Mitel Technical Configuration Notes – HO3620

January 21, 2020

Configuration of Algo 8301 Paging Adapter Scheduler with MiVoice Business 9.0 SP3

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

Environment: MiVoice Business 9.0 SP3 (9.0.3.15), Algo 8301 Paging Adapter (1.7.9)

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TRADEMARKS

Mitel Technical Configuration Notes – Configuration of Algo 8301 Paging Adapter Scheduler with MiVoice Business 9.0 SP3.

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Overview

This document provides a reference to Mitel Authorized Solutions Providers for configuring the MiVoice Business 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 with required options setup.

Interop History

Version	Date	Reason
1	November, 2019	Algo 8301 Paging Adapter with MiVB 9.0 SP3 (9.0.3.15)

Interop Status

The Interop of the Algo 8301 Paging Adapter has been given a Certification status. This device will be included in the SIP CoE Reference Guide. The status Algo 8301 Paging Adapter achieved is:

	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.

Algo 8301 Paging Adapter with MiCloud Flex/MiVoice Business has been certified under Teritary category. The three levels of certification corresponding to different deployment models are explained below –

BRONZE – Bronze status indicates the device is interoperable with MiVB on-premise deployments and MiCloud Flex MPLS/SD-WAN solution, but UDP/TCP as the primary transport protocol as primary and with minimal coverage for TLS. Corresponds to the Tertiary Topology Diagram

SILVER - Silver is the next level of compliance where the device can only be used against MPLS and SD-WAN variants of Flex but not as OTT Teleworker. Both UDP/TCP and TLS can be used as transport protocols. And the device is also compatible with MiVB on-premise solution. Corresponds to the Secondary and Tertiary Topology Diagrams

GOLD – Gold is the premium level of compliance that the device can be used across MiVB onpremise and MiCloud Flex deployments (OTT Teleworker, MPLS, SD-WAN). Corresponds to the Primary, Secondary and Tertiary Topology Diagram.

Software & Hardware Setup

The test setup generated basic SIP calls between Algo 8301 Paging Adapter and the MiVoice Business.

Note: Although this testing was performed on the below tested variants, the scope of this testing can be extended to other product variants that work with the same firmware. The list of components for which this testing can be considered applicable is given in the "Additional Applicable Variants" column of the following table –

Manufacturer	Tested Variant	Software Version	Additional Applicable Variants
Mitel	MiVoice Business Platform	9.0 SP3 (9.0.3.15)	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
Mitel	68xx/69xx Mode	5.1.0.2051/01.05.00.075	NA

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 Pans for detailed test cases.

Feature	Feature Description	Issues
Basic Call	Making and receiving a call	☑
Codec	G711 and G722 codec	☑
TLS/SRTP	Basic incoming/outgoing call.	☑
Resiliency	Device able to handle resiliency when primary MiVB goes down.	☑

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

Resiliency

The following table lists the scenarios of resilience supported by this device when connected to the Mitel MiVoice Business.

Device	Basic	Advanced
Algo 8301 Paging Adapter	☑	⚠

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

Note: Refer to list of device limitations and known issues later in the document for recommendations.

The various scenarios are described below. The scenario names are a convenience for understanding this section of the configuration guide.

Basic: Resiliency is achieved by utilizing the ability of DNS servers to provide multiple IP addresses against a single FQDN. This is generally achieved by using DNS SRV or A records. This scenario requires nothing from a SIP Endpoint except that it supports standard DNS behavior. It can also be done by manually setting up back proxy on the phone.

Using REGISTER-301 Moved Permanently message to redirect registration to an alternate MiVoice Business element.

At a minimum, a 32-second timeout for the REGISTER, SUBSCRIBE, INVITE or OPTIONS messages should trigger a Failover

After Failover/Failback – the device must restart all subscriptions (message-summary

Advanced: There are different ways to detect the failure in this category.

P-Alternate-Server:
Use the P-Alternate-Server header in the REGISTER-200 OK message to store the HE and SE addresses.
Heartbeat
Use a light-weight heartbeat to periodically monitor the health of the MiVoice Business element to which the device is connected. This allows for the device to recover from failures faster without overloading the controlling element.
Survival Mode
Continue existing conversations when a failure is detected until at least the Session Timer expires or the user takes an action which causes termination. Displaying a message on the set is also recommended.
First Call after Failure
Implement a policy to time out a new call early if no 18x/2xx message is received.

Note: *ALGO uses SIP NOTIFY & OPTIONS method to detect the failure between the servers.*

While testing resillency, we have used SIP OPTIONS method to poll servers in order to detect their availability.

For example, Initially Algo is registered to primary server and calls are working fine. Now we make primary server down, SIP OPTIONS method will poll from Algo (OPTIONS will be sent by Algo every 2 mins) and detect the available server. From the time we make primary down and next SIP OPTION method sent in-between if we make calls, calls will fail. Once SIP OPTIONS get success response from active server, calls will work immediately and on register expire Algo will directly send register to the active server.

Device Limitations

This is a list of problems or not supported features when the Algo 8301 Paging Adapter is connected to MiVoice Business.

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. information. **Recommendation:** Contact Algo for further information.
FAX	Algo 8301 Paging Adapter does not support FAX. **Recommendation**: Contact Algo Support for further information.
G729 Codec	Algo 8301 does not support G729 codec. **Recommendation**: Contact Algo Support for further information.
Resiliency	Initially Algo will be registered to Primary MiVB. During resillency testing when primary is down and make call immediately call will fail. Since SIP OPTIONS will be sent by Algo every 2 mins (minimum value) which is used to poll the active server. With in this time gap calls will fail. Once OPTIONS sent and got success response, calls will work. Note that the primary resiliency method of SRV Record is fully supported. **Recommendation**: Contact Algo Support for further information.

Network Topology

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

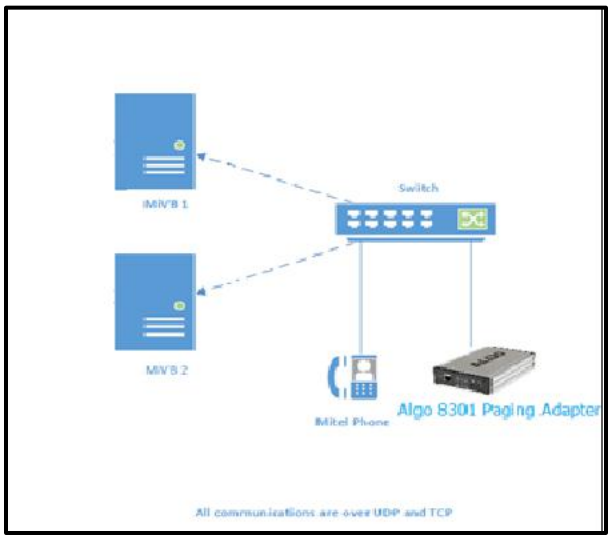


Figure 1 – Network Topology

The Algo 8301 Paging Adapter is configured as a SIP endpoint where a persistent connection is created for each SIP user. Each connected device has a separate SIP connection to the SIP server.

