Spectralink Versity Smartphone

Cisco Unified Communications Manager (CUCM)

Interoperability Guide: Version 12
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<thead>
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About This Guide

This interoperability guide describes the procedures for configuring Spectralink Versity handsets with the Cisco Unified Communications Manager. The overall objective of the interoperability compliance testing is to verify that Spectralink Versity Wireless Telephones function in an environment comprised of a Cisco Unified Communications Manager and various Cisco telephones and PSTN connections. All testing was performed in Spectralink laboratories.

Product Support

Spectralink wants you to have a successful installation. If you have questions, please contact the Customer Support Hotline at 1-800-775-5330.

The hotline is open Monday through Friday, 6 a.m. to 6 p.m. Mountain Time.

For Technical Support: mailto:technicalsupport@Spectralink.com
For Knowledge Base: http://support.Spectralink.com
For Return Material Authorization: mailto:nalarma@Spectralink.com

Spectralink References

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Specific Documents
Spectralink Versity software and support documents are available on the Spectralink support site at http://support.spectralink.com/versity.
Spectralink SAM software and support documents are available on the Spectralink support site at http://support.spectralink.com/sam.

Release Notes accompany every software release and provide the new and changed features and resolved issues in the latest version of the software. Please review these for the most current information about your software.

Spectralink Versity Deployment Guide provides a high-level overview of the deployment process for Spectralink Versity smartphones. This includes the interface with an EMM, the method for getting Versity connected to the wireless LAN, and the interface with the Spectralink Application Management (SAM) server.

Spectralink Applications Management Guide The Spectralink Applications Management (SAM) Guide provides information about every setting and option for the Spectralink applications that are available to the administrator on the SAM server. Time-saving shortcuts, troubleshooting tips and other important maintenance instructions are also found in this document.

The Spectralink Versity User Guide offers comprehensive instructions for using each of the Spectralink Applications deployed on the handsets.

For information on IP PBX and soft switch vendors, see the Spectralink Call Server Interoperability Guide.

Technical Bulletins and Feature Descriptions explain workarounds to existing issues and provide expanded descriptions and examples.

AP Configuration Guides explain how to correctly configure access points and WLAN controllers (if applicable) and identify the optimal settings that support Spectralink Versity smartphone. You can find them on the VIEW Certified webpage.

White Papers
For details on RF deployment please see *The Challenges of Ensuring Excellent Voice Quality in a Wi-Fi Workplace* and *Deploying Enterprise-Grade Wi-Fi Telephony*.

These White Papers identify issues and solutions based on Spectralink’s extensive experience in enterprise-class Wi-Fi telephony. They provide recommendations for ensuring that a network environment is adequately optimized for use with Spectralink devices.

**Cisco Documentation**

Interoperability testing between the Spectralink Versity handsets and the CUCM was conducted using version 12 of the Cisco Unified Communications Manager. Other recent versions (10.0 – 11.x) of CUCM are expected to interoperate successfully with the Versity handsets but may need to be field verified. This document covers only a small subset of the features and functionality available in the Cisco Unified Communications Manager in conjunction with the Spectralink Versity handset interoperability. Please navigate to the Cisco documentation site for the latest Cisco branded documentation:


**Conventions Used In This Document**

**Typography**

A few typographic conventions, listed next, are used in this guide to distinguish types of in-text information.

<table>
<thead>
<tr>
<th>Convention</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><strong>Bold</strong></td>
<td>Highlights interface items such as menus, softkeys, file names, and directories. Also used to represent menu selections and text entry to the handset.</td>
</tr>
<tr>
<td><strong>Italics</strong></td>
<td>Used to emphasize text, to show example values or inputs, and to show titles of reference documents available from the Spectralink Support Web site and other reference sites.</td>
</tr>
<tr>
<td><strong>Underlined blue</strong></td>
<td>Used for URL links to external Web pages or documents. If you click text in this style, you will be linked to an external document or Web page.</td>
</tr>
<tr>
<td><strong>Bright orange text</strong></td>
<td>Used for cross references to other sections within this document. If you click text in this style, you will be taken to another part of this document.</td>
</tr>
<tr>
<td><strong>Fixed-width-font</strong></td>
<td>Used for code fragments and parameter names.</td>
</tr>
</tbody>
</table>

This guide also uses a few writing conventions to distinguish conditional information.
<table>
<thead>
<tr>
<th>Convention</th>
<th>Description</th>
</tr>
</thead>
<tbody>
<tr>
<td><code>&lt;MACaddress&gt;</code></td>
<td>Indicates that you must enter information specific to your installation,</td>
</tr>
<tr>
<td></td>
<td>handset, or network. For example, when you see <code>&lt;MACaddress&gt;</code>, enter your</td>
</tr>
<tr>
<td></td>
<td>handset’s 12-digit MAC address. If you see <code>&lt;installed-directory&gt;</code>, enter</td>
</tr>
<tr>
<td></td>
<td>the path to your installation directory.</td>
</tr>
<tr>
<td><code>&gt;</code></td>
<td>Indicates that you need to select an item from a menu. For example,</td>
</tr>
<tr>
<td></td>
<td><code>Settings&gt; Basic</code> indicates that you need to select <code>Basic</code> from the</td>
</tr>
<tr>
<td></td>
<td><code>Settings</code> menu.</td>
</tr>
</tbody>
</table>
Chapter 1: Overview

System Diagram

Below is a system diagram depicting the lab setup used to test the Spectralink Versity interoperation with the Cisco Unified Communications Manager.
Test Infrastructure Version Information

- Cisco Unified Communications Manager (CUCM) Software Version: 12.0.1.21900-7
  Note: Spectralink has also performed testing of the Versity handset against version 10.0 and 11.0 of the CUCM and will support the integration of Versity with versions of CUCM within that range. This document details the configuration process against CUCM 12.0, though the configuration described herein should apply to both 10.x and 11.x versions of CUCM as well.
- Cisco Unity Connection Software Version: 12.0.1.21900-10
- Spectralink Versity Handset Software Version: 1.0.0.784 (or later)
- Spectralink BizPhone Application Version: 3.4.4592 (or later)
- Motorola 6532 Access Point Software Version: 5.2.3.0-023
Feature Configuration and Test Summary

A description of each feature tested and comments about feature functionality can be found in SIP Feature Configuration and Configuration Parameter Test Details.

