Spectralink Versity Smartphone

Mitel MiVoice Business 3300 ICP

Interoperability Guide

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About This Guide

This interoperability guide describes the procedures for configuring Spectralink Versity handsets with the Mitel MiVoice Business software running on a 3300 ICP platform. The overall objective of the interoperability compliance testing is to verify that Spectralink Versity smartphones function in an environment comprised of a Mitel MiVoice Business Server and various Mitel 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.

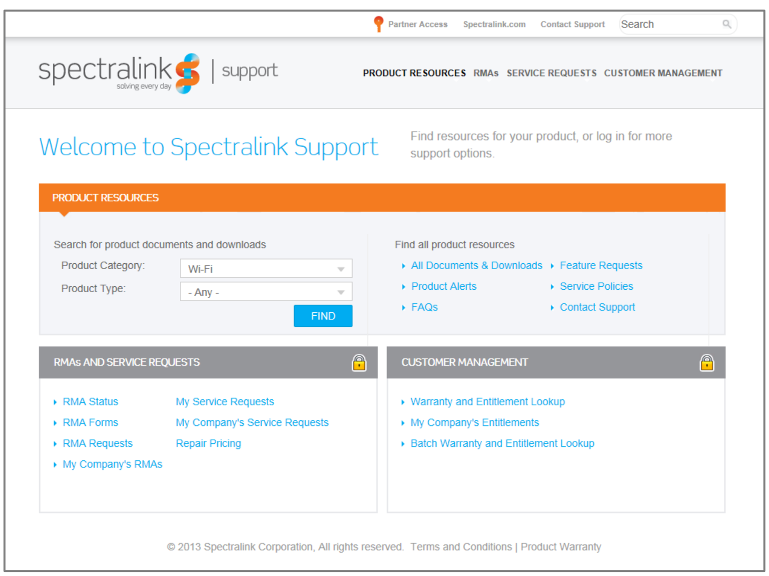
For Technical Support: mailto:technicalsupport@Spectralink.com

For Knowledge Base: http://support.Spectralink.com

For Return Material Authorization: mailto:nalarma@Spectralink.com

Spectralink References

All Spectralink documents are available at http://support.Spectralink.com.



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Select the Product Category and Product Type from the dropdown lists and then select the product from the next page. All resources for that particular product are displayed by default under the All tab. Documents, downloads and other resources are sorted by the date they were created so the most recently created resource is at the top of the list. You can further sort the list by the tabs across the top of the list to find exactly what you are looking for. Click the title to open the link.

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.

Mitel Documentation

This document does not attempt to cover even a small subset of the features and functionality available in the Mitel MiVoiceSystem. Please navigate to the Mitel documentation site for the latest Mitel branded documentation:

http://edocs.mitel.com/default.htm

Conventions Used In This Document

Typography

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

Convention	*Description*
Bold	Highlights interface items such as menus, softkeys, file names, and directories. Also used to represent menu selections and text entry to the handset.
Italics	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.
Underlined blue	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.
Bright orange text	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.
`Fixed-width-font`	Used for code fragments and parameter names.

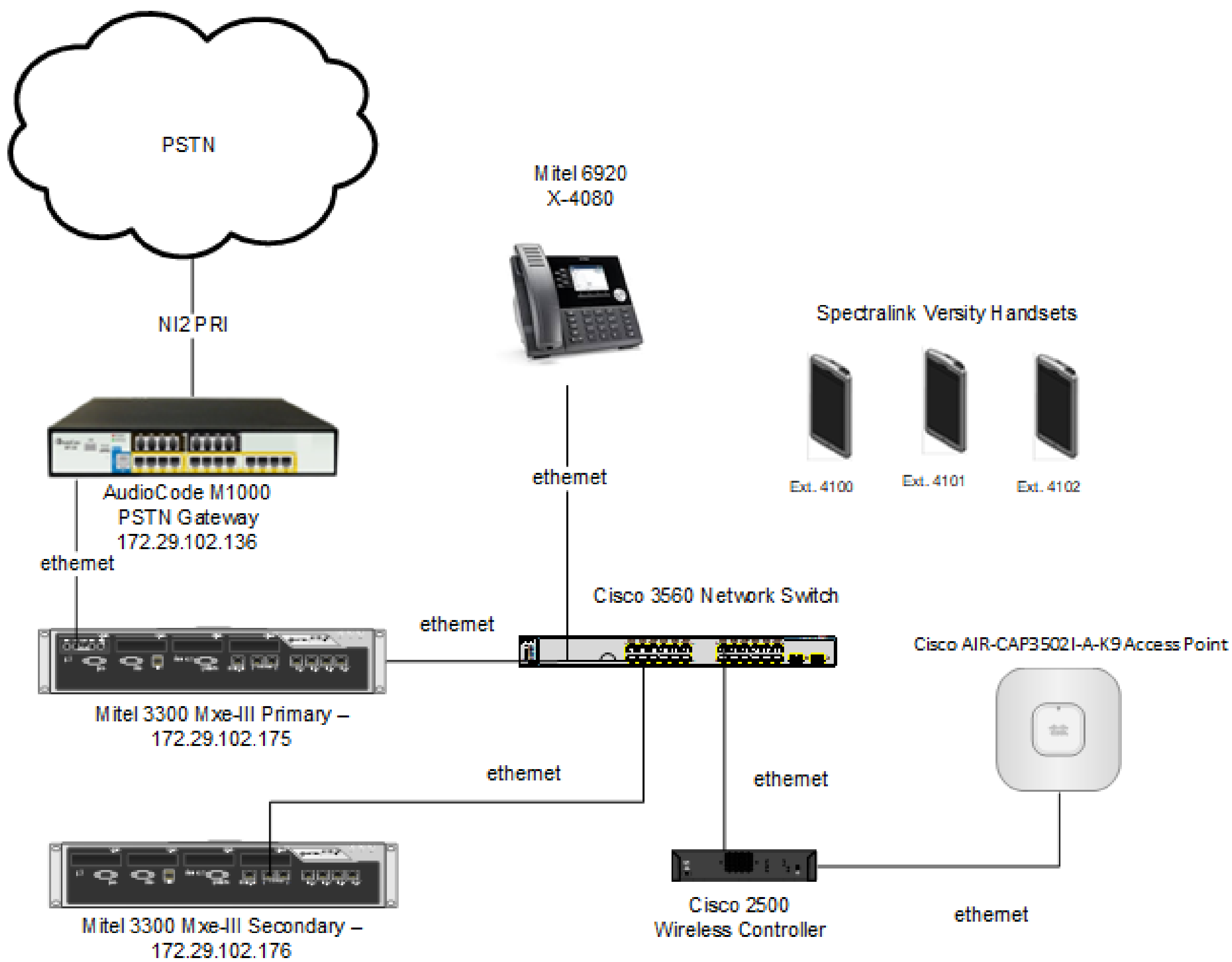
This guide also uses a few writing conventions to distinguish conditional information.

Convention	*Description*
<MACaddress>	Indicates that you must enter information specific to your installation, handset, or network. For example, when you see *<MACaddress>*, enter your handset's 12-digit MAC address. If you see *<installed-directory>*, enter the path to your installation directory.
>	Indicates that you need to select an item from a menu. For example, **Settings> Basic** indicates that you need to select **Basic** from the **Settings** menu.

Chapter 1: Overview

System Diagram

Below is a system diagram depicting the lab setup used to test the Spectralink Versity interoperation with the Mitel MiVoiceSystem.



Test Infrastructure Version Information

- Mitel MiVoice Business: Release 9.0 SP2, Active Version: 9.0.2.16
- Spectralink Versity Handset Software Version: 1.6.0.1212
- Cisco AIR-CAP3502I-A-K9 Access Point Software Version: 8.5.140.0

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.

Features Tested	Supported
Direct to Mitel MiVoice System SIP Registration	Y
SIP Digest Authentication	Y
Basic Calls	Y
Voicemail Integration	Y
Message Waiting Indication (MWI)	Y
Call Waiting	Y
Multiple Calls Per Line Key (or per registration)	Y
Conference: 3-way	Y
Transfer: Blind	Y
Transfer: Announced	Y
Transfer: Attended	N
Caller ID	Y
Hold and Resume	Y
Music On Hold	Y
Call Reject	Y
Do Not Disturb	Y
Call Park	Y
DTMF via RFC2833	Y
Call Forward	Y
Feature Access Codes	Y
SIP Using TCP or TLS	N
G.711u, G.711a, and G.729A Codecs	Y
G.722 Codec	N
Multiple Line Keys (or registrations) per handset	N
‘Paired’ lines (shared line, bridged line)	N
Ring Multiple Phones Simultaneously / Ring Group	Y
Trunk Calling	Y
Failover / Fail-back / Redundancy / Resiliency	Y

Configuration Sequence Overview

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

1. Ensure adequate licenses are available in the MiVoice Business System to support the Versity handset
2. Create a Class of Service for The Spectralink Versity Phones
3. Define a SIP Device Capabilities Number for the Spectralink Versity Phones
4. Configure the Call Rerouting First Alternatives Location
5. Add the User and the Device
6. Add a Voicemail Box (If Desired)
7. Configure Call Re-routing for your extension
8. Configure the Spectralink Versity handset to use the Mitel MiVoice System as the SIP Server
9. Verify Registration Status
10. Test Basic Calling Features and Functionality

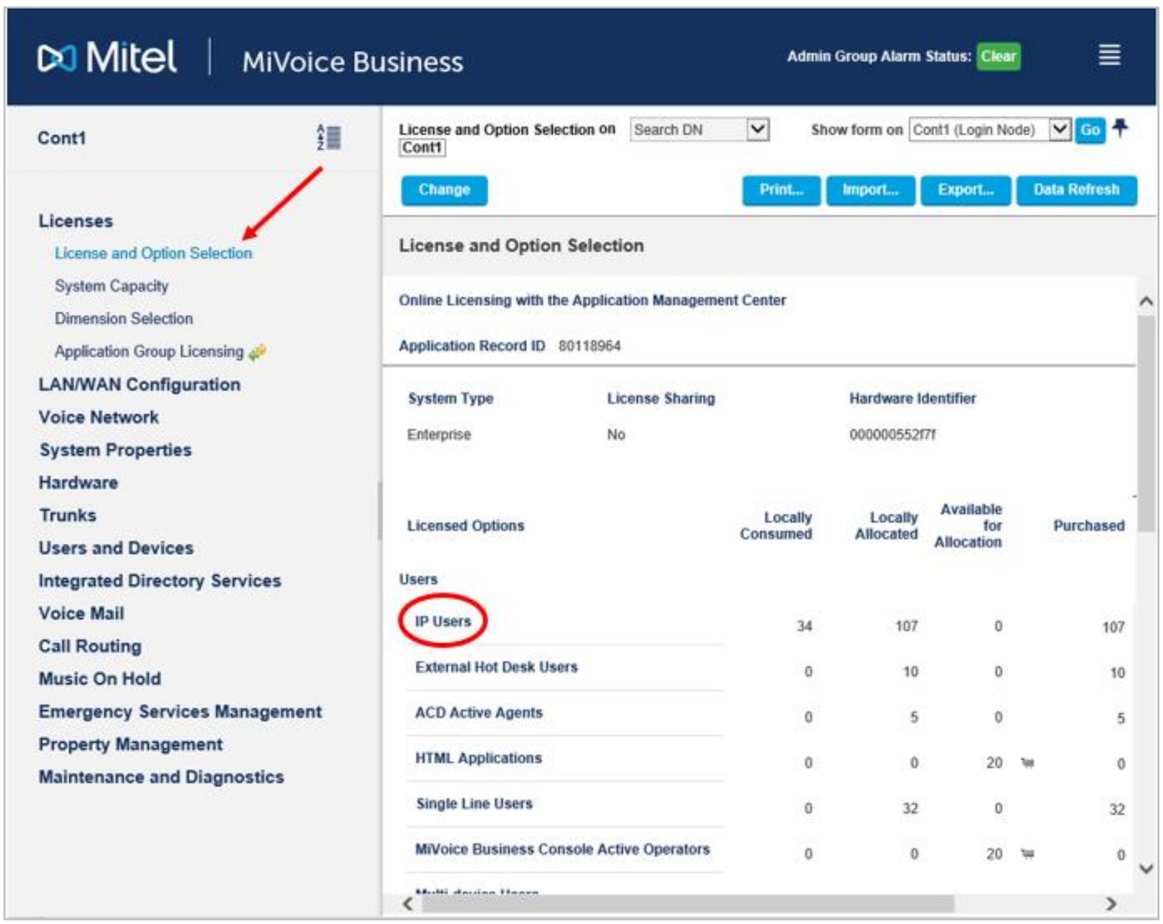
Chapter 2: Configuration Steps

The intent of this section of the guide is to provide a minimum series of steps necessary to create the configuration on the Mitel MiVoice System 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 Chapter 3 for configuration details regarding more advanced features and functionality.

1. Licenses

Ensure adequate licenses are available in the MiVoice Business System to support the Versity handset.