MiVoice Business - Configuration Notes

The following steps show how to program a MiVoice Business to connect with the Algo 8301 Paging Adapter.

Network Requirements

- There must be adequate bandwidth to support the voice over IP. As a guide, the Ethernet bandwidth is approx 85 Kb/s per G.711 voice session and 29 Kb/s per G.729 voice session (assumes 20ms packetization). As an example, for 20 simultaneous SIP sessions, the Ethernet bandwidth consumption will be approx 1.7 Mb/s for G.711 and 0.6Mb/s. Almost all Enterprise LAN networks can support this level of traffic without any special engineering. Please refer to the MiVoice Business Engineering guidelines for further information.
- For high quality voice, the network connectivity must support a voice-quality grade of service (packet loss <1%, jitter < 30ms, one-way delay < 80ms).

Assumptions for the MiVoice Business Programming

- The SIP signaling connection uses UDP on Port 5060.

Software License – SIP Licensing

Ensure that the MiVoice Business is equipped with enough Mode licenses for the connection of SIP end points. This can be verified within the Software License Feature section form.

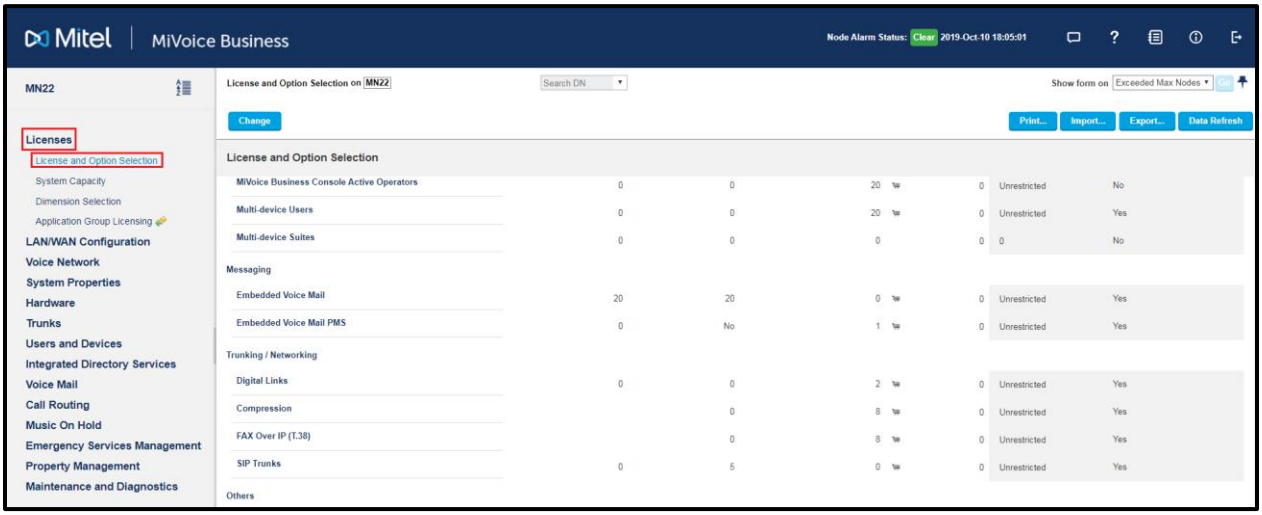


Figure 2 – Software License

Multiline IP Set Configuration

On the MiVoice Business, a SIP device can be programmed either in the User Configuration form or the Multiline IP Set Configuration form and are programmed as a “Generic SIP Phone”.

The User PIN is the SIP authentication password and the Number is the Directory Number (DN is a telephone number). All other field names should be programmed according to the site requirements or left at default.

