Powering connections

Mitel Technical Configuration Notes – HO3630

February 5, 2020

Configuration of Algo 8301 Paging Adapter & Scheduler with MiVoice Connect

Description: This document provides a reference to Mitel Authorized Solutions Providers for configuring the MiVoice Connect to host the Algo 8301 Paging Adapter.

Environment: MiVoice Connect R19.1 (22.10.7600.0), Algo 8301 Paging Adapter (1.7.9)

NOTICE

TRADEMARKS

Mitel Technical Configuration Notes – Configuration of Algo 8301 Paging Adapter & Scheduler with MiVoice Connect

January 2020 – HO3630

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Overview

This document provides a reference to Mitel Authorized Solutions Providers for configuring the MiVoice Connect to host the Algo 8301 Paging Adapter. The different devices can be configured in various configurations depending on your VoIP solution. This document covers a basic Algo 8301 Paging Adapter setup as Endpoint gateway with required options setup and explicitly excludes all Algo 8301 Paging Adapter 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 Algo 8301 Paging Adapter (1.7.9)

Interop Status

The Interop of the Algo 8301 Paging Adapter has been given a Certification status. This device will be included in the Mitel Interop Reference Guide (IRG). The status of Algo 8301 Paging Adapter 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 Algo 8301 Paging Adapter 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
Algo	Algo 8301 Paging Adapter	1.7.9	8201,8180,8186,8188,8189,8190,8190S,8301,8373, 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	☑
Codec	All test cases were performed using G711 and G722 codecs.	☑
DTMF	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 Algo 8301 Paging Adapter phone is connected to MiVoice Connect.

Feature	Problem Description
Call transfer/forward/conference	The Algo 8301 Paging adapter fully supports transfer/forward, and conference scenarios performed by the other phone involved in the call. Calls may be transferred to it and it can be brought into a conference. Note, however, that the 8301 cannot initiate these actions, as it is not a telephone and does not have a keypad. **Recommendation:** Contact Algo for further information.
PRACK	PRACK cannot be controlled on MiVoice Connect system. **Recommendation:** Please contact Mitel for more information.
SIP INFO	SIP INFO is not supported by MiVoice Connect. **Recommendation:** Please contact Mitel for more information.
G729 Codec	Algo 8301 does not support G729 codec. **Recommendation:** Contact Algo Support for further information.

Network Topology

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

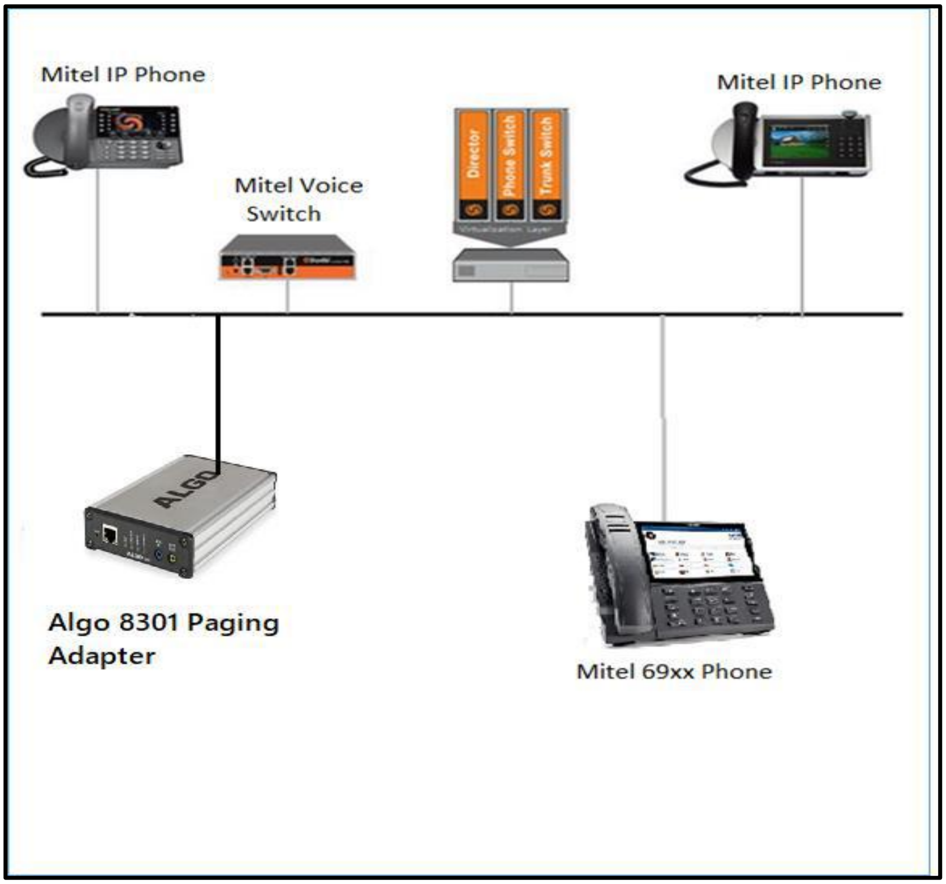


Figure 1 – Network Topology

The Algo 8301 Paging Adapter is configured as endpoint gateway where a persistent connection is created for each SIP user. Each connected device 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
- Algo 8301 Paging Adapter
- 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 Algo 8301 Paging Adapter was configured in our test environment.