Verify licenses by navigating to the **Licenses> License and Option Selection** form in the MiVoice Business Controller. Each Spectralink handset will consume one ***IP Users*** License.



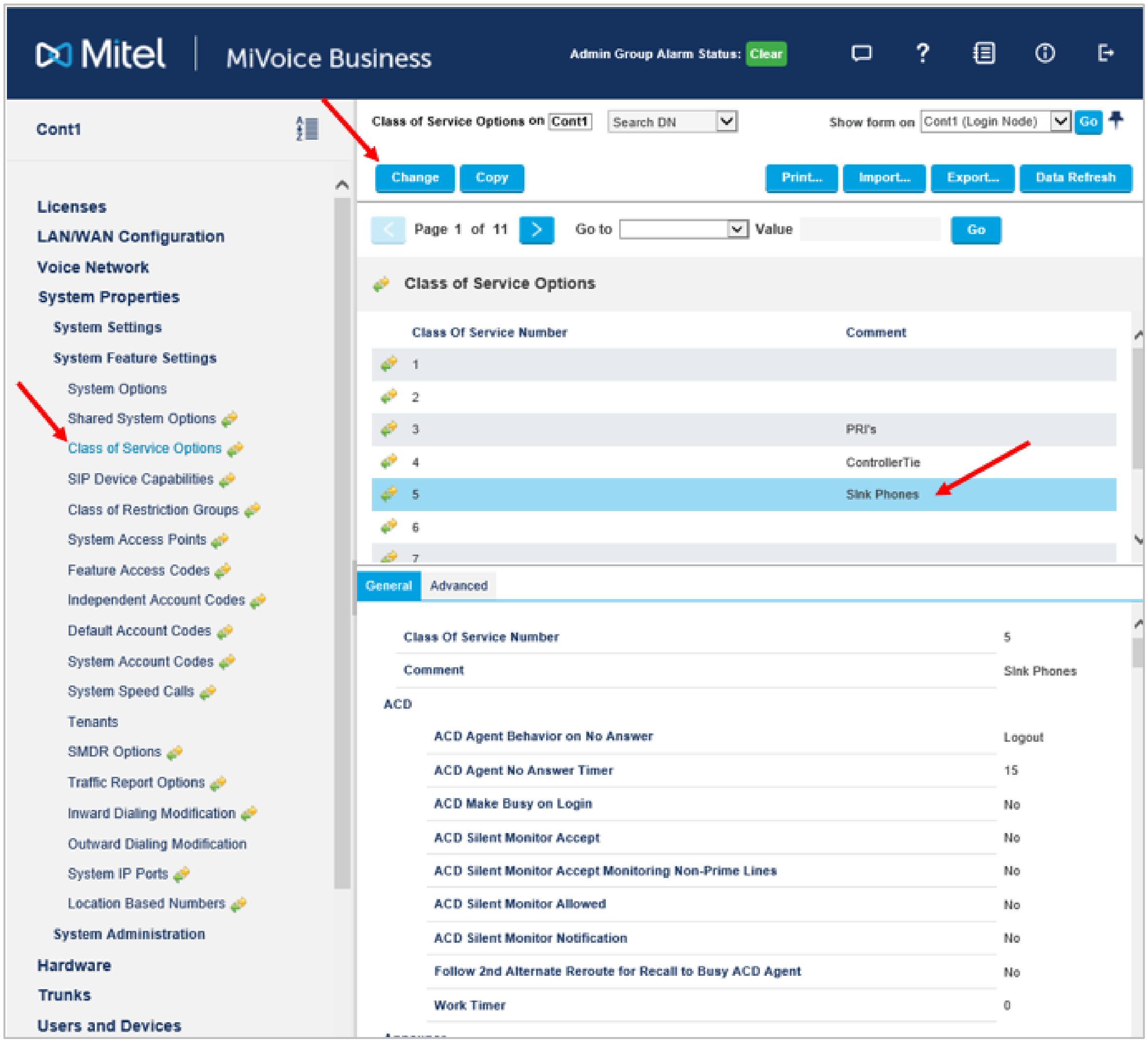
2. Class of Service

Build a Class of Service (COS) for the Spectralink Versity phones and specify its options.

The Class of Service Options form can be found by navigating to **System Properties> System Feature Settings> Class of Service Options.** 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 COS for the Spectralink Versity phones or utilize an existing COS as long as it conforms to the recommended values below.

1. Select the Class of Service Number you wish to modify and click **Change**. For purposes of our example, we will build a custom COS for the Spectralink Versity phones. There were only three options we found needed to be modified from the defaults:
2. In the **General** tab, scroll to and change the following three options.
   - **Public Network Access via DPNSS** set to **Yes.**
   - **Auto Campon Timer** is Blanked (Clear the existing value).
   - **Call Park – Allowed To Park** set to **Yes** (if you wish to Park) –please see the Call Park Feature Configuration section of this guide for known caveats.

Example: Class of Service



3. Define SIP Device Capabilities

Define a SIP Device Capabilities Number for the Spectralink Versity phones and specify its options.

The SIP Device Capabilities form allows us to customize the features and options the Mitel MiVoice System will use and accept when communicating with the Spectralink Versity phones. Spectralink recommends creating a unique SIP Device Capabilities number for the Versity phones. In our below example we create SIP Device Capabilities number 5, and list field values that were modified from the system defaults:

1 Navigate to **System Properties> System Feature Settings> SIP Device Capabilities**.

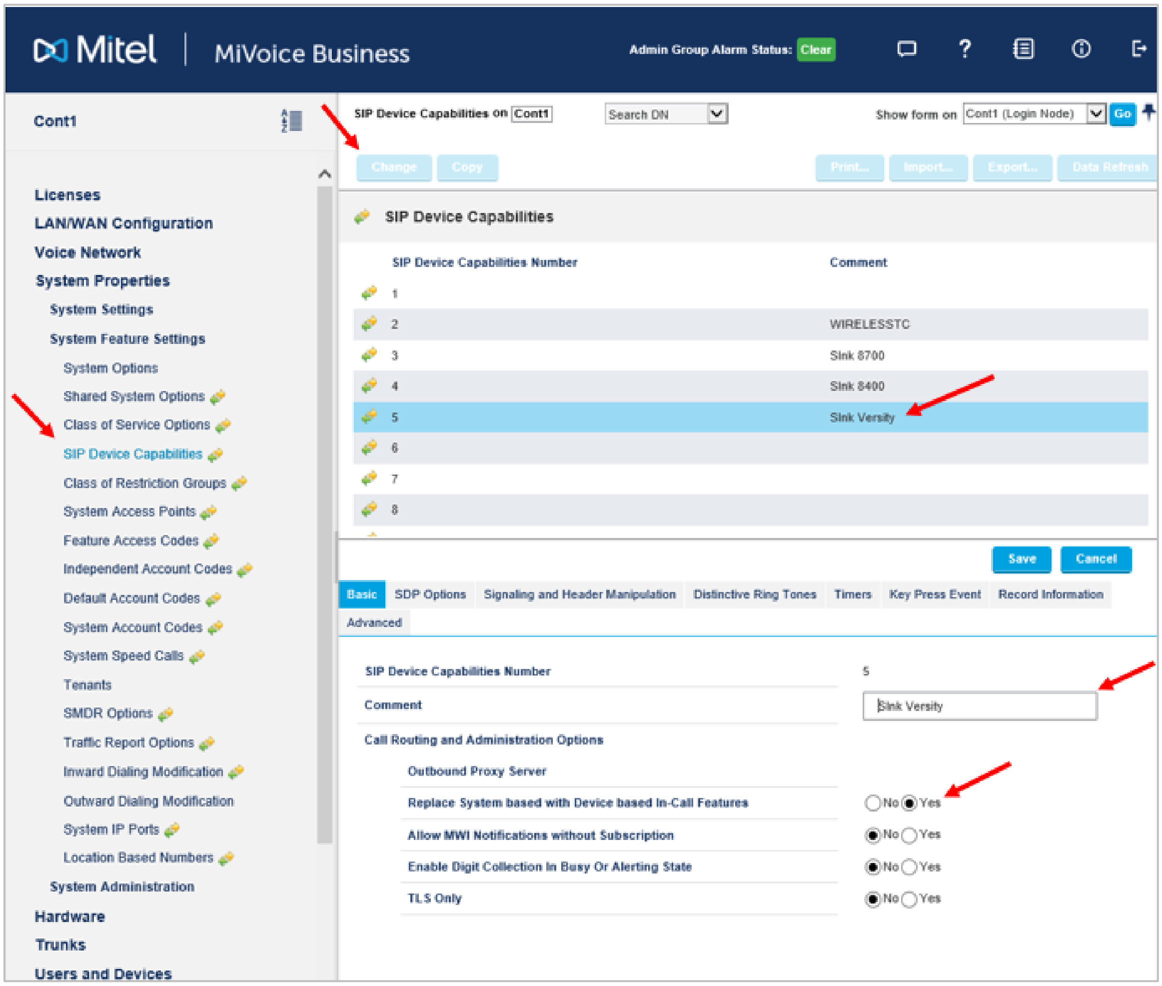
2 Select an unused **SIP Device Capabilities Number**, and select the **Change** softkey.

a In the **Comment** field on the Basic tab, give this SIP Device Capabilities Number a name, such as **SInk Versity.**

b Set the **Replace System based with Device based In-Call Features** value to **Yes.**

c Other values may be left to utilize their default values.

Example: Define SIP capabilities number



3 Next, Select the **SDP Options** tab on the SIP Device Capabilities form.

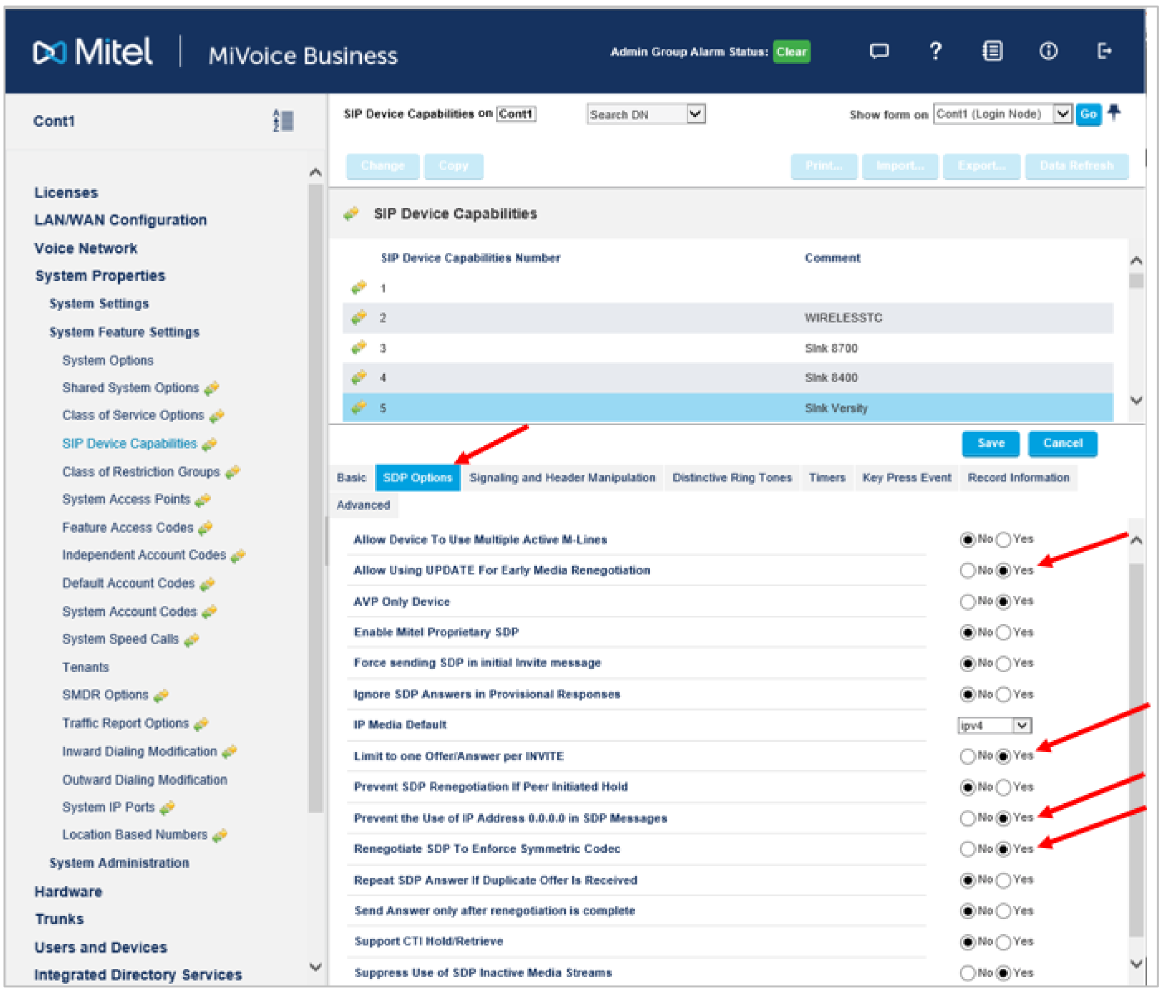
a Set the **Allow Using UPDATE for Early Media Renegotiation** value to **Yes**.

b Set the **Limit to one Offer / Answer per Invite** value to **Yes**.

c Set the **Prevent the Use of IP Address 0.0.0.0 in SDP Messages** value to **Yes**.

d Set the **Renegotiate SDP to Enforce Symmetric Codec** value to **Yes**.

