Mitel Technical Configuration Notes – HO3633

February 5, 2020

Configure the MiVoice Connect for use with Spectralink 8440 series phones

Description: This document provides a reference to Mitel Authorized Solutions Providers for configuring the MiVoice Connect to host the Spectralink 8440 phones.

Environment: MiVoice Connect R19.1 22.10.7600.0/ Spectralink 8440 6.2.0.2221

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Overview

This document provides a reference to Mitel Authorized Solutions Providers for configuring the MiVoice Conect to host the Spectralink 8440 phones. the different devices can be configured in various configurations depending on your VoIP solution. This document covers a basic Spectralink 8440 phones setup as Endpoint gateway with required options setup and explicitly excludes all Spectralink 8440 phones setup as Trunk gateway testing.

Interop History

Version	Date	Reason
1	January, 2020	Initial Interop with MiVoice Connect R19.1 (22.10.7600.0) and Spectralink 8440 6.2.0.2221

Interop Status

The Interop of the Spectralink 8440 has been given a Certification status. This device will be included in the Mitel Interop Reference Guide (IRG). The status of Spectralink 8440 achieved is:

COMPATIBLE	The most common certification which means the device/service has been tested and/or validated by the Mitel SIP CoE team. Product support will provide all necessary support related to the interop, but issues unique or specific to the 3rd party will be referred to the 3rd party as appropriate.

Software & Hardware Setup

The test setup generated basic SIP calls between the Spectralink 8440 phone and the MiVoice Connect.

Manufacturer	Variant	Software Version	Additional Applicable Variants
Mitel	MiVoice Connect	Release 19.1 (22.10.7600.0)	NA
Mitel	IP480	804.1905.1300.0	NA
Mitel	230 IP Phones	SEV.3.9.13	NA
Mitel	69xx Phones	5.2.1.133	NA
Spectralink	Spectralink 8440	6.2.0.2221	8441,8450,8452,8453,8180(G2),8128,8128(G2), 8028, 8028(G2) and 8138

Tested Features

Listed below is an overview of the features tested during the Interop test cycle and not a detailed view of the test cases. Please see the SIP Line Side Interoperability Test Plans for detailed test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call	☑
Registration/Authentication	Device registration w/o authentication	☑
Call Hold	Putting a call on hold	☑
Call Transfer	Transferring a call to another destination	☑
Call Forward	Forwarding a call to another destination	☑
Conference	Conferencing multiple calls together	☑
Redial	Last Number Redial	☑
Call Park	Parking a call on the system for retrieval	☑
Caller ID	Making and receiving basic calls between MiVoice Connect and Spectralink 8440 phone	☑
Codec	All test cases were performed using G711ulaw, G729 and G722.	☑
Voicemail	Terminating calls to voicemail boxes and DTMF detection.	☑

☑ *- No issues found* ✗ *- Issues found, cannot recommend to use* ⚠ *- Issues found*

Device Limitations

This is a list of problems or not supported features when the Spectralink 8440 phone is connected to MiVoice Connect.

Feature	Problem Description
Conference	No Audio between Mitel phone and third-party SIP phone when Mitel 69XX phone originates the conference. Please refer MIVC-659 JIRA ticket for more details. We have seen this issue in 5.2.1.133 build but same was not reproduced in 5.2.1.1065 build. **Recommendation:** Please contact Mitel Product Support referencing JIRA ticket MIVC-659 for any updates to this issue.
Hunt group	If the caller hung up while hunt group was ringing and when the call was answered, there would be dead air. To avoid this issue, need to add ***'XferFailureNotSupported=1'*** parameter in SIP Profile. **Recommendation:** Please refer to SIP Peer profile programming section here.
PRACK	PRACK cannot be controlled on MiVoice Connect system. **Recommendation:** No recommendations available but should you require this feature, please contact your Mitel account team to open a design change request.
SIP INFO	SIP INFO is not supported by MiVoice Connect. **Recommendation:** No recommendations available but should you require this feature, please contact your Mitel account team to open a design change request.
4 – Party Conference	4-Party conference is not supported by Spectralink 8440 phones **Recommendation**: Please contact Spectralink to add this feature.

Network Topology

This diagram shows how the testing network is configured for reference.

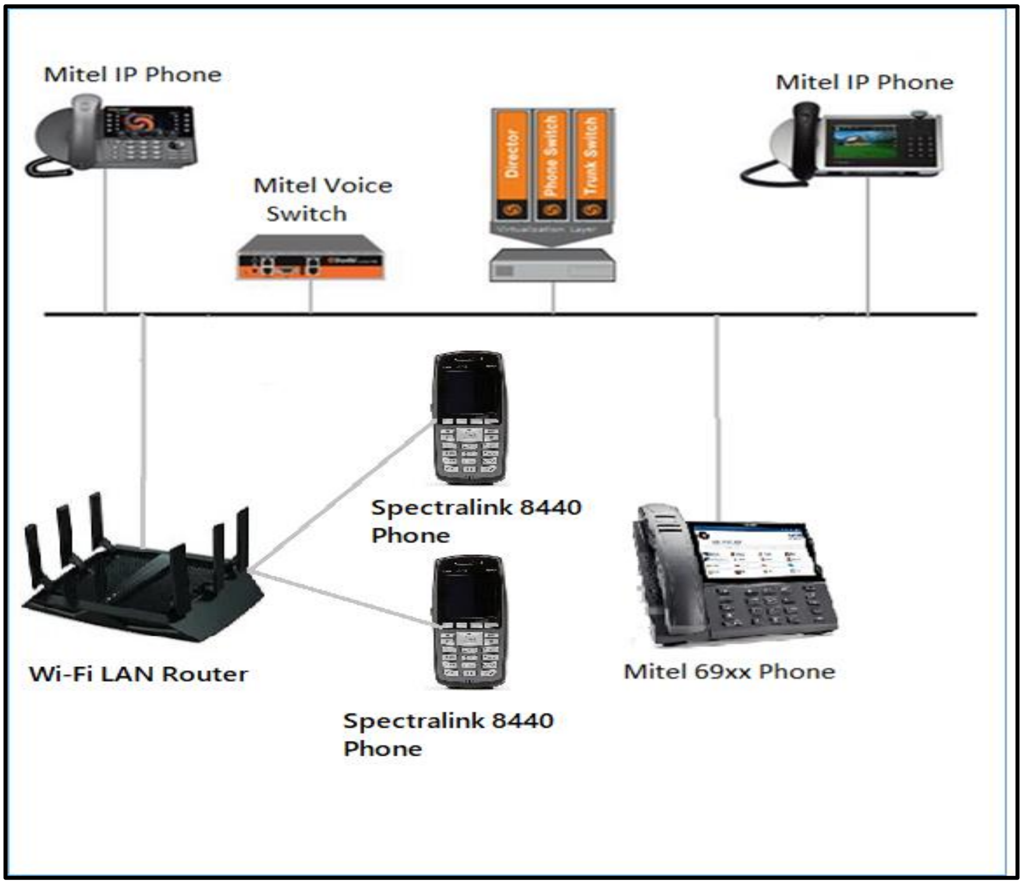


Figure 1 – Network Topology

The Spectralink 8440 is configured as endpoint gateway where a persistent connection is created for each SIP user. Each device connected to has a separate SIP connection to the SIP server.

