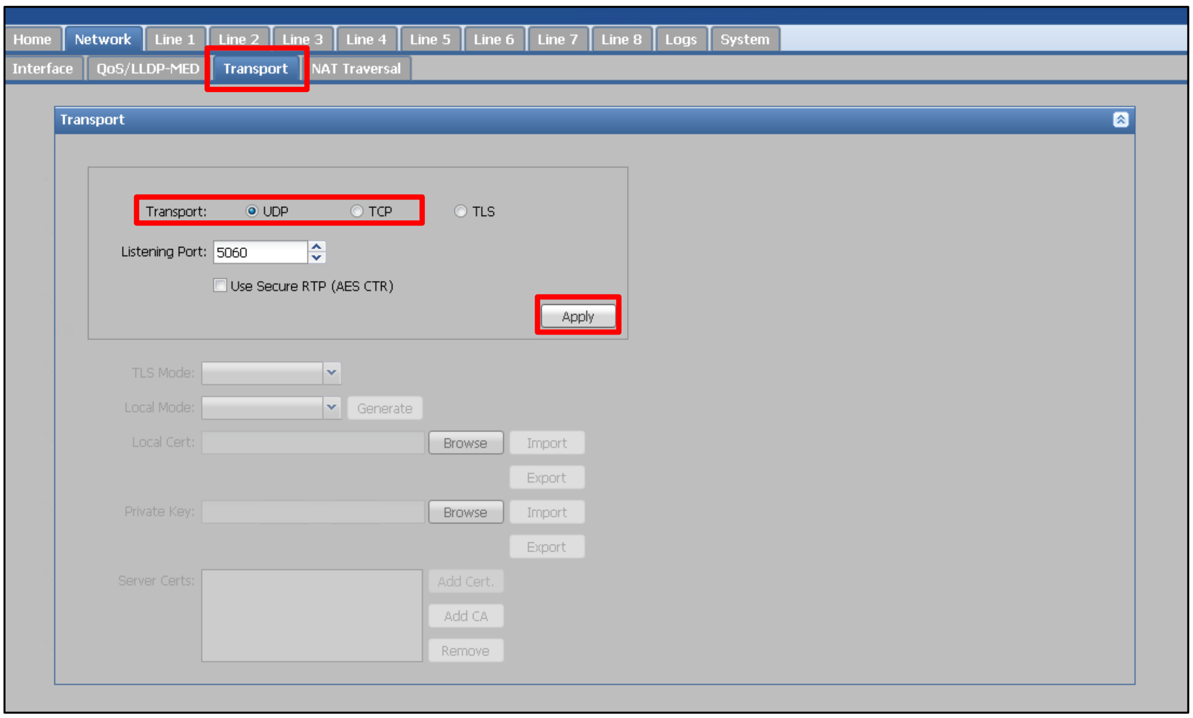
Click **Apply** after making any changes to restart the networking services on the device.



3.2 Transport Configuration

Click on the **Transport** tab to access signaling transport configuration. Set the transport to either UDP or TCP as per the CUCM configuration in Section 2.1. The default transport type for the DMP is UDP. **Note:** At the time of writing, TLS may be used only on Line 1 (a future firmware update will address this issue).

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 numeric DN specified in Section 2.5.

2) **Authentication User Name:** Set this to match the User ID specified Section 2.2.

3) **Authentication Password:** Set to match the password as specified in Section 2.2.