We recommend that the Algo 8301 Paging Adapter 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 Algo 8301 Paging Adapter.

Call Control Options

This section describes the SIP settings required on the Mitel system to work with Algo 8301 Paging Adapter 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**

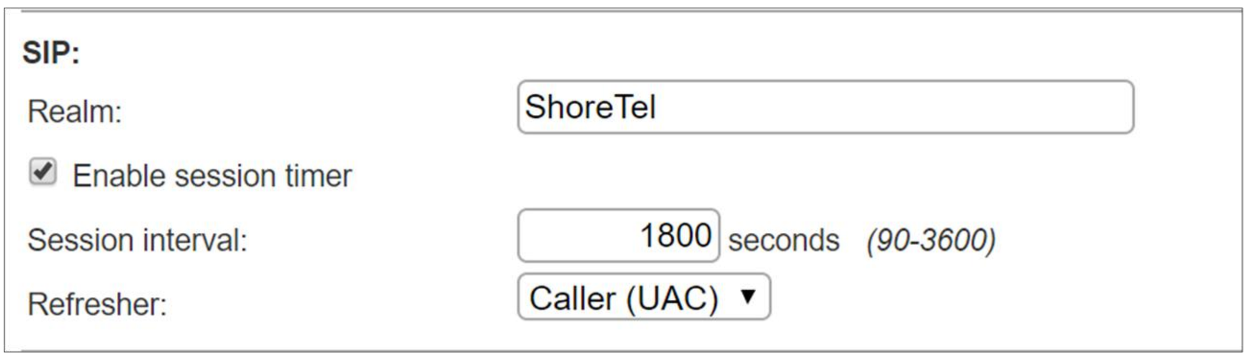


Figure 2 – Call Control Options

SIP Proxy Settings – Allocating Ports for SIP Extensions

This section describes the Switch configuration required on the Mitel system to work with the Algo 8301 Paging Adapter. 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 Algo 8301 Paging Adapter. The following example shows Port allocation designated on a Mitel SG-90 for IP Phones and SIP Proxy resources

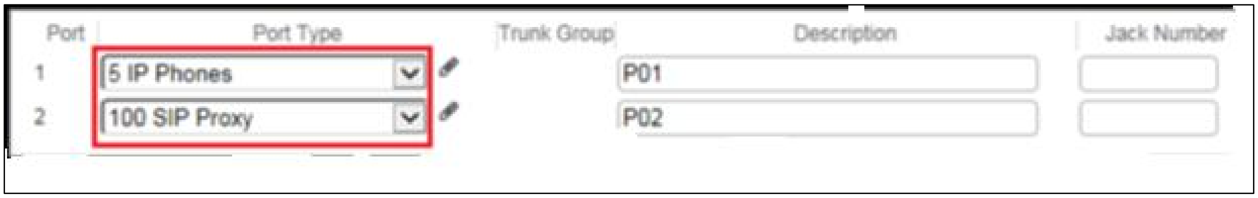
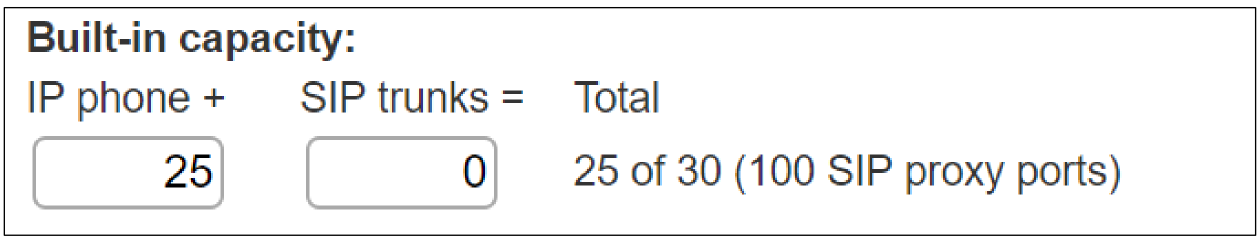