Test Environment

- Mitel Connect ONSITE Server
- Mitel Voice Switch
- Mitel ShoreGear Switch
- Mitel 230 IP Phones
- Mitel 69xx Phones
- Spectralink 8440 phones
- Mitel Virtual Phone Switch
- Mitel Virtual Trunk Switch
- Mitel Collaboration Service Appliance
- Mitel Connect Client

Configuration Notes

This section is a description of how the SIP Interop was configured. These notes should give a guideline as to how a device can be configured in a customer environment and how the Spectralink 8440 phone was configured in our test environment.

We recommend that the Spectralink 8440 phone is configured in Device Based mode. You will configure the Device Based mode in the SIP Device Capabilities Form as described in this section.

Disclaimer: Although Mitel has attempted to setup the interop testing facility as closely as possible to a customer premise environment, implementation setup could be different onsite. YOU MUST EXERCISE YOUR OWN DUE DILIGENCE IN REVIEWING, planning, implementing, and testing a customer configuration.

MiVoice Connect Configuration Notes

The following steps show how to program a MiVoice Connect to connect with the Spectralink 8440 phone.

Call Control Options

This section describes the SIP settings required on the Mitel system to work with Spectralink 8440 phone This is accomplished from Mitel Connect Director.

1. Navigate to Administration > Features > Call Control > Options

2. Verify the parameters located under the **SIP** section

3. **Realm:** The realm is used in authenticating all SIP devices. Changing this value will require a reboot of switches serving as SIP extensions. It is not necessary to modify this parameter

4. **Enable SIP Session Timer:** Ensure this parameter is checked

5**. Session interval**: Session interval value indicates the SIP session registration period. There is no need to modify the default value of 1800 seconds.

6. **Refresher:** The refresher setting decides if user agent client or user agent server refreshes the session. There is no need to modify the default value of “Caller (UAC).”

7. Click **SAVE**

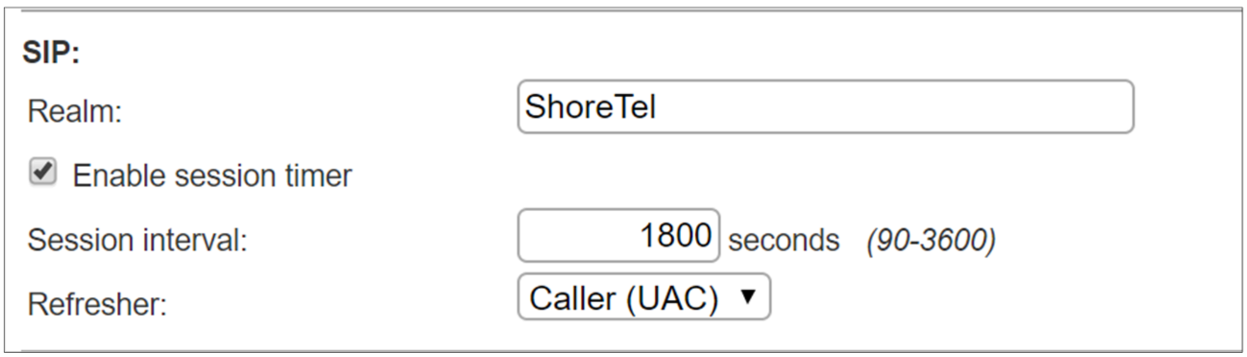


Figure 2 – Call Control Options

SIP Proxy Settings – Allocating Ports for SIP Extensions

This section describes the Switch configuration required on the Mitel system to work with the Spectralink 8440 phone Depending on the switch type, Mitel Voice Switches, and Virtual Phone Switches support variable numbers of SIP Proxies and IP Phones, and can be verified on the Switch Edit page of Mitel Connect Director.

Mitel ShoreGear Switches with processing resources that support Digital and Analog ports can be reallocated to support 100 SIP Proxies. The Mitel Administrator can define one of the “Port Type” settings from the available ports to “100 SIP Proxy”, as well as sufficient “IP Phone” ports to support the total number of Spectralink 8440 phone. The following example shows Port allocation designated on a Mitel SG-90 for IP Phones and SIP Proxy resources

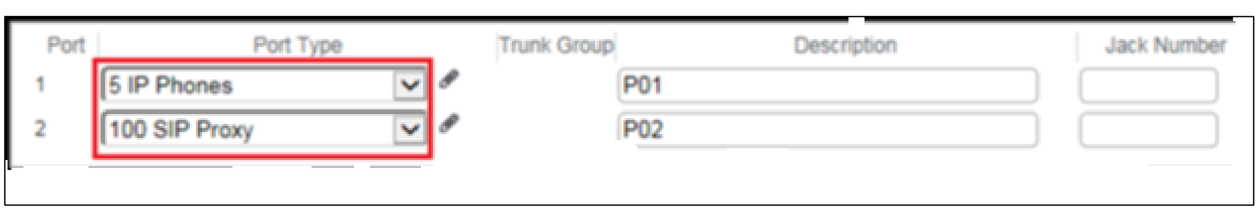
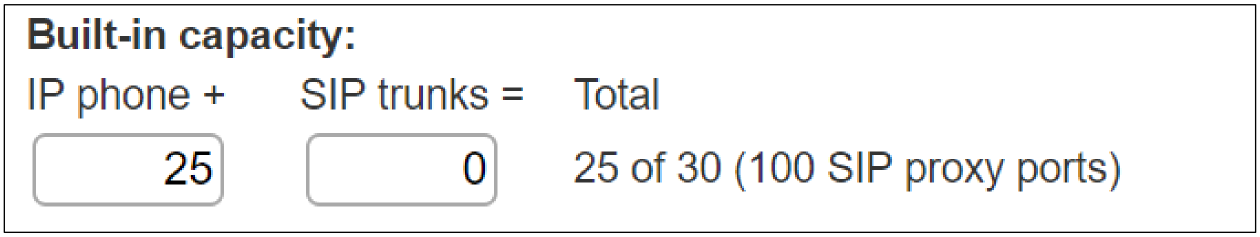


Figure 3 – Multiline IP Set Configuration

If the Mitel ShoreGear Switch that you have selected has “built-in” capacity (i.e., ShoreGear 50/90/220T1/E1, etc.) for IP phones and SIP trunks, you can also remove 5 ports from the total number available to provide the "100 SIP Proxy" configuration necessary. Every 5 ports you remove from the total available will result in "100 SIP Proxy" ports being made available. The following example shows 5 ports removed from total available resulting in 100 SIP Proxy ports being available.