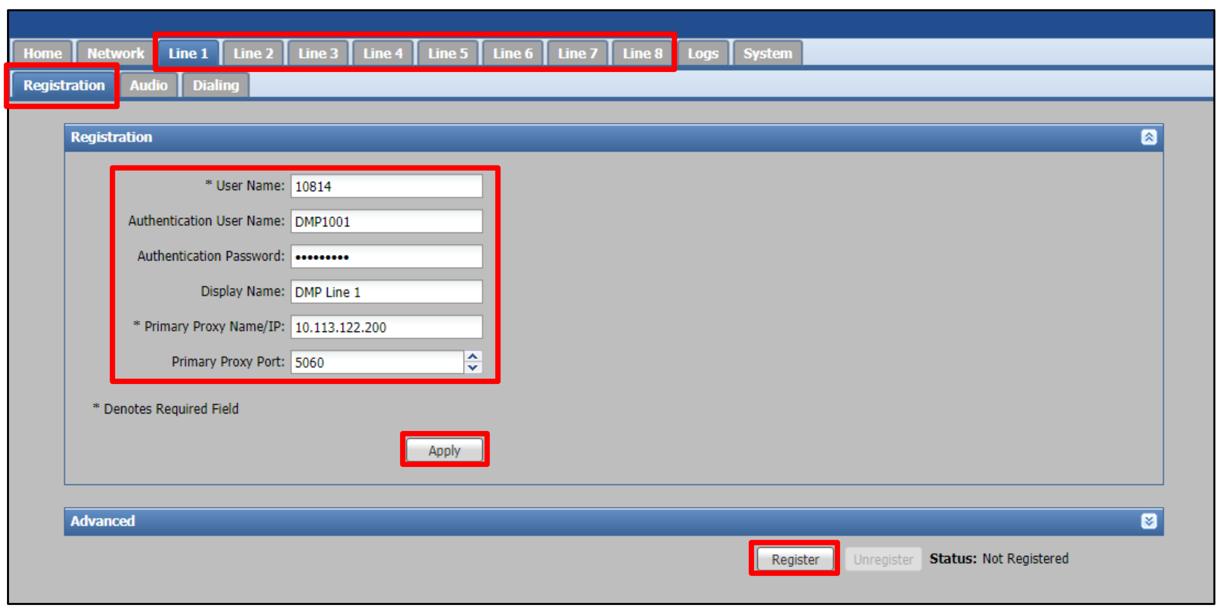
4) **Display Name:** Optional. Specify an identifier for the line if required.

5) **Primary Proxy Name/IP:** Specify either the IP address or domain name of CUCM.

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.

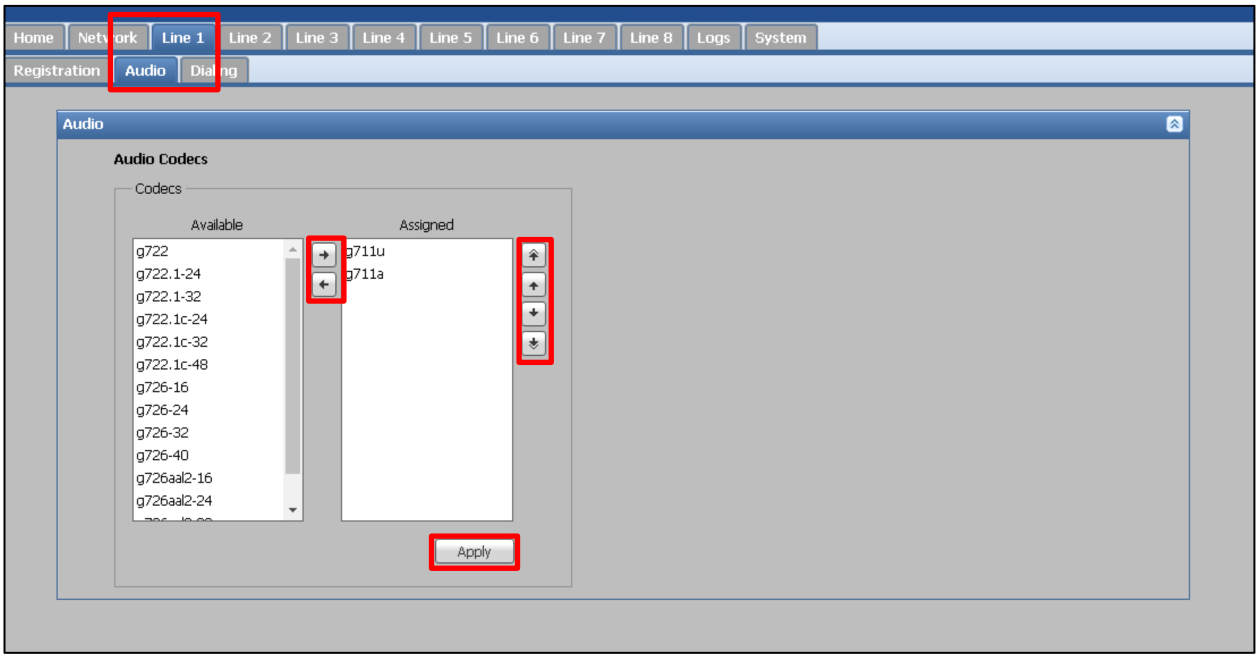
Click the **Register** button to initiate registration to CUCM. If successful, the registration status to the right of the Register/Unregister buttons will indicate *Registered – Primary*.



