

DMP 128 Plus C V

DMP 128 Plus C V AT

ShoreTel Configuration Guide

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Revision Log

Date	Version	Notes
Sep. 21 st 2018	1.0	First Release: Applies to Firmware 1.01.0010
Dec. 4 th 2018	1.1	Updated ShoreTel screen shots and annotation
Dec. 6 th 2018	1.1.1	Layout and language changes for emphasis
July 26 th 2019	1.1.2	Added Appendix C



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1.0 Introduction

This document provides essential instructions for registering DMP 128 Plus C V (AT) VoIP lines with ShoreTel Connect ONSITE systems from **Version 13** onwards.

DMP 128 Plus C V / C V AT Firmware Version **1.02.0001-b001** or higher is required.

2.0 Configuring ShoreTel for DMP 128 Plus C V (AT) VoIP Registration

Required: Prior to proceeding with this guide, contact ShoreTel or the VoIP administrator in order to add or purchase SIP extensions for use with the DMP 128 Plus C V (AT) as required. The DMP behaves as a **3rd party SIP device**.

2.1 SIP Licensing

To verify that the number of required SIP licenses are available on the system, log on to the ShoreTel Connect ONSITE administration page and navigate to **Administration – System Parameters – Licenses – Requirements**. Verify the number of SIP Phone Licenses available.

License Requirements Help		
License Requirement List		
License Registration	Request License Keys	
Name	Required	Purchased
Keyed Licenses:		
ShoreWare System License (unknown) - Missing System Key	1	1
ShoreWare Additional Site License	6	10
ShoreWare Extension License	551	600
ShoreWare Mailbox License	424	600
ShoreWare SoftPhone License	11	50
ShoreWare Additional Language License	0	13
ShoreWare Remote Web Reporting License	1	1
ShoreWare Mobile Call Manager License	52	55
ShoreWare SIP Phone License	0	10
ShoreWare SIP Trunk License	50	54
ShoreWare Standard Resolution Video License	5	10
ShoreWare High Resolution Video License	2	0
ShoreWare Operator Call Manager License	24	0
ShoreWare Professional Call Manager License	15	19
ShoreWare Workgroup Agent Call Manager License	5	0
ShoreWare Workgroup Supervisor Call Manager License	4	0
Self-audited Licenses:		
ShoreWare Personal Call Manager Software	504	700
ShoreWare Remote Server Software	2	100
ShoreWare TAPI Application Server	1	1
Shoreware Phone API License	8	10

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2.2 Creating a New User

Navigate to **Administration – Users – Individual Users**. (NOTE: If multiple sites are available, select the desired location from the corresponding dropdown menu and click **Go** before continuing).

On the **Edit User** page, shown below, complete the following fields:

- 1) **Number** – The extension number required for the VoIP line
- 2) **First Name** – Use the extension number specified in (1)
- 3) **Last Name** – Specify an additional identifier for the line (e.g. DMP 128)
- 4) **License Type** – Select **Extension Only**
- 5) **Access License** – Select **Personal**
- 6) **Caller ID** – Set the desired caller ID for the line

Users
Edit User

New Copy Save Delete Reset

General Personal Options Distribution Lists Workgroups Refresh

First Name: 10814

Last Name: DMP 128

Number: 10814

License Type: Extension Only

Access License: Personal Enable Contact Center Integration

Caller ID: +1 (555) 123-4567 (e.g. +1 (613) 946-1000)

DID Range: View System Directory

DID Number: (Range: +14083351000 - 14083351999)

PSTN Failover: None

Define matching passwords for the line in the **Client Password** and **SIP Password** fields.

Client Username:

Client Password:

Voice Mail Password:

SIP Password:

Click the **Save** button at the top of the page to complete the account creation process.

2.3 Setting the Call Stack (Number of Appearances)

The DMP 128 Plus supports up to a maximum of **8** appearances on a single VoIP line. It is important to specify the desired number of appearances within the ShoreTel user configuration.

Navigate to **Edit User – Personal Options**. In the **Current Call Stack Size** field, specify the required number of appearances, e.g. 4.



The screenshot shows the 'Users' configuration interface. At the top, there are buttons for 'New', 'Copy', 'Save', 'Delete', and 'Reset'. Below these are tabs for 'General', 'Personal Options', 'Distribution Lists', and 'Workgroups'. The 'Personal Options' tab is active. Under this tab, the 'User Name' is '10814 DMP 128' and the 'Extension' is '10814'. Under the 'Call Control Options' section, the 'Current Call Stack Size' is set to '4', which is highlighted with a red rectangular border.