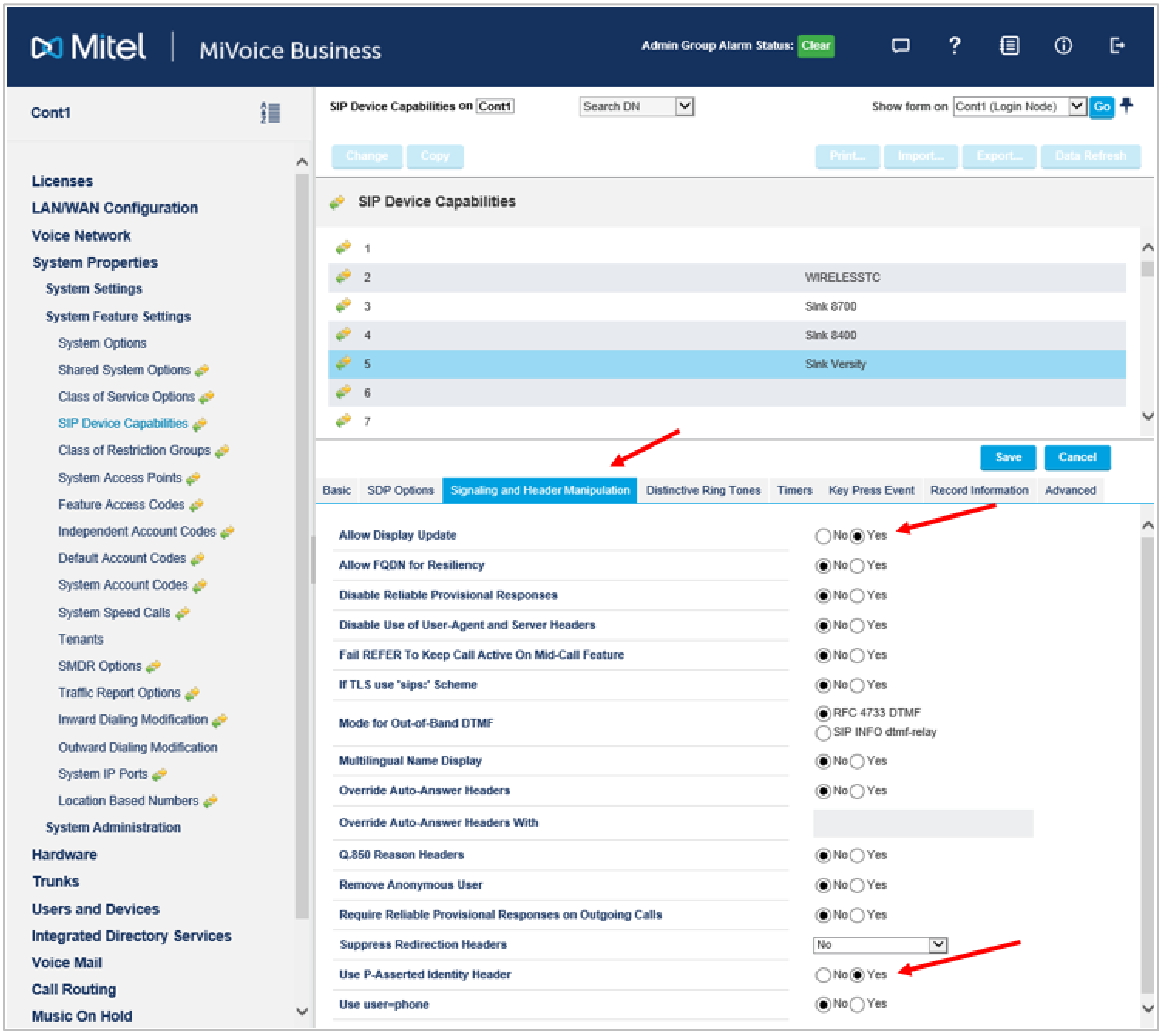
Example: SIP device capabilities options



4 Next, Select the **Signaling and Header Manipulation** tab on the **SIP Device Capabilities** form.

a Set the **Allow Display Update** value to **Yes.**

b Set the **Use P-Asserted Identity Header** value to **Yes.**

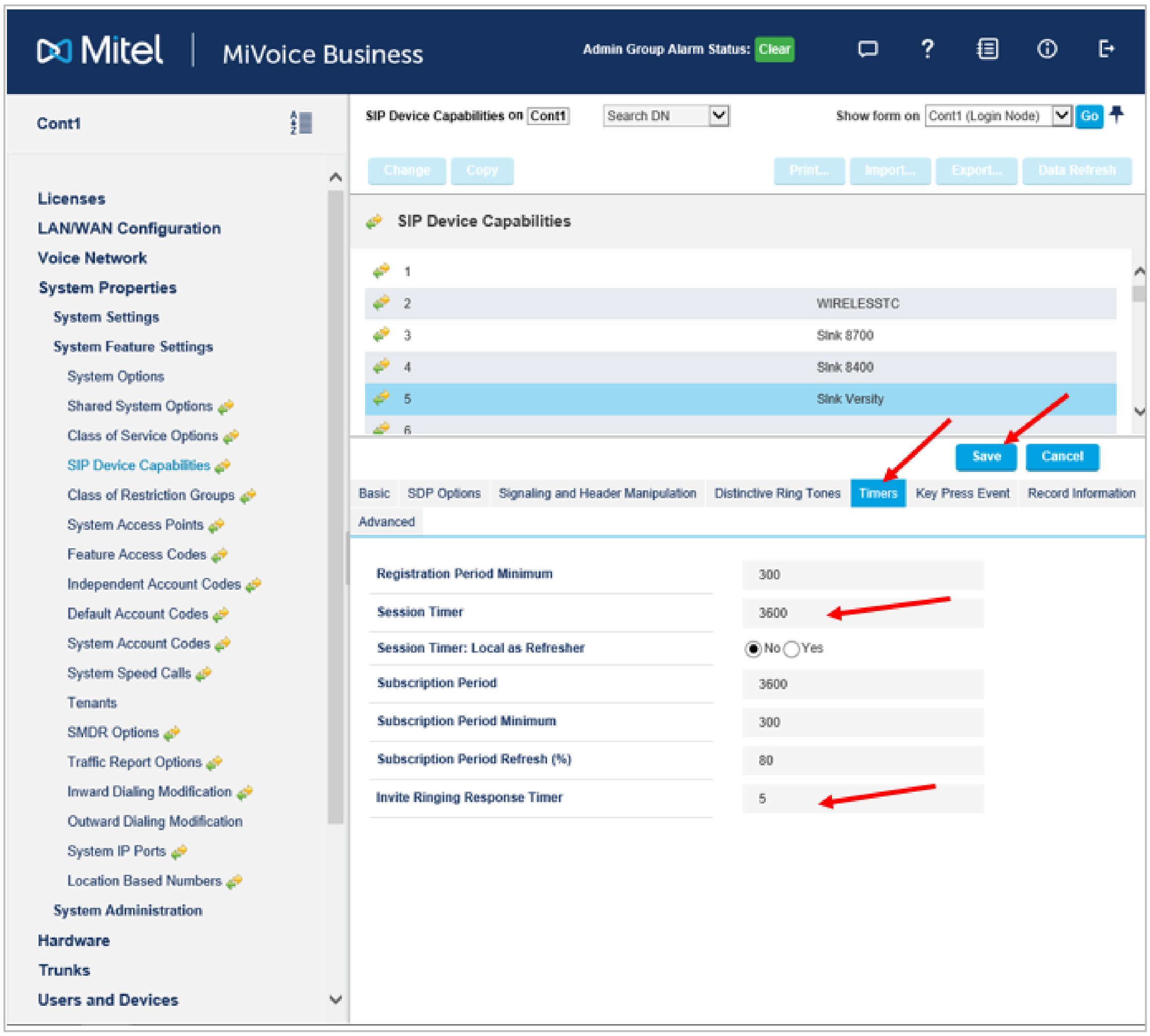


5 Next, Select the **Timers** tab on the **SIP Device Capabilities** form.

a Modify the **Session Timer** value to **3600.** (This will cause the system to require that phones refresh the session every hour. The system will tear the calls down if it does not receive a response from the phone. This could help to “free up” any calls that were not torn down correctly.)

b Modify the **Invite Ringing Response Timer** to a value of **5s.** (This will cause calls to stations that are currently unavailable because they have been powered off or left the wireless LAN’s coverage area before the registration has expired to receive the Out Of Service Handling treatment after 5s.)

6 Save the SIP Device Capabilities selections by clicking the **Save** button.



4. Configure Call Rerouting First Alternatives

Call Forwarding for No Answer, Busy, or Unregistered conditions should be configured on the Mitel MiVoice System itself. Whereas the Spectralink Versity does support a Call Forward All feature implemented from the handset, if the handset is powered off or not connected to the wireless network, the handset will not be able to respond to SIP Call Invitations with a message to indicate that the call should be forwarded. Therefore the Mitel system is a better alternative.

Configure the Call Rerouting First Alternatives Location

Below we will create a Call Rerouting First Alternative number (*3*) that will point all Calls for various unanswered / unavailable situations to our voicemail system's main number (*6000*). If you already have a Call Rerouting First Alternatives number that forwards calls to the location you desire you may skip this step, but below we chose an unused Call Rerouting First Alternatives Number (3) and modified it to direct all calls to the main Voicemail number (6000).

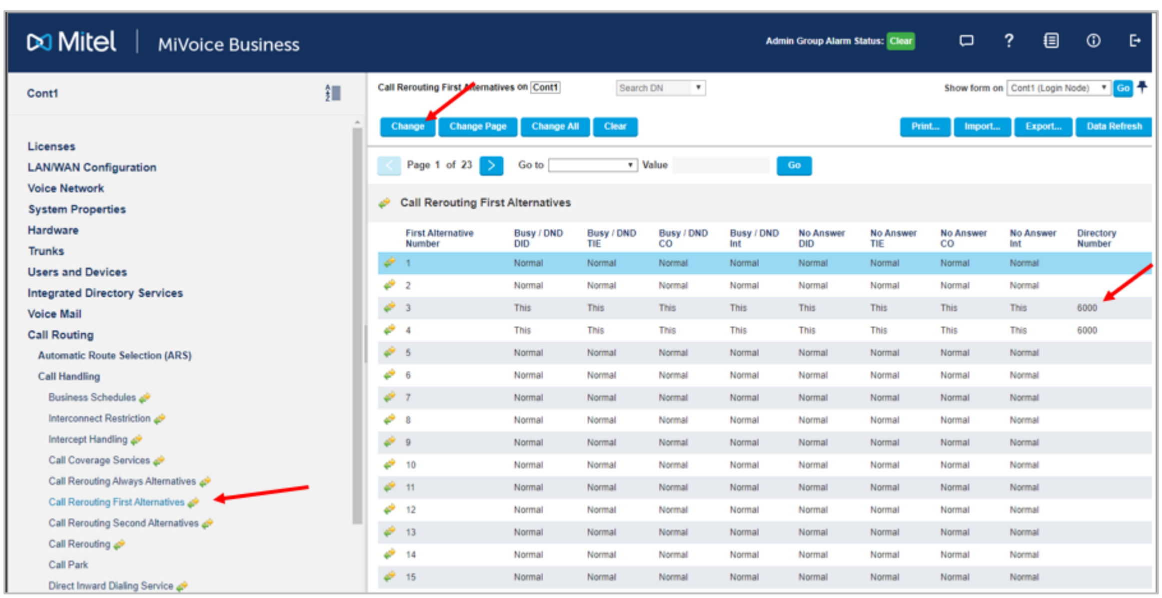
1 Navigate to **Call Routing> Call Handling> Call Rerouting First Alternatives** form.

2 Select the **Call Rerouting First Alternatives Number** you wish to use.

3 Click **Change**.

4 In the form that appears, modify the Call Rerouting First Alternatives Locations to point to the desired destination. In our example we pointed all Call Forwarding First Alternatives destinations to the destination of 6000 (our Voice Mail Pilot number), by selecting **This** as the destination for each, and then specifying the destination number **6000** in the Directory Number field.