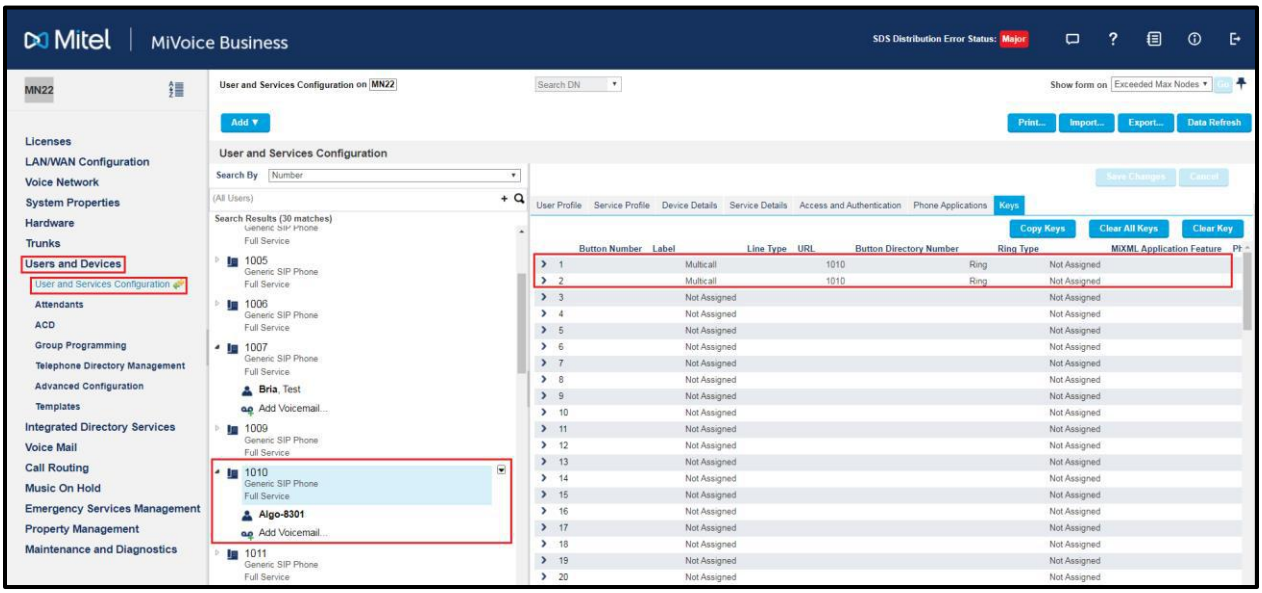
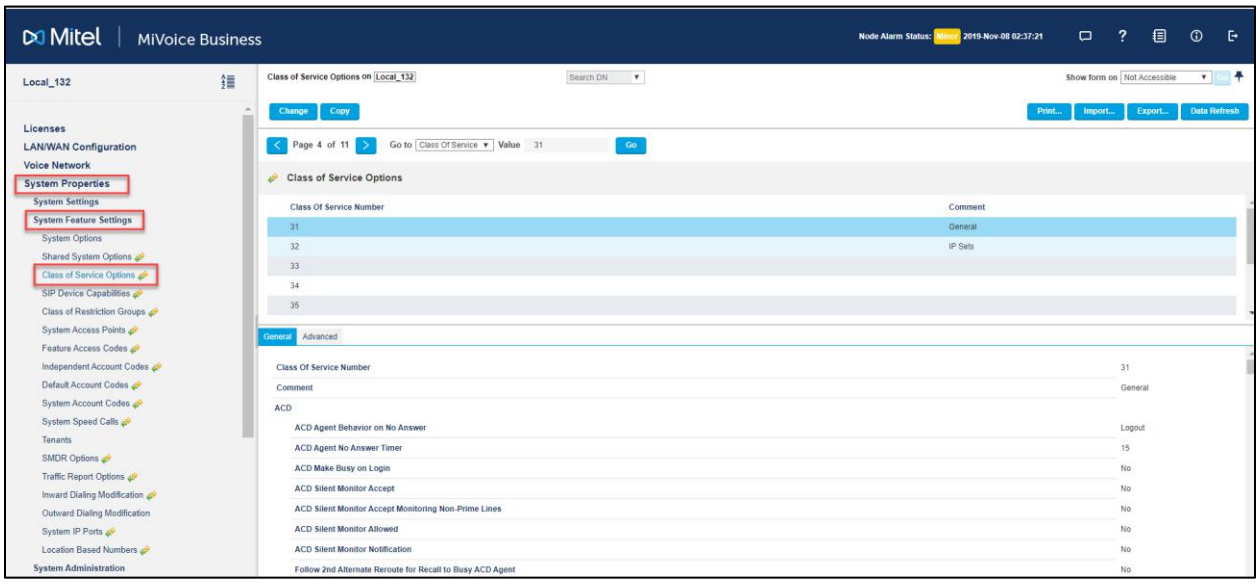


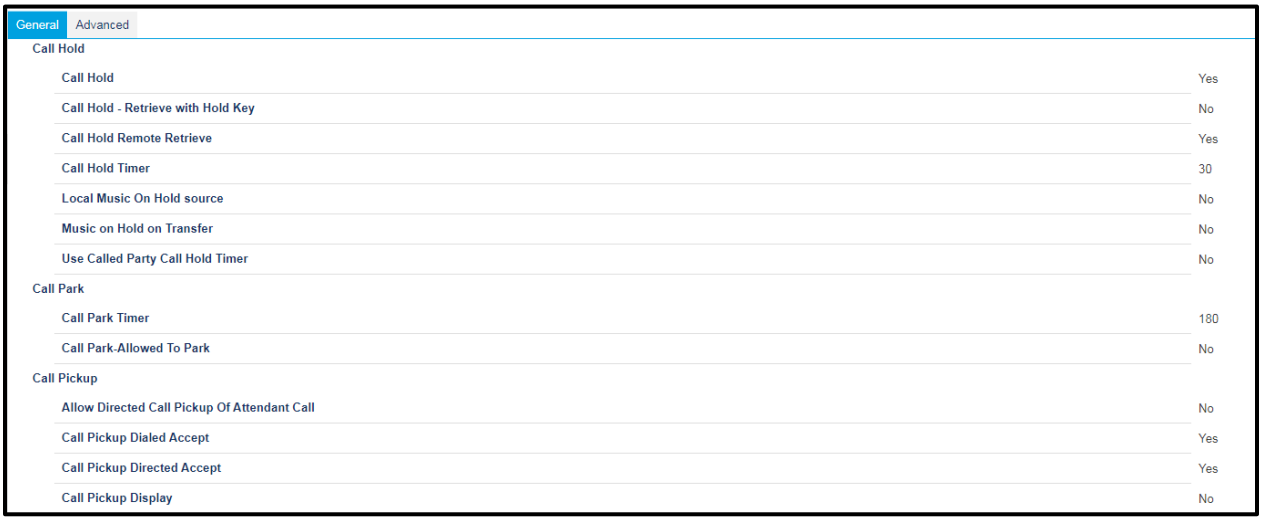
Figure 3 – Create SIP Extension

Class of Service Assignment

The Class of Service Options form is used to create or edit the Class of Service and specify its options. Classes of Service, identified by Class of Service numbers, are referenced by the Station Attributes form for the SIP device.

Many different options may be required for your site deployment, but the options below are required to be changed from the default for a Generic SIP Device to work with the 3300 ICP.





General | Advanced

Call Privacy	
Call Privacy	No
Calling Party Name Substitution	No
Name Suppression on outgoing Trunk Call	No
Privacy Released	No
Public Network Identity Provided	No
Call Waiting	
Call Waiting Swap	No
ONS CLASS/CLIP: Visual Call Waiting	Yes
Campon	
Auto Campon Timer	10
Campon Recall Timer	10
Direct Voice Call	
Direct Voice Call - Accept	No
Direct Voice Call - Allow	No
Direct Voice Call - Maximize Volume	No

General | Advanced

Display	
After Answer Display Time	
Calling Name Display - Internal - ONS	Yes
Calling Number Display - Internal - ONS	Yes
Display ANI/DNIS/ISDN Calling/Called Number	No
Display ANI/ISDN Calling Number Only	No
Display Caller ID on multicall/keylines	No
Display Caller ID On Multicall/Keylines Timer	5
Display Caller ID On Single Line Displays For Forwarded Calls	No
Display Dialed Digits during Outgoing Calls	No
Display DNIS/Called Number Before Digit Modification	No
Display DNIS on Key Label	No
Display Held Call ID on Transfer	No
Display Transfer Destination on Recall	No
Hot Desk External User - Display Internal Calling ID	No
Maintain Ringing Party During Recall	No
Non-Prime Public Network Identity	No