Figure 3 – Multiline IP Set Configuration

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

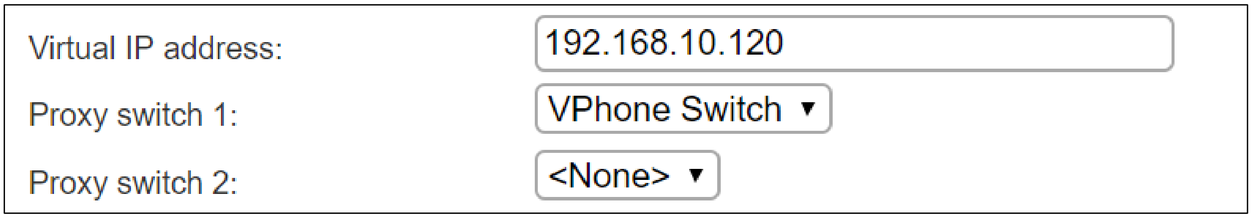


Figure 4 – Site Settings

Configure a SIP Profile

This section describes the steps required to configure the “SIP Profiles” for the Algo 8301 Paging Adapter. By default, the Algo 8301 Paging Adapter 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

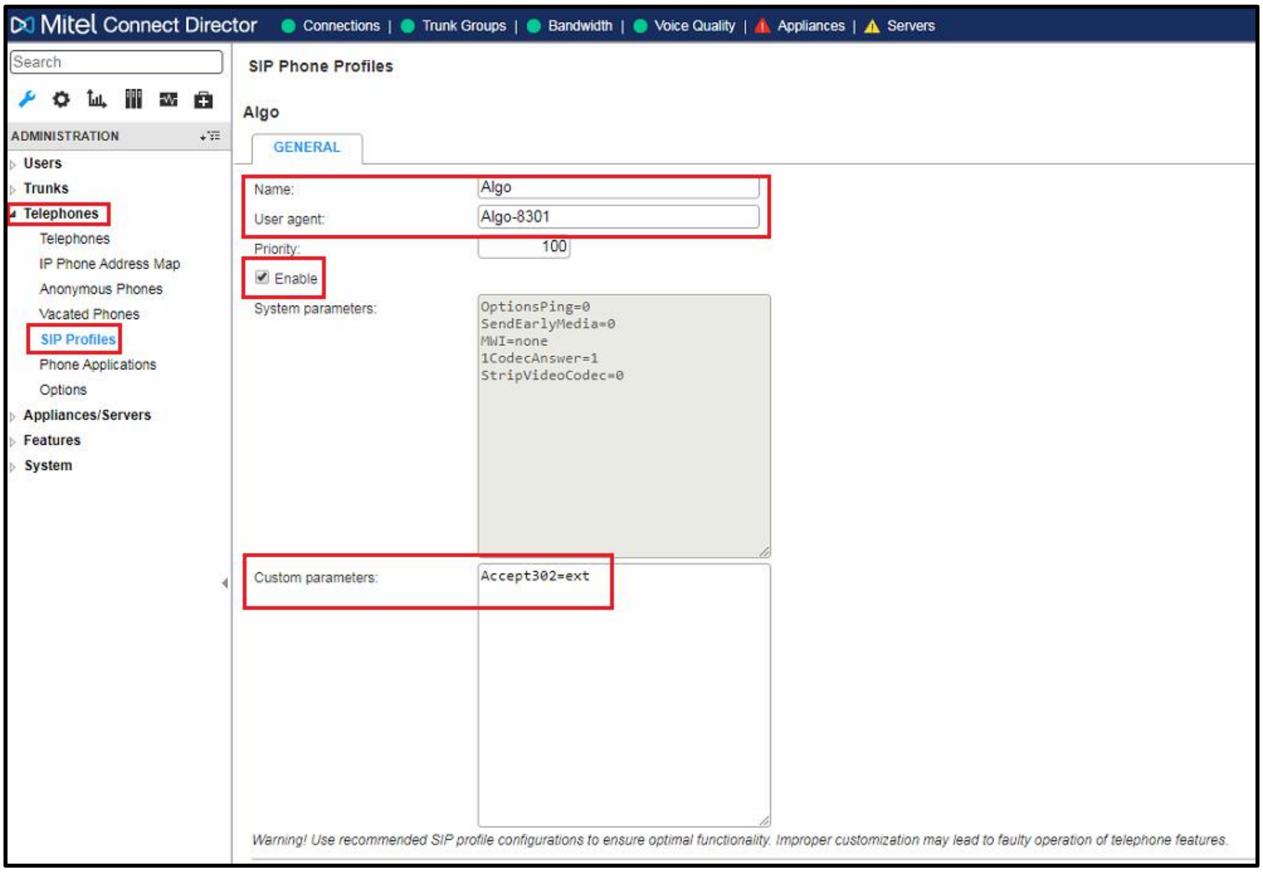


Figure 5 – Configuring a SIP Profile

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

4. For the parameter User agent: enter “Algo-8301” (without quotes)

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

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

7. If you want to add any parameter’s manualy. please add parameters under Custom Parameters

Sections.