<table>
<thead>
<tr>
<th>Features Tested</th>
<th>Supported</th>
</tr>
</thead>
<tbody>
<tr>
<td>Direct to CUCM SIP Registration</td>
<td>Y</td>
</tr>
<tr>
<td>SIP Digest Authentication</td>
<td>Y</td>
</tr>
<tr>
<td>Basic Calls</td>
<td>Y</td>
</tr>
<tr>
<td>Voicemail Integration</td>
<td>Y</td>
</tr>
<tr>
<td>Message Waiting Indication (MWI)</td>
<td>Y</td>
</tr>
<tr>
<td>Call Waiting</td>
<td>Y</td>
</tr>
<tr>
<td>Multiple Calls Per Line Key or Maximum Calls Per Line</td>
<td>Y</td>
</tr>
<tr>
<td>Conference: 3-way</td>
<td>Y</td>
</tr>
<tr>
<td>Transfer: Blind</td>
<td>Y</td>
</tr>
<tr>
<td>Transfer: Announced</td>
<td>Y</td>
</tr>
<tr>
<td>Transfer: Attended</td>
<td>Y</td>
</tr>
<tr>
<td>Caller ID</td>
<td>Y</td>
</tr>
<tr>
<td>Hold and Resume</td>
<td>Y</td>
</tr>
<tr>
<td>Music On Hold</td>
<td>N</td>
</tr>
<tr>
<td>Call Reject</td>
<td>Y</td>
</tr>
<tr>
<td>Do Not Disturb</td>
<td>Y</td>
</tr>
<tr>
<td>Call Park</td>
<td>Y</td>
</tr>
<tr>
<td>DTMF via RFC2833</td>
<td>Y</td>
</tr>
<tr>
<td>Call Forward</td>
<td>Y</td>
</tr>
<tr>
<td>Feature Access Codes</td>
<td>N</td>
</tr>
<tr>
<td>SIP Using TCP</td>
<td>Y</td>
</tr>
<tr>
<td>G.711u, G.711a, G.722 and G.729A Codecs</td>
<td>Y</td>
</tr>
<tr>
<td>Multiple Line Keys (or registrations) per handset</td>
<td>N</td>
</tr>
<tr>
<td>‘Paired’ lines (shared line, bridged line )</td>
<td>Y</td>
</tr>
<tr>
<td>Call Pickup</td>
<td>N</td>
</tr>
<tr>
<td>Trunk Calling</td>
<td>Y</td>
</tr>
<tr>
<td>Failover / Fallback / Redundancy / Resiliency</td>
<td>Y</td>
</tr>
<tr>
<td>TLS / SRTP</td>
<td>N</td>
</tr>
<tr>
<td>Busy Lamp Field</td>
<td>N</td>
</tr>
<tr>
<td>Barge-In</td>
<td>N</td>
</tr>
<tr>
<td>Presence</td>
<td>N</td>
</tr>
</tbody>
</table>
### Features Tested

<table>
<thead>
<tr>
<th>Feature</th>
<th>Supported</th>
</tr>
</thead>
<tbody>
<tr>
<td>Reset and Restart through the Cisco UCM</td>
<td>N</td>
</tr>
<tr>
<td>Cisco's Dial Plan File</td>
<td>N</td>
</tr>
<tr>
<td>Centralized Cisco TFTP Integration</td>
<td>N</td>
</tr>
<tr>
<td>Cisco XML Applications</td>
<td>N</td>
</tr>
<tr>
<td>Cisco Phone Directory</td>
<td>N</td>
</tr>
<tr>
<td>Cisco Ad-Hoc Conferencing</td>
<td>N</td>
</tr>
</tbody>
</table>

Y—Yes  
N—No

### Configuration Sequence Overview

Steps required to support a Spectralink Versity Handset on the CUCM. Each item on this list links to the corresponding step information later in this document.

1. Ensure adequate licenses are available in the CUCM to support the Versity handset  
2. Build a Phone Security Profile appropriate for the Versity handset (if one does not exist)  
3. Add the End User  
4. Add the Phone  
5. Add a Directory Number  
6. Configure the Spectralink Versity handset to register with the CUCM  
7. Verify Registration Status  
8. Test Basic Calling Features and Functionality
Chapter 2: Configuration Steps

The intent of this section of the guide is to provide a minimum series of steps necessary to create the configuration on the CUCM to support the Spectralink Versity handsets, and then connect the Versity handsets to the network and achieve registration. Your environment may require that some additional fields or configuration be completed to ensure the handset works as desired. Please consult SIP Feature Configuration and Configuration Parameter Test Details for configuration details regarding more advanced features and functionality.

1. Licenses

Ensure adequate licenses are available to support the Versity handset.

The Spectralink Versity phone was configured as a Third-Party Advanced SIP Endpoint and consumed one CUCL Enhanced License unit per device deployed. An additional license unit will be consumed in the messaging system per voicemail box built.

License usage and availability can be confirmed in the Cisco CUCM License Management screens, or in Cisco’s Smart Software Manager.

Example: License Usage for 17 endpoints
2. **Phone Security Profile**

Build a Phone Security Profile Appropriate for the Versity handsets.

1. Navigate to **System > Security > Phone Security Profile**. Each deployment is unique and may require options other than those recommend below due to site policy or administrative requirements. You may build a unique Phone Security Profile for the Spectralink Versity phones or utilize an existing Phone Security Profile if it conforms to the recommended values below.

2. Navigate to **System > Security > Phone Security Profile**.

3. Select the **Add New** soft key:

   ![Example: Phone Security Profiles Page](image)

4. In the Phone Security Profile Type Drop-Down box, select the **Third-party SIP Device (Advanced)** option.

5. Select the **Next** soft key:
Example: Phone Security Profile Configuration Page

6 On the **Phone Security Profile Information** form
   a  Give the new profile a name, such as **Spectralink Wireless Phones**
   b  Select the **Enable Digest Authentication** checkbox.
   c  Select the **Save** key: Save
Example: Phone Security Profile Information Page

![Phone Security Profile Information](image)

<table>
<thead>
<tr>
<th>Product Type</th>
<th>Third-party SIP Device (Advanced)</th>
</tr>
</thead>
<tbody>
<tr>
<td>Device Protocol</td>
<td>SIP</td>
</tr>
<tr>
<td>Name</td>
<td>Spectrlink Wireless Phones</td>
</tr>
<tr>
<td>Description</td>
<td></td>
</tr>
<tr>
<td>Nonce Validity Time</td>
<td>600</td>
</tr>
<tr>
<td>Transport Type</td>
<td>TCP+UDP</td>
</tr>
<tr>
<td>Enable Digest Authentication</td>
<td></td>
</tr>
</tbody>
</table>