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.

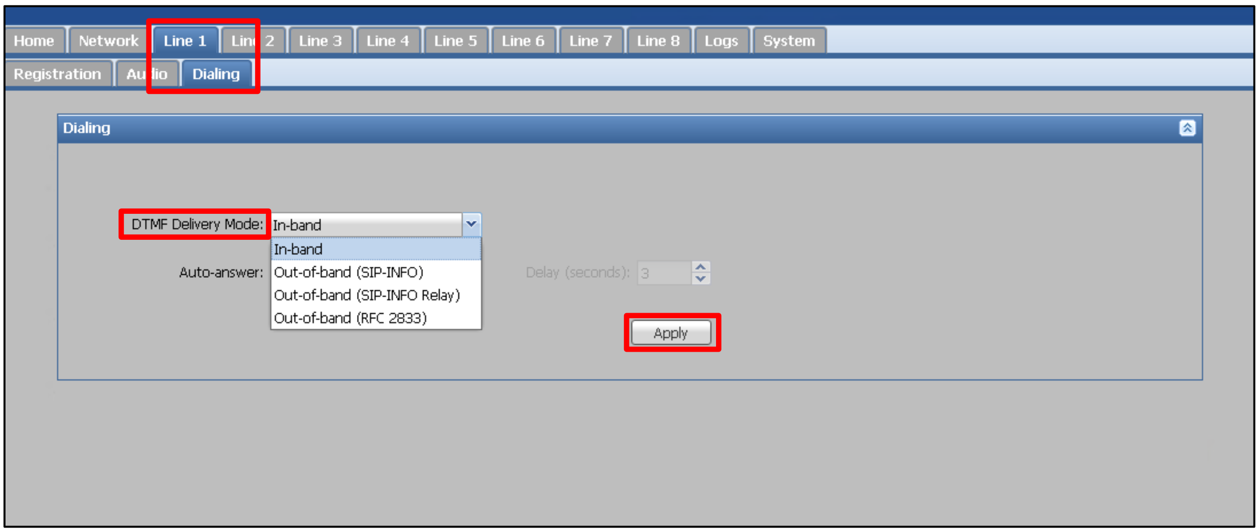


3.5 Dialing

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

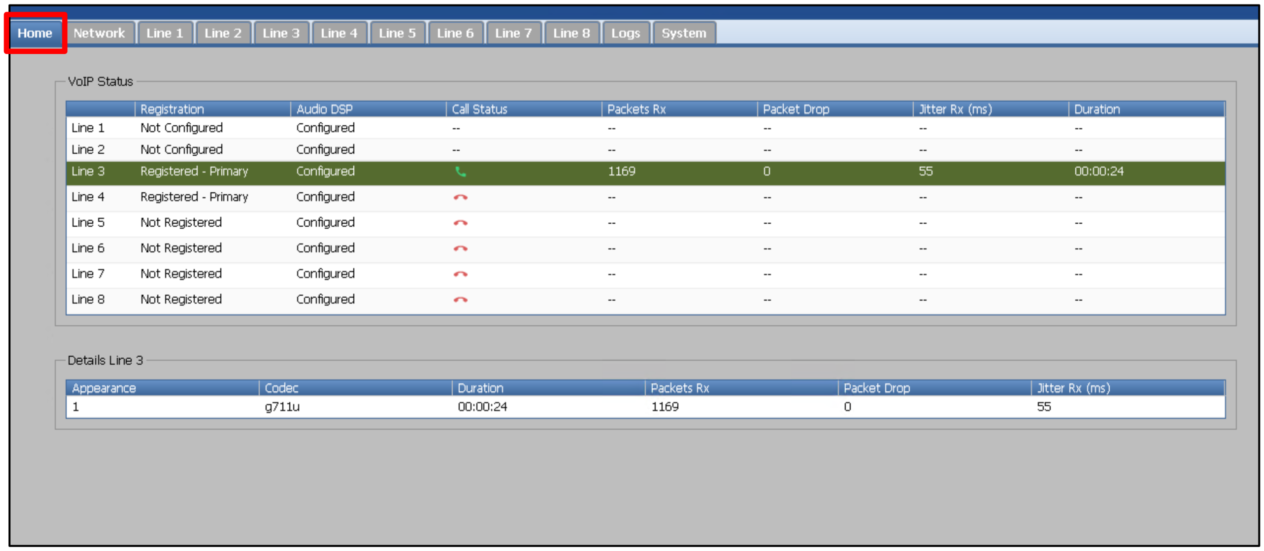
- Out of Band – SIP INFO
- Out of Band – SIP INFO (RELAY)
- Out of Band – RFC 2833

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 CUCM, 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.



3.7 Troubleshooting

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

- Check that the credentials specified as part of the CUCM 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*.