Site Settings

The next settings to address are the administration of Sites. The Mitel Administrator can designate up to two Proxy switches per site for redundancy and reliability: one switch is assigned as the primary Proxy server, and the other switch acts as the backup Proxy server in case the primary fails. A Virtual IP Address is the IP Address of the switch that is configured as the SIP Proxy server for the Site. The Virtual IP Address must be static. If you choose not to define a "Virtual IP Address," you can only define one proxy switch, and there will be no redundancy or failover capabilities. The switches available in the "Proxy Switch 1 / 2" will only be shown if proxy resources have been enabled on the switch. This is accomplished from Mitel Connect Director.

1. Navigate to Administration > System > Sites

2. Select the name of the Site in which SIP Proxies will be assigned

3. In the General Tab, set Proxy switch 1: Select the Mitel switch configured with SIP Proxies for the Site

4. Click SAVE

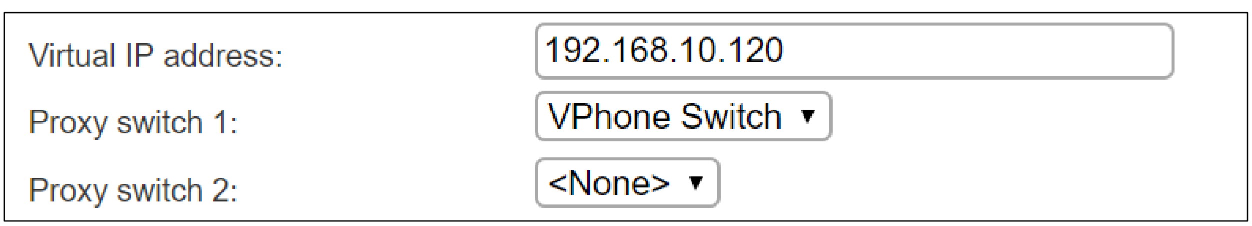


Figure 4 – Site Settings

Configure a SIP Profile

This section describes the steps required to configure the “SIP Profiles” for the Spectralink 8440 phone. By default, the Spectralink 8440 phone utilize the “System” profile. In order to optimize the functionality, you will need to add a custom profile. This is accomplished from Mitel Connect Director.

1. Navigate to Administration > Telephones > SIP Profiles

2. Click New, to create a new SIP Profile

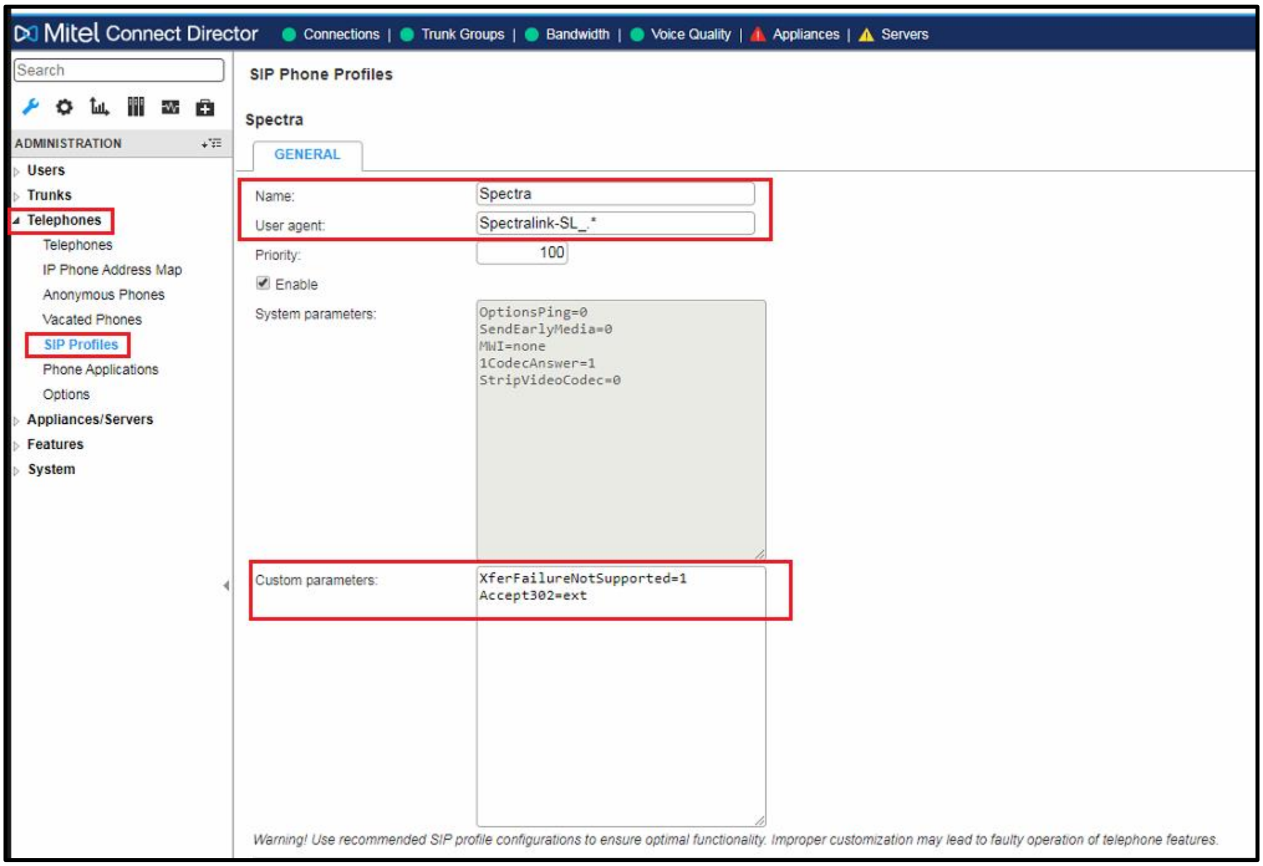


Figure 5 – Configuring a SIP Profile

3. In the General Tab, define a Name: we recommend a name that describes the SIP endpoint.

4. For the parameter User agent: enter “Spectralink-SL_. *” (without quotes)

5. The parameter “Priority:” defaults to 100, no change is required.

6. Enable the profile by checking (enabling) the Enable option.

7. Please add the following parameter’s manualy under Custom Parameters Sections :

XferFailureNotSupported=1
Accept302=ext

8. Click SAVE

Configure Spectralink 8440 phone as a SIP Extension

1. Navigate to Administration > Users > Users

2. Click New, to create a new user

3. Define the First name: and Last name: Enter the appropriate user information

4. Define an Extension: Mitel Connect Director will automatically assign the next available extension number, but it can also be modified to any available extension number

5. Define the License type: and Access license: In our example, we chose “Extension and Mailbox”, and “Connect Client” for Access license

6. Define a SIP phone password: There is no default SIP phone password configured, it is masked with the appearance that there is a default password and must be defined by the Mitel Director Administrator. Make certain to type the password in both fields.