Click the **Save** button at the top of the page to confirm any changes.

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>

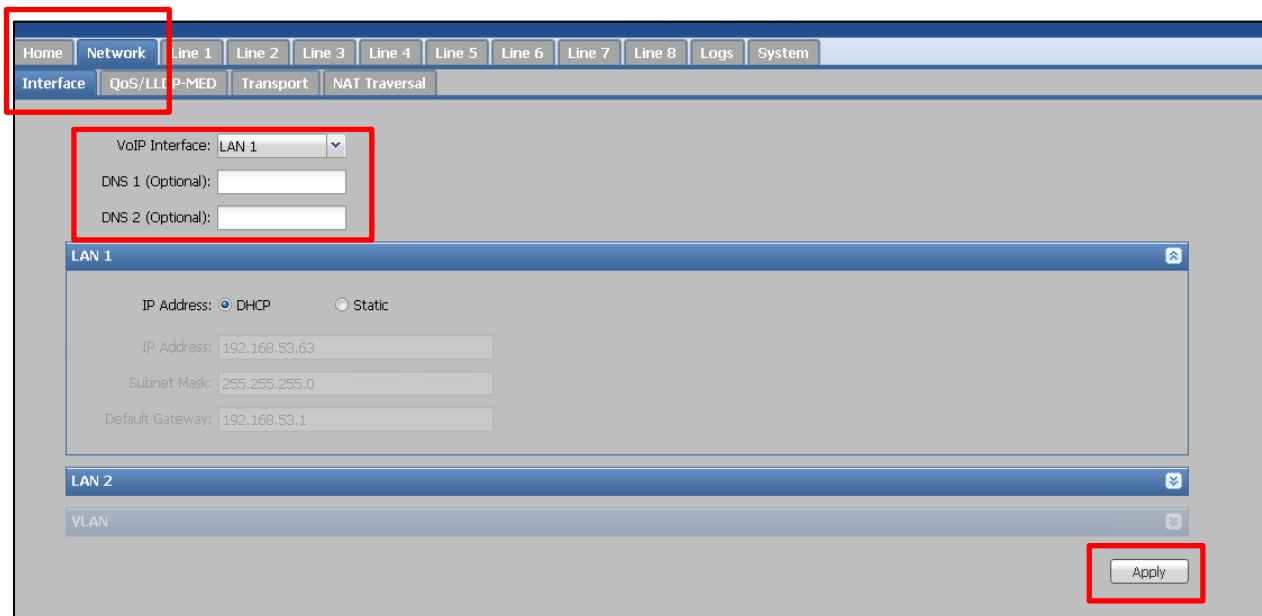
- where 192.168.254.254 in this example is the default IP address of the DMP 128 Plus device.

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 ShoreTel configuration process (Section 2.0).

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.