**Parameters used in Phone**

- **SIP Phone Port**: 5066

* indicates required item.
3. **Add the End User**

1. Navigate to User Management > End User
2. Select the Add New softkey:
3. On the User Information page that appears Spectralink recommends setting the following fields at a minimum:
   a. **User ID: jdoe** (Enter a user ID that complies with your system and account policies. This value will correspond with the Username field on the SIP phone form of the Versity handset.)
   b. **Password: 1234** (Enter a password for this user that complies with your system and account policies. If you are LDAP integrated, this field will be grayed out and unavailable, and you would create or modify this password through the Active Directory Server. This password is not used by the Versity phone, but it is good practice to assign a password for each user.)
   c. **Confirm Password: 1234** (Repeat the value you entered in the last step)
   d. **Self-Service User ID: 5011** (We may use the extension number we intend for the device. This is not used by the Versity phone, but the user might wish to utilize this to enter the Self-Service Web portal)
   e. **Pin: 1234** (Enter a pin if you wish the user to take advantage of pin enabled features such as user web login.)
   f. **Confirm Pin: 1234** (Repeat the value you entered in the last step)
   g. **Last Name: Doe** (Enter the User’s last name)
   h. **First Name: John** (Enter the User’s first name)
   i. **Digest Credentials: 9876** (Enter the Digest Authentication Password you would like the phone to use to register. This will correspond with the Password value that you enter in the Admin settings > SIP phone menu on the Versity handset.)
   j. **Confirm Digest Credentials: 9876** (Repeat the value you entered in the last step)
   k. Enter other End User field values as required by your site’s system and account policies.
   l. Select the Save key:
Example: User Information Page

![User Information Page](image)

<table>
<thead>
<tr>
<th>User Status</th>
<th>Enabled Local User</th>
</tr>
</thead>
<tbody>
<tr>
<td>User ID</td>
<td>doe</td>
</tr>
<tr>
<td>Password</td>
<td>****</td>
</tr>
<tr>
<td>Confirm Password</td>
<td>****</td>
</tr>
<tr>
<td>Self-Service User ID</td>
<td>5031</td>
</tr>
<tr>
<td>PIN</td>
<td>****</td>
</tr>
<tr>
<td>Confirm PIN</td>
<td>****</td>
</tr>
<tr>
<td>Last name</td>
<td>Doe</td>
</tr>
<tr>
<td>Middle name</td>
<td></td>
</tr>
<tr>
<td>First name</td>
<td>John</td>
</tr>
<tr>
<td>Display name</td>
<td></td>
</tr>
<tr>
<td>Title</td>
<td></td>
</tr>
<tr>
<td>Directory URI</td>
<td></td>
</tr>
<tr>
<td>Telephone Number</td>
<td></td>
</tr>
<tr>
<td>Home Number</td>
<td></td>
</tr>
<tr>
<td>Mobile Number</td>
<td></td>
</tr>
<tr>
<td>Pager Number</td>
<td></td>
</tr>
<tr>
<td>Mail ID</td>
<td></td>
</tr>
<tr>
<td>Manager User ID</td>
<td></td>
</tr>
<tr>
<td>Department</td>
<td></td>
</tr>
<tr>
<td>User Locale</td>
<td>&lt;None&gt;</td>
</tr>
<tr>
<td>Associated PC/Site Code</td>
<td>****</td>
</tr>
<tr>
<td>Digest Credentials</td>
<td>****</td>
</tr>
<tr>
<td>Confirm Digest Credentials</td>
<td>****</td>
</tr>
<tr>
<td>User Profile</td>
<td>Use System Default Standard (Factory Default) UnView Details</td>
</tr>
</tbody>
</table>
4. **Add the Phone**

1. Navigate to **Device > Phone**.

2. Select the **Add New** soft key:

   ![Add New Soft Key](image)

**Example: Find and List Phones Page**

3. On the **Add a New Phone** page, in the Phone Type pull-down box, select **Third-party SIP Device (Advanced)**.

4. Select the **Next** soft key:
Example: Add a New Phone Page

5 In the Device Information section of the Phone Configuration page, Spectralink recommends configuring the following fields at a minimum. We should mention that Cisco offers many configuration features and functionality that we will not detail in this example, but instead, we will attempt to illustrate a minimal configuration that should allow us to make and receive calls and address the major configuration requirements of the CUCM and the Versity handset. Enter additional fields as required by your site policies and procedures for new phone additions:

a MAC Address: 00907AA7DCC8 (Enter the MAC address of the Versity phone you are configuring. This is not utilized by the CUCM since this is a third-party SIP phone, but we must populate this field, so we might as well use the phones’ actual MAC address.)

b (Optional) Description: John Doe Versity (Enter a description for the phone in this field that will help you to identify this unique device.)

c Device Pool: Default (Select a Device Pool appropriate for the Codec Region you wish to use, the Date / Time group, and Call Manager Group for your site.)

d Phone Button Template: Third-party SIP Device (Advanced).

e Calling Search Space: None (Select a Calling Search Space appropriate for your phone and installation. The Calling Search Space determines how or if a dialed number can be routed. In our lab environment the Default Calling Search Space is unrestricted, but in a production environment the Calling Search Space must be configured such that it will be able to route to any numbers that are part of your dial plan just as you would configure any standard Cisco SCCP station.)
f **Location: Hub_None** (Select a Location for your phone. In our lab, we selected Hub_None, which means that Call Admission Control will not be used. Note that if you do specify a location, and you wish to call devices located within other locations, you may need to enable the G.729 codec on the Spectralink Versity handset. Cisco’s default inter-region codec is G.729.)

g **Owner User ID: jdoe** (Select the End User you created in Step 3)

Example: Device Information Section of the Phone Configuration Page
6 Scroll Down to the **Protocol Specific Information** section of the **Phone Configuration** page. Spectralink recommends configuring the following fields at a minimum:

a **Device Security Profile: Spectralink Wireless Phones** (Select the Phone Security Profile you created in Step 2)

b **Re-routing Calling Search Space: None** (Select a Calling Search Space with permissions appropriate for dialing any call forward or transfer destination you may use. In our lab environment a CSS of None was used, but in your environment the CSS chosen must have appropriate permissions to allow the phone to find the route to the refer-to-target.)

c **SIP Profile: Standard SIP Profile**

d **Digest User: jdoe** (Select the End User you created in Step 3.)

e Select the **Save** soft key:

f You will likely receive a pop-up notifying you that you need to **Apply the Configuration**. Select **OK** in response to this pop-up.