7. Click SAVE

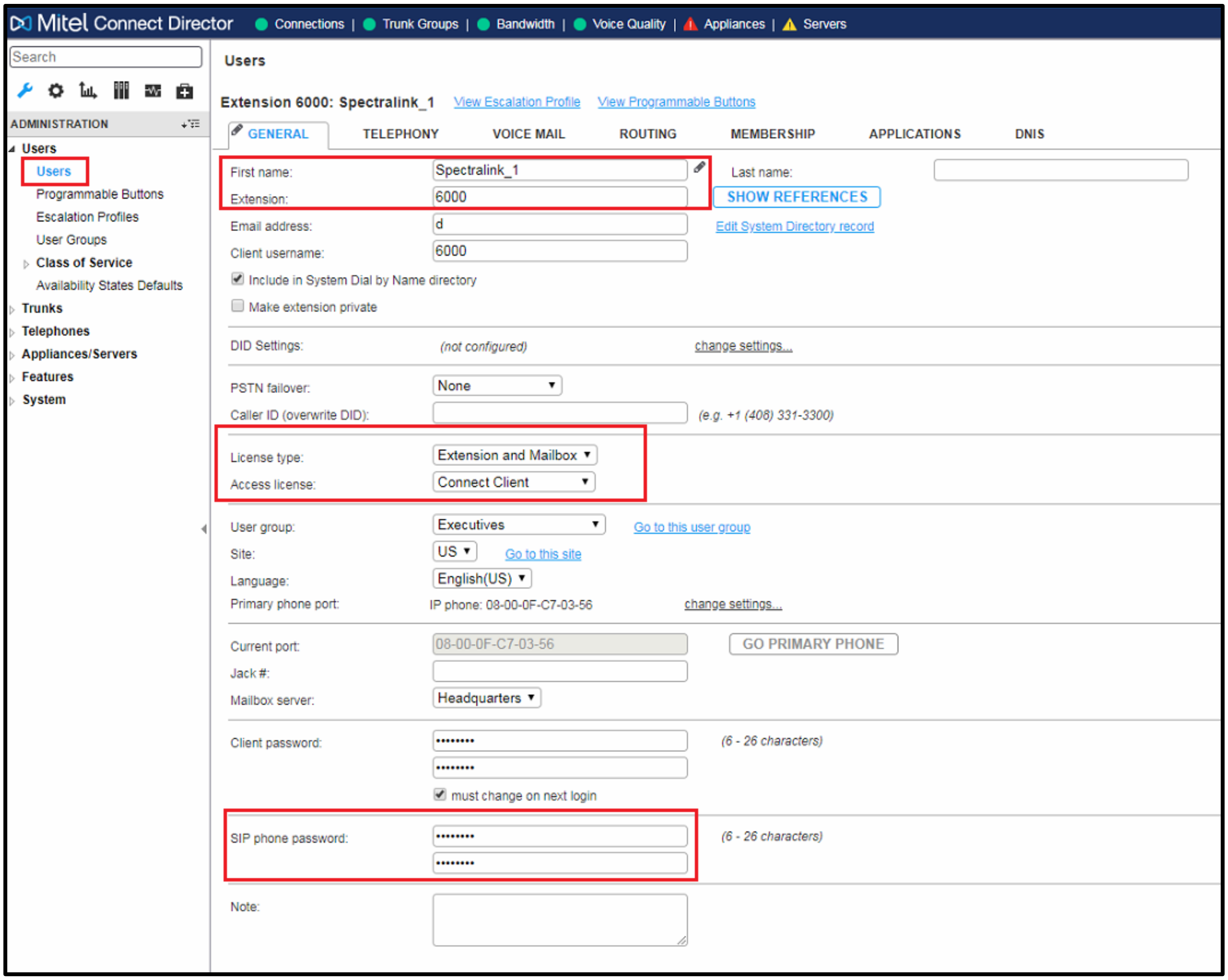


Figure 6 – Create a User

Spectralink 8440 phone Configuration Notes

This section outlines the basic instruction on how to program Spectralink 8440 phone to interconnect with MiVoice Connect. This is by no means a comprehensive guideline. We assume that Spectralink 8440 phone has been upgraded to the latest software release as found in https://support.spectralink.com/products/wi-fi/spectralink-84-series-wireless-telephone. Please note that your phone must have been upgraded to current software release.

Note: *In our test lab we utilized the web configuration utility to configure the 8440 test handsets as outlined below. However, for enterprise deployments Spectralink strongly recommends a QNC (Quick Network Connect) device be used for initial Wi-Fi configuration and a provisioning server be used to supply device configuration and software updates in order to support deployments in a more scalable manner.*

Home Page Login

Using your favorite browser, enter the phone's IP address as the browser address. You will be asked for login credentials. The default username and passwors is **admin / 456**.

Figure 7 – Spectralink 8440 phone home page login

Upgrade Software

For upgrading latest software, you can follow below procedure:

Go to Utilities→Software Upgrade, then Click on Check for Updates as shown below.