8. Click SAVE

Configure Algo 8301 Paging Adapter 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

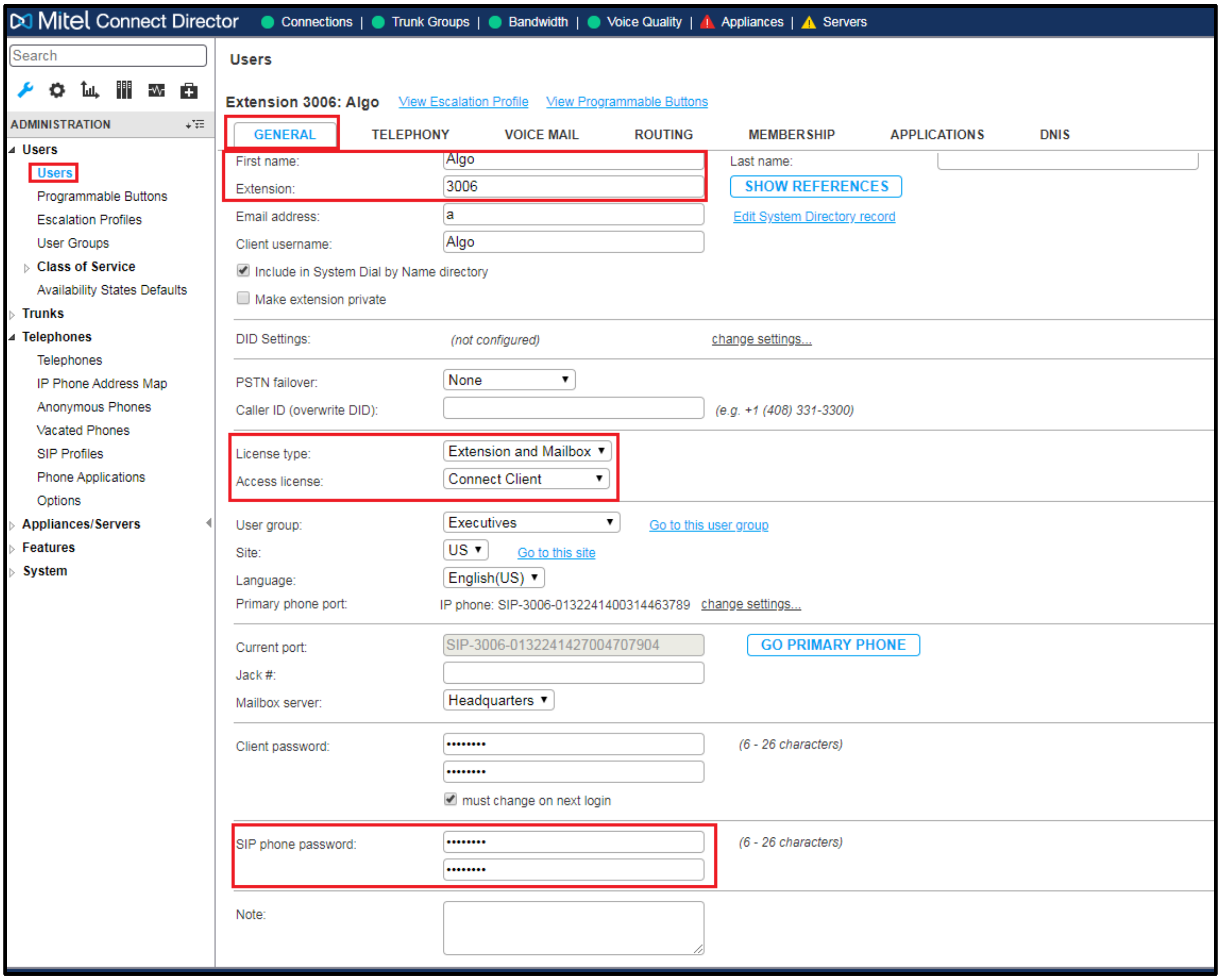


Figure 6 – Create a User

Algo 8301 Paging Adapter & Schedular Configuration Notes

This section outlines the basic instruction on how to program Algo 8301 Paging Adapter to interconnect with MIVB. This is by no means a comprehensive guideline. We assume that Algo 8301 Paging Adapter has been upgraded to the latest software release as found in ***http://www.algosolutions.com/support/firmware.html***. Please note that your phone must have been upgraded to current software release.

Home Page Login

Access the 8301 Paging Adapter & Scheduler web page by entering the IP address into a browser (Chrome, IE, Firefox etc) and login using the default password ***algo***.