Example: Protocol Specific Information Section of the Phone Configuration Page
5. **Add a Directory Number**

1. Still on the *Phone Configuration* screen, In the *Association Information* area on the left side of the *Phone Configuration* window at the top, click the *Line [1] – Add a new DN link.*

![Association Information](image)

2. In the *Directory Number Configuration* window that appears, Spectralink recommends configuring the following fields at a minimum, though additional fields may be required by your site policies and procedures for new extension provisioning:

   a. **Directory Number:** 5011 (Enter the Extension number, or Directory Number you wish to use for your handset’s deployment. This value will correspond with the Extension number value entered in the BizPhone Settings menu on your Versity handset.)

   b. **Description:** John Doe Reg 1 (Enter a description for this particular Directory Number)

   c. **Alerting Name:** John Doe (Enter a name that will be displayed to callers)

   d. **ASCII Alerting Name:** John Doe (Typically we use the same name here that we use in the Alerting Name field)

   e. **Voice Mail Profile:** Cisco_UUnity_Connection_Profile (If this Directory Number will utilize voicemail, specify a Voice Mail Profile that will allow callers to be directed to the Voice Mail pilot number. This should be provisioned in the same manner as other Cisco phones.)

   f. **Calling Search Space:** None (Select a Calling Search Space with partitions that include any numbers you may dial from this line. In our lab environment Calling Search Spaces are not utilized, but in a production environment a CSS appropriate for your deployment should be chosen.)
g Configure the Call Forward Settings as Desired for your environment. In our sample configuration, we have configured Call Forward for all Unavailable, No Answer, or Busy scenarios to forward calls to the Cisco Unity Connection Voicemail server. You might also specify a different, unique call Forward Destination for any of the Call Forward scenarios listed. Note that if the CUCM system is using Partitions and Calling Search Spaces, Cisco recommends that you configure the Call Forward Calling Search Spaces as well (shown at right in the below screen shot). Failure to configure a Call Forward CSS may result in Call forward failures. In our lab environment, Calling Search Spaces are not utilized so we are able to leave the Call Forward CSS’s with a value of None, but your deployment will likely require that these are populated with appropriate CSS’s to achieve proper Call Forwarding.
**Example: Call Forward and Call Pickup Settings on the Directory Number Configuration Page**

<table>
<thead>
<tr>
<th>Call Forward and Call Pickup Settings</th>
<th>Voice Mail</th>
<th>Destination</th>
<th>Calling Search Space</th>
</tr>
</thead>
<tbody>
<tr>
<td>Calling Search Space Activation Policy</td>
<td></td>
<td></td>
<td>Use System Default</td>
</tr>
<tr>
<td>Forward All</td>
<td>or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Secondary Calling Search Space for Forward All</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Busy</td>
<td>or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Internal</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>External</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Answer Internal</td>
<td>or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Answer External</td>
<td>or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Coverage Internal</td>
<td>or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward No Coverage External</td>
<td>or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward on CFI Failure</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered Internal</td>
<td>or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Forward Unregistered External</td>
<td>or</td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>No Answer Ring Duration (seconds)</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
<tr>
<td>Call Pickup Group</td>
<td></td>
<td></td>
<td>&lt; None &gt;</td>
</tr>
</tbody>
</table>

- **h** Display: John Doe (This name will be presented to internal called parties)
- **i** ASCII Display: John Doe (Typically we use the same name here that we use in the Display field)
- **j** Maximum Number of Calls: 4 (Four calls are the maximum number of calls the Spectralink Versity can place or receive per registration.)
- **k** Busy Trigger: 4 (Four calls is the maximum number of calls the Spectralink Versity can place or receive per registration.)

- **l** Select the **Save** softkey:
Example: Caller ID and Call Waiting Settings on the Directory Number Configuration Page
6. Configure the Spectralink Versity Handset to Register with the CUCM

The first step in connecting the Spectralink Versity handset to the CUCM is to get the handset connected to the wireless LAN. This can be done manually on each handset by configuring Wi-Fi settings on the Administrative menus. See Spectralink References for pertinent references. Visit the Spectralink Support site for additional technical information. Information regarding WLAN interoperability and configuration procedures specific to different WLAN vendor’s infrastructure can be found on the Spectralink support web site: https://support.spectralink.com/versity.

Once the handset is connected to the WLAN we can proceed to configure the SIP parameters for the handset so that it can connect to and register with the CUCM.

The SIP configuration fields are basically the same whether provisioned through the Spectralink Application Management (SAM) system or through the handset’s BizPhone menus manually, so you may use either interface to specify the required parameters.

Example: Manual SIP configuration

Below, we will illustrate a manual configuration example with the sample user (John Doe) and handset extension (5011) we built in the preceding steps. We will show the resulting configuration in the handset itself. However, if you have more than a few handsets to deploy, Spectralink strongly recommends use of a SAM server. Some comments and details about how to tailor these fields for your unique environment are also provided below.

First, navigate to the BizPhone App and open the Overflow (three dots) menu, then select the Settings option.

Enable SIP: On. Ensure this is set to On or Enabled. If you disable this, The BizPhone app will not function.
The below settings should be configured under the Registration 1 heading:

SIP server: **172.29.103.100**. This value should be replaced with the IP address of your CUCM. You may also use a DNS A-Name record or a DNS SRV record to specify the server address.

SIP server port: **leave blank**. Spectralink’s lab server uses UDP port 5060 for SIP communications and that is the default, so there is no need to populate this field if your site is utilizing port 5060 for SIP communications. If you specify a port number here, the Versity handset will not query for a SRV record at all, but instead will only look for an A-name record (or an IP address) for the SIP server address. So, if your installation uses a different port for SIP communication you could either specify this value here or you could specify the SIP server port number using a DNS SRV record.

Transport: **UDP**. The CUCM will accept UDP or TCP and support is specified in the Transport Type field within the CUCM Phone Security Profile created for Versity in Step 2. Versity is capable of using either transport type as well. The Versity default is UDP.

Extension number: **5011**. This value should be replaced with your phone’s extension number. This value corresponds with the Directory Number value you added to your phone using the Add a New DN link in the CUCM.

Username: **jdoe**. This value corresponds with the User ID value you entered on the End User Configuration form in the CUCM. In SIP terms, this is the SIP digest authentication username.
Password: **9876.** This value should be replaced with your End User’s SIP digest authentication password. This value corresponds with the **Digest Credentials** value entered on the End User Configuration page in the CUCM.

