### Revision Log

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1.0 Introduction

This document provides essential instructions for registering the VoIP lines of DMP Plus Series, C V and C V AT models, as GoToConnect cloud-based SIP extensions.

DMP Plus Relates to the following products:

- DMP 128 Plus C V / C V AT – Requires Firmware Version 1.04.0001 or higher
- DMP 128 FlexPlus C V AT
- DMP 64 Plus C V / C V AT

2.0 Configuring GoToConnect for DMP Plus Series VoIP Registration

Prior to proceeding with this guide, contact GoToConnect in order to add or purchase SIP extensions for use with the DMP Plus Series. The DMP Plus Series behaves as a 3rd party SIP device. The following credentials are required for each line that is to be used on the system –

1) SIP Server and Port Number
2) Outbound Proxy and Port Number
3) Username (Authorization ID)
4) Password
3.0 Configuring DMP Plus Series 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:


- where 192.168.254.254 in this example is the default IP address of the DMP 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 IP Office 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 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 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 provided by GoToConnect (Section 2.0).

1) **User Name**: Set this to match the **User Name** from GoToConnect.

2) **Authentication Name**: Same as the **Username** from GoToConnect

3) **Authentication Password**: Set to match the **Password** from GoToConnect.

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

5) **Primary Proxy Name/IP**: Enter the **SIP Server** from GoToConnect

6) **Primary Proxy Port**: Specify the **SIP Domain Port Number - 5060**.

Once the above settings have been entered, click the **Apply** button to save to the device. **Do not attempt to register the line at this stage.**
3.4 Outbound Proxy

NOTE:
The following steps must be carried out in order to set the **Outbound Proxy** and **Port Number** required for GoToConnect registration.

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

2. Enter the following text into the document, replacing the ‘1’ in ‘line1’ with the required DMP Plus Series line ID (1 – 8):

   ```json
   {"users":[{"id":"line1","outbound_proxy":"sip10.GoToConnect.com","outbound_proxy_port":"5060"}]
   ```

   - Replace “sangardemo.GoToConnect.rtcfront.net” with the **Outbound Proxy Address** provided by GoToConnect (Section 2.0), if different.
   - Change “5060” to the **Outbound Proxy Port** provided by GoToConnect (Section 2.0), if different.

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 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 Plus Series with the new outbound proxy settings. A notification will appear once the settings have applied successfully.

8. Return to the **Line - Registration** tab and click **Register** to complete the registration process.
### 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.

Currently GoToConnect supports **G.711u**, **G.711a** and **G.722**

Click the **Apply** button to commit any changes to the device.
3.6 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

GoToConnect will require the use 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.
3.7 System Overview

Once all required lines have been registered to GoToConnect, 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.

3.8 Troubleshooting

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

- Check that the credentials provided by GoToConnect 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 Plus Series 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 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 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 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 –

   ```json
   {"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.

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}}