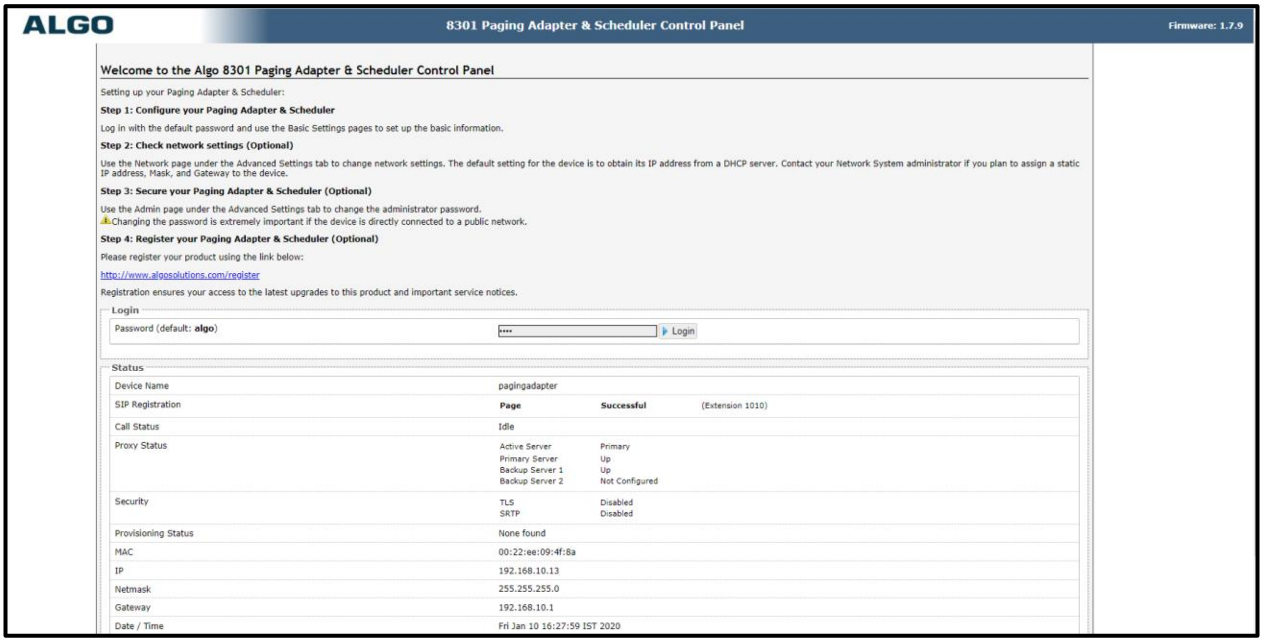


Figure 7 – Algo 8301 Paging Adapter Home page login

Configuration details

The 8301 Paging Adapter & Scheduler can be registered as a third-party SIP extension with a hosted or enterprise Communications Server supporting 3rd party SIP endpoints. To register the adapter with the SIP server, use the ***Basic Settings > SIP*** tab in the web interface to enter the Communication Server IP address, extension, username, and password.

This information will be available from the IT Administrator. If VLAN is used, navigate to the Advanced Settings > Network tab to set VLAN options. (Note, once the adapter is using VLAN you will need to be on the same VLAN to access the web interface). Navigate to the Status tab and ensure the extension(s) are successfully registered. The adapter may now be accessed by dialling its assigned extension from a telephone, device, or client. The adapter will auto-answer, play the default WAV pre-announce tone, and allow voice paging until disconnected.

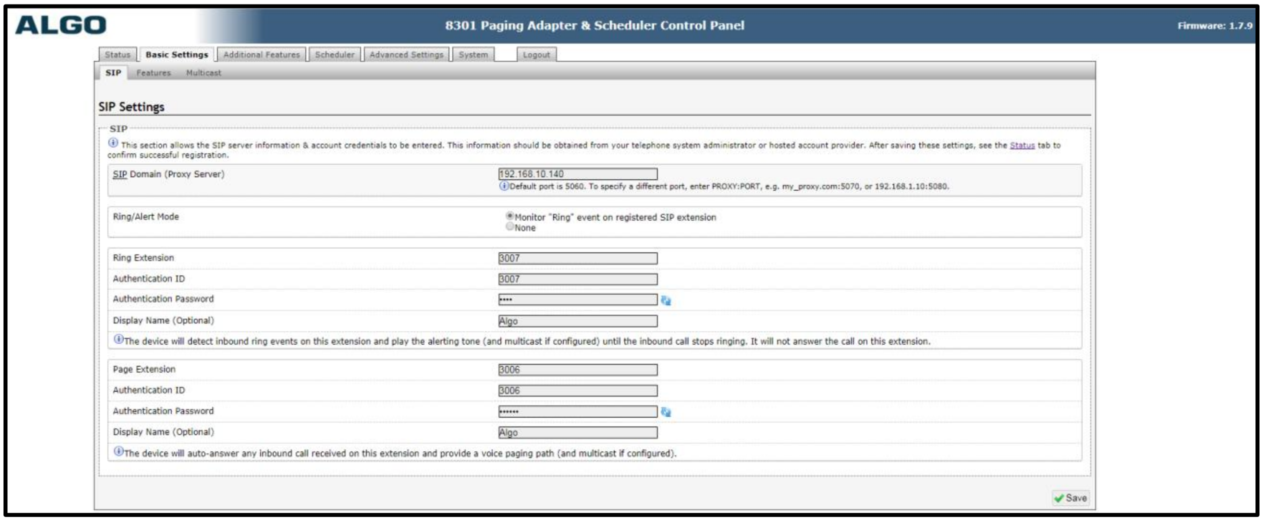
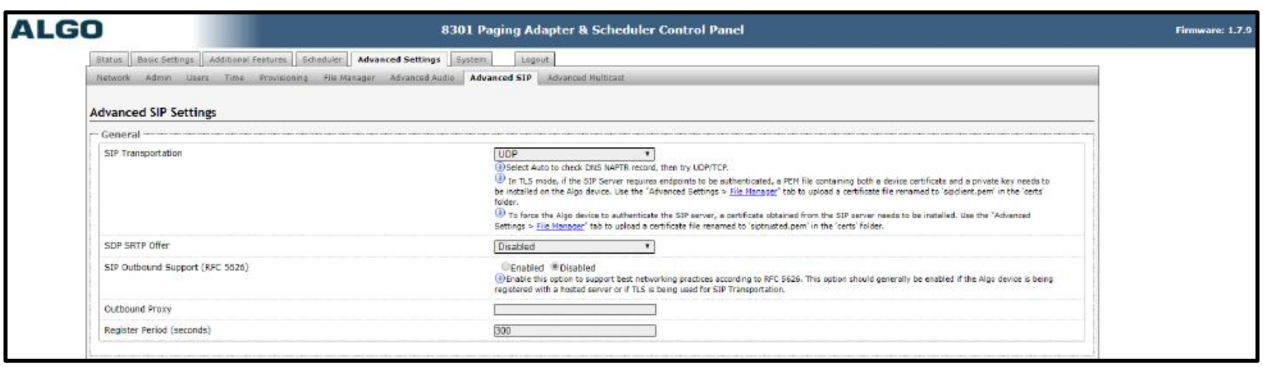