- If the Logs indicate a registration failure, edit the Cisco account so that User ID and Digest User are the same as the Directory Number (DN). Then, update both the User Name **and** Authentication Name fields on the DMP with the Directory Number (DN). Although this is not a prerequisite for registration, it may assist in identifying configuration issues.

Appendix A: RTP Port Range

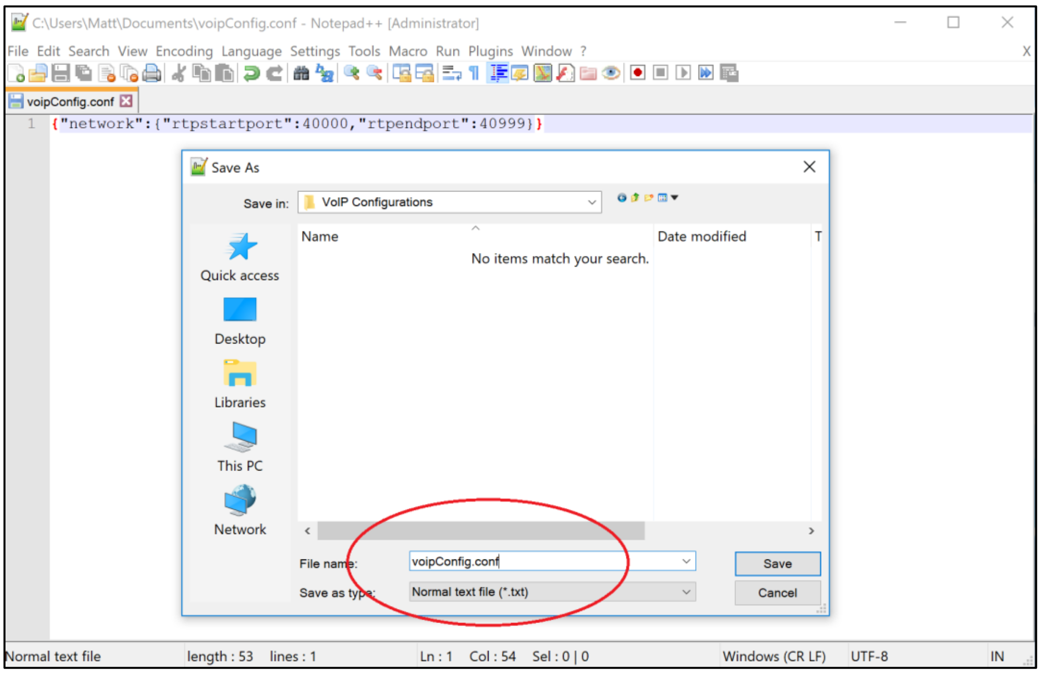
The default port range for VoIP RTP traffic on the DMP is **50000 – 50999**. To change this range, the following steps must be carried out.