General | Advanced

Originator's Display Update In Call Forwarding/Rerouting	No
Prefer Call Forwarding/Rerouting Information	No
Prefer Name for Call Information	No
Suppress Delivery of Caller ID Display between Sets	No
Suppress Delivery of Caller ID Display between Sets - Override	No
Suppress Display Of Account Code Numbers	No
Suppress Redial Display	No
Fax	
Campon Tone Security	No
External Trunk Standard Ringback	No
Fax Capable	No
Return Disconnect Tone When Far End Party Clears	No
HCI	
HCI/CTI/TAPI Call Control Allowed	Yes
HCI/CTI/TAPI Monitor Allowed	Yes

General | Advanced

Hot Desk	
Green BLF Lamp for Logged in Hotdesk User	No
Hot Desk Auto Logout Timer	0
Hot Desk External User - Allow DTMF Dialing	Yes
Hot Desk External User - Allow Mid-Call Features	Yes
Hot Desk External User - Answer Confirmation	Yes
Hot Desk External User - Dial Tone on Call Complete	Yes
Hot Desk External User - Permanent Login	No
Hot Desk External User - Remote MWI Enable Feature Access Code	
Hot Desk External User - Remote MWI Disable Feature Access Code	
Hot Desk Login Accept	Yes
Hot Desk Remote Logout Enabled	No
Miscellaneous	
Backlighting - Enabled	Yes
Clear All Features Remote	No
Enbloc Dialing - Enabled	No

General | Advanced

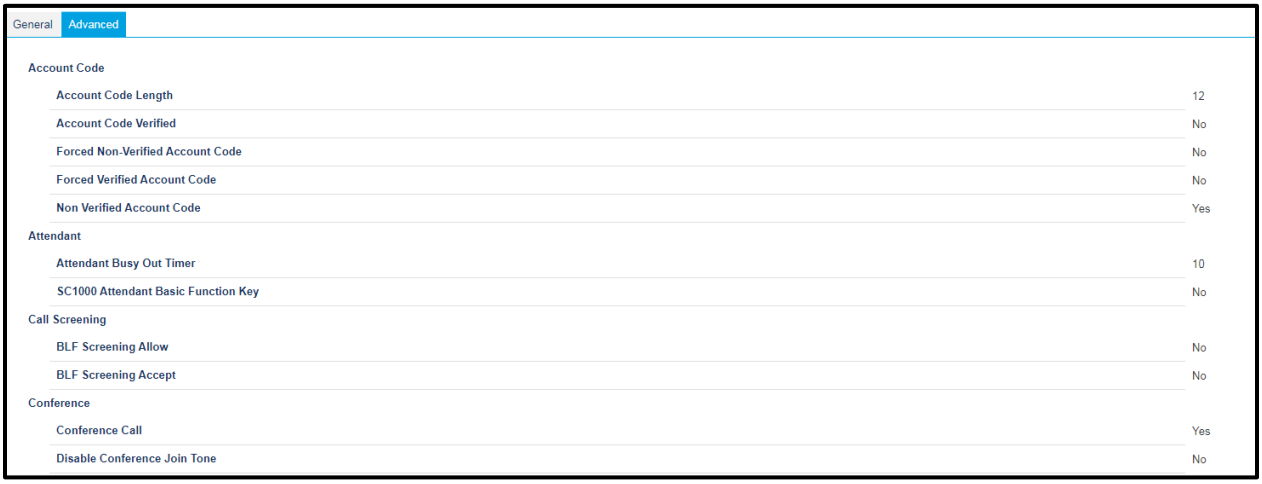
Handset Volume Adjustment Saved	No
Head Set Switch Mute	No
Integrated DECT High Power - Enabled	Yes
Integrated DECT Wideband - Enabled	Yes
Enable Device Configuration	0
Multi-Color LED Support - Disable	No
Phone Lock	No
Reseize Timer	180
Timed Reminder Allowed	Yes
User Inactivity Timer	0
Paging	
Group Page Accept	No
Group Page Allow	No
Loudspeaker Pager Equivalent Zone Override Security	No
Loudspeaker Pager Override	Yes
Pager Access All Zones	Yes

General | Advanced

Pager Access Individual Zones	No
PC Port	
PC Port On IP Device - Disable	No
RAD	
Answer Plus Delay To Message Timer	20
Answer Plus Expected Off-hook Timer	30
Answer Plus Message Length Timer	10
Answer Plus System Reroute Timer	0
Recorded Announcement Device	No
Recorded Announcement Device - Advanced	No
Ringing	
Allow Recall after Transfer	No
Delay Ring Timer	10
No Answer Recall Timer	17
Ringing Line Select	No
Ringing Timer	180