Example: BizPhone upper Registration 1 SIP phone settings screen

On the lower portion of the Registration 1 settings screen ensure the following are configured:

Voice mail retrieval address: **2999.** This value should be replaced with your Voice Mail Pilot Number, or the number you would dial to retrieve Voicemail messages.

Use SIP Standard Hold Signaling: **On.** Leave this setting On to utilize RFC3261 style hold.

Force subscription to message waiting notifications: **Unchecked.** Testing with Cisco Unity Connection in Spectralink’s lab environment showed that Third Party SIP Endpoints were subscribed to Message Waiting Indications (MWI’s) automatically and will receive SIP Notify messages without forcing a subscription.

Allow contact header updates: **Off.** If enabled, will replace the Contact Header with the received ip address and rport values contained in the Via of the 200 OK response to SIP Registration. PIVOT will then renew the SIP Registration using the updated Contact information, and subsequent Invites and Registration requests will be made using the updated Contact information for call control messaging subscription.
Specify new TCP port in contact header: Off. If enabled, causes the phone to open a new listening port for TCP and put that port number in the contact header field. This parameter should remain disabled for integration with the CUCM.

Navigate away from the Registration 1 Settings using the back arrow and configure the following under the Common settings configuration screen:

Audio DSCP: 46. This value is the default. It should not be necessary to modify this default unless specifically advised to do so under the requirements of the Spectralink VIEW deployment instructions.

Call Control DSCP: 40. This value is the default. It should not be necessary to modify this default unless specifically advised to do so under the requirements of the Spectralink VIEW deployment instructions.

G.711u, G.711a, G.722 and G.729A codec priorities: G.722 = 1, G.711u =2 and G.711a = 3. G.729A=4. Spectralink recommends enabling the G.722 codec as the first-choice codec to realize high definition audio when calling high definition capable endpoints. G.729A should also be enabled as the fourth-choice codec. If your site uses different CUCM Regions this will allow the Versity phone to accept G.729A calls if offered by the CUCM (and G.729 is the default inter-region codec for CUCM). Note that this could also be handled by modifying the CUCM to use a different codec (i.e. G.711u) as the default inter-region codec but enabling G.729A here will
allow us to gracefully handle this scenario when it does occur without necessitating programming changes on the CUCM side.

**DTMF relay payload type: 96.** This value is the default. This parameter allows the RFC2833 DTMF telephone-event payload type advertised in the SDP of SIP messaging to be modified. The handset will honor the SDP telephone-event value returned by the called party for DTMF payload negotiation and will match the SDP telephone-event payload value type of received invites but modifying this parameter will change the telephone-event payload type the endpoint advertises in initial invites. Spectralink recommends leaving this value at the default setting (96) unless the far end will not accept DTMF payload types of 96 and fails to respond to invites with the SDP payload value it requires.

Example: BizPhone upper Common Settings screen

On the lower portion of the Common settings screen ensure the following are configured:

**Force in-band DTMF tones: Off.** If enabled will disable advertisement of the RFC2833 telephony event parameters and will result in In-band DTMF tones rather than RFC2833 tones. It should also be noted that Versity will revert to In-band DTMF if telephony event packages are not advertised in the SDP of the far end regardless of the setting of this parameter.
Override automatic switch from UDP to TCP: **Off.** If enabled and the handset is configured with Transport type of UDP, this parameter will force the handset to continue to send SIP packets that exceed 1300 bytes in size using UDP, rather than switching to TCP for large packet transmissions as RFC3261 section 18.1.1. mandates. For interoperability with the Cisco UCM this setting should remain Off.

Example: BizPhone lower Common Settings screen
7. **Verify Registration Status**

Once the handset has successfully connected to the wireless LAN and you have entered the SIP credentials and submitted them by exiting the BizPhone settings menus using the back key, or by clicking the Save Configuration button if using the CMS, then you will want to confirm whether the registration has been successful.

**Successful registration status examples:** The below screenshots show an example of a phone that has achieved a successful registration with the Callserver. This shows the idle screen icon and the BizStatus screen information (available under Apps or by selecting the SIP registered notification) you might expect to see if the phone has successfully registered with the CUCM.

The **200 OK** is the call server’s success response to the registration request. If you do not observe a **200 OK** in this area of the screen, then the registration request is failing. The error code returned by the server may provide some additional hints as to the reason for the failure.

**Example: Versity Successful Registration Confirmation Screens**

![Successful Registration Screenshots]

**Failed Registration:** The below screenshots show a registration failure. In this case we can observe that the registration has failed (because there is not a 200 OK here). The 403 response
typically indicates that the username and password provisioned in the Versity handset do not match those expected by the call server:

Example: Versity Failed Registration Screens
Registration Status in the CUCM: You can check to see the status of the Versity handset according to the CUCM by navigating to **Device > Phone > Find and List Phones**, and then using one of the search mechanisms to find the phone of interest. If the handset has successfully contacted the CUCM it should show you the current registration status as **Registered** and allow you to observe the IP address the CUCM thinks the registration is coming from. An example of this is shown below:

Example: Versity Successful Registration Confirmation in the CUCM
8. **Test Basic Calling Features and Functionality**

Once the device's registration has been confirmed, a basic functionality test should be performed. Spectralink recommends running the following tests at a minimum to verify proper Versity handset / CUCM interaction.

- Basic Call to and from the Versity handset to another CUCM System device.
- Call Transfer the Versity handset to another device and use the Versity handset to conduct a transfer.
- Perform a conference call with the Versity handset, using the Versity handset as the conference initiator and test using the Versity handset as a conference participant.
- Hold and resume a call (note that MOH is not supported by the Versity handset).
- Leave a voicemail for the Versity handset (if equipped) – Ensure Message Waiting Indication is delivered. Call the voicemail system from the Versity handset and retrieve the call.
- Place a call to a PSTN number equipped with a menu system and verify the functionality of DTMF tones to navigate the menus.
- Verify other functionality of interest.
Chapter 3: SIP Feature Configuration and Configuration Parameter Test Details

Direct to CUCM SIP Registration

Spectralink Varsity handsets register directly to the CUCM System.

SIP Digest Authentication

The configuration instructions in 2. Phone Security Profile of this manual detail how to configure a Device Security Profile on the CUCM to support the Varsity handset using Digest Authentication.

The Username value in the Varsity SIP Settings menu corresponds with the User ID value entered on the End User Configuration form in the CUCM.

The Password value in the Varsity SIP Settings menu corresponds with the Digest Credentials value entered on the End User Configuration form in the CUCM.