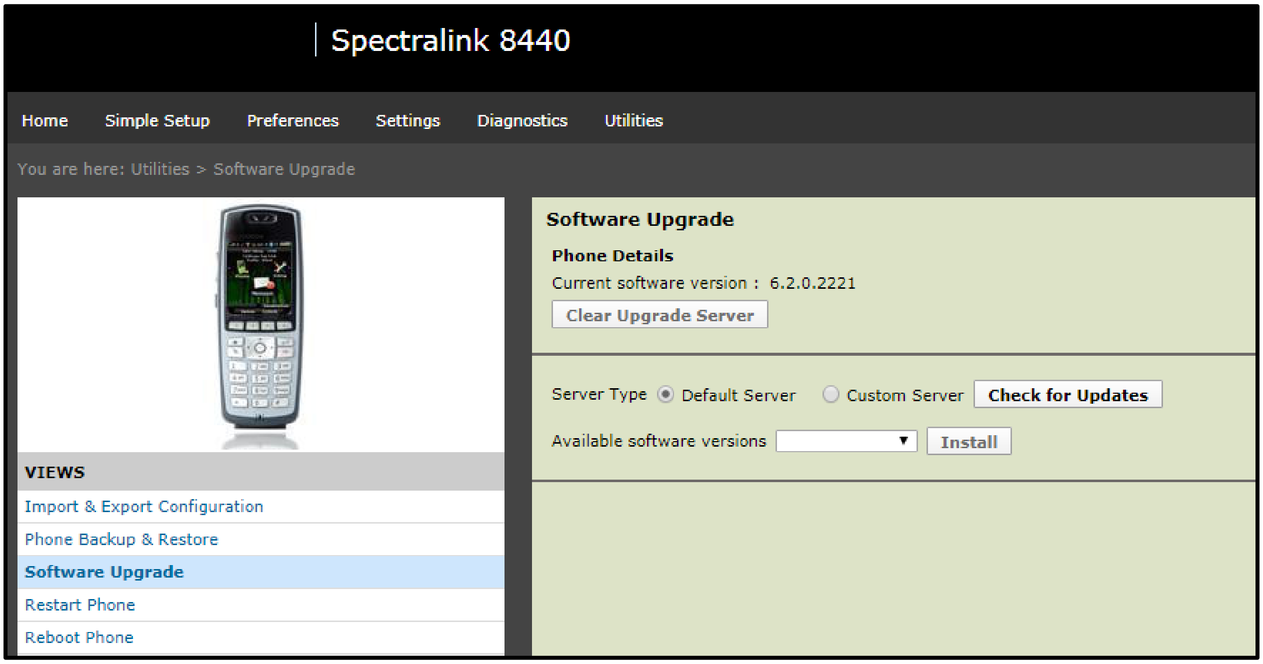


Figure 8 – Spectralink 8440 phone software upgrade

After that you need to install the Latest Software by clicking Install button as shown below:

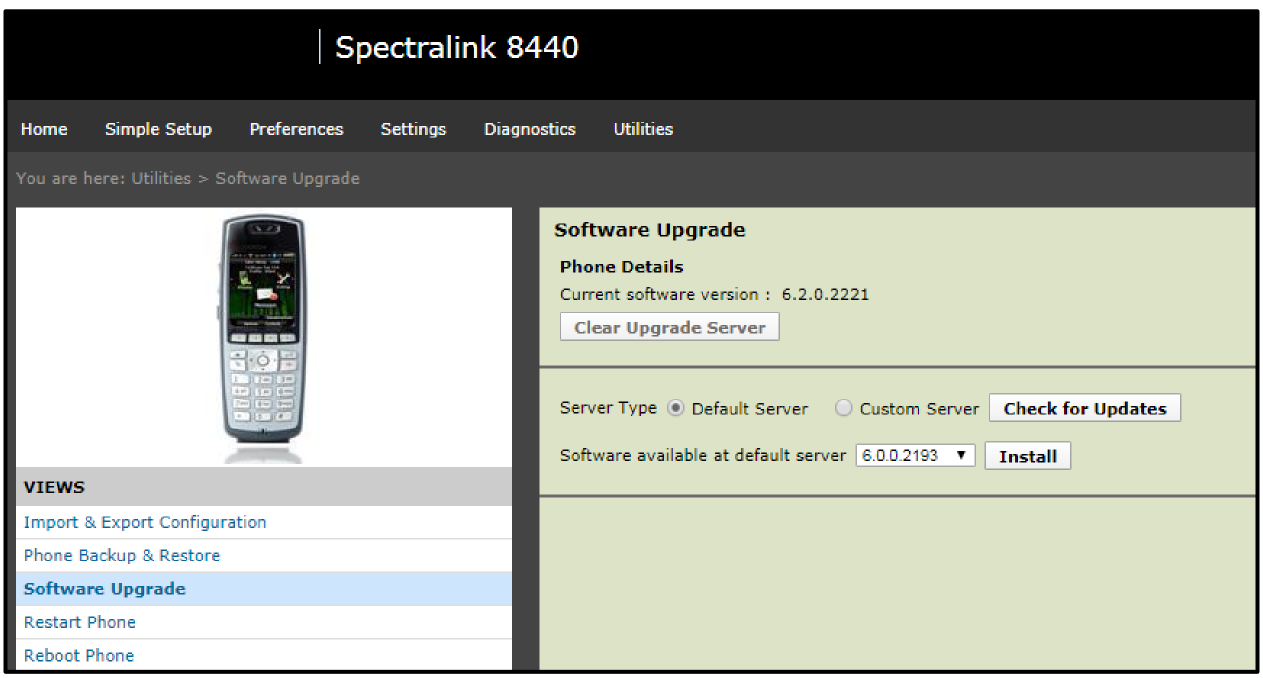


Figure 9 – Spectralink 8440 phone software upgrade

Line Based Configuration

UC Software Application lines could be programmed individually.

Line based Proxy Server Provisioning

Select Settings > Lines > Line 1 > Server 1 to program proxy server information including registration expiry time, at line level.

