DMP 128 Plus C V
DMP 128 Plus C V AT

Interactive Intelligence Configuration Guide

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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 1.02.0001-b001 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**.

![Basic Station License and Enable Licenses checkboxes](image)

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

![Phone Number Classifications](image)
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 -


- 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.

<table>
<thead>
<tr>
<th>Line</th>
<th>Registration</th>
<th>Audio CIP</th>
<th>Call Status</th>
<th>Packets Rx</th>
<th>Packet Drop</th>
<th>Bit_rx (mb)</th>
<th>Duration</th>
</tr>
</thead>
<tbody>
<tr>
<td>1</td>
<td>Not configured</td>
<td>Configured</td>
<td>--</td>
<td>--</td>
<td>--</td>
<td>--</td>
<td>--</td>
</tr>
<tr>
<td>2</td>
<td>Not configured</td>
<td>Configured</td>
<td>--</td>
<td>--</td>
<td>--</td>
<td>--</td>
<td>--</td>
</tr>
<tr>
<td>3</td>
<td>Registered</td>
<td>Configured</td>
<td>--</td>
<td>1149</td>
<td>0</td>
<td>53</td>
<td>00:00:24</td>
</tr>
<tr>
<td>5</td>
<td>Not Registered</td>
<td>Configured</td>
<td>--</td>
<td>--</td>
<td>--</td>
<td>--</td>
<td>--</td>
</tr>
<tr>
<td>6</td>
<td>Not Registered</td>
<td>Configured</td>
<td>--</td>
<td>--</td>
<td>--</td>
<td>--</td>
<td>--</td>
</tr>
<tr>
<td>7</td>
<td>Not Registered</td>
<td>Configured</td>
<td>--</td>
<td>--</td>
<td>--</td>
<td>--</td>
<td>--</td>
</tr>
<tr>
<td>8</td>
<td>Not Registered</td>
<td>Configured</td>
<td>--</td>
<td>--</td>
<td>--</td>
<td>--</td>
<td>--</td>
</tr>
</tbody>
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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) -

   ```json
   {"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 -

   ```json
   \{"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:

```json
{"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 –

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

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