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}}
   ```

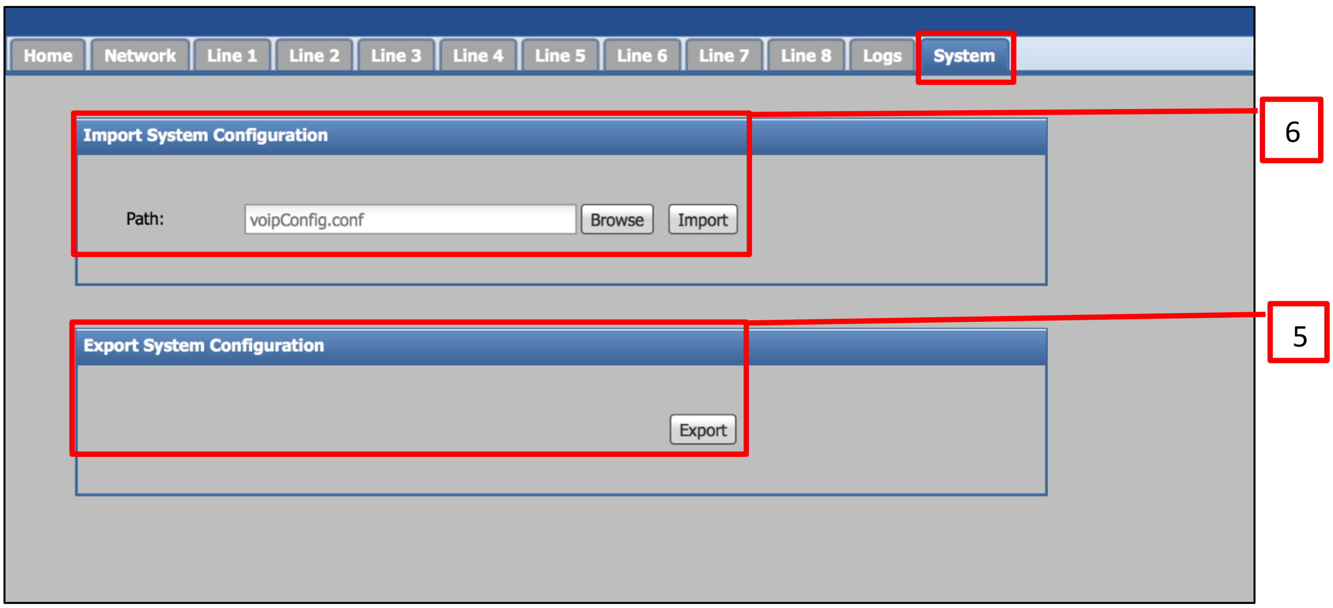
3. Save the file as **voipConfig.conf**.



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.

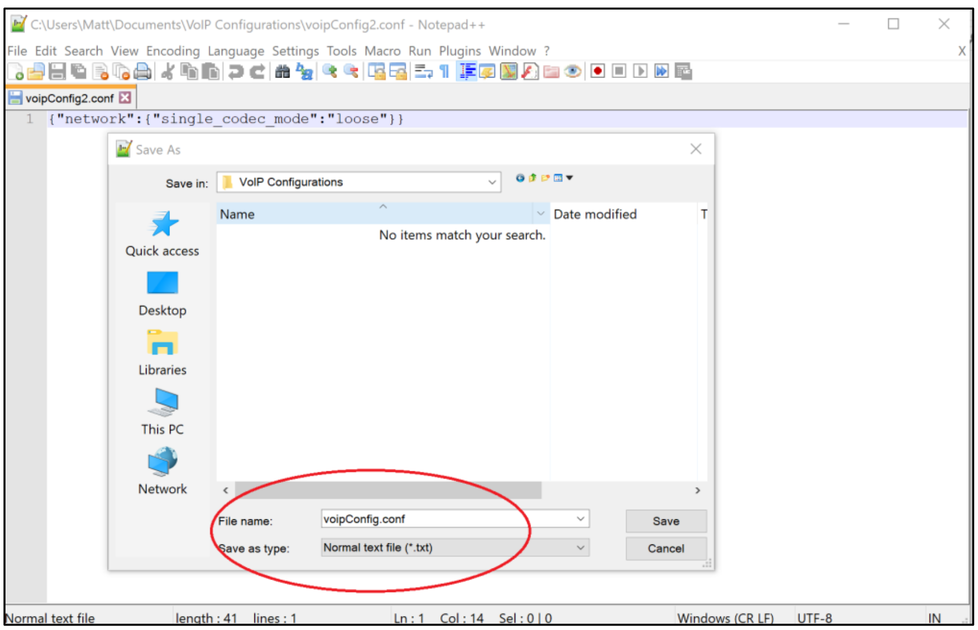
To change the outgoing call terminate mode from BYE to CANCEL (default is BYE), the following steps must be carried out. **Requires FW 1.01.0007-b004 or later.**

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}}
```