Figure 8 – SIP Settings



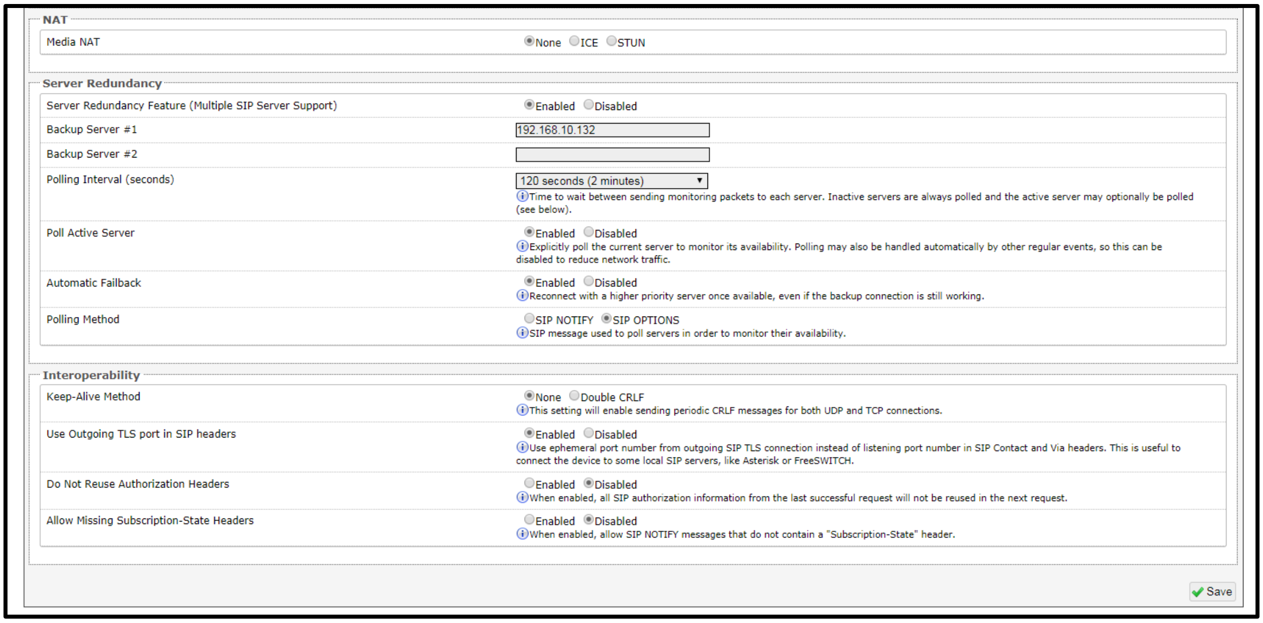


Figure 9 – Advanced SIP Settings

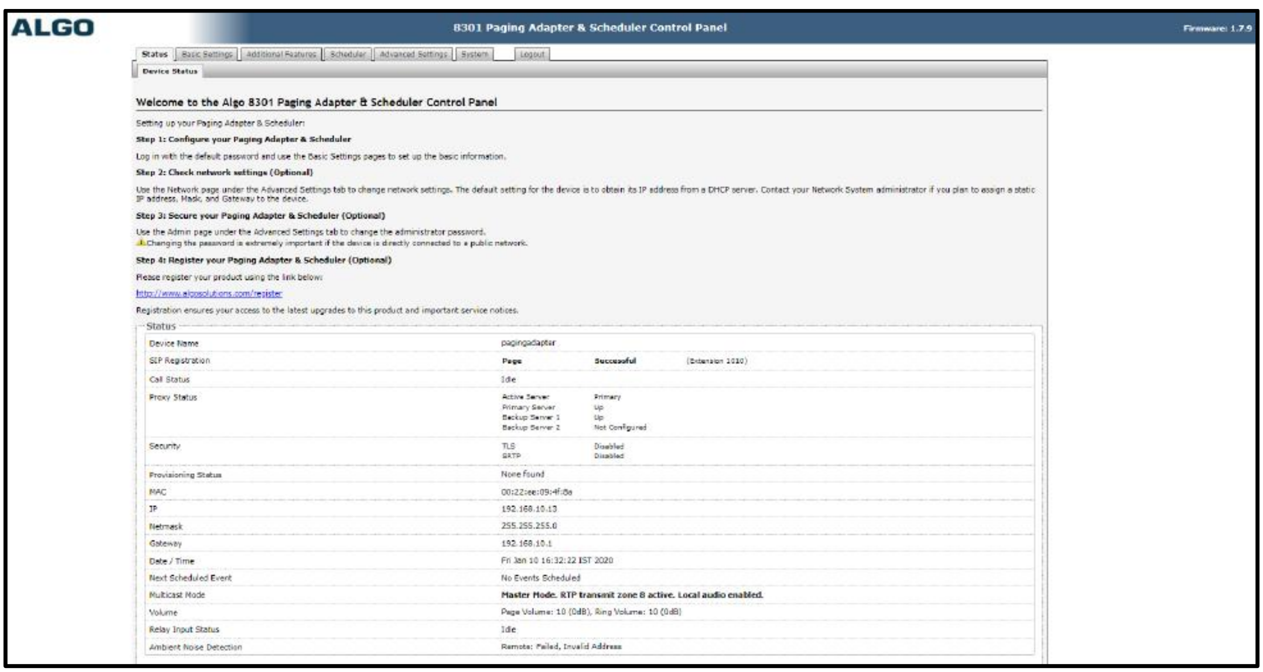


Figure 10 – Device status

Summary of Tests and Results

Primary Switch (ST Virtual Phone Switch)

N/A = Not Applicable 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	PASS	Device initialization with DHCP	Verify successful startup and initialization of the device up to a READY/IDLE state using DHCP	
1.4	PASS	Device reset – idle (for dynamic configurations)	Verify successful re-initialization of device after power loss while device is idle	
1.5	N/T	Verify Diffserv Code Point support	Verify the ability to set Diffserv Code Point from SIP DUT (device under test)	
1.6	PASS	Verify Date and Time Update support	Verify setting of Date and Time Update on SIP DUT	
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	N/A	Place call - redial	Verify successful call placement using re-dial to SIP Reference	Algo 8301 does not have a redial feature, as it is not a telephone.
1.10	N/A	Place call – speed dial	Verify successful call placement using programmed speed dial	Algo 8301 does not have a speed dial feature, as it is not a telephone.

ID	Result	Name	Description	Notes
1.11	PASS	CODEC support (DUT to ShoreTel Phone)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	Algo 8310 support only G711 & G722. G729 is not supported
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)	Algo 8310 support only G711 & G722. G729 is not supported
1.13	PASS	CODEC negotiation	Verify successful negotiation between devices configured with different default CODECs (G.711-Ulaw and G.729)	Algo 8310 support only G711 & G722. G729 is not supported
1.14	N/A	Hold DUT to SIP reference	Verify successful hold and resume of connected call	Algo 8301 does not have a call hold feature, as it is not a telephone. Calls may be put on hold by the other party.
1.15	N/A	Hold DUT to ShoreTel	Verify successful hold and resume of connected call	Algo 8301 does not have a call hold feature, as it is not a telephone. Calls may be put on hold by the other party.
1.16	N/A	Forward	Verify successful forwarding of incoming calls	Algo 8301 does not have a call forward feature, as it is not a telephone. Calls may be transferred by the other party.
1.17	N/A	Forward from SIP DUT	Verify successful forwarding of incoming calls	Algo 8301 does not have a call forward feature, as it is not a telephone. Calls may be transferred by the other party.
1.18	N/A	Mute	Verify device's mute function	Algo 8301 does not have a mute feature.
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	
1.20	N/A	Missed call notification	Verify that device notifies the user about missed calls	Algo 8301 does not have a missed notification feature.