Basic Calls

Call functionality was tested by calling between Spectralink Varsity handsets as well as to and from a Cisco 7960 phone. No special Varsity configuration parameters should be required to realize this ability.

Voicemail Integration

Testing was performed with Cisco Unity Connection version 12.0.1.21900-10. There is no special configuration required for the Varsity handsets within the Cisco Unity Connection configuration. Mailboxes should be built and configured as with other Cisco extensions.

Please note also that Cisco Unity Connection does not provide notifications including the number of waiting messages. MWI notifications are delivered to SIP endpoints with a simple yes / no status, and as such, the Spectralink Varsity phone cannot provide a message count to the user but will provide an indicator that it has message(s) waiting.

The below BizPhone configuration parameter was found to help optimize the Cisco Unity Connection voicemail integration with the Varsity phones:

Voice mail retrieval address: **2999**. This value should be replaced with your Voice Mail Pilot Number, or the number you would dial to retrieve voicemail messages. The voicemail server
address was not sent in the Message-Account field of the SIP Notify messages from the Cisco Unity Connection system, so this field must be populated with the main voicemail number to allow notification and speed dial dialing of the voicemail system. Entering this number will allow you to dial the voicemail system by opening the dialer and long-pressing the 1 key on the dial pad, or by tapping the Message Waiting Notification from the notification drawer.

**Message Waiting Indication (MWI)**

Parameters described in the Voicemail Integration section above were all that we found to be required to realize successful Message Waiting Indications. There is no need to Force Subscribe for MWI notifications, as endpoints are automatically subscribed in the CUCM upon successful registration.

**Call Waiting**

By default, when you build a DN on a phone in the CUCM, it will allow two calls to that number. Additional calls will be sent to Forward Busy treatments defined on the Directory Number Configuration page in the CUCM. However, in our example configuration, we recommended modifying this value to the maximum value supported by the Versity phone; four calls. To verify current call waiting settings on the CUCM navigate to the phones' configuration in the CUCM, and then select the DN you wish to modify in the Line Association Information section of the Phone Configuration Form. Now scroll down to the Multiple Call / Call Waiting Settings on Device portion of the Directory Number Configuration Window and edit the Maximum Number of Calls and Busy Trigger fields as shown below:

Example: Directory Number Call Waiting Configuration Settings
Multiple Calls per Line Key or Maximum Calls per Line

The guidelines specified in the Call Waiting section of this document apply to Multiple Calls and Maximum Calls per Line Key.

Conference 3-way

In a three-way conference, the Versity handset will merge the appropriate audio streams locally. No special treatment is required from the CUCM. It should be noted that if the Versity handset is the conference initiator and ends the conference by hanging up, the Versity phone will drop the other two conference participants, and they will no longer be in a call.

Transfer: Blind

This type of transfer occurs when Phone A calls Phone B and Phone B has answered, resulting in a two-way call. Phone B then presses the transfer button, placing Phone A on Hold, and dials the number for Phone C, followed by pressing the transfer button again. Phone B never talks to Phone C and Phone C begins ringing with the call from Phone A. If Phone C answers he will be in call with Phone A. Blind transfer was successfully tested in Spectralink’s labs.

Troubleshooting Call Transfer Issues

When SIP smartphones on a CUCM perform a Call Forward or a Call Transfer, they are rerouting the call. If Call Transfers cannot be completed as expected, please ensure the Rerouting Calling Search Space on the Phone Configuration page and / or the Call Forward Calling Search Spaces on the Directory Number Configuration page are set such that the Forward destination is included in the Calling Search spaces you define. (You might try configuring this field with the same Calling Search Space you use for the device itself to validate that this is the problem you are facing.)

Transfer: Announced

This type of transfer occurs when Phone A calls Phone B and they are in call. Phone B then presses the add call button, placing Phone A on Hold, and dials the number for Phone C, followed by pressing the Send key. Phone C begins ringing with the call from Phone B, and if Phone C answers he will be in call with Phone B. Phone B can then “announce” that he is going to connect Phone C to Phone A. Phone B then presses the transfer key and taps the call with Phone A as the call to receive the transfer. The result is that Phone C and Phone A are in call and Phone B is no longer in the call. Announced transfer was successfully tested in Spectralink’s labs.
**Transfer: Attended**

This type of transfer is really a conference, where the conference initiator drops out of the call after the conference has been established and is not supported by the Versity Handset.

**Caller ID**

Calling Party and Called Party name and number are supported by the Spectralink Versity handsets. The Versity handsets make use of the Cisco provided Remote-Party-ID field which allows the phone to use PBX supplied messages to update the called and calling party names when SIP re-invites, refers, or progress messages occur such as during a transfer.

Caller ID returned to the calling party is typically controlled by the Alerting Name, and ASCII Alerting Name fields on the Directory Number Configuration page of the extension of interest.

**Example: Caller ID Returned to the Calling Party**

![Caller ID Configuration](image)

Caller ID Displayed to the Called Party is typically controlled by the Display (Caller ID), and ASCII Display (Caller ID) fields on the Directory Number Configuration page of the extension of interest.
Example: Caller ID Sent to the Called Party

Hold and Resume
Spectralink Versity handsets are capable of hold and resume, however, clients placed on hold by a Spectralink Versity handset will not hear (MOH) Music On Hold. Spectralink Versity handsets placed on hold by a Cisco SCCP or SIP client will hear MOH if configured.

Music On Hold
Spectralink Versity handsets are capable of hold and resume, however, clients placed on hold by a Spectralink Versity handset will not hear (MOH) Music On Hold. Spectralink Versity handsets placed on hold by a Cisco SCCP or SIP client will hear MOH if configured.

Call Reject
Call Reject allows a caller to decline an inbound call. For purposes of this test we ensured that when an inbound call was rejected, the calling party would be sent directly to voicemail or to the presently defined Call Forward Busy location. When this occurs, the call log on the Versity handset will show rejected calls as missed calls. The Spectralink Versity phone sends a 486 Busy SIP message back to the CUCM when the user rejects an offered call.

Do Not Disturb
The BizPhone application will honor the Android Oreo implementation of Do Not Disturb (DND). If DND is enabled, the result for SIP calls will be that the smartphone will send back a 486 Busy
Here to any incoming SIP Invites, which will cause the CUCM to forward the calling party to the Call Forward Busy location defined in the CUCM for the called Directory Number. This implementation does not honor the CUCM-side DND settings for Ringer Off, Call Reject, and DND Incoming Call Alert Options.