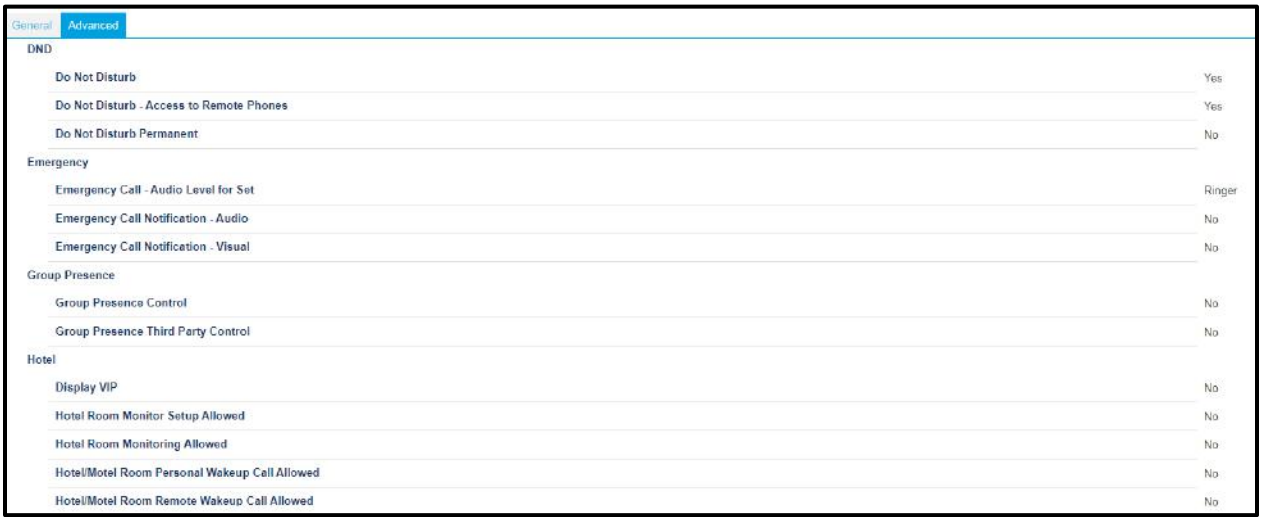
General | Advanced

SMDR	
SMDR External	No
SMDR Internal	No
Trunk	
ANI/DNIS/ISDN Number Delivery Trunk	No
DASS II OLI/TLI Provided	No
Public Network Access via DPNSS	Yes
Public Network To Public Network Connection Allowed	Yes
Public Trunk	Yes
R2 Call Progress Tone	No
Suppress Simulated CCM after ISDN Progress	No
Trunk Calling Party Identification	Yes
Trunk Flash Allowed	No
Two B-Channel Transfer Allowed	No
Voice Mail	
COV/ONS/E&M Voice Mail Port	No
ONS VMail-Delay Dial Tone Timer	5



General | Advanced

Account Code	
Account Code Length	12
Account Code Verified	No
Forced Non-Verified Account Code	No
Forced Verified Account Code	No
Non Verified Account Code	Yes
Attendant	
Attendant Busy Out Timer	10
SC1000 Attendant Basic Function Key	No
Call Screening	
BLF Screening Allow	No
BLF Screening Accept	No
Conference	
Conference Call	Yes
Disable Conference Join Tone	No



General | Advanced

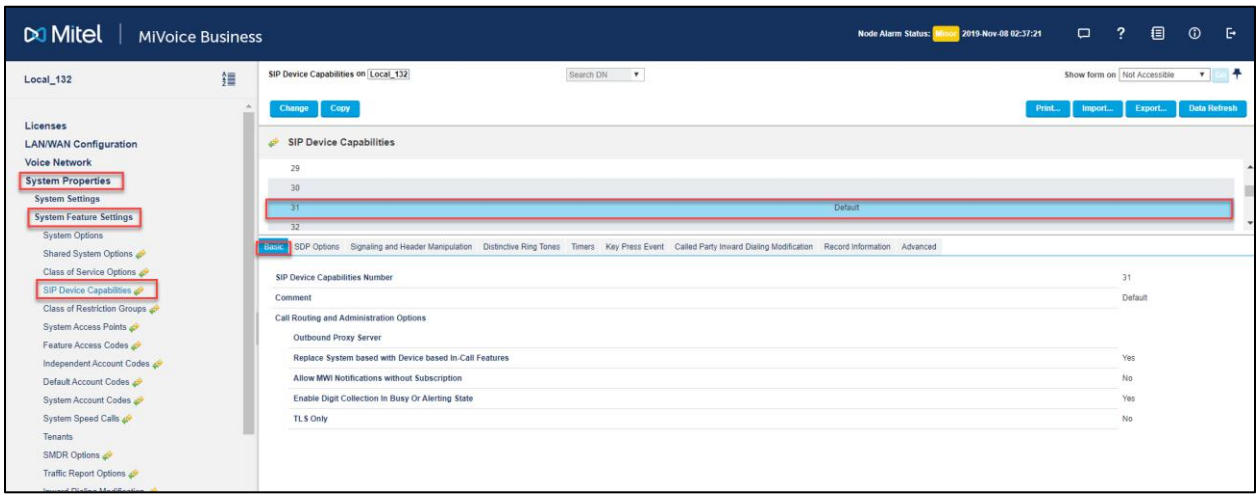
DND	
Do Not Disturb	Yes
Do Not Disturb - Access to Remote Phones	Yes
Do Not Disturb Permanent	No
Emergency	
Emergency Call - Audio Level for Set	Ringer
Emergency Call Notification - Audio	No
Emergency Call Notification - Visual	No
Group Presence	
Group Presence Control	No
Group Presence Third Party Control	No
Hotel	
Display VIP	No
Hotel Room Monitor Setup Allowed	No
Hotel Room Monitoring Allowed	No
Hotel/Motel Room Personal Wakeup Call Allowed	No
Hotel/Motel Room Remote Wakeup Call Allowed	No

Figure 4 – Class of Service Options

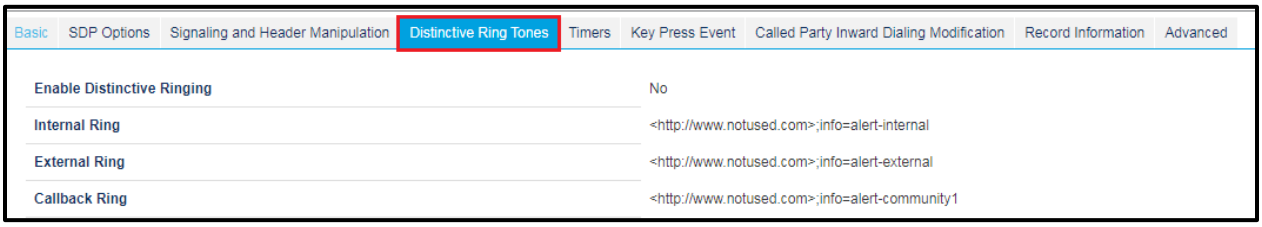
SIP Device Capabilities

This form provides configuration options that can be applied to various types of SIP devices. The association between the SIP device and the form is like how the Class of Service options work. The SIP Device Capabilities number provides a SIP profile that can be applied to SIP devices to allow for alternate capabilities as recommended through the Mitel interop process.

In the SIP Device Capabilities form, program a SIP Device Capabilities Number for the Algo 8301 Paging Adapter. Ensure that “Replace System based with Device based In-Call Features” is set to ‘**Yes**’.