ID	Result	Name	Description	Notes
1.21	PASS	Volume	Verify the device's volume adjustment function	
2.1	N/A	Call waiting	Verify appropriate notification and successful connection of incoming call while busy with another party	Algo 8301 does not support call waiting.
2.2	N/A	Park	Verify successful park and retrieval of connected call	Algo 8301 does not support call Park.
2.3	N/A	Extended forward	Verify extended call forwarding options – busy forwarding, ring no answer forwarding	Algo will automatically answer all incoming calls, so call forwarding no answer is not supported.
2.4	N/A	Extended forward from SIP DUT	Verify extended call forwarding options – busy forwarding, ring no answer forwarding	
2.5	N/A	Transfer – blind	Verify successful blind transfer of connected call	
2.6	N/A	Transfer – monitored	Verify successful monitored transfer of connected call	Algo 8301 supports call transfer performed by the other phone involved in the call.
2.7	N/A	Conference – ad hoc	Verify successful ad hoc conference of three parties	Algo 8301 supports conference scenarios performed by the other phone involved in the call.
2.8	N/T	Place call – secondary line	Verify successful call placement using secondary line	
2.9	N/T	Receive call – secondary line	Verify successful connection of incoming call on secondary line	
2.10	N/A	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	

ID	Result	Name	Description	Notes
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.	Only tested with G711 as G729 not supported by Algo.
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.	Only tested with G711 as G729 not supported by Algo.
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.	(Call will be automatically answered by Algo when called to hunt group number)
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	N/A	Call Forward – "FindMe"	Verify that calls are forwarded to DUT's "FindMe" destination. Verify that DUT works properly when it's a "FindMe" destination	From algo we don't have dial pad to enter digits as per IVR.
2.18	N/A	ShoreTel Converged Conferencing Server	Verify that calls are properly forwarded to the ShoreTel Converged Conferencing Server and it properly accepts the access code and you're able to participate in the conference.	From algo we don't have dial pad to press digits to join the conference.
2.19	N/T	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.	Call hold/Transfer not supported
2.20	PASS	Additional Phones (Simulring)	Verify that calls ring simultaneously on DUT and ShoreTel IP Phone	

ID	Result	Name	Description	Notes
2.21	N/T	Account Codes	Verify outbound calls when Account Codes is enabled on the system.	
2.22	N/T	Place call to an International Number	Verify an outbound call to the international number	
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 ShoreTel Phone)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)	(Tested G711 & G722. G729 is not supported by Algo)
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)	(Tested G711 & G722. G729 is not supported by Algo)
1.13	Pass	CODEC negotiation	Verify successful negotiation between devices configured with different default CODECs (G.711-Ulaw and G.729)	(Tested G711 & G722. G729 is not supported by Algo)
1.14	N/A	Hold DUT to SIP reference	Verify successful hold and resume of connected call	
1.15	N/A	Hold DUT to ShoreTel	Verify successful hold and resume of connected call	
1.16	N/A	Forward	Verify successful forwarding of incoming calls	Algo will automatically answer all incoming calls, so call forwarding no answer is not supported.

ID	Result	Name	Description	Notes
1.17	PASS	Forward from SIP DUT	Verify successful forwarding of incoming calls	Algo will automatically answer all incoming calls, so call forwarding no answer is not supported.
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	

ID	Result	Name	Description	Notes
2.2	N/A	Park	Verify successful park and retrieval of connected call	
2.4	PASS	Extended forward from SIP DUT	Verify extended call forwarding options – busy forwarding, ring no answer forwarding	Algo will automatically answer all incoming calls, so call forwarding no answer is not supported.
2.5	N/A	Transfer – blind	Verify successful blind transfer of connected call	
2.7	N/A	Conference – ad hoc	Verify successful ad hoc conference of three parties	Algo 8301 supports conference scenarios performed by the other phone involved in the call.
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.	Only tested with G711 as G729 not supported by Algo.
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	N/A	ShoreTel Converged Conferencing Server	Verify that calls are properly forwarded to the ShoreTel Converged Conferencing Server and it properly accepts the access code and you're able to participate in the conference.	From algo we don't have dial pad to to press digits to join the conference.
2.20	PASS	Additional Phones (Simulring)	Verify that calls ring simultaneously on DUT and ShoreTel 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)	(Tested G711 & G722. G729 is not supported by Algo)
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)	(Tested G711 & G722. G729 is not supported by Algo)
1.13	PASS	CODEC negotiation	Verify successful negotiation between devices configured with different default CODECs (G.711-Ulaw and G.729)	(Tested G711 & G722. G729 is not supported by Algo)

ID	Result	Name	Description	Notes
1.14	N/A	Hold DUT to SIP reference	Verify successful hold and resume of connected call	
1.15	N/A	Hold DUT to Mitel	Verify successful hold and resume of connected call	
1.16	N/A	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
2.2	N/A	Park	Verify successful park and retrieval of connected call	
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	N/A	Transfer – blind	Verify successful blind transfer of connected call	
2.7	N/A	Conference – ad hoc	Verify successful ad hoc conference of three parties	
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.	(Tested G711 & G722. G729 is not supported by Algo)
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.	(Call will be automatically answered by Algo when called to hunt group number)

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	N/A	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.	From algo we don't have dial pad to to press digits to join the conference.
2.20	PASS	Additional Phones (Simulring)	Verify that calls ring simultaneously on DUT and Mitel IP Phone	