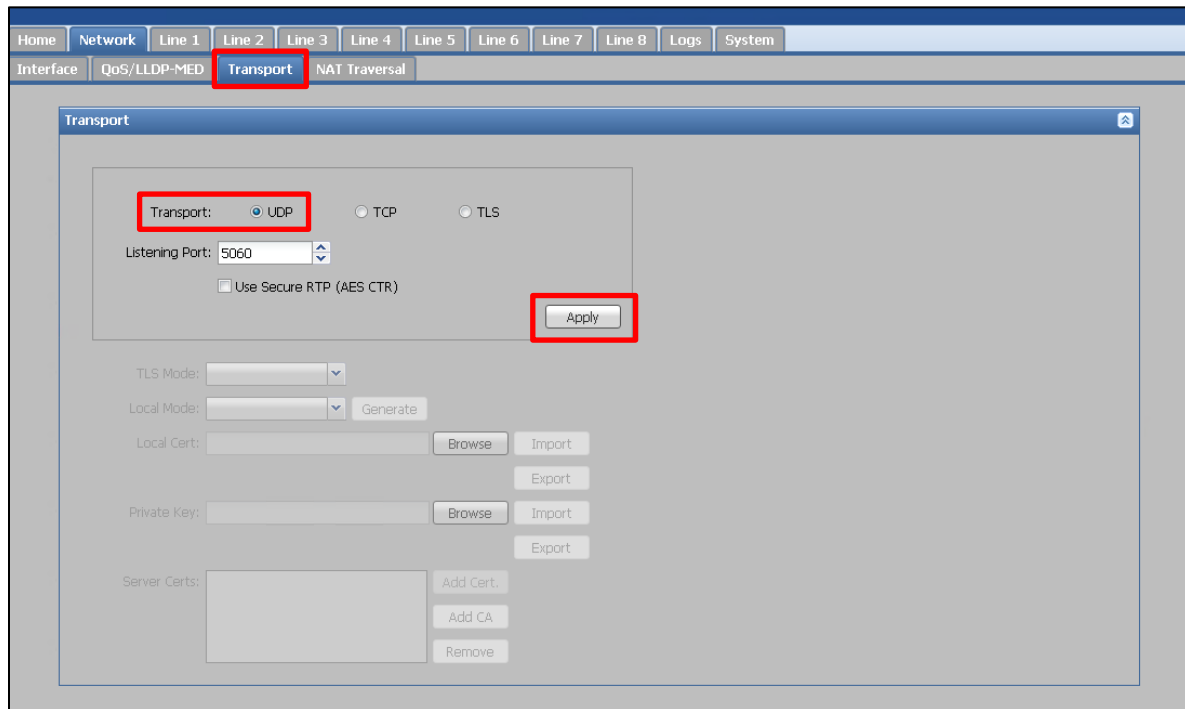
Click **Apply** after making any changes in order to restart the networking services on the device.



3.2 Transport Configuration

Click on the **Transport** tab to access signaling transport configuration. Check that the transport is set to 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**. Refer to the credentials specified in Section 2.0.

- 1) **User Name:** Set this to match the **Number** from Section 2.2.
- 2) **Authentication Name:** Set this to match the **Number** from Section 2.2.
- 3) **Authentication Password:** Set to match the **SIP/Client Password** from 2.2.
- 4) **Display Name:** Optional. Specify an identifier for the line if required.
- 5) **Primary Proxy Name/IP:** Enter the address of the ShoreTel server.
- 6) **Primary Proxy Port:** Specify the SIP port number of the ShoreTel server (default **5060**).

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

The screenshot shows the ShoreTel configuration interface. At the top, there is a navigation bar with tabs for Home, Network, Line 1, Line 2, Line 3, Line 4, Line 5, Line 6, Line 7, Line 8, Logs, and System. The 'Line 1' tab is selected and highlighted with a red box. Below this, there are sub-tabs for Registration, Audio, and Dialing, with 'Registration' also highlighted by a red box. The main content area is titled 'Registration' and contains the following fields:

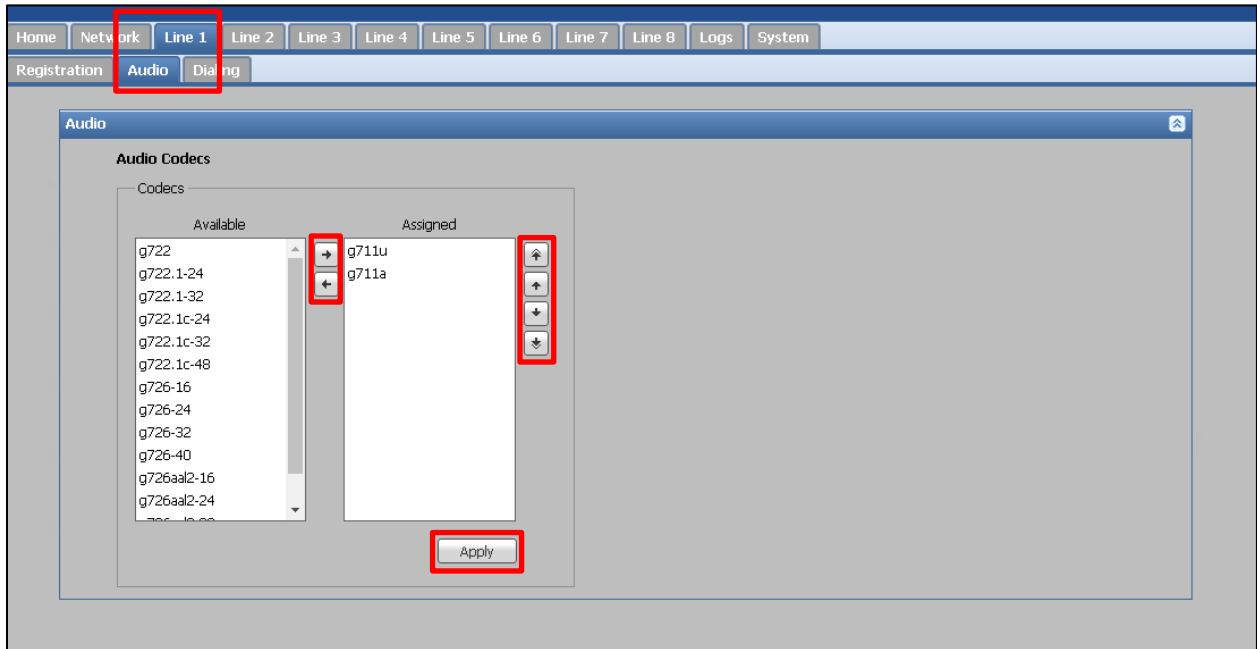
- * User Name: 10814
- Authentication User Name: 10814
- Authentication Password: [masked]
- Display Name: DMP Line 1
- * Primary Proxy Name/IP: 10.113.122.200
- Primary Proxy Port: 5060