Example: Call rerouting



5 Click **Save** when you are finished.

5. Add the User and the Device

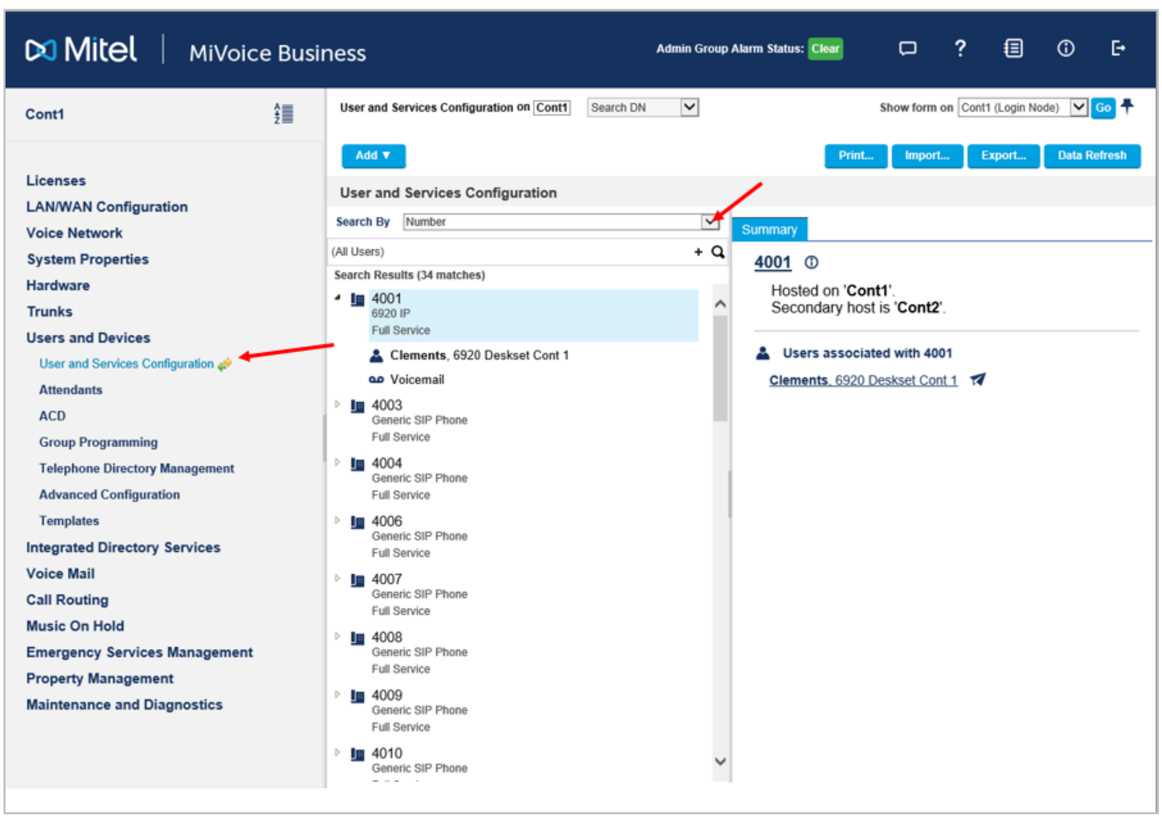
Devices may be programmed in the IP Telephones forms, but for our example we will construct the User and specify the Device Details by navigating to **Users and Devices> User and Services Configuration**.

Choose a Number for your new Device

1 If you already know the number you will use for your Device's primary extension number, you may skip to the next step, but for our example below, we will find an appropriate extension number by modifying the **Search By** field to show: **Number.**

2 Note the existing extension numbers currently in use and choose one that is currently unused and that aligns with your site's dial plan. For our example we will be adding Extension Number 4107.

Example: Searching for available extensions



Add the User

When you have identified the Extension Number you will add the User:

1 From the **User and Services Configuration** form, click **Add**.

2 Select **Default User and Device** from the pull down menu that appears when you click **Add**.

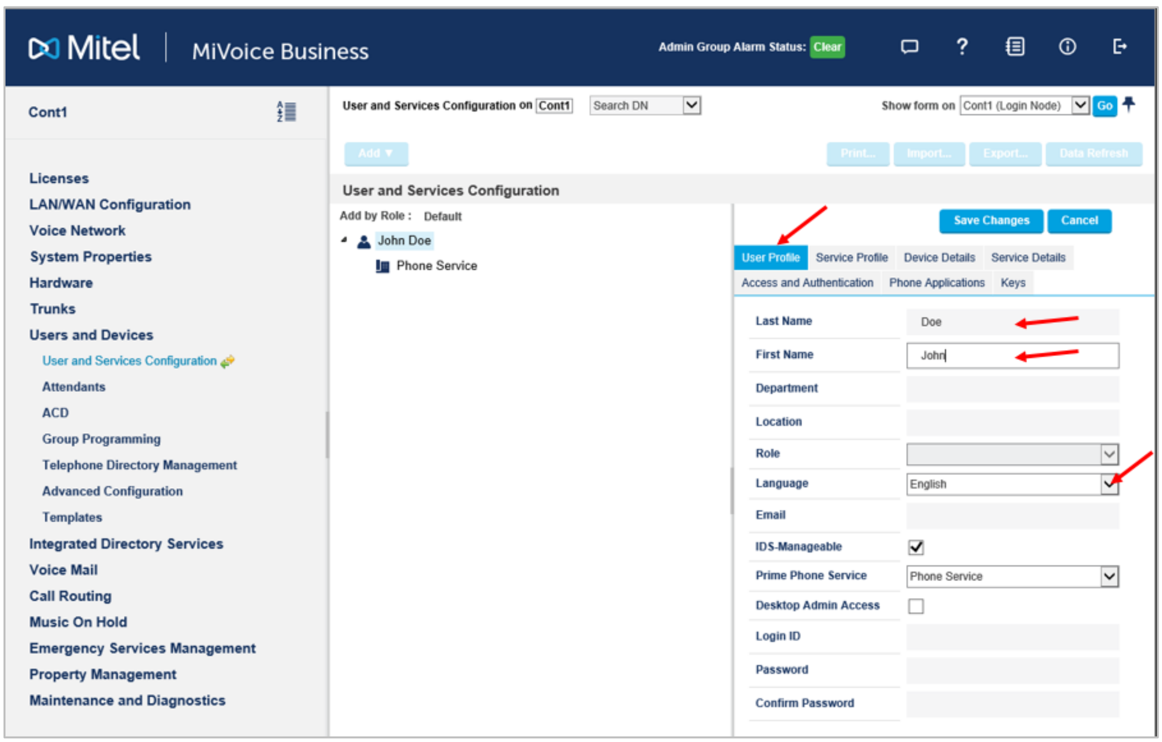
3 On the **User Profile** tab fill in the following fields at a minimum.

4 **Last Name**: Fill in a last name for the user (*Doe*).

5 **First Name**: Fill in a first name for the user (*John*).

6 **Language**: Select the language (*English*).

Example: Adding a user

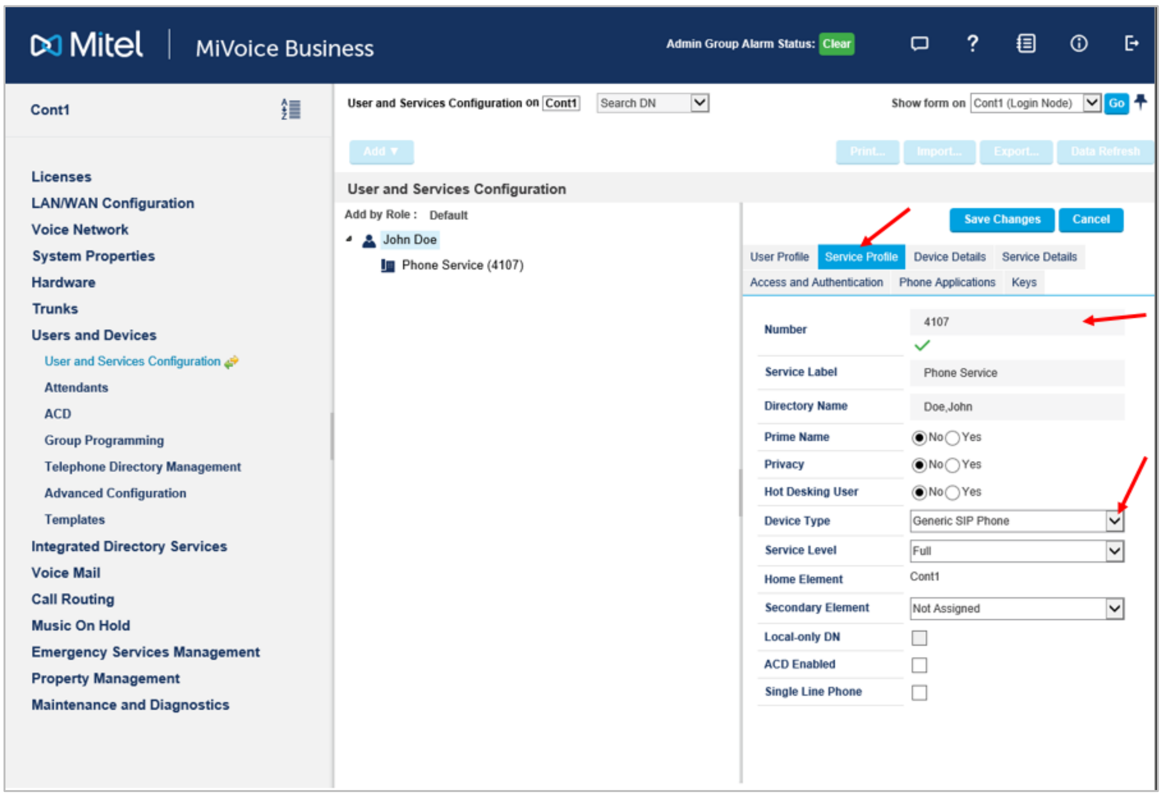


Define the Service Profile

On the **Service Profile** tab fill in the following fields at a minimum.

- **Number**: Enter the main extension number you wish to use for your device (*4107*). This will correspond with the Extension Number value in the Spectralink Versity' SIP configuration parameters.
- **Device Type**: Select **Generic SIP Phone**.

Example: Service profile

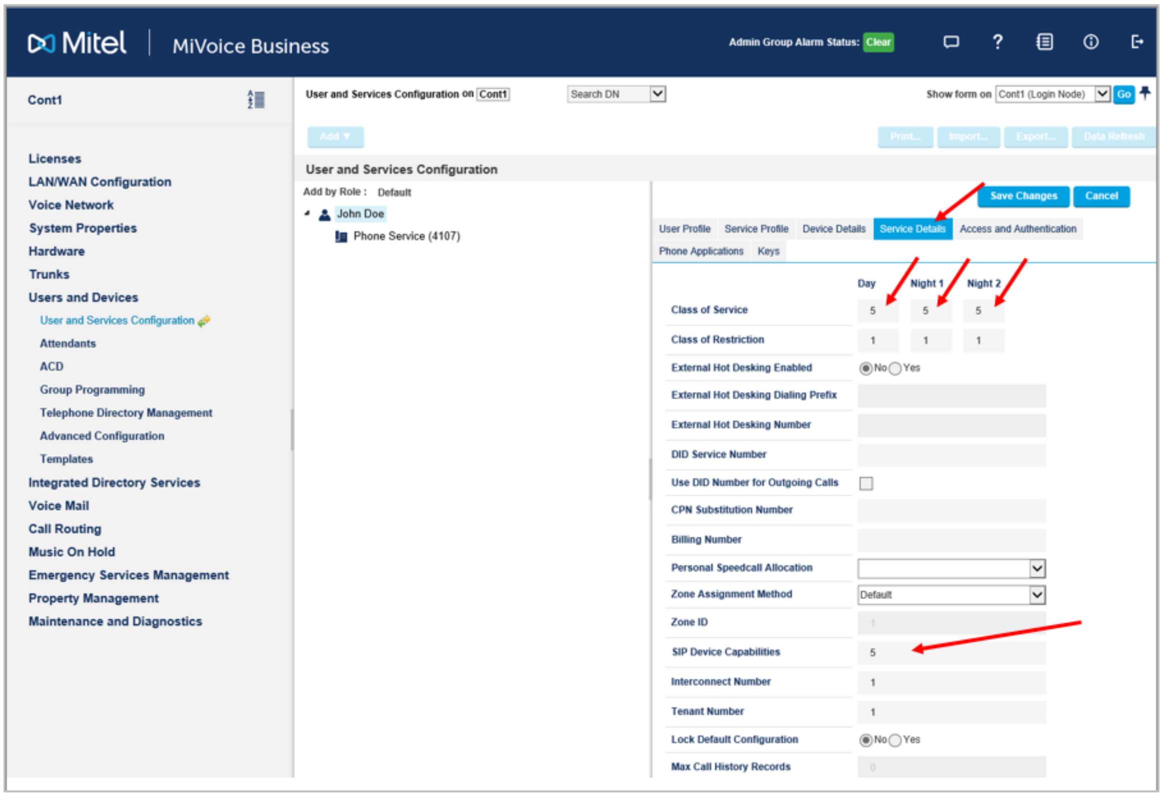


Configure Service Details

On the **Service Details** tab fill in the following fields at a minimum

- **Class Of Service**: Enter the Class of Service Number you created for the Versity phones in the Class of Service section of this document. (*5*)
- **SIP Device Capabilities**: Enter the SIP Device Capabilities number you created for the Versity phones in the SIP Device Capabilities section of this document. (*5*)