**Call Park**

Cisco supports several variants of the Call Park feature. For purposes of our testing we configured and tested against the Directed Call Park feature. This works on the Versity handset; however, users MUST utilize the announced transfer feature rather than the blind transfer feature to park the call. (If the Blind Transfer feature is utilized, callers will be unable to retrieve the parked call.) Spectralink notes that the CUCM Features and Services Guide states, “The system does not support directed call park when the blind transfer softkey is used on Cisco Unified IP Phones 7940 and 7960 that are running SIP.” Third party SIP phones seem to be subject to the same constraint.

**DTMF via RFC2833**

The Spectralink Versity handset utilizes RFC2833 to support delivery of DTMF tones. There is no special configuration required for the handset to utilize RFC2833, and RFC2833 was verified to function correctly during Spectralink lab testing through the manipulation of Cisco Unity Voice Messaging menus and trunk calls to PSTN IVR services.

**Call Forward**

There are several different ways to implement a call forwarding solution for the Versity phones. You may implement call forwarding on the Versity handset itself, or program Call Forward through a Cisco provided interface. Some comments about each of these methods can be found below.

**Call Forward All Calls Using the Handset**

This method of implementing Call Forward was tested by navigating to the Spectralink SIP Dialer and tapping the Overflow button (three dots), then selecting the Call Forwarding Setup menu. One advantage to this method is that it posts a user-friendly indication in the dialer that the phone is in the forwarding state any time you have enabled call forwarding. This method will also log calls that are forwarded as missed calls in the call logs.

The disadvantage to this method of call forward implementation is that the phone must remain powered on and connected to the WLAN to successfully redirect any offered calls to the Call Forward destination. So, a user that set call forward and then powered the handset off would
not, in fact, still be forwarding calls since the handset must be available to respond to offered calls with the forwarding destination.

Call Forward All Calls Using the Cisco UCM

This method of implementing Call Forward was tested by utilizing the Cisco Unified Communications Self Care Portal (on CUCM 12 or CUCM 11.) or End User Web page login (on earlier CUCM versions) to set call forwarding on the extension of interest, though an administrator could also configure this through the phones’ Directory Number Configuration page. One advantage to this method of implementing call forward is that the forward remains in effect regardless of whether the handset remains powered on or in range of the wireless network. The disadvantage to this method of call forward implementation is that the phone does not provide any user-friendly indication that the phone is in the forwarded state when call forwarding is set. The call forwarding state is maintained by the CUCM itself and the CUCM will simply never offer calls to the phone until the call forward is cancelled through the CUCM’s web UI. As such, calls forwarded using this mechanism will not show as missed calls in the call log of the Spectralink Versity device.

Call Forward Busy and Call Forward No Answer

The Spectralink Versity handset does not provide settings for Call Forward Busy or No Answer on the handset itself. Forward Busy and No Answer destinations should be programmed on the Cisco UCM’s Directory Number Configuration settings page or the Self Care Portal rather than on the handset.

Call Forward Timers

The Global Forward No Answer timer can be set under System> Service Parameters> Cisco Call Manager> Clusterwide Parameters (Feature - Forward) => Forward No Answer Timer (Default =12s).

The per station Forward No Answer timer can be set on the Directory Number Configuration page under the No Answer Ring Duration (seconds) parameter.

Troubleshooting Call Forward Issues

If call forwarding is not working as expected, please ensure the Rerouting Calling Search Space on the Phone Configuration page and / or the Call Forward Calling Search Spaces on the Directory Number Configuration page are set such that the Forward destination is included in the Calling Search spaces you define.

Feature Access Codes

Feature Access Codes are not supported by the CUCM.
**SIP Using TCP**

The CUCM can utilize the TCP transport mechanism in conjunction with the Spectralink Versity phone. Though the Versity phone defaults to utilizing UDP, the transport mechanism can be modified to TCP in the Spectralink handset by toggling the **SIP Phone > Transport** setting. In our example configuration, we created a Phone Security Profile in the CUCM that supported both TCP and UDP, and we assigned that profile to the Versity phones on the Phone Configuration page. This allows the device to control whether TCP or UDP is used as the transport mechanism. However, a Phone Security Profile that supports only TCP or only UDP could also be implemented and applied to the Device on the Phone Configuration page in the CUCM.

**G.722, G.711u, G.711a, G.729A codecs**

The Spectralink Versity handset was tested using each of the above codecs when deployed against the CUCM.

**Default Versity Advertised Codec List**

The Spectralink Versity phones’ will advertise G.711u first, and G.711a second by default. However, Spectralink recommends enabling the G.722 codec as the first choice to achieve high definition audio when calling to or from other high definition capable endpoints. Spectralink also recommends enabling the G.729 codec as the fourth choice. If your site uses different CUCM Regions this will allow the Versity phone to accept G.729A calls if offered by the CUCM (and G.729 is the default inter-region codec for CUCM). Note that this could also be handled by modifying the CUCM to use a different codec (i.e. G.711u) as the default inter-region codec but enabling G.729A here will allow us to gracefully handle this scenario when it does occur without necessitating programming changes on the CUCM side.

G.711u, G.711a, G.722, and G.729A codec priorities: \[G.722 = 1, G.711u = 2 \text{ and } G.711a = 3 \text{ and } G.729A = 4.\] A value of 1 is the highest priority and enabled, 2 is second priority and enabled and so on. A value of 0 disables the codec. You may modify the order or enable / disable codecs as required for your installation.

**Multiple Line Keys (or Registrations) per Handset**

The Spectralink Versity SIP application provides full support for only one SIP registration per handset. So, there is not a way to configure the handset to register to multiple accounts or “lines” with the ability to select any of those registered accounts for outbound calls Second registration support in Versity currently allows only for the receipt of calls on the second registration and does not provide a UI mechanism to allow the selection of the second registration for call initiation. For the above reason, Spectralink considers this feature
unsupported, though current functionality could be utilized to allow the receipt of calls on a second registration or Directory Number if the need to use that registration to place outbound calls did not exist. For a more detailed discussion of the Second Registration feature support, please consult the Spectralink Application Management Guide available on the Spectralink support site.

‘Paired’ Lines (Shared Line, Bridged Line)

Since Versity provides full support for only one registration, we cannot configure a second line appearance as a bridged line. We may configure the same Directory Number as the first registration on multiple phones though and calls to that number will ring all phones configured in this manner. This is because the credentials associated with the registrations are specified in the CUCM’s End User’s profile rather than at the Directory Number level, so each Versity phone will still use unique Username and Password values. However, the phones that use this same "bridged" number can only receive a total number of calls equal to the Max Call Value specified on the Directory Number Configuration page of the phone in the CUCM. So, if we followed the sample configuration and specified a Max Calls value equal to four, then if we had two phones each with this Directory Number configured, we could receive four calls to one of the phones, or two simultaneous calls to both phones, but subsequent callers would receive the Call Forward Busy treatment.