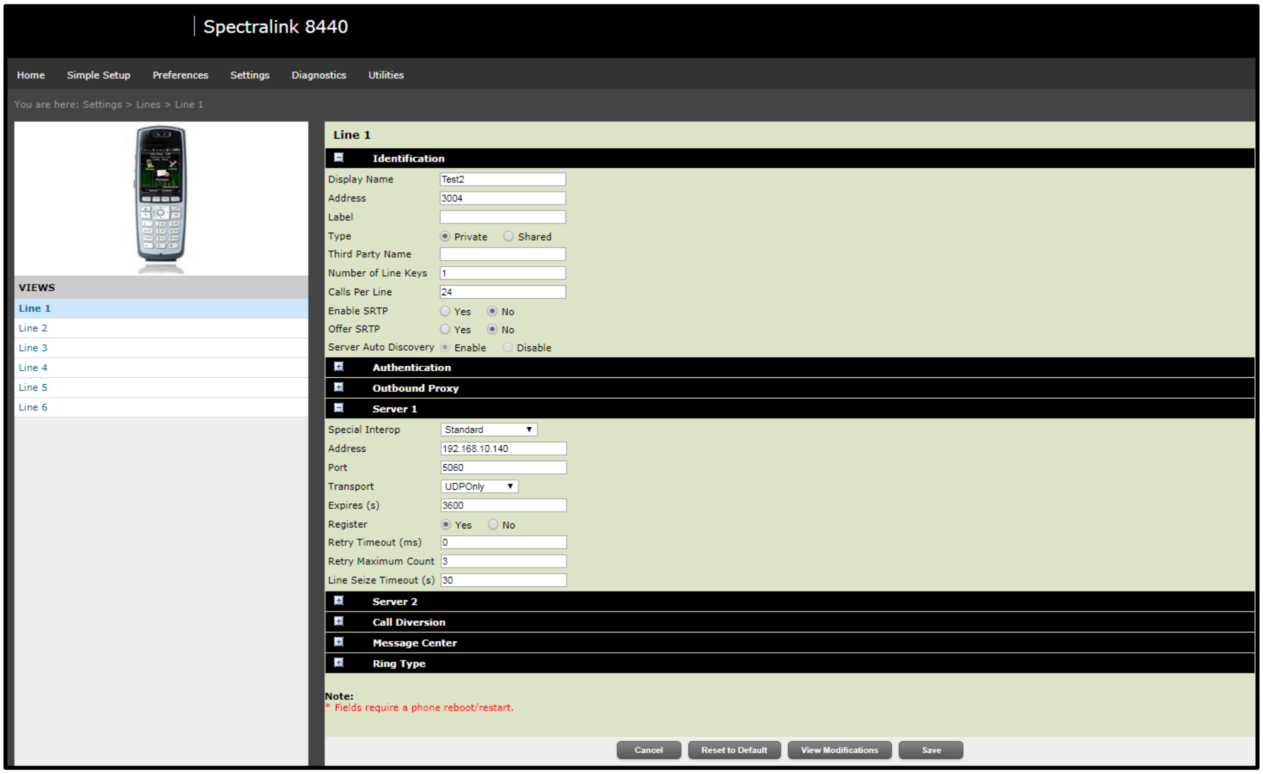


Figure 10 – Spectralink 8440 phone Line Setting

Call Diversion Setup

Select Settings > Lines > Line 1 > Call Diversion, to setup Device based Call Forwarding Always, No Answer or Busy.

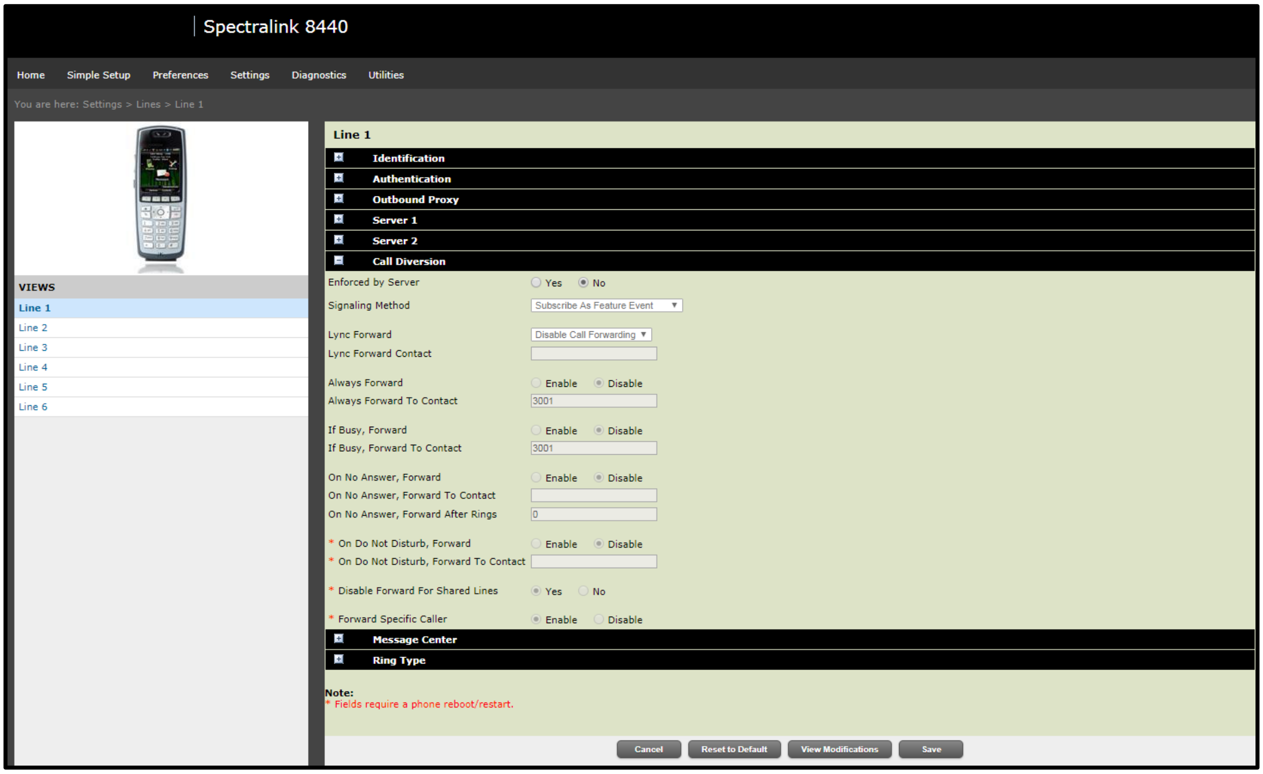


Figure 11 – Line based Call Diversion Setup

Codec Setting

Go to Settings→Audio Codec Priority, Then Set codec priority as per requirement as shown below in figure 12

