DMP 128 Plus C V
DMP 128 Plus C V AT
Cisco CUCM Configuration Guide

REVISION: 1.2.3
DATE: JULY 26, 2019
## DMP 128 Plus C V / C V AT – Cisco CUCM

### Revision Log

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

```
<table>
<thead>
<tr>
<th>SIP Profile Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Name</td>
</tr>
<tr>
<td>Description</td>
</tr>
<tr>
<td>Default MTP Telephony Event Payload Type</td>
</tr>
<tr>
<td>Early Offer for G.Clear Calls</td>
</tr>
<tr>
<td>User-Agent and Server header information</td>
</tr>
<tr>
<td>Version in User Agent and Server Header</td>
</tr>
<tr>
<td>Dial String Interpretation</td>
</tr>
<tr>
<td>Confidential Access Level Headers</td>
</tr>
</tbody>
</table>

Redirect by Application

Disable Early Media on 160

Outgoing T.38 INVITE include audio mline

Offer valid IP and Send/Receive mode only for T.38 Fax Relay

Use Fully Qualified Domain Name In SIP Requests

Assured Services SIP conformance

Enable External QoS

SDP Information

SDP Session-level Bandwidth Modifier for Early Offer and Re-invites | TIAS and AS |
| SDP Transparency Profile | Pass all unknown SDP attributes |
| Accept Audio Codec Preferences in Received Offer | Default |

Require SDP Inactive Exchange for Mid-Call Media Change

Allow RR/RS bandwidth modifier (RFC 3556) |
6) When configuring the Trunk, **Early Offer Support must be set to Mandatory.** An MTP should be used if required:

<table>
<thead>
<tr>
<th>Trunk Specific Configuration</th>
</tr>
</thead>
<tbody>
<tr>
<td>Reroute Incoming Request to new Trunk based on*</td>
</tr>
<tr>
<td>Resource Priority Namespace List</td>
</tr>
<tr>
<td>SIP Rel1XX Options*</td>
</tr>
<tr>
<td>Video Call Traffic Class*</td>
</tr>
<tr>
<td>Called Party Identification Presentation*</td>
</tr>
<tr>
<td>Session Refresh Method*</td>
</tr>
<tr>
<td>Early Offer Support for Voice and Video Calls*</td>
</tr>
</tbody>
</table>

- Enable ANAT
- Deliver Conference Bridge Identifier
- Allow Passthrough of Configured Line Device Caller Information
- Reject Anonymous Incoming Calls
- Reject Anonymous Outgoing Calls
- Send ILS Learned Destination Route String
- Connect Inbound Call before Playing Queuing Announcement

<table>
<thead>
<tr>
<th>SIP OPTIONS Ping</th>
</tr>
</thead>
<tbody>
<tr>
<td>Ping Interval for In-service and Partially In-service Trunks (seconds)*</td>
</tr>
<tr>
<td>Ping Interval for Out-of-service Trunks (seconds)*</td>
</tr>
<tr>
<td>Ping Retry Timer (milliseconds)*</td>
</tr>
<tr>
<td>Ping Retry Count*</td>
</tr>
</tbody>
</table>

<table>
<thead>
<tr>
<th>SDP Information</th>
</tr>
</thead>
<tbody>
<tr>
<td>Send send-receive SDP in mid-call INVITE</td>
</tr>
<tr>
<td>Allow Presentation Sharing using BFCP</td>
</tr>
<tr>
<td>Allow IX Application Media</td>
</tr>
<tr>
<td>Allow multiple codecs in answer SDP</td>
</tr>
</tbody>
</table>

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 -


- 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.

![System Overview Table](image)

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) -

   ```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. 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.

   ![Import and Export System Configuration](image)

   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:

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