A red box highlights the 'Apply' button at the bottom of the registration form. Below the form, there is an 'Advanced' section with a 'Register' button (highlighted with a red box), an 'Unregister' button, and a 'Status: Not Registered' indicator.

3.5 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.



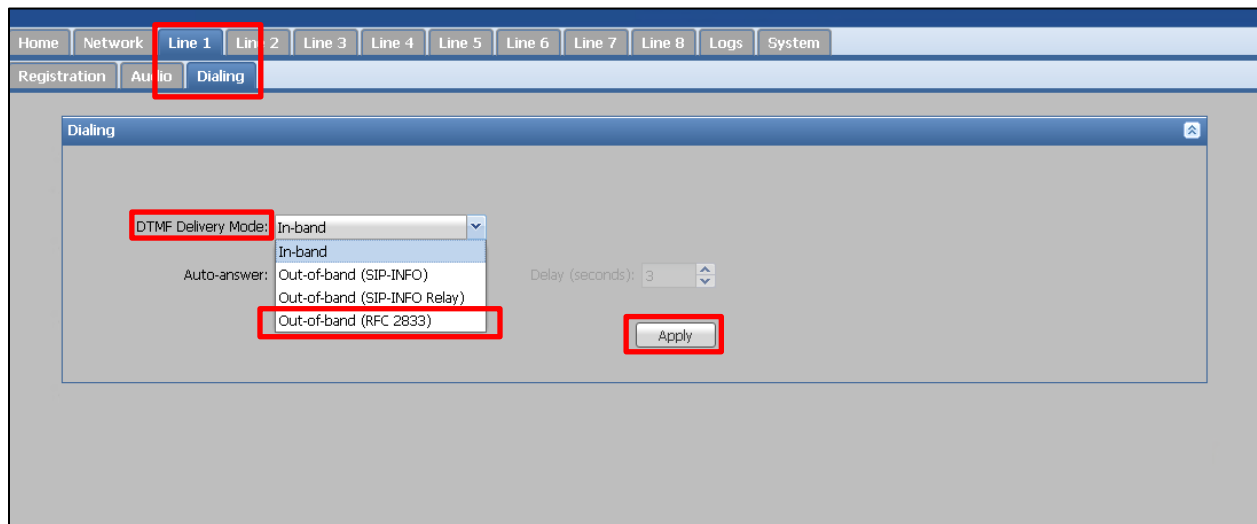
3.6 Dialing

Use the **Dialing** tab to specify the DTMF method. The options are as follows:

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

The DTMF Delivery Mode **must** be set to **Out-of-band (RFC 2833)**.

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



3.7 System Overview

Once all required lines have been registered to the ShoreTel 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.

The screenshot shows the ShoreTel management interface. At the top, there is a navigation bar with tabs for Home, Network, Line 1 through Line 8, Logs, and System. The 'Home' tab is selected and highlighted with a red box. Below the navigation bar, the 'VoIP Status' section displays a table with the following data:

	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		--	--	--	--

Below the VoIP Status table, the 'Details Line 3' section shows a table with the following data:

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

3.8 Troubleshooting

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

- Check that the credentials specified in Section 2.0 are correctly entered into the registration fields for each line.
- Check network interface settings, including DNS fields.
- 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*.