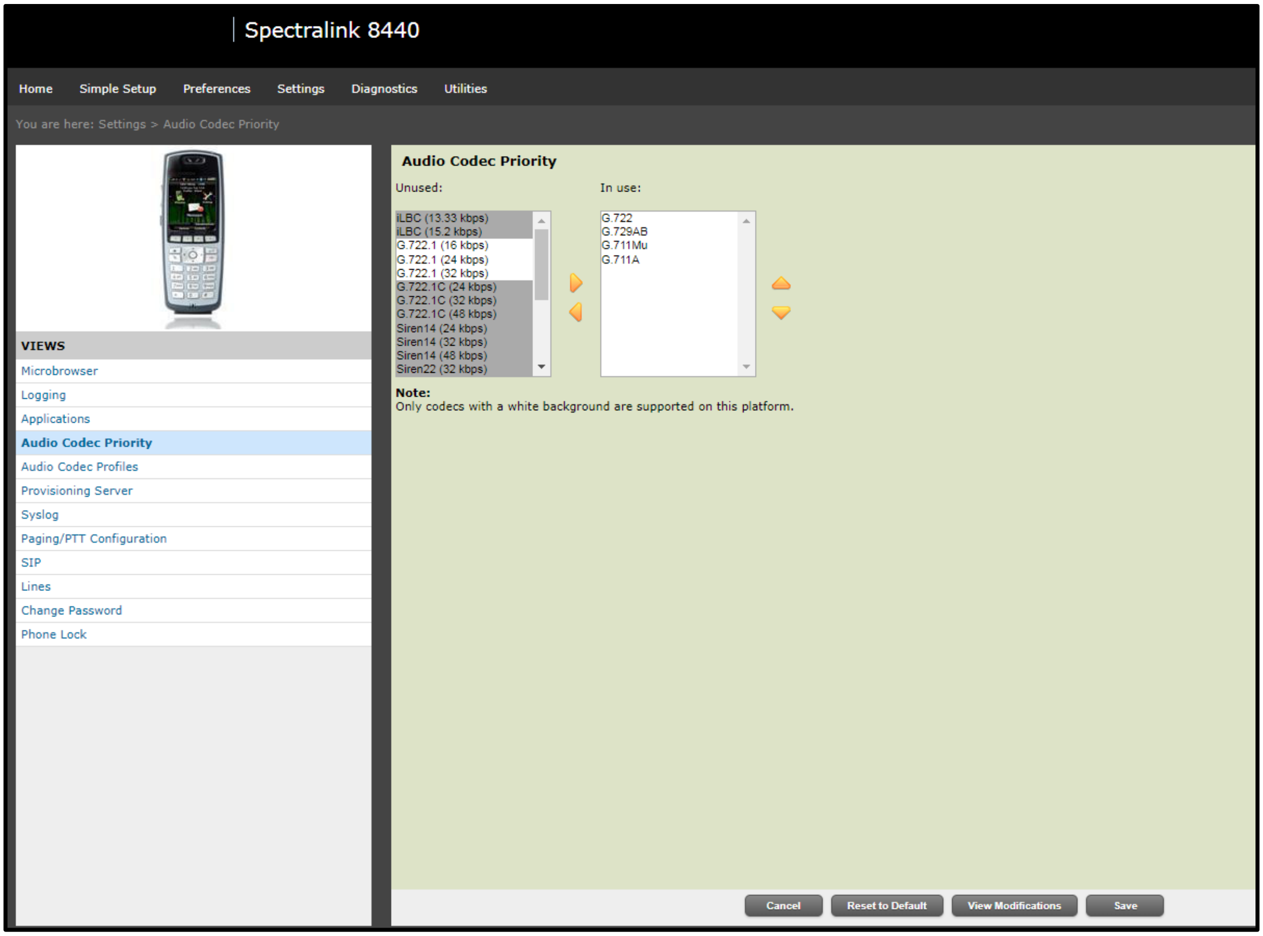


Figure 12 – Audio Codec Priority

Summary of Tests and Results

Primary Switch (ST Virtual Phone Switch)

N/S = Not Supported N/T= Not Tested N/A= Not Applicable C. PASS= Conditional Pass

ID	Result	Name	Description	Notes
1.1	PASS	Device initialization with static IP address	Verify successful startup and initialization of the device up to a READY/IDLE state using a static IP address	
1.2	PASS	Device reset – idle (for static configurations)	Verify successful re-initialization of device after power loss while device is idle	
1.3	N/T	Device initialization with DHCP	Verify successful startup and initialization of the device up to a READY/IDLE state using DHCP	
1.4	N/T	Device reset – idle (for dynamic configurations)	Verify successful re-initialization of device after power loss while device is idle	
1.5	N/A	Verify Diffserv Code Point support	Verify the ability to set Diffserv Code Point from SIP DUT (device under test)	
1.6	C.PASS	Verify Date and Time Update support	Verify setting of Date and Time Update on SIP DUT	Spectralink 8440 phone fetches the time and date from NTP server. But not from Director
1.7	PASS	Place call	Verify successful call placement with normal dialing to a variety of terminating phones	
1.8	PASS	Receive call	Verify successful call placement with normal dialing to a variety of terminating phones	
1.9	PASS	Place call - redial	Verify successful call placement using re-dial to SIP Reference	Internal Redial List on Spectralink Handset
1.10	Pass	Place call – speed dial	Verify successful call placement using programmed speed dial	
1.11	PASS	CODEC support (DUT to Mitel Phone)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	
1.12	PASS	CODEC support (DUT to SIP reference)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	

ID	Result	Name	Description	Notes
1.13	PASS	CODEC negotiation	Verify successful negotiation between devices configured with different default CODECs (G.711-Ulaw and G.729)	
1.14	PASS	Hold DUT to SIP reference	Verify successful hold and resume of connected call	
1.15	PASS	Hold DUT to Mitel	Verify successful hold and resume of connected call	
1.16	PASS	Forward	Verify successful forwarding of incoming calls	
1.17	PASS	Forward from SIP DUT	Verify successful forwarding of incoming calls	Accept302=ext on SIP Profiles
1.18	PASS	Mute	Verify device's mute function	
1.19	PASS	Out-of-band DTMF Transmission	Verify successful transmission of out-of- band digits (RFC2833) for calls placed to and from the DUT with a variety of other devices	SIP INFO is not Supported by MiVoice Connect
1.20	PASS	Missed call notification	Verify that device notifies the user about missed calls	
1.21	PASS	Volume	Verify the device's volume adjustment function	During the call. Volume Adjustment Function has been tested
2.1	PASS	Call waiting	Verify appropriate notification and successful connection of incoming call while busy with another party	
2.2	PASS	Park	Verify successful park and retrieval of connected call	*11+ext to park the call There was no indication on the DECT handset that there is a parked call present. User must dial *12+their own extension to answer the call. but the user must retrieve the call via the handset. This is expected behavior.
2.3	PASS	Extended forward	Verify extended call forwarding options – busy forwarding, ring no answer forwarding	

ID	Result	Name	Description	Notes
2.4	PASS	Extended forward from SIP DUT	Verify extended call forwarding options – busy forwarding, ring no answer forwarding	Accept302=ext on SIP Profiles
2.5	PASS	Transfer – blind	Verify successful blind transfer of connected call	
2.6	PASS	Transfer – monitored	Verify successful monitored transfer of connected call	
2.7	PASS	Conference – ad hoc	Verify successful ad hoc conference of three parties	4 way or N-way conference is not supported by Spectralink 8440 phone.
2.8	PASS	Place call – secondary line	Verify successful call placement using secondary line	
2.9	PASS	Receive call – secondary line	Verify successful connection of incoming call on secondary line	
2.10	PASS	Callback	Verify successful connection of a call using the missed- call callback feature of the device	
2.11	PASS	Caller ID	Verify that Caller ID name and number is sent and received from SIP endpoint device	
2.12	PASS	SIP Device Generates Busy Tone	Verify that SIP DUT generates busy tone when calling a busy extension	
2.13	PASS	Initiate call to a Hunt Group	Initiate a call from DUT and verify that calls route to the proper Hunt Group and are answered by an available hunt group member with audio in both directions using G.729 and G.711 codecs.	If the caller hung up while hunt group was ringing and when the call was answered, there would be dead air. To avoid this issue, need to add 'XferFailureNotSupported=1' parameter in SIP Profile.