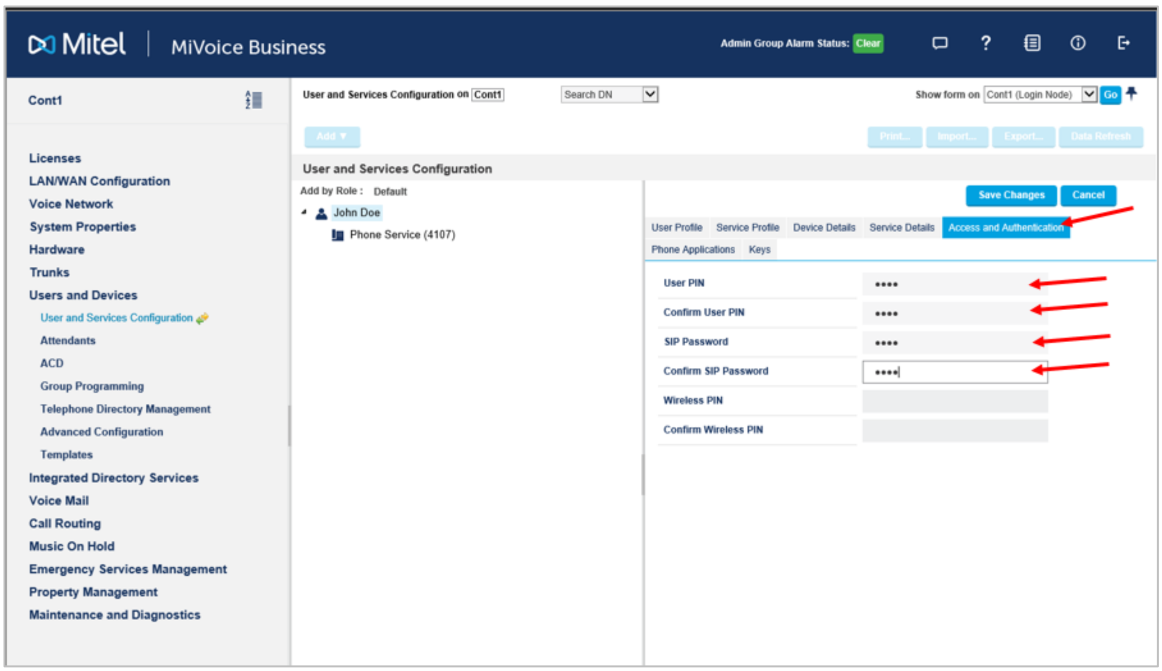
Example: Service details



Configure Access

On the **Access and Authentication** tab fill in the following fields:

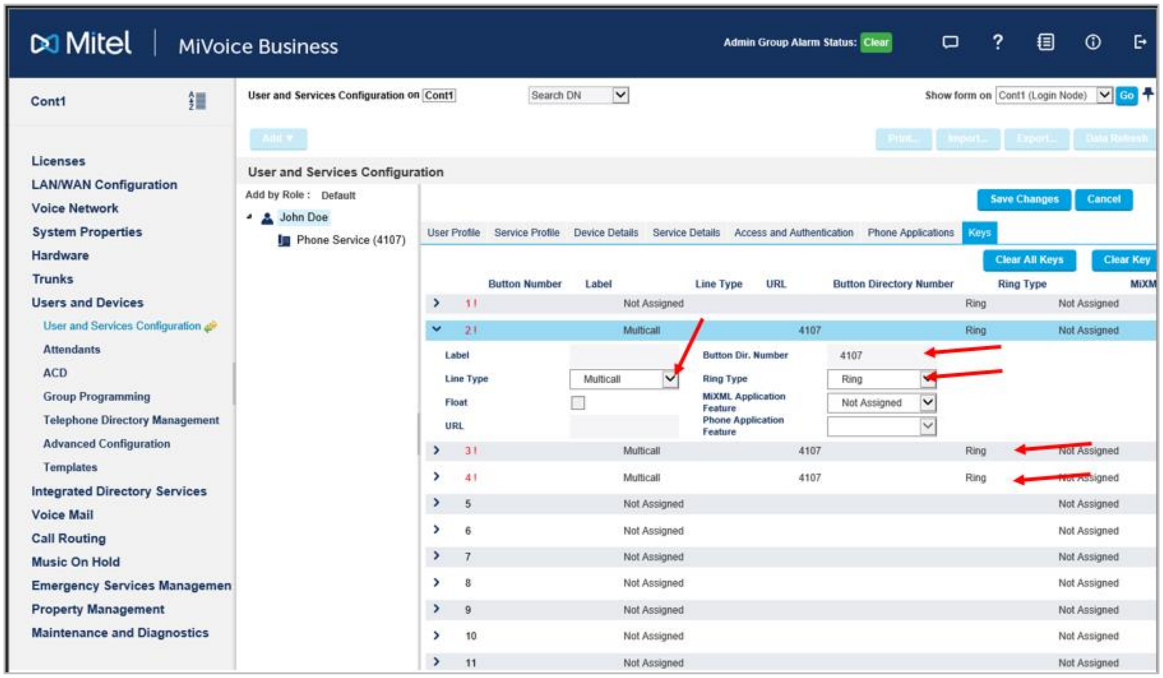
- **User PIN**: Normally this is used for Hot-desking, and other Mitel specific features, but may be used for the SIP Password after a system restore or an upgrade, and as such, Spectralink recommends making the User PIN have the same value as the SIP Password. (*1234*). See Example: Manual SIP configuration.
- **Confirm User PIN**: Enter the value above (*1234*).
- **SIP Password**: This should correspond to the value used as the Password in the SIP phone menu of the Versity phone. (*1234*).
- **Confirm SIP Password**: Enter the value above (*1234*).



Add Multicall Buttons

On the **Keys** tab add up to 3 Multicall Buttons (numbers 2, 3 and 4) to allow the phone to place and receive more than one call. (Spectralink Versity supports up to 4 calls and the first “call” is already configured.)

1. Select the radio button for the additional call you want this extension to support.
2. **Line Type**: Select **Multicall***.*
3. **Button Dir**. **Number**: enter <the extension number being configured>. (*4107*)
4. **Ring Type**: Select **Ring**
5. If more calls are desired, add buttons 3 and 4 the same way.



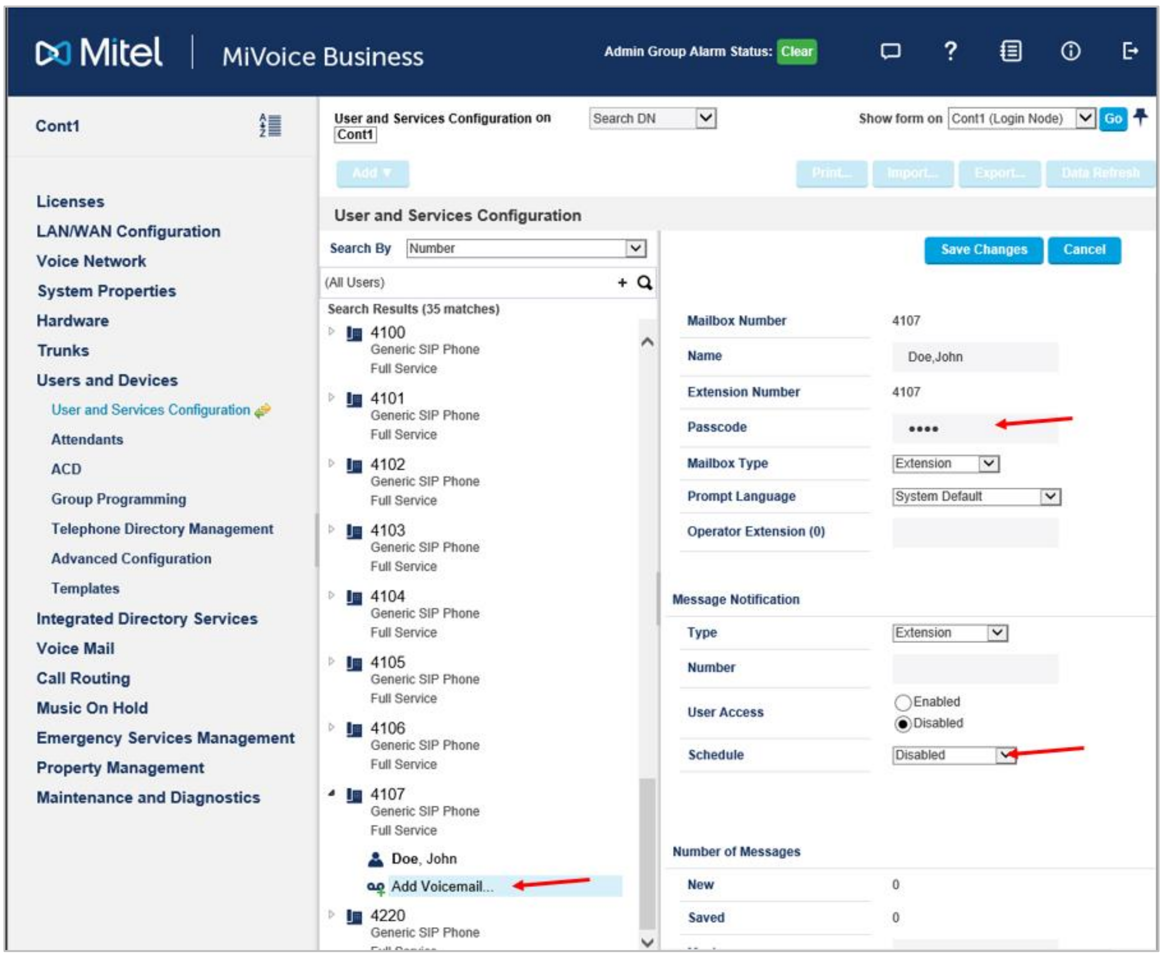
6. Click **Save Changes** to ensure that your User and Device are saved in the system.

6. Add a Voicemail Box (if desired)

1 From the **User and Services Configuration** form, expand the **Phone Service** selection under the User's name.

2 Select **Add Voicemail**. Voicemail boxes should be configured about the same as any other user's Voicemail box would be on your system. For our purposes, we changed only the following fields:

- Passcode: The password to be used the first time the user logs into the Voicemail system.
- Schedule: **Disabled** (Unless you want the system to notify you via a callback every time a message is left for your device).



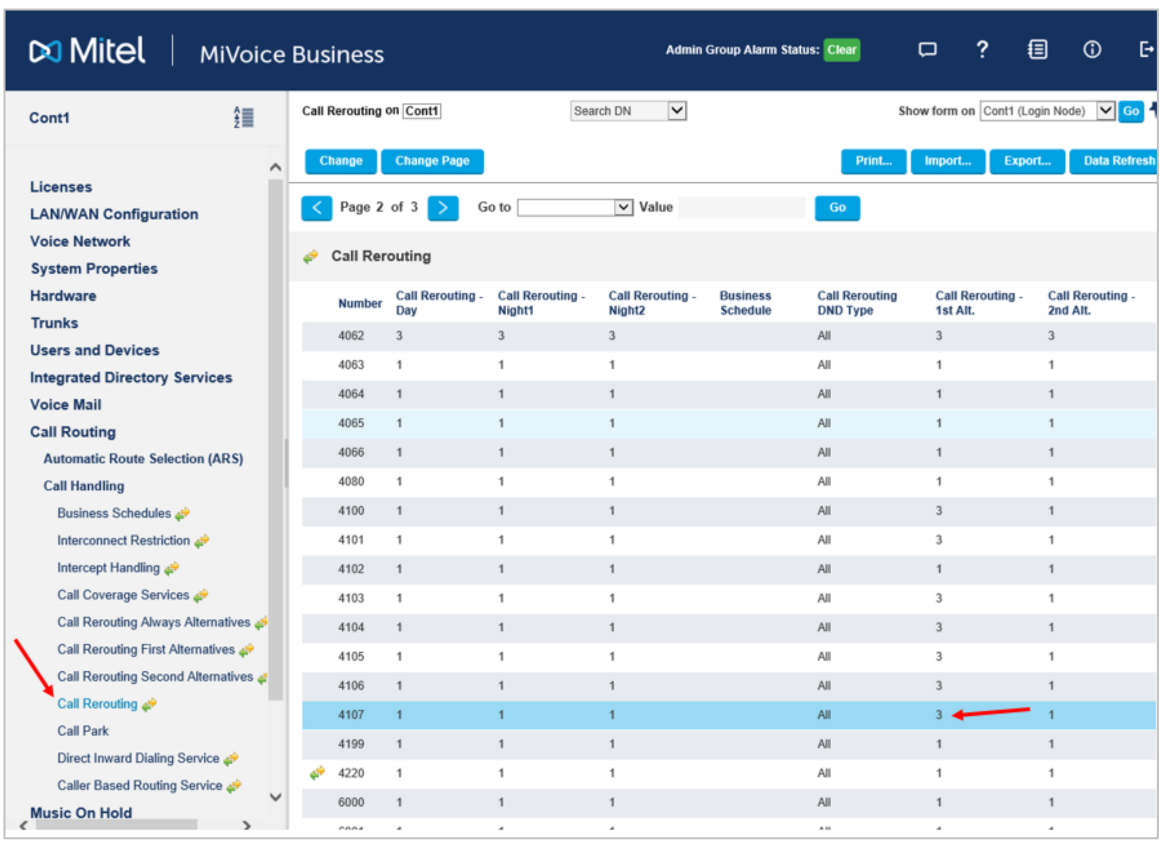
7. Configure Call Re-Routing For Your Extension

Earlier, in the Configure Call Rerouting First Alternatives section, we created a Call Rerouting First Alternatives number to specify the location we would like calls to be sent in various no answer, busy and unavailable scenarios. In this section, we will assign that number to our individual extension.