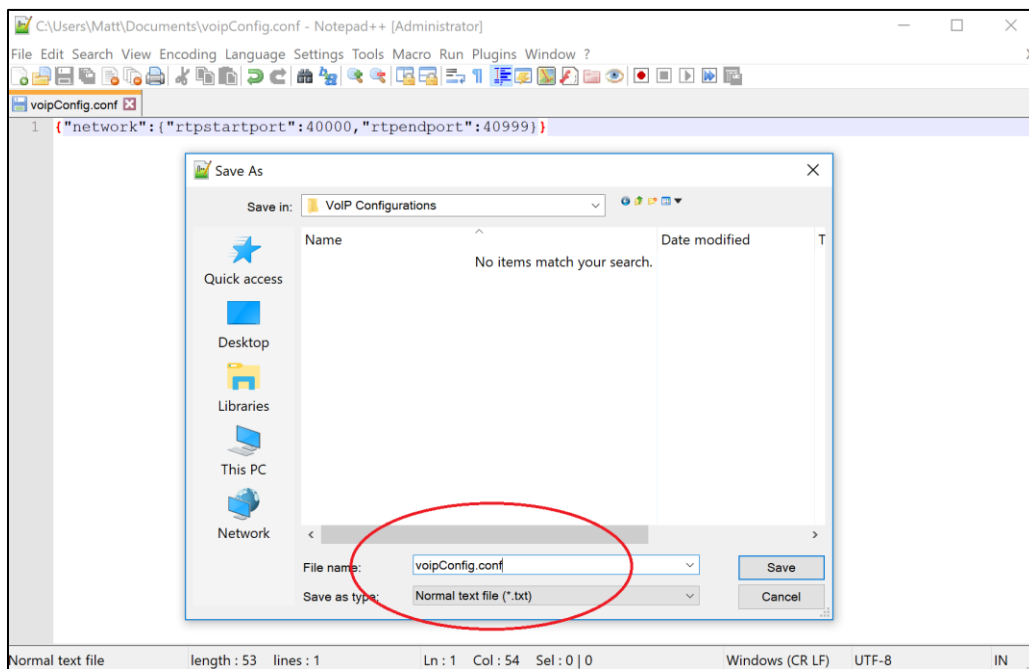
Appendix A: RTP Port Range

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

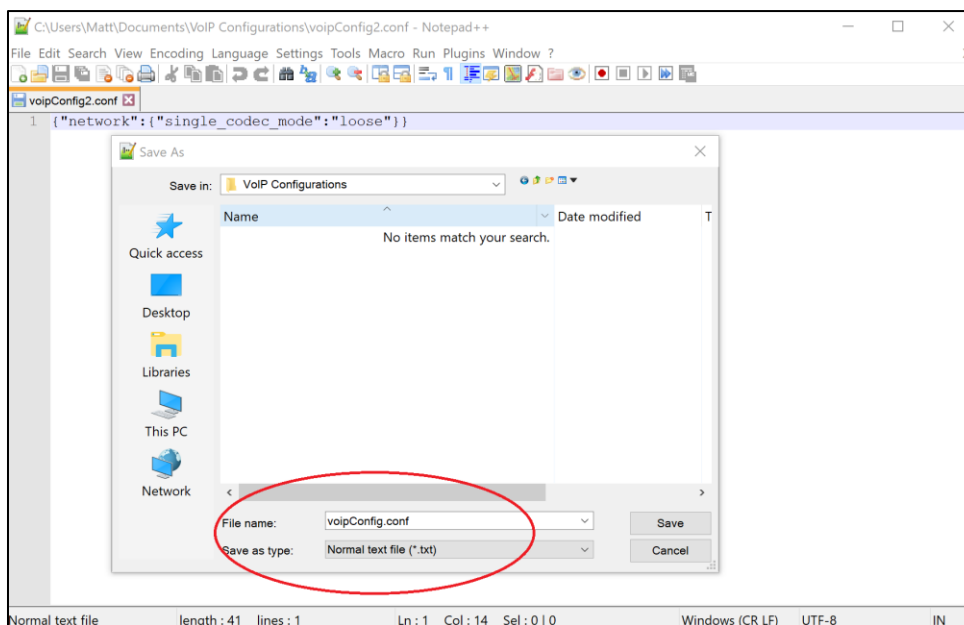
Some call server configurations require the use of a CANCEL SIP message to terminate unanswered outgoing calls, rather than a traditional BYE message. This requirement will become apparent if far end devices continue to ring after the DMP has ended the call.