Basic
SDP Options
Signaling and Header Manipulation
Distinctive Ring Tones
Timers
Key Press Event
Called Party Inward Dialing Modification
Record Information
Advanced
Allow Display Update
Yes
Allow FQDN for Resiliency
No
Disable Reliable Provisional Responses
Yes
Disable Use of User-Agent and Server Headers
No
Fail REFER To Keep Call Active On Mid-Call Feature
No
If TLS use 'sips:' Scheme
No
Mode for Out-of-Band DTMF
RFC 4733 DTMF
Multilingual Name Display
No
Override Auto-Answer Headers
No
Override Auto-Answer Headers With
Q.850 Reason Headers
No
Remove Anonymous User
No
Require Reliable Provisional Responses on Outgoing Calls
No
Suppress Redirection Headers
No
Use P-Asserted Identity Header
Yes
Use user=phone
No


Basic
SDP Options
Signaling and Header Manipulation
Distinctive Ring Tones
Timers
Key Press Event
Called Party Inward Dialing Modification
Record Information
Advanced
Enable Distinctive Ringing
No
Internal Ring
<http://www.notused.com>;info=alert-internal
External Ring
<http://www.notused.com>;info=alert-external
Callback Ring
<http://www.notused.com>;info=alert-community1


Basic
SDP Options
Signaling and Header Manipulation
Distinctive Ring Tones
Timers
Key Press Event
Called Party Inward Dialing Modification
Record Information
Advanced
Registration Period Minimum
300
Session Timer
3600
Session Timer: Local as Refresher
No
Subscription Period
3600
Subscription Period Minimum
300
Subscription Period Refresh (%)
80
Invite Ringing Response Timer
0


Basic
SDP Options
Signaling and Header Manipulation
Distinctive Ring Tones
Timers
Key Press Event
Called Party Inward Dialing Modification
Record Information
Advanced
Allow Out Subscriptions for Remote Digit Monitoring
No
Force Out Subscriptions for Remote Digit Monitoring
No


Basic
SDP Options
Signaling and Header Manipulation
Distinctive Ring Tones
Timers
Key Press Event
Called Party Inward Dialing Modification
Record Information
Advanced
Add Member
Delete Member
Index
Digits to Match
Digit Length Operator
Digit Length
Number of Digits to Absorb
Digits to be Inserted

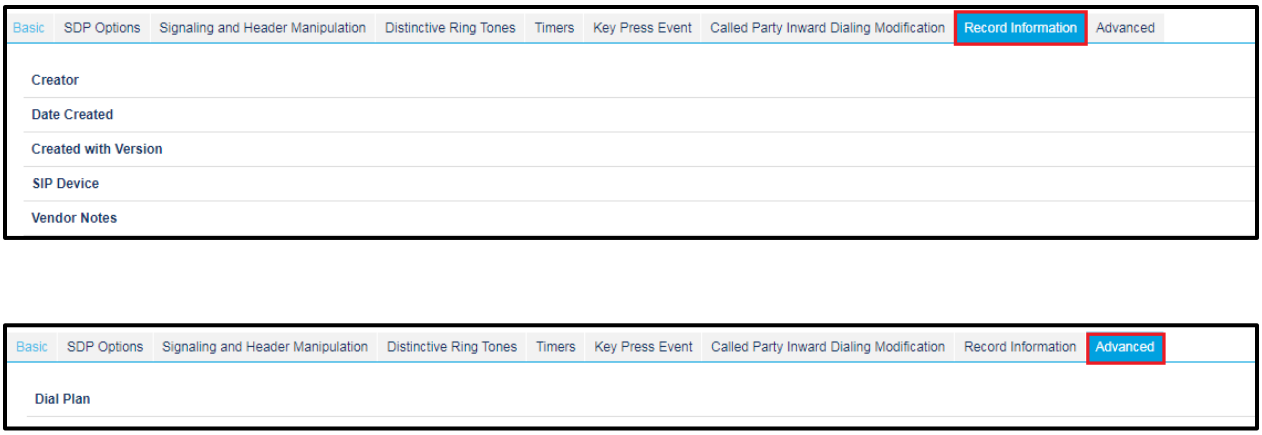


Figure 5 – SIP Device Capabilities

Station Attributes

Use the Station Attributes form to assign the previously configured Class of Service and SIP Device Capability number to each of the Algo 8301 Paging Adapter in the MiVoice Business. This form utilizes Range Programming.

Select the Algo 8301 Paging Adapter number then select Change. Enter the previously configured SIP Device Capability number (**31**) and Class of Service for Day, Night 1 & Night 2 (**31**).
See an example in **Figure 6** below.

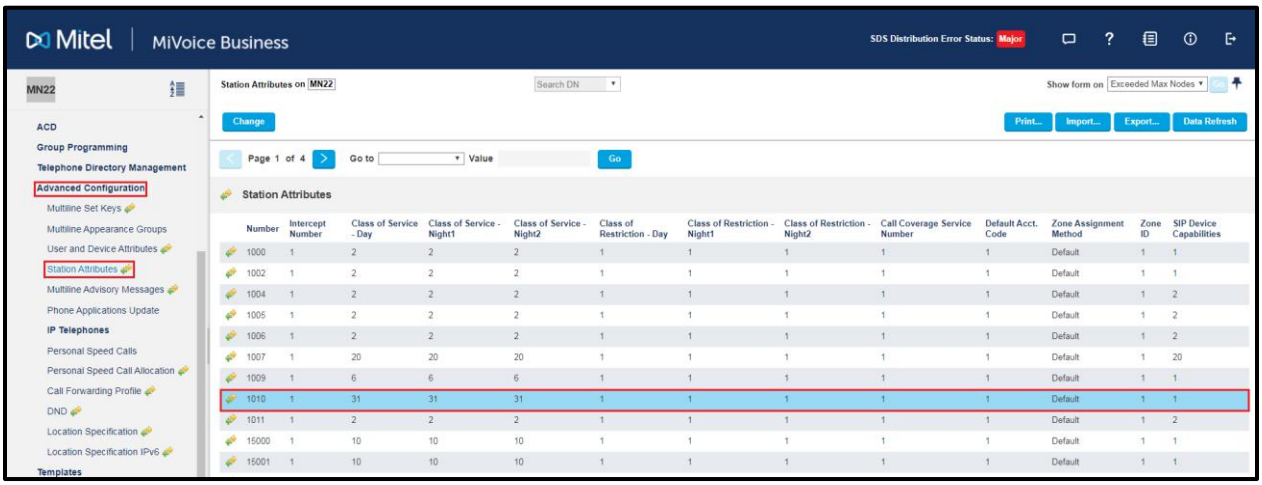


Figure 6 – Station Attributes

Enabling Voice Encryption in MIVB

Set Voice Encryption Enabled and Voice/Video SRTP Encryption Enabled to YES as in ***Figure 7***