Configure the Call Rerouting Assignment

In this example, we assign the Call Rerouting First Alternatives Location number (*3*) to the Spectralink Versity extension (*4107*).

1. Select the **Call Routing> Call Handling> Call Rerouting** form.
2. Highlight the extension you need to modify and select the **Change** button.
3. Enter the Call Rerouting 1st Alternative number you created above in the Call Rerouting – 1st Alt field for your extension (*3*).
4. Click the **Save** button when you are finished.



8. Configure the Spectralink Versity Handset to Register with the Mitel MiVoice Server

The first step in connecting the Spectralink Versity smartphone to the Mitel MiVoice system is to get the smartphone connected to the wireless LAN. This can be done manually on each smartphone in several ways:

- Manually configure Wi-Fi settings in the settings menus,
- Use the Android Setup Wizard if the phone is in a factory default configuration,
- Use the NFC “bump” from a Relay Agent at the initial Android Google Setup Wizard (see the *Versity Deployment Guide*).

Additional 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 open the Biz Phone app and configure the SIP parameters for the handset so that it can connect to and register with the Mitel MiVoice system.

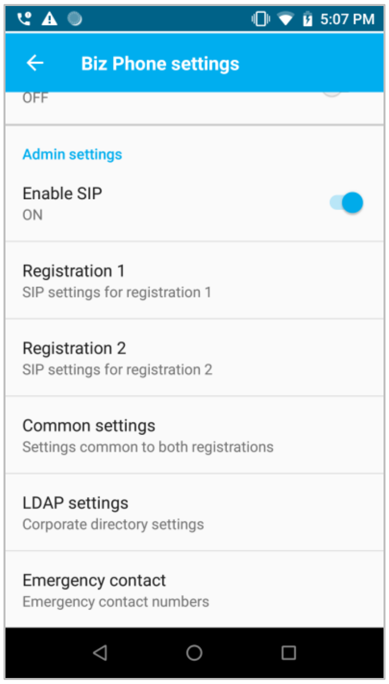
The SIP configuration fields are basically the same whether provisioned through the Spectralink Application Management (SAM) server or by using the Biz Phone app settings menu on Versity. You may use either interface to specify the required parameters.

Caution: Ensure no other SIP dialer is active

The Biz Phone app must be the only active SIP dialer.

Example: Manual SIP configuration

Below, we will illustrate a manual configuration example with the sample user (John Doe) and handset extension (4107) we built in the preceding steps. This will be accomplished through the Biz Phone application, under settings. We will show the resulting configuration in the handset itself. If you have more than a few handsets to deploy, Spectralink strongly recommends use of SAM. Some comments and details about how to tailor these fields for your unique environment are also provided below.



Under Admin Settings:

Enable SIP**: *On***. Ensure this is set to On. If you disable this, The Spectralink Biz Phone app will not function (This setting is called Enable SIP in SAM, and needs to be checked to function properly).

Under Registration 1:

SIP server**: *172.29.102.175.*** This value should be replaced with the IP address of your Mitel MiVoice System. You may also use a DNS A-Name record or a DNS SRV record to specify the server address.

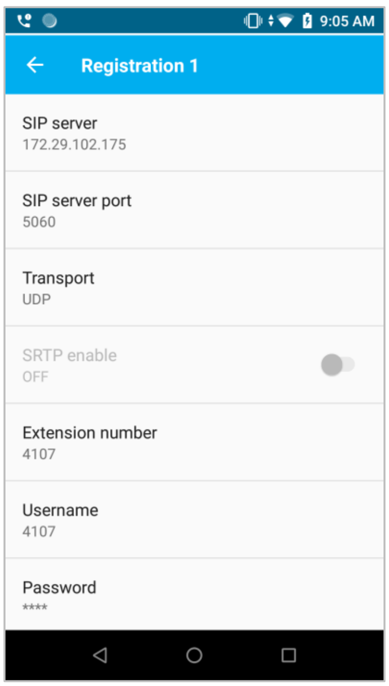
SIP server port**: *5060.*** Spectralink's lab server uses UDP port 5060 for SIP communications and that is the default, so there is no need to change this field if your site is utilizing port 5060 for SIP communications. If your installation uses a different port for SIP communication you could either specify this value here or you could specify the SIP server port number through the use of a DNS SRV record.

Transport: ***UDP*.** Spectralink strongly suggests using UDP as the selected Transport Protocol. Problematic out-of-range behavior was observed in the Spectralink lab when using TCP (and by extension, TLS) with the Mitel MiVoice system. Please see the ***SIP Using TCP or TLS*** section of this document for additional information.

Extension number**: *4107.*** This value should be replaced with your phone's extension number. This value corresponds with the Number value you entered for your device on the Service Profile tab of the User and Services Configuration form.

Username**: *4107.*** This value should also be the same as your Extension number. This value also corresponds with the **Number** value you entered for your device on the Service Profile tab of the User and Services Configuration form. In SIP terms, this is the SIP digest authentication username.

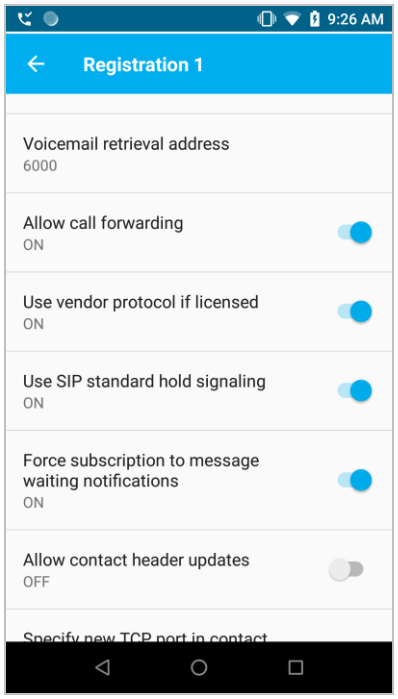
Password**: *1234.*** This value should be replaced with your End User's SIP digest authentication credentials. This value corresponds with the SIP Password value you specified on the Access and Authentication tab of the User and Services Configuration form on the Mitel MiVoice System. See Chapter 2.8. Configure the Spectralink Versity Handset to Register with the Mitel MiVoice Server.



Voice mail retrieval address**: *6000.*** 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 (the default) to utilize rfc3261 style hold.

Force subscription to message waiting notifications: ***On.*** Using the configuration settings described in this guide, the user must send SIP Subscribe messages to the Mitel MiVoice System in order to obtain SIP Notify messages indicating Message Waiting Status.



Under Common Settings:

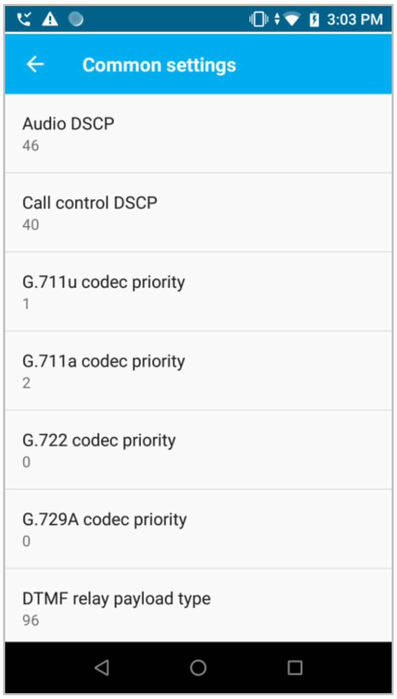
Audio DSCP**:** ***46.*** This is the default value. It should not be necessary to modify this default unless specifically advised to do so under the requirements of the Spectralink VIEW deployment instructions.

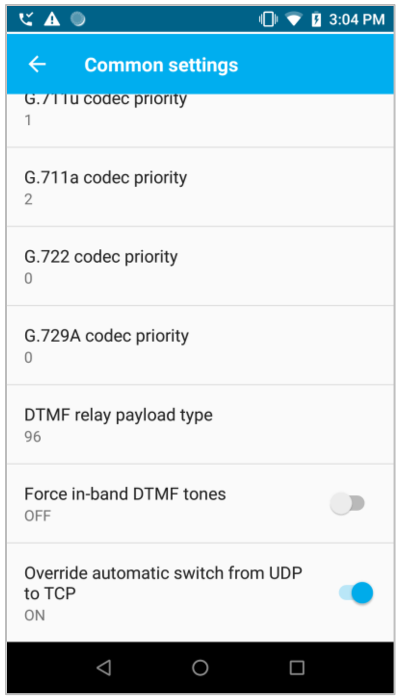
Call Control DSCP**:** ***40.*** This is the default value. It should not be necessary to modify this default unless specifically advised to do so under the requirements of the Spectralink VIEW deployment instructions.

G.711u, G.711a, and G.729A codec priorities**:** ***G.711u =1 and G.711a = 2. G.729A=0***. These values are the defaults. 1 is the highest priority and enabled, 2 is second priority and enabled. A value of 0 disables the codec. Modify as desired for your site.

G.722 codec: **G.722=0.** Some interoperability issues with the G.722 codec and the Mitel MiVoice System were discovered in the course of Spectralink labs' testing, so at the time of this writing, Spectralink does not recommend the G.722 codec be enabled on the Versity Handsets.

Override automatic switch from UDP to TCP: ***On***. This setting overrides the RFC 3261 requirement to use a congestion controlled transport protocol when a packet becomes larger than 1300 MTU. As noted in the guide, Spectralink has observed problematic out-of-range behavior with the Mitel MiVoice System when utilizing TCP. As such, this setting will allow the protocol selection to remain UDP.


3:03 PM
Common settings
Audio DSCP
46
Call control DSCP
40
G.711u codec priority
1
G.711a codec priority
2
G.722 codec priority
0
G.729A codec priority
0
DTMF relay payload type
96

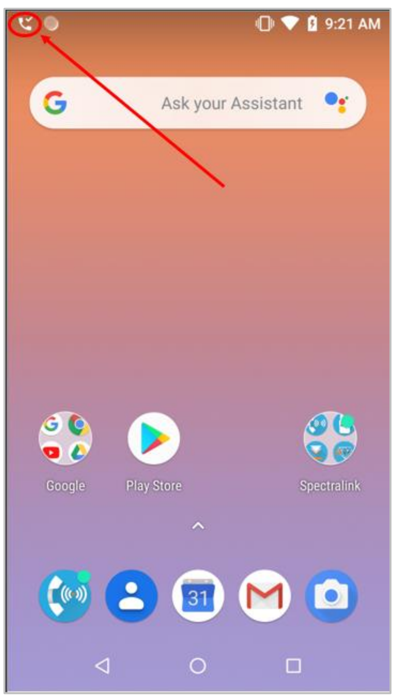

3:04 PM
Common settings
1
G.711a codec priority
2
G.722 codec priority
0
G.729A codec priority
0
DTMF relay payload type
96
Force in-band DTMF tones
OFF
Override automatic switch from UDP to TCP
ON

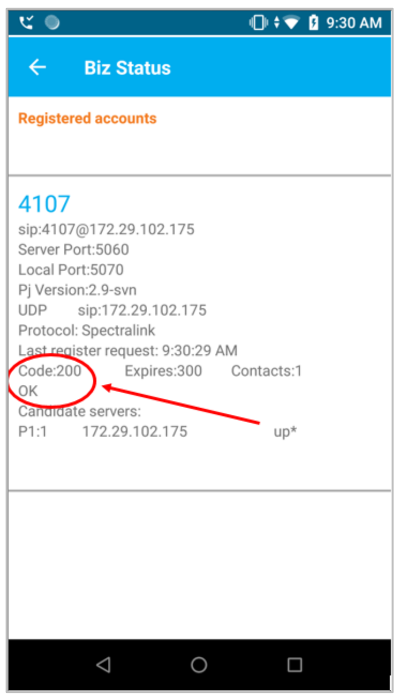
9. Verify Registration Status

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

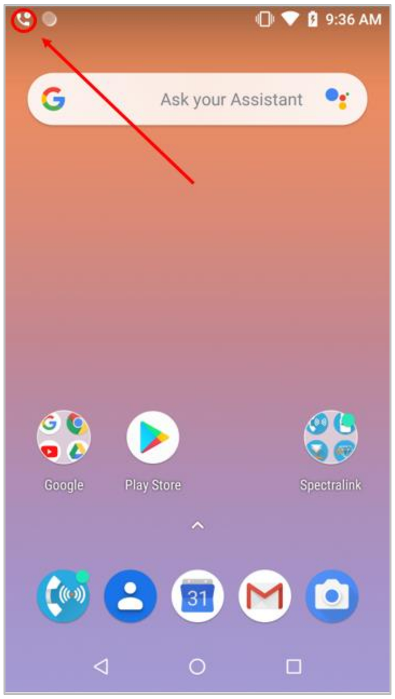
Successful registration status examples: The below screenshots show an example of a phone that has achieved a successful registration with the call server. This shows the idle screen icon and the Biz Status screen information (available within the Biz Phone application) you might expect to see if the phone has successfully registered with the Mitel MiVoice System.