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. 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

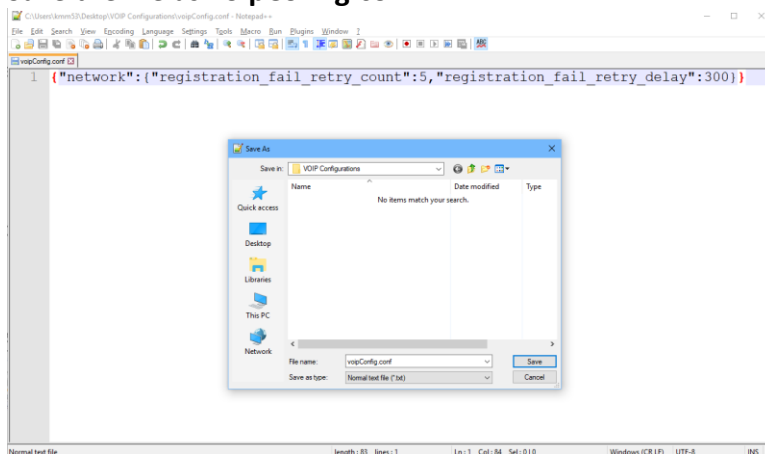
- i. Example above is set to five (5) reconnections attempts
- ii. If this is set to zero (0), the feature is disabled
- iii. Valid Range of values: 0 - 99

- b. registration_fail_retry_delay":300

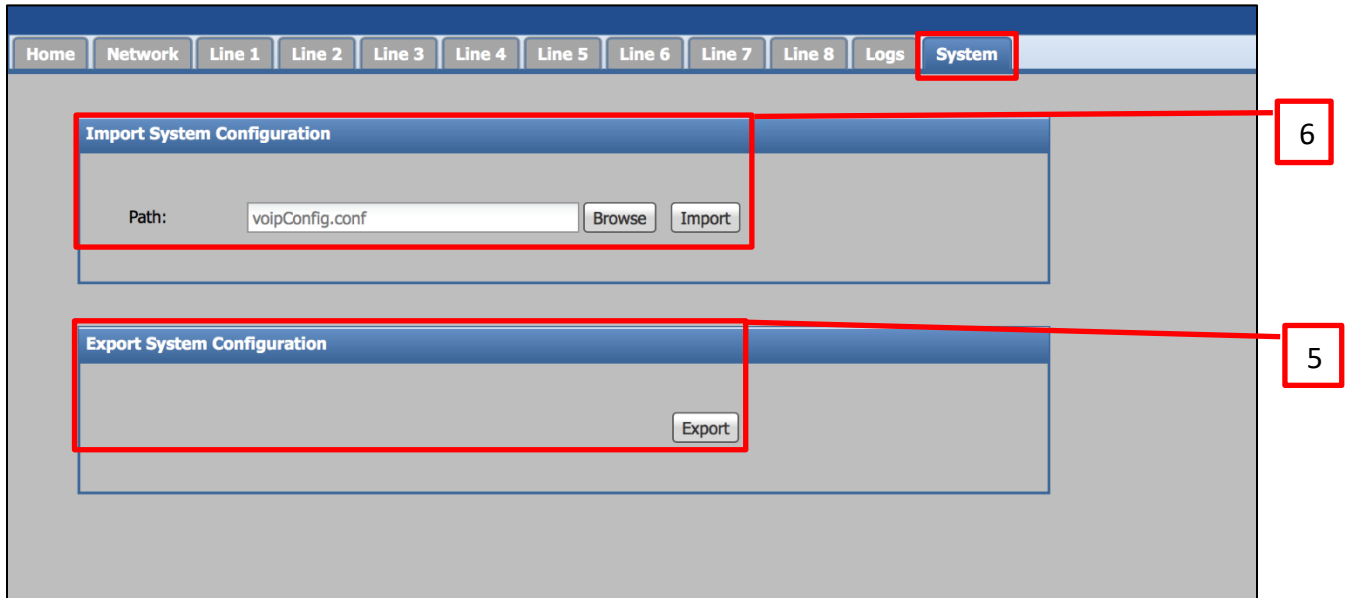
Amount time between registration attempts in seconds

- 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}}
```