ID	Result	Name	Description	Notes
2.14	PASS	Initiate call to a Workgroup	Initiate a call from DUT and verify that calls route to the proper Workgroup and are answered successfully by an available workgroup agent with audio in both directions using G.729 and G.711 codecs.	
2.15	PASS	Hunt Group Member	Verify that incoming calls to a hunt group can be answered properly when DUT is a member of the hunt group.	
2.16	PASS	Workgroup Agent	Verify that incoming calls to a workgroup can be answered properly when DUT is an agent of the workgroup.	
2.17	PASS	Call Forward – “FindMe”	Verify that calls are forwarded to DUT’s “FindMe” destination. Verify that DUT works properly when it’s a “FindMe” destination	
2.18	PASS	Mitel Converged Conferencing Server	Verify that calls are properly forwarded to the Mitel Converged Conferencing Server and it properly accepts the access code and you’re able to participate in the conference.	
2.19	N/S	Bridged Call Appearance (BCA) extension	Verify that DUT can initiate calls properly to a BCA extension and the call is presented to all of the phones that have BCA configured. Verify that the call can be answered, placed on-hold and then transferred.	There is no option to configure Programmable Button on Spectralink 8440 phones.
2.20	PASS	Additional Phones (Simulring)	Verify that calls ring simultaneously on DUT and Mitel IP Phone	
2.21	PASS	Account Codes	Verify outbound calls when Account Codes is enabled on the system.	
2.22	PASS	Place call to an International Number	Verify an outbound call to the international number	Tested with PSTN Trunk Call

ID	Result	Name	Description	Notes
2.23	PASS	Place a private call using *67	Verify private call from DUT using *67	

Secondary Switch (ST Voice Switch ST50A)

ID	Result	Name	Description	Notes
1.7	PASS	Place call	Verify successful call placement with normal dialing to a variety of terminating phones	
1.8	PASS	Receive call	Verify successful call placement with normal dialing to a variety of terminating phones	
1.11	PASS	CODEC support (DUT to Mitel Phone)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	
1.12	PASS	CODEC support (DUT to SIP reference)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	
1.13	PASS	CODEC negotiation	Verify successful negotiation between devices configured with different default CODECs (G.711-Ulaw and G.729)	
1.14	PASS	Hold DUT to SIP reference	Verify successful hold and resume of connected call	
1.15	PASS	Hold DUT to Mitel	Verify successful hold and resume of connected call	
1.16	PASS	Forward	Verify successful forwarding of incoming calls	
1.17	PASS	Forward from SIP DUT	Verify successful forwarding of incoming calls	Accept302=ext on SIP Profiles
1.19	PASS	Out-of-band DTMF Transmission	Verify successful transmission of out-of- band digits (RFC2833) for calls placed to and from the DUT with a variety of other devices	SIP INFO is not Supported by MiVoice Connect

ID	Result	Name	Description	Notes
2.2	PASS	Park	Verify successful park and retrieval of connected call	*11+ext to park the call There was no indication on the DECT handset that there is a parked call present. User must dial *12+their own extension to answer the call. but the user must retrieve the call via the handset. This is expected behavior.
2.4	PASS	Extended forward from SIP DUT	Verify extended call forwarding options – busy forwarding, ring no answer forwarding	Accept302=ext on SIP Profiles
2.5	PASS	Transfer – blind	Verify successful blind transfer of connected call	
2.7	PASS	Conference – ad hoc	Verify successful ad hoc conference of three parties	4 way or N-way conference is not supported by SIP DECT.
2.11	PASS	Caller ID	Verify that Caller ID name and number is sent and received from SIP endpoint device	
2.13	PASS	Initiate call to a Hunt Group	Initiate a call from DUT and verify that calls route to the proper Hunt Group and are answered by an available hunt group member with audio in both directions using G.729 and G.711 codecs.	
2.14	PASS	Initiate call to a Workgroup	Initiate a call from DUT and verify that calls route to the proper Workgroup and are answered successfully by an available workgroup agent with audio in both directions using G.729 and G.711 codecs.	
2.15	PASS	Hunt Group Member	Verify that incoming calls to a hunt group can be answered properly when DUT is a member of the hunt group.	
2.16	PASS	Workgroup Agent	Verify that incoming calls to a workgroup can be answered properly when DUT is an agent of the workgroup.	

ID	Result	Name	Description	Notes
2.18	PASS	Mitel Converged Conferencing Server	Verify that calls are properly forwarded to the Mitel Converged Conferencing Server and it properly accepts the access code and you're able to participate in the conference.	
2.20	PASS	Additional Phones (Simulring)	Verify that calls ring simultaneously on DUT and Mitel IP Phone	

Tertiary Switch (ShoreGear Switch SG90V)

ID	Result	Name	Description	Notes
1.7	PASS	Place call	Verify successful call placement with normal dialing to a variety of terminating phones	
1.8	PASS	Receive call	Verify successful call placement with normal dialing to a variety of terminating phones	
1.11	PASS	CODEC support (DUT to Mitel Phone)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	
1.12	PASS	CODEC support (DUT to SIP reference)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	
1.13	PASS	CODEC negotiation	Verify successful negotiation between devices configured with different default CODECs (G.711-Ulaw and G.729)	
1.14	PASS	Hold DUT to SIP reference	Verify successful hold and resume of connected call	
1.15	PASS	Hold DUT to Mitel	Verify successful hold and resume of connected call	
1.16	PASS	Forward	Verify successful forwarding of incoming calls	

ID	Result	Name	Description	Notes
1.17	PASS	Forward from SIP DUT	Verify successful forwarding of incoming calls	Accept302=ext on SIP Profiles
1.19	PASS	Out-of-band DTMF Transmission	Verify successful transmission of out-of- band digits (RFC2833) for calls placed to and from the DUT with a variety of other devices	SIP INFO is not Supported by MiVoice Connect
2.2	PASS	Park	Verify successful park and retrieval of connected call	*11+ext to park the call There was no indication on the DECT handset that there is a parked call present. User must dial *12+their own extension to answer the call. but the user must retrieve the call via the handset. This is expected behavior
2.4	PASS	Extended forward from SIP DUT	Verify extended call forwarding options – busy forwarding, ring no answer forwarding	Accept302=ext on SIP Profiles
2.5	PASS	Transfer – blind	Verify successful blind transfer of connected call	
2.7	PASS	Conference – ad hoc	Verify successful ad hoc conference of three parties	4 way or N-way conference is not supported by SIP DECT.
2.11	PASS	Caller ID	Verify that Caller ID name and number is sent and received from SIP endpoint device	
2.13	PASS	Initiate call to a Hunt Group	Initiate a call from DUT and verify that calls route to the proper Hunt Group and are answered by an available hunt group member with audio in both directions using G.729 and G.711 codecs.	
2.14	PASS	Initiate call to a Workgroup	Initiate a call from DUT and verify that calls route to the proper Workgroup and are answered successfully by an available workgroup agent with audio in both directions using G.729 and G.711 codecs.	

ID	Result	Name	Description	Notes
2.15	PASS	Hunt Group Member	Verify that incoming calls to a hunt group can be answered properly when DUT is a member of the hunt group.	
2.16	PASS	Workgroup Agent	Verify that incoming calls to a workgroup can be answered properly when DUT is an agent of the workgroup.	
2.18	PASS	Mitel Converged Conferencing Server	Verify that calls are properly forwarded to the Mitel Converged Conferencing Server and it properly accepts the access code and you're able to participate in the conference.	
2.20	PASS	Additional Phones (Simulring)	Verify that calls ring simultaneously on DUT and Mitel IP Phone	