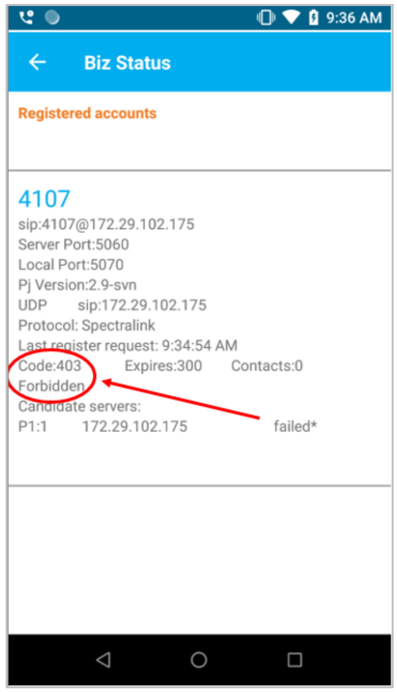
The ***200 OK*** is the call server's success response to the registration request. If you do not observe a ***200 OK*** in this area of the screen, then the registration request is failing. The error code returned by the server may provide some additional hints as to the reason for the failure.



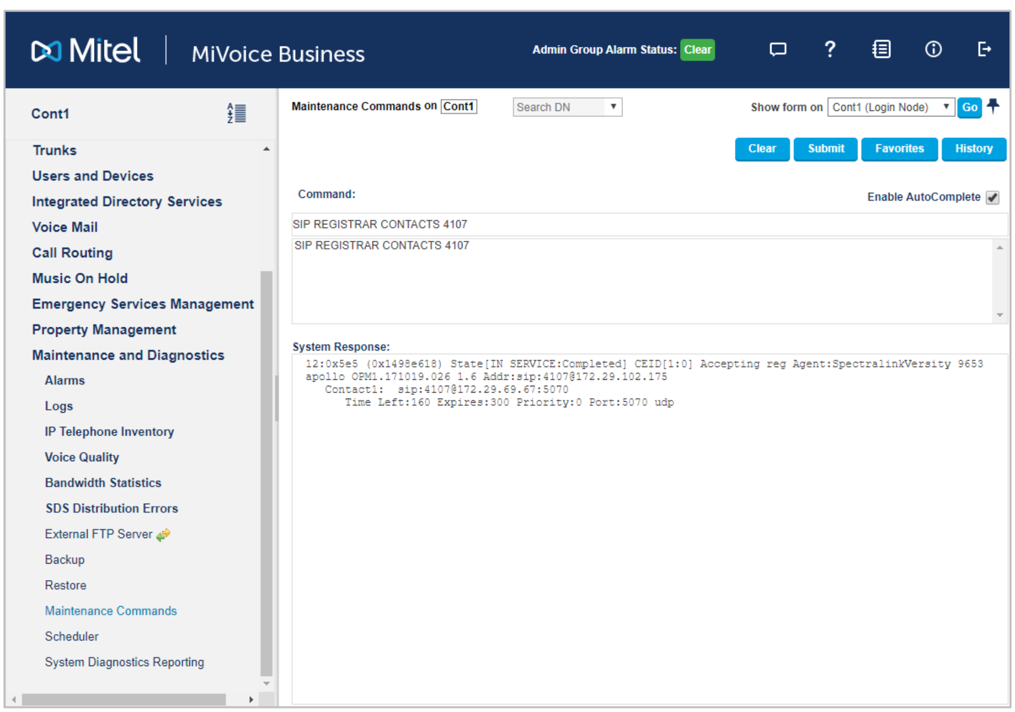


Failed Registration: The below screenshots show a registration failure. In this case we can observe that the server thinks the registration has failed (because there is not a 200 OK here). The 403 response typically indicates that the username and password provisioned in the Versity handset do not match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





Registration Status in the MiVoice Business System: You can check the status of the Versity handset according to the MiVoice Business System by navigating to **System Administration> Maintenance and Diagnostics> Maintenance Commands**. Then enter the command SIP REGISTRAR CONTACTS XXXX (where xxxx is your extension number). The below shows an example of a successful registration and details the device's IP address.



10. Test

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

- Basic Call to and from the Versity handset to another MiVoice Business 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.
- 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

Mitel MiVoice System SIP Registration

Spectralink Versity handsets register directly to the Mitel MiVoice System.

SIP Digest Authentication

The Mitel MiVoice System requires the use of SIP Digest Authentication. The Username and Password fields in the Biz Phone Settings -> Registration menu control the credentials the Versity phone will utilize for these parameters.

The Password value in the Biz Phone Settings -> Registration menu corresponds with the SIP Password value you specified on the Access and Authentication tab of the User and Services Configuration form on the Mitel MiVoice System.

The Username value corresponds with the Number value you entered for your device on the Service Profile tab of the User and Services Configuration form.

Basic Calls

Call functionality was tested by calling between Spectralink Versity handsets as well as to and from a Mitel 6920 IP phone. No special Versity configuration parameters should be required in order to realize this ability.

Voicemail Integration

The below parameters were found to help optimize the Mitel MiVoice System's voicemail integration with the Versity phones:

Force subscription to message waiting notifications: ***On.*** In many PBX integrations, SIP devices are automatically subscribed to receive Message Waiting Indicators (MWI's) when they register. While there are configuration parameters in the Mitel MiVoice system that would allow us to achieve this, we found that a more reliable integration was possible by requiring the extensions to subscribe for MWI status. Setting this parameter will help to mitigate the problem where MWI could be lost when the phone is out of wireless range.

Please note also that the MiVoice Business System 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 Versity phone cannot provide a message count to the user, but will provide an indicator that it has message(s) waiting.

Voice mail retrieval address: ***6000.*** 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 MiVoice Business 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. It should be noted that in the course of testing, Spectralink labs observed that at times, it took up to 60 seconds to deliver MWI updates to the handset after a message had been left for the user or deleted from the user's voicemail box for the server.

Call Waiting

By default, when you build an extension, the Mitel MiVoice System places only one key with that extension number on each phone and any additional calls will receive the Call Re-Routing First Alternatives Assignment defined for your extension. If you have added Multicall Keys per the Add Multicall Buttons section of this guide then your phone will be able to receive up to four calls total. Instead of being re-routed, incoming calls that occur during an active call will cause the phone to prompt the user with an in-ear tweedle and the phone will display the additional call being offered.

To verify current Call Waiting configuration navigate to **System Administration> Users and Devices> User and Services Configuration**, and the **Keys** tab of the Device you are interested in. There should be a Multicall Key for each additional Call Waiting appearance you have defined up to the Versity supported maximum of three additional Multicall buttons, or four simultaneous calls per device.

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 Mitel MiVoice System. 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 they are in 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.

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. Additionally, the Versity handsets support the p-asserted identity header 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 in the course of a transfer.

Hold and Resume

Spectralink Versity handsets are capable of hold and resume and utilize the RFC 3261 hold mechanisms (setting SDP to sendonly). Note that a call that is placed on hold from both ends will be dropped by the ICP. This is referred to as a double hold scenario and is a Mitel documented behavior for any SIP Line Side device connected to the MiVoice platform. If other failures or issues are experienced with hold and resume behavior, please verify that the SIP Device capabilities form the extension references has the **Prevent the Use of IP Address 0.0.0.0 in SDP Messages** value set to **Yes**.

Music On Hold

Spectralink Versity handsets are capable of hold and resume, and in Spectralink's lab environment, clients placed on hold by a Spectralink Versity handset were able to hear the system supplied (MOH) Music On Hold.

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 called phone no longer rang. The Spectralink Versity phone sends a 486 Busy message back to the Mitel MiVoice System when the user rejects an offered call. Many servers will immediately forward callers to the Call Forward Busy location in this scenario, however, the Mitel SIP Line Side Interop document specifies that the Mitel MiVoice System determines if a device is busy or not based on the number of lines programmed. The result is, that when the Spectralink Versity phone rejects a call and sends a 486 Busy message to the Mitel MiVoice System, the Mitel MiVoice System continues to play ringback to the calling party until such time as the Call Forward No Answer (CFNA) Timer expires and the call is sent to the Call Rerouting First Alternatives CFNA destination specified after that time.

Do Not Disturb

Do Not Disturb Using the Handset

The Versity handset will honor the Android Do Not Disturb function and preferences designated in the Do Not Disturb settings. If a call is generated to the Versity handset with the Android Do Not Disturb feature activated, the phone will return a '486 Busy Here' to the Mitel call server. Mitel will still play ringback to the originating caller until the configured Call Forward No Answer (CFNA) Timer expires, and will then send the call to the Call Rerouting First Alternatives destination.

Users wishing to direct only Biz Phone traffic to a Do Not Disturb location might also consider leveraging the Call Forward functionality to direct all incoming callers to voicemail by implementing Call Forward, and populating the Call Forward destination with the voicemail pilot number (or whatever other number they would like calls to be directed to), essentially achieving the same result as Do Not Disturb. When a call forward occurs, the forwarding phone will vibrate slightly to notify the user they have missed a call, and the call log on the Versity handset will show the forwarded call as a missed call.

Do Not Disturb Using Feature Access Codes (FACs)

This method of implementing Do Not Disturb was tested by dialing the Do Not Disturb and Do Not Disturb – Cancel FACs. One advantage to this method of implementing Do Not Disturb is that the Do Not Disturb functionality remains in effect regardless of whether the handset remains powered on or in range of the wireless network. The disadvantage to this method of Do Not Disturb implementation is that the phone does not provide any user friendly indication that the phone is in the Do Not Disturb state when Do Not Disturb has been activated on the Mitel MiVoice System using a FAC. The Do Not Disturb state is maintained by the Mitel MiVoice System itself and the Mitel MiVoice System will simply never offer calls to the phone until the Do Not Disturb is cancelled using the Do Not Disturb - Cancel FAC. As such, calls given the Do Not Disturb treatment using this mechanism will not show as missed calls in the call log of the Spectralink Versity device. Administrators that would like their users to utilize the FAC method for Do Not Disturb may wish to implement the Do Not Disturb Feature Access Code as a Contact and program it as a Speed Dial for their users.

Call Park

Calls may be parked and retrieved using Feature Access Codes (FACs), however, problems were found to exist using Paging in conjunction with the Call Park feature, and as such Spectralink cannot recommend or support Call Park with any Paging functionality configured in conjunction with the Page. Default Paging type configured for Call Park must be set to No Paging in order for Call Park to function. Please consult the FACs section of this manual for ideas about how to make Feature Access Code-based features easier for end users to utilize.

DTMF via RFC2833

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

Call Forward

The Spectralink Versity handset was tested using what Mitel refers to as Call Forward Follow Me functionality. Many vendors refer to this functionality as Call Forward All Calls. The following two methods for implementing this feature were both tested in Spectralink's labs:

Call Forward All Calls Using the Handset

This method of implementing Call Forward was tested by navigating to the Spectralink Biz Phone and tapping the Overflow button (three dots), then selecting the **Call Forwarding** 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 in order 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 if they implement call forwarding through the phone's UI.

Call Forward All Calls Using Feature Access Codes (FACs)

This method of implementing Call Forward was tested by dialing the Call Forward Follow Me FAC and Cancel All Forwarding FAC's programmed in the Mitel MiVoice System. 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 Mitel MiVoice System itself and the MiVoice Business System will simply never offer calls to the phone until the call forward is cancelled using the Call Forward Cancel FAC. As such, calls forwarded using this mechanism will not show as missed calls in the call log of the Spectralink Versity device. Administrators that would like their users to utilize the FAC method for call forwarding may wish to implement the Call Forward Feature Access Code as a Contact and program it as a Speed Dial for their users.

Feature Access Codes

Feature Access Codes (FACs) were utilized in Spectralink's labs for the testing of several features (Call Forward and Call Park to name a few.) Since users may struggle to remember the digits for a Feature Access Code, Spectralink recommends that Feature Access Codes be added as Contacts with the Feature Name saved as the contact name. These "contacts" may subsequently be added as speed dials to the phone in order to simplify the use of these features.