Call Pickup

Call pickup is not supported by third party SIP phones on the CUCM.

Trunk Calling

In and outbound trunk calling were tested utilizing an Audiocodes Mediant 1000 Gateway connected to the CUCM using SIP and connected to the PSTN through an ISDN PRI circuit. The Spectralink Versity handset was able to make and receive calls through this configuration as well as to pass DTMF digits through to IVR style menus on the PSTN.

Failover, Fallback, Redundancy or Resiliency

Spectralink Versity phones do not utilize the same configuration mechanisms and detection mechanisms that Cisco branded phones use to achieve Redundancy or Failover but do support a redundancy mechanism that will allow the phones to failover to and fallback from a secondary, tertiary, or quaternary server. For a detailed discussion of Redundancy and Failover behavior and configuration methods for the Spectralink Versity phones, please consult the Spectralink Application Management Guide available on the Spectralink support site.
**TLS & SRTP**

TLS and SRTP for SIP transport are not currently supported by the Spectralink Versity handsets.

**Busy Lamp Field**

Busy Lamp field is a CUCM Presence feature that allows a user to monitor the status (in-call or idle) of another user. This feature is not supported by the Spectralink Versity handset.

**Barge-In**

Barge-In is not supported by the Spectralink Versity handset.

**Presence**

The Presence feature is not supported by the native Spectralink Versity SIP application. Additional testing using an Android-based Jabber client may be interesting to better understand the possibility of the handset to convey Presence information. However, it should be noted that the telephony portion of the Android based Jabber client would not currently benefit from Spectralink’s Wi-Fi voice optimization, and that is the primary reason Spectralink recommends telephony integration using the Versity’ native SIP client.

**Reset and Restart through the Cisco UCM**

SpectraLink Versity handsets will not reset or restart using messages sent by the Cisco UCM. SpectraLink strongly recommends a SAM (Spectralink Application Management) server be utilized to allow phone configuration updates without requiring each phone to be collected and reprogrammed through the UI. Spectralink also recommends that a System Update Server be utilized and that the handsets be set to a periodic polling interval such that they will find and download new software revisions when the code is made available on the provisioning server. For details on the SAM and Update Servers please see Spectralink Application Management Guide.

**Cisco’s Dial Plan File**

SpectraLink Versity handsets do not support Cisco’s dial plan file and do not currently offer support for a local dial plan implementation.
Centralized Cisco TFTP Integration

Spectralink phones do not support the centralized Cisco TFTP provisioning server. Spectralink recommends the use of a SAM (Spectralink Application Management) server for efficient provisioning of the Versity handsets. See Spectralink Application Management Guide.

Cisco XML Applications

SpectraLink Versity handsets support XML API/XHTML related applications, but do not specifically support Cisco branded XML Applications. Please see Spectralink Application Management Guide for additional details.

Cisco Phone Directory

SpectraLink Versity handsets do not support the Cisco Phone directory but do allow LDAP integration with Active Directory such as to provide a comprehensive directory search tool. The Versity handset also maintains detailed inbound, outbound, and missed call lists, as well as the ability to save and add contacts to a local phone directory.

Cisco Ad-Hoc Conferencing

Cisco branded Ad-Hoc conferencing and controls are not specifically supported, though the Versity handset does support Ad-Hoc Conferencing and merging of Audio streams locally on the Versity handset itself. See Spectralink Application Management Guide.
Chapter 4: Troubleshooting

SIP Traces on the CUCM

If call setup or signaling failures are suspected, a Wireshark trace of the SIP messaging is often one of the most useful tools for diagnosing the issue. Obtaining an ADB logcat, bug report, or verbose level syslog capture from the Versity phone should also provide the data necessary for our support organization to take a detailed look at SIP messaging to and from the phone itself (though not in a Wireshark form, this is still useful information.) However, you may encounter situations where we believe the phone is sending packets out to the network but does not seem to be receiving responses from the call server. In this case, we may want to analyze a capture from the CUCM to help determine whether the server received messages sent by the phone, and how it responded. The CUCM does have the ability to gather SIP traces, through a tool called the Cisco Unified Real Time Monitoring Tool (RTMT) that is freely available for download from any installed CUCM. Please consult Cisco’s RTMT Administration Guide for an in-depth discussion of the RTMT and a detailed description of how to utilize the tool to capture SIP traffic to or from the CUCM.

To download a copy of the RTMT from your CUCM:

1. Log into the CUCM
2. Navigate to Applications> Plug-ins
3. Use the Find tool to locate the Cisco Unified Real Time Monitoring Tool appropriate for your environment (there should be both a Windows and Linux based tool offered) and select and install it.
4. Consult Cisco’s documentation regarding the RTMT to allow you to capture SIP signaling traces of your device.

DSCP Values

The default DSCP value for Audio from the CUCM is aligned with the Spectralink recommendation of a decimal DSCP value of 46. That said, if wireless analysis determines that packets are not “getting through” to the handset, or are arriving with an incorrect DSCP tag, it may be worth verifying that the CUCM is tagging Audio packets with appropriate DSCP values.

Audio DSCP

To modify or check the Audio DSCP Values Sent by the CUCM
Log into the CUCM, then navigate to: System> Service Parameters
Next, select the active CUCM server form the pull-down list.

In the Service Window that appears, select **Cisco CallManager (Active)** from the pull-down list.

**Example: Service Parameter Configuration**

![Service Parameter Configuration](image)

Navigate down to the **Clusterwide Parameters (System – QOS)** portion of the page that appears and look for the **DSCP for Audio Calls** parameter. The recommended value is 46 (as shown below).
Clusterwide Parameters (System - QoS)

<table>
<thead>
<tr>
<th>Priority Class</th>
<th>Normal Priority</th>
<th>Normal Priority</th>
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</thead>
<tbody>
<tr>
<td>DSCP for Audio Calls</td>
<td>46 (101110)</td>
<td>46 (101110)</td>
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<tr>
<td>DSCP for Video Calls</td>
<td>34 (100010)</td>
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<tr>
<td>DSCP for Audio Portion of Video Calls</td>
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<tr>
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<tr>
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</tr>
<tr>
<td>DSCP for Flash Audio Calls</td>
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<tr>
<td>DSCP for ICCP Protocol Links</td>
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