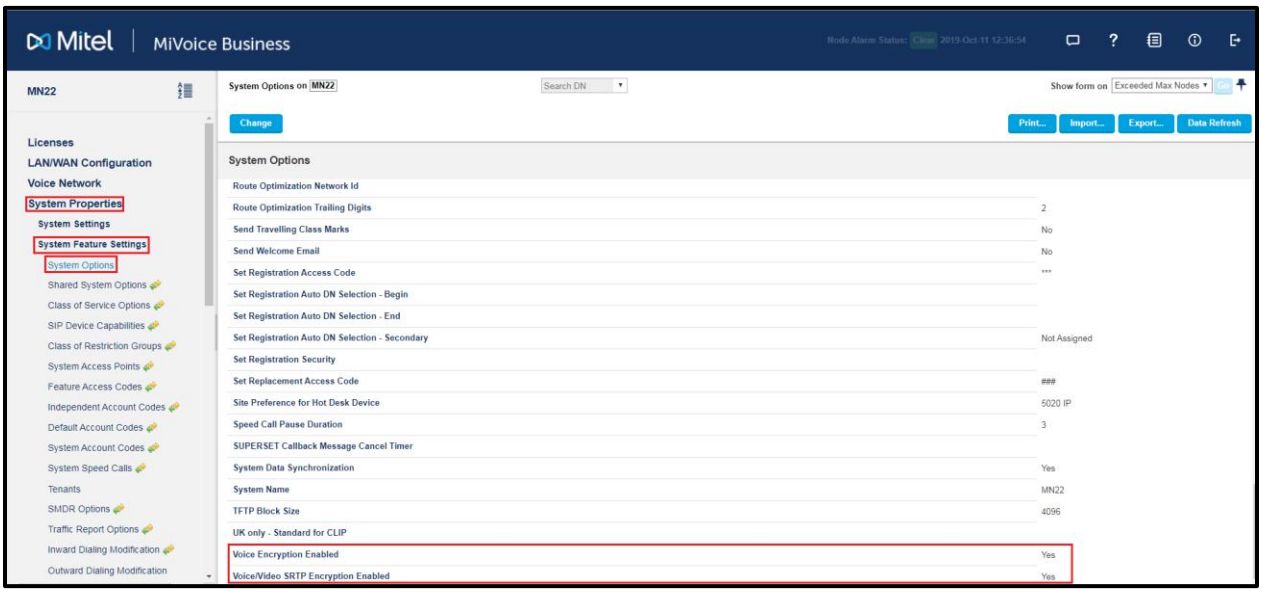


Figure 7 – System Options

Algo 8301 Paging Adapter Software Setup 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***.

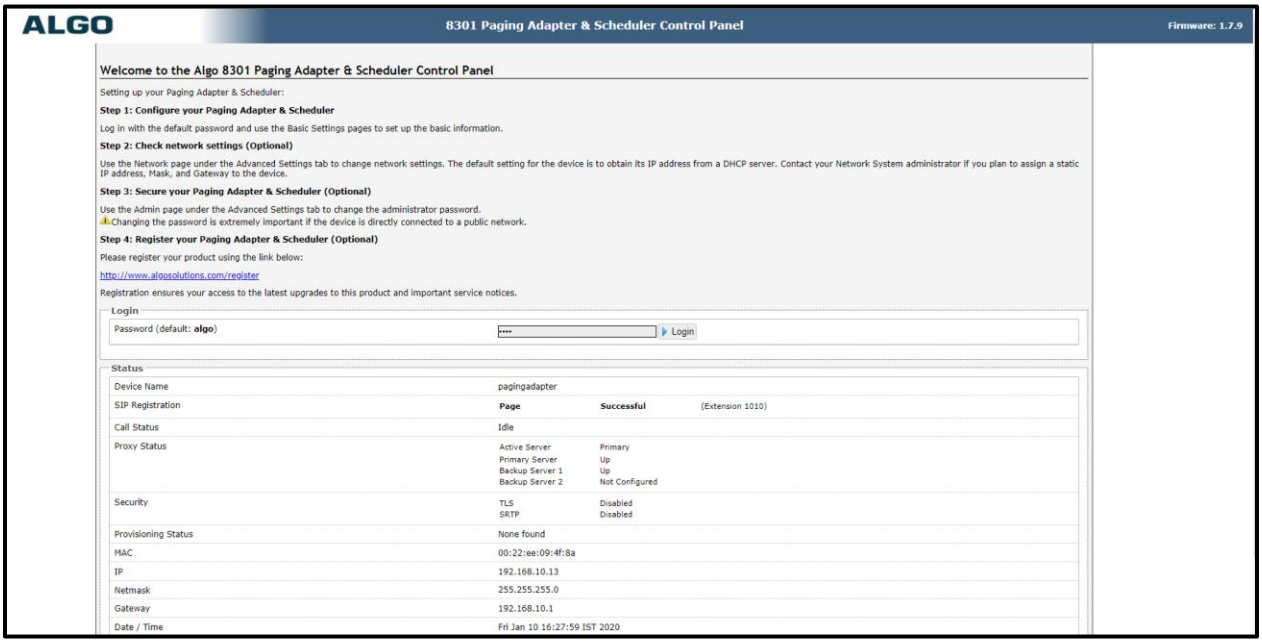


Figure 8 – Algo 8301 Paging Adapter Home page login

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.