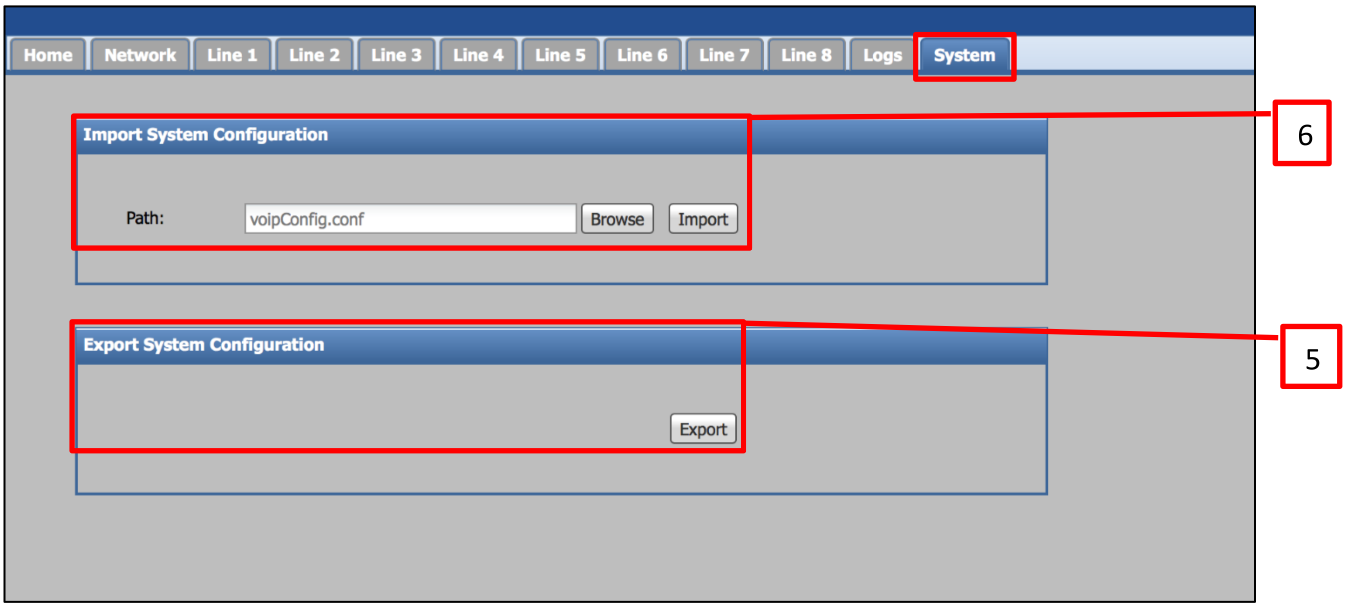
3. Save the file as **voipConfig.conf**.