SIP Using TCP or TLS

TCP (and by extension, TLS) are not recommended Transport protocol selections on the Versity handset when registering to the Mitel MiVoice system. In the lab environment, Spectralink observed call maintenance trouble when one of the handsets goes out of range of a Wireless Access Point (WAP). When TCP was selected as a transport protocol, the Mitel MiVoice system would not allow the call to resume when a Versity handset would go out of range for a short duration. This appears to be related to how the Mitel call server interprets the subsequent SIP messaging and port information when the Versity handset attempts to re-initiate a connection. As a result, the recommended transport protocol is UDP.

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

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

Default Versity Advertised Codec List

The Spectralink Versity phones' will advertise G.711u first, and G.711a second by default.

G.711u, G.711a, and G.729A codec priorities: ***G.711u =1 and G.711a = 2. G.729A = 0***. These values are the defaults. 1 is the highest priority and enabled, 2 is second priority and enabled. A value of 0 disables the codec. You may modify the order or enable / disable codecs as required for your installation.

G.722 Codec: ***G.722 =0***. The Mitel MiVoice System supports the G.722.1 32kbps variant of the G.722 codec and the Spectralink Versity phone supports the traditional G.722 codec, so codec support for G.722 is not aligned between the handset and the Mitel MiVoice System. For this reason, Spectralink does not recommend the G.722 codec be enabled on the Versity Handsets.

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)

The Spectralink Versity phone does not support a Shared Line or Bridged Line appearance. If you wish to be able to dial one number and ring multiple phones, that functionality is available through the Mitel Ring Group feature. However, Ring Groups do not allow member phones to select the Ring Group number as a line available for outbound calls.

Ring Multiple Phones Simultaneously / Ring Group

The ring group is basically a unique number that you can program to ring multiple devices simultaneously. Again, there is no way to select the ring group number for use as an outbound call, but inbound calls to a ring group number can be programmed to ring all member devices simultaneously. They should also supply the called party with the information that it is the Ring Group number that is being called rather than the Ring Group member’s primary line. There is no special configuration on the Spectralink Versity phone to support this feature and the Spectralink phone’s extension number should be added to the Ring Group just as you would program any other existing Mitel extension.

Programming of Ring Groups can be performed through the **System Administration> Users and Devices> Group Programming> Ring Groups** page.

Trunk Calling

In and outbound trunk calling were tested utilizing a SIP trunk connected between the Mitel MiVoice system and an Audiocodes Mediant Gateway configured with an ISDN PRI connecting to the PSTN. 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. There is no special configuration on the Versity handset required to allow this functionality.

Failover, Fail-back, Redundancy or Resiliency

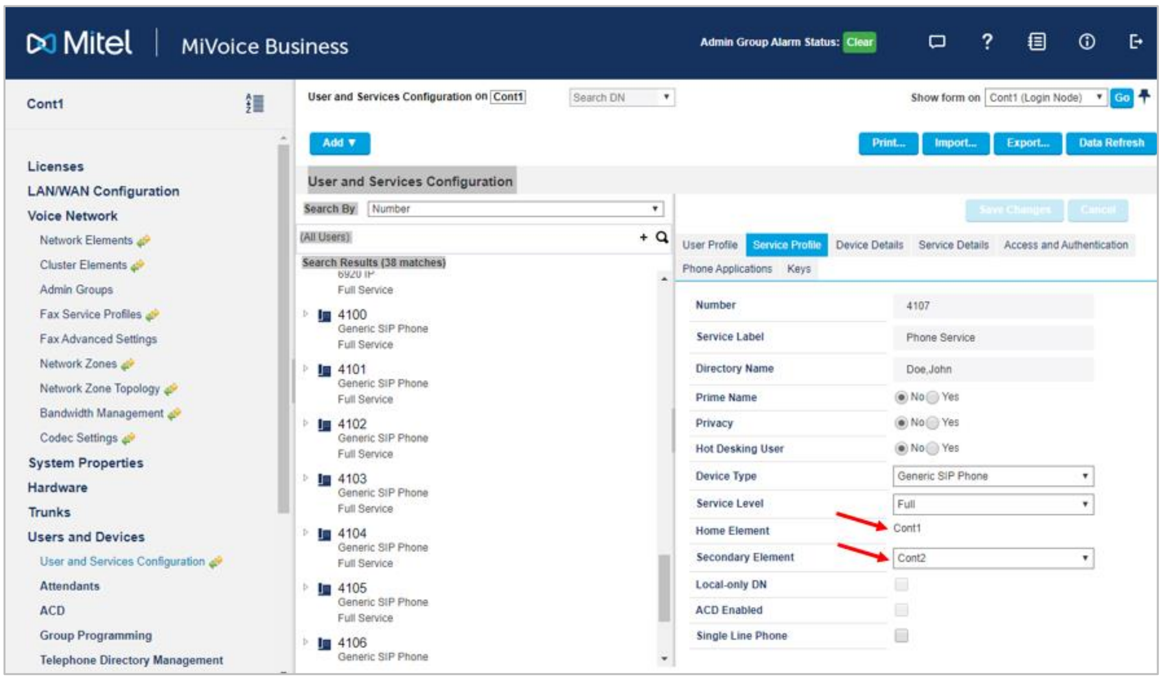
Spectralink Versity smartphones do not utilize the same configuration mechanisms and detection mechanisms that Mitel branded phones use to achieve resiliency, but do support a redundancy mechanism that will allow the phones to failover to and fail-back from a secondary server.

For a detailed discussion of Redundancy and Failover behavior and configuration methods for the Spectralink Versity phones, please consult the Spectralink Versity Call Server Redundancy CS-18-10 document available on the Spectralink support site.

Please note, the Spectralink Biz Phone application does not currently recognize some of the Mitel-specific header information in regard to resiliency, and as such, failover cannot be achieved without the primary controller being unreachable (i.e. not responding to INVITE or REGISTER requests).

Call Server Setup

In the Spectralink lab, we configured a resilient cluster among primary and secondary Mitel 3300 Mxe-III controllers, with data sharing enabled. Please refer to the Mitel documentation for a detailed explanation of how these can be configured within your network. Additionally, within **Users and Devices-> User and Services Configuration**, we configured each extension's Service Profile with a Home element configured as the primary controller (Cont1), and a Secondary Element programmed to the secondary controller (Cont2):



Method 1: Using IP addresses (without the use of DNS)

Using this method, we set the domain name and the IP addresses of the two servers, putting the IP addresses in the preferred order:

SIP server: **mitelslnk.local; 172.29.102.175; 172.29.102.176.**

During testing, once the Versity handset detected the primary controller (172.29.102.175) was unreachable (using INVITE or REGISTER), Versity would send the registration request to the secondary controller (172.29.102.176). Once registration was complete, all subsequent SIP messages were sent to the secondary controller, until a fail-back condition was achieved.

Method 2: Leveraging DNS

Using this method, we simply set the domain name in the SIP server field:

SIP server: **mitelslnk.local**

DNS Server Configuration: **mitelslnk.local** should be defined as a DNS SRV record that lists the desired Mitel registrars in prioritized order.

Versity will then perform a DNS SRV query for this name and honor the prioritized list it receives as a result from the DNS server. Note that all INVITE and REGISTER requests will be directed to the ip address of the server of highest priority that Versity detects as available, and will be in the form of xxxx@domainname (xxxx@mitelslnk.local in this example).

Call Resiliency

Call Resiliency refers to the ability of an active call to survive when the primary controller becomes unreachable, as well as the ability to properly tear down the active call. Spectralink was able to verify call survivability in failover testing (call remained active when primary controller became unreachable), but the call teardown would often cause an error to display in the Versity handset, stating the call server is not responding, due to the fact that the BYE messages would receive a 200OK response from the original call server. In this scenario, the Versity handset will provide an option to end the current call.

Fail-back

Fail-back refers to the process of all call flow operations to restore to the primary controller after failover occurs. The Versity handset will continue to send OPTIONS requests to the primary controller after failover takes place. If the primary controller does not respond within roughly 32 seconds, Versity will continue to send OPTIONS requests roughly every 268 seconds until the primary controller responds (this matches the registration interval for Versity). Once the primary controller responds, the Versity handset will re-register back to the primary controller.

In the Spectralink lab, we observed that the secondary Mitel controller would automatically detect when the primary was back online. Once the primary controller was operational, any requests sent from the Versity handset to the secondary controller would be met with a '301 Moved Permanently' response, containing a Mitel-proprietary header directing registration back to the primary controller. Since Versity does not currently accommodate this header information, the registration attempt to the secondary controller would fail, and the handset would remain unregistered until the next OPTIONS request was answered by the primary controller. This allows a worst-case scenario where a device could remain unregistered for a period of time lasting up to around 5 minutes, depending on when the secondary controller detects the primary is operational. During our lab testing, the unregistered handset was not able to place outbound calls, but was still able to receive calls within this time period.

Chapter 4: Troubleshooting

SIP Traces on the MiVoice Business System

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 or bug report from the Versity phone should provide a detailed look at SIP messaging to and from the phone itself (though not in a Wireshark form, this is still useful information.) However, we 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 Mitel Server to help determine whether the server received messages sent by the phone, and how it responded. The Mitel server does have the ability to gather SIP traces, though we should caution that this should probably be performed during low traffic times both for ease of reading the resultant trace, and to prevent excessive load on the Mitel Server's CPU.

To create a SIP Trace on the Mitel Server

1. Log Into **System Administration> Maintenance and Diagnostics> Maintenance Commands**.
2. Enter the Command: `SIP TRACE ON`, to start the SIP trace.
3. Now, perform the experiment of interest.
4. Enter the Command: `SIP TRACE OFF`, to stop the SIP trace.

To gather the SIP Trace from the Mitel Server

1. FTP to the controller using the FTP tool of your choice (command line, ftp client or browser).
2. Login with the administrator's login and password.
3. Change to the `/db` directory.
4. Get the `SipTrace.rtf` file. This will contain a trace of all of the SIP packets to or from the Mitel Server during the course of your experiment.

DSCP Values

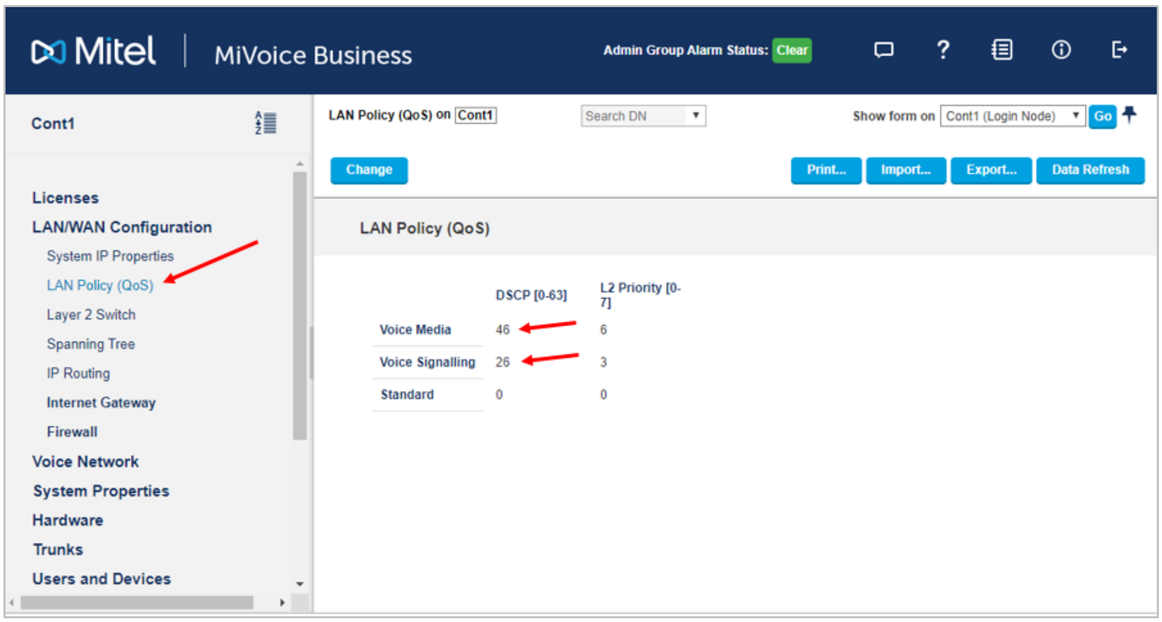
The default DSCP values for Call Control and Audio are typically aligned with Spectralink recommendations. That said, if wireless analysis determines that packets are not "getting through" to the handset, it may be worth verifying that the MiVoice Business System is tagging Audio and SIP Control packets with appropriate DSCP values.

Audio and Call Control DSCP

The system default Call Control value for SIP Signaling packets sent by the Mitel MiVoice System is 26. While the Versity phone typically sends SIP Signaling packets with a default DSCP value of 40. Though these differ, a value of 26 is not necessarily an issue and as such, Spectralink believes that the default values are acceptable. The Mitel MiVoice System also defaults to an audio DSCP value of 46 which matches the Spectralink recommendation.

To modify or check the Call Control and Audio DSCP Values

Log into the Mitel MiVoice System, then navigate to: **System Administration> LAN/WAN Configuration> LAN Policy (QOS)**.



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