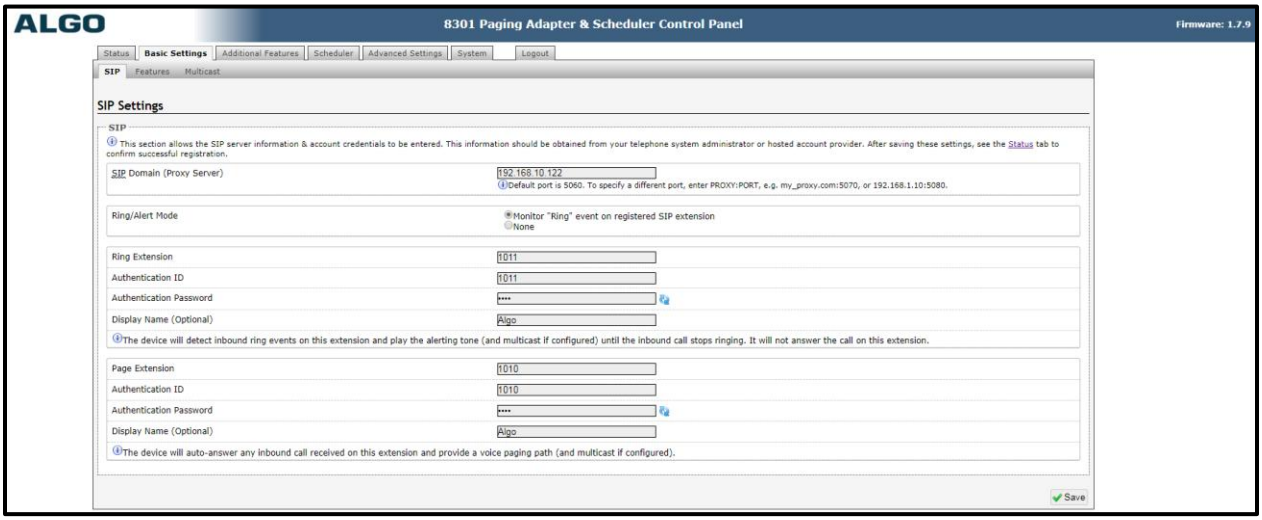


Figure 9 – SIP Settings

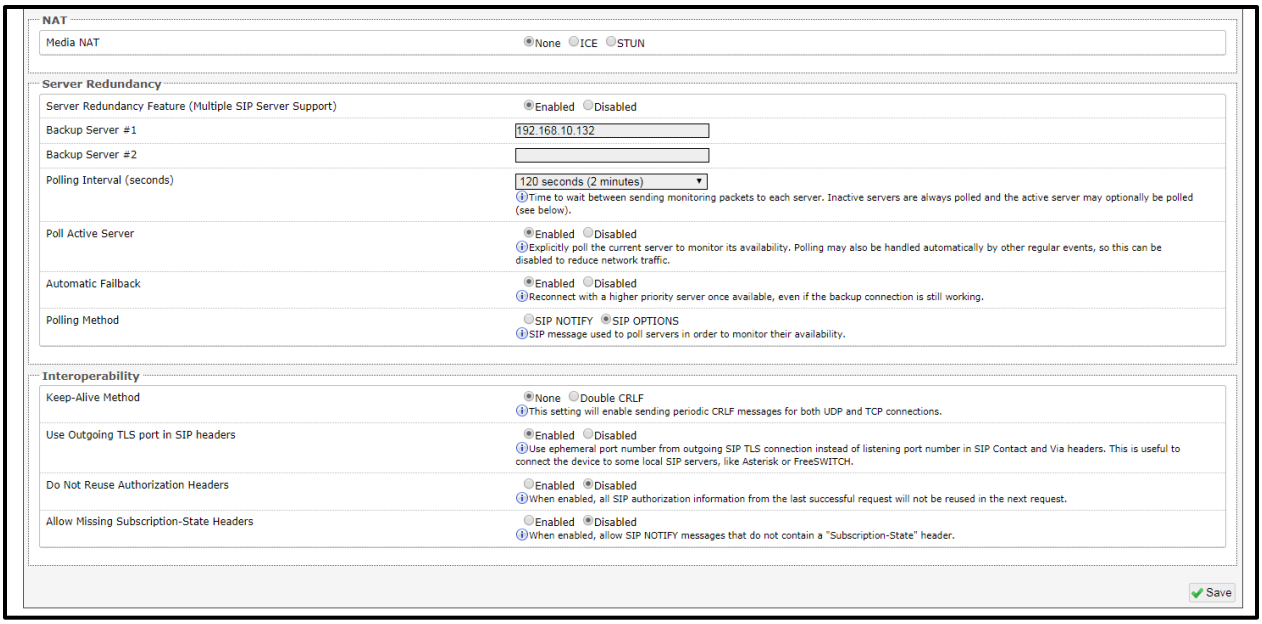


Figure 10 – Advanced SIP Settings

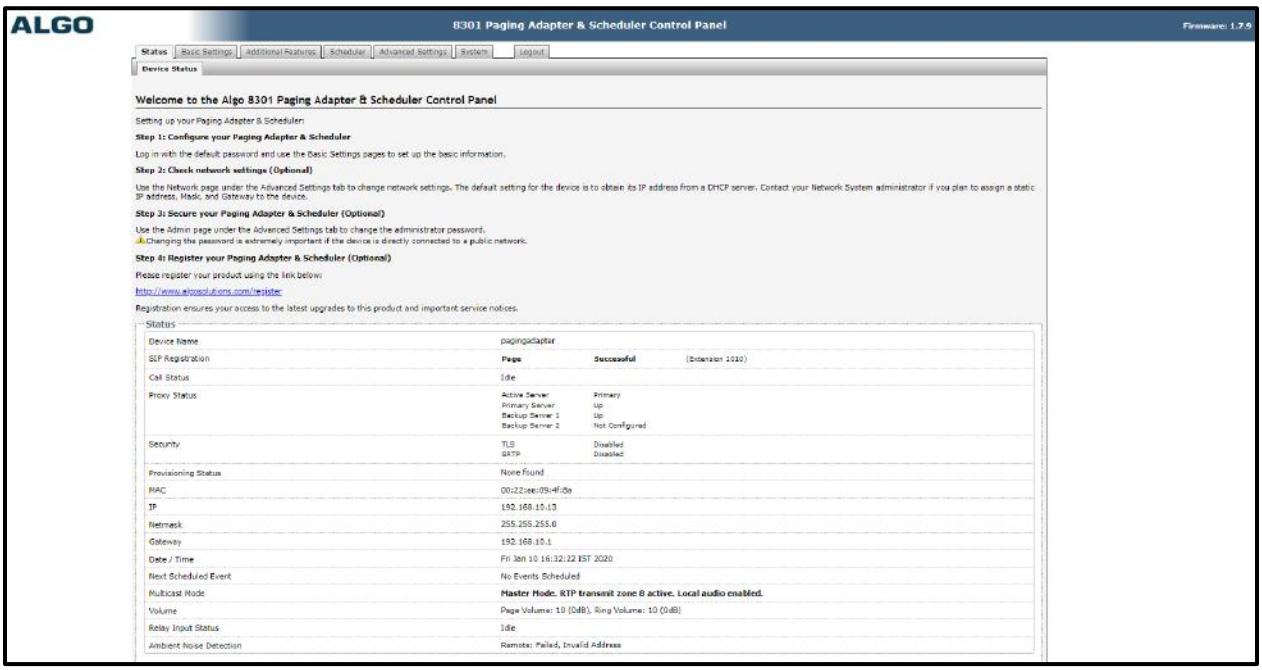


Figure 11 – Device status

Glossary

MiVoice Business	MiVB
MiCollab	MiCollab
MiNET Interface	MiNET
Mitel Solutions Alliance	MSA
Knowledge Management System	KMS
Class of Service	COS
Automatic Route Selection	ARS