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

5. 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}}
```

Appendix C: 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 FW 1.02.0001-b001 or later.**

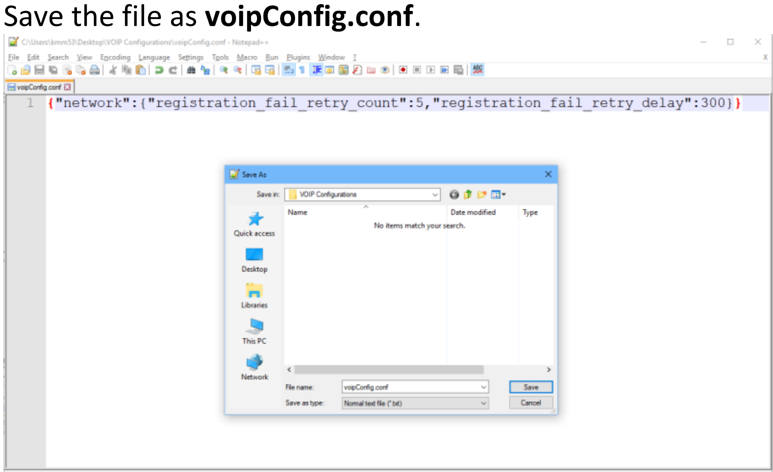
1. Create a new blank text file using a suitable basic text editor

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

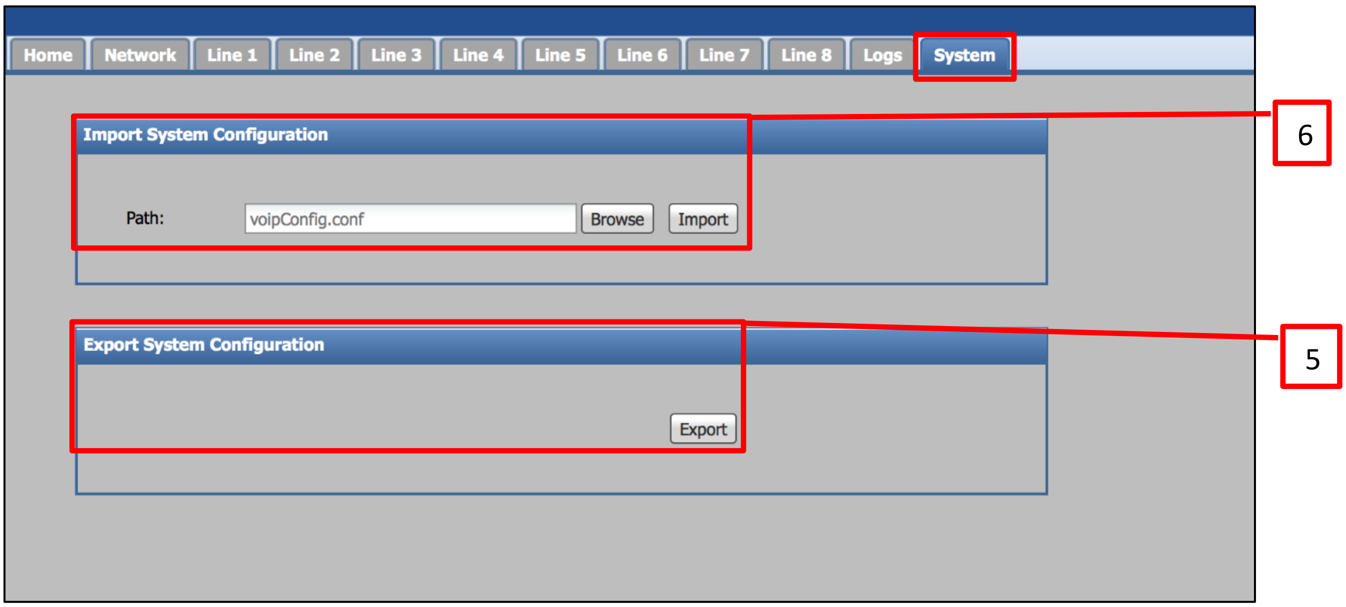
3. Save the file as **voipConfig.conf**.



4. Navigate to the DMP 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.



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

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

```
{"network":{"registration_fail_retry_count":0,"registration_fail_retry_delay":200}}
```