SCOPTEL IP PBX Software - Managing Incoming Lines

Contents

Incoming Lines

Background Information

Incoming Lines types are typically:

- **“Extension (DNIS)” which are received numbers from SIP/IAX2 or PRI trunks. “Block” (a configured list of DNIS numbers).**
  - **DNIS (Dialed Number Information Service). The service is provided by the TELCO and refers to the Called Party Number.**
    - **On ISDN PRI trunks the number is received in the Q.931 SETUP on the D channel.**

      PRI Span: 1 < Message Type: SETUP (5)
      PRI Span: 1 < Called Party Number (len= 7) [Ext: 1 TON: Subscriber Number (4) NPI: ISDN/Telephony Numbering Plan (E.164/E.163) (1) ‘0312’]
      Executing [0312@zap-incoming:5] Set(“DAHDI/i1/5796301651-d0f0”, “__INCOMING_DNIS=0312”) in new stack

    - **On SIP trunks the DNIS is derived by default from the SIP INVITE and in some fringe cases To Header routing must be enabled.**

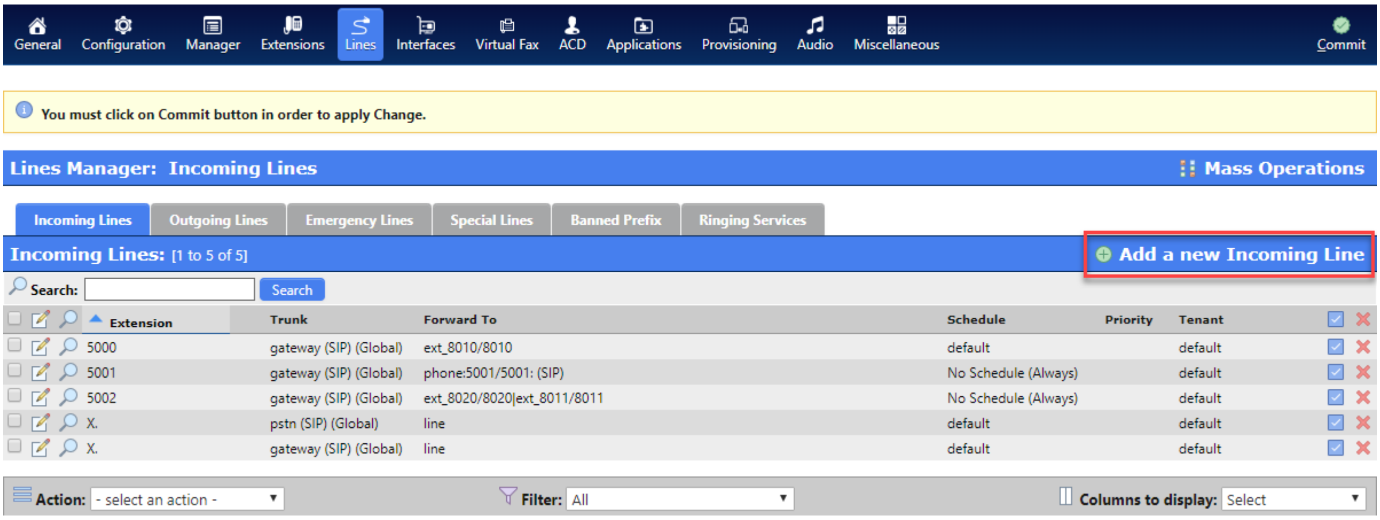
      INVITE sip:211@192.168.192.88;user=phone SIP/2.0
      Via: SIP/2.0/UDP 192.168.192.78:5060;branch=z9hG4bK6d5aafcc
      Max-Forwards: 70
      From: “Extension 8010” <sip:5555558011@192.168.192.78>;tag=as1ce199ae
      To: sip:211@192.168.192.88;user=phone

Executing [211@all-gateway-incoming:5] Set(“SIP/gateway-00000086”, “__INCOMING_DNIS=211”) in new stack

- “Port (TDM)” which are analog FXO ports supported by Sangomaor Digiumcards.

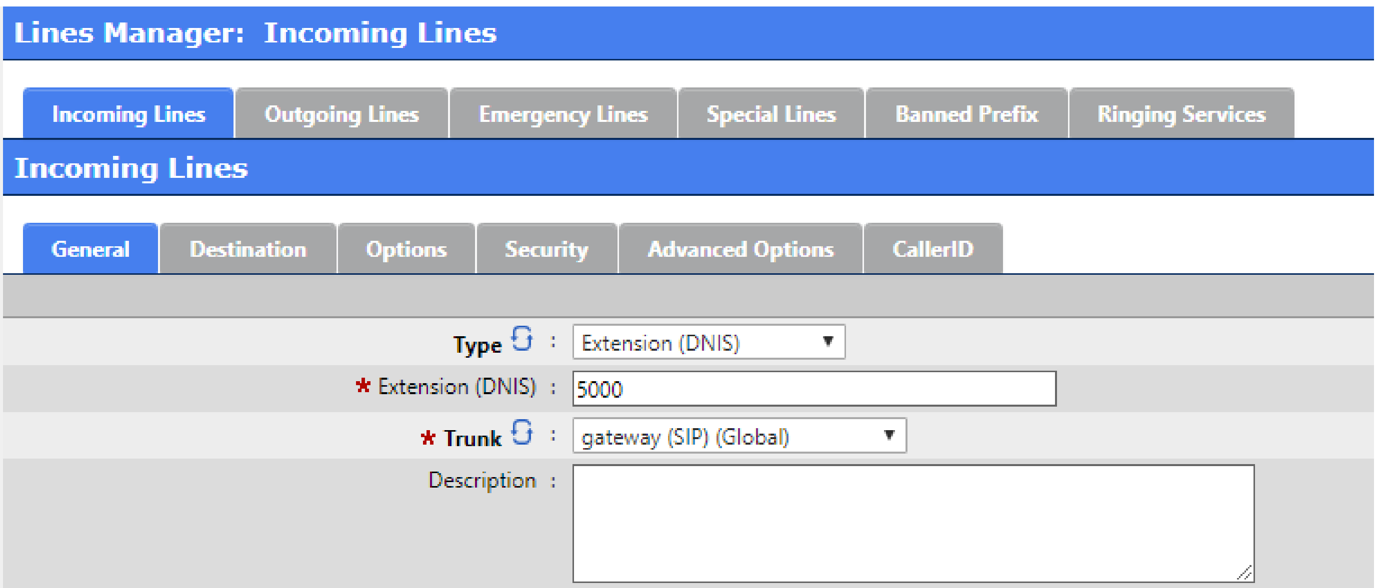
Add a new Incoming Line

- Incoming Lines must be created to Route incoming calls to required destinations.
- From Configuration > Telephony > Lines > Incoming Lines.
- Click on “Add a new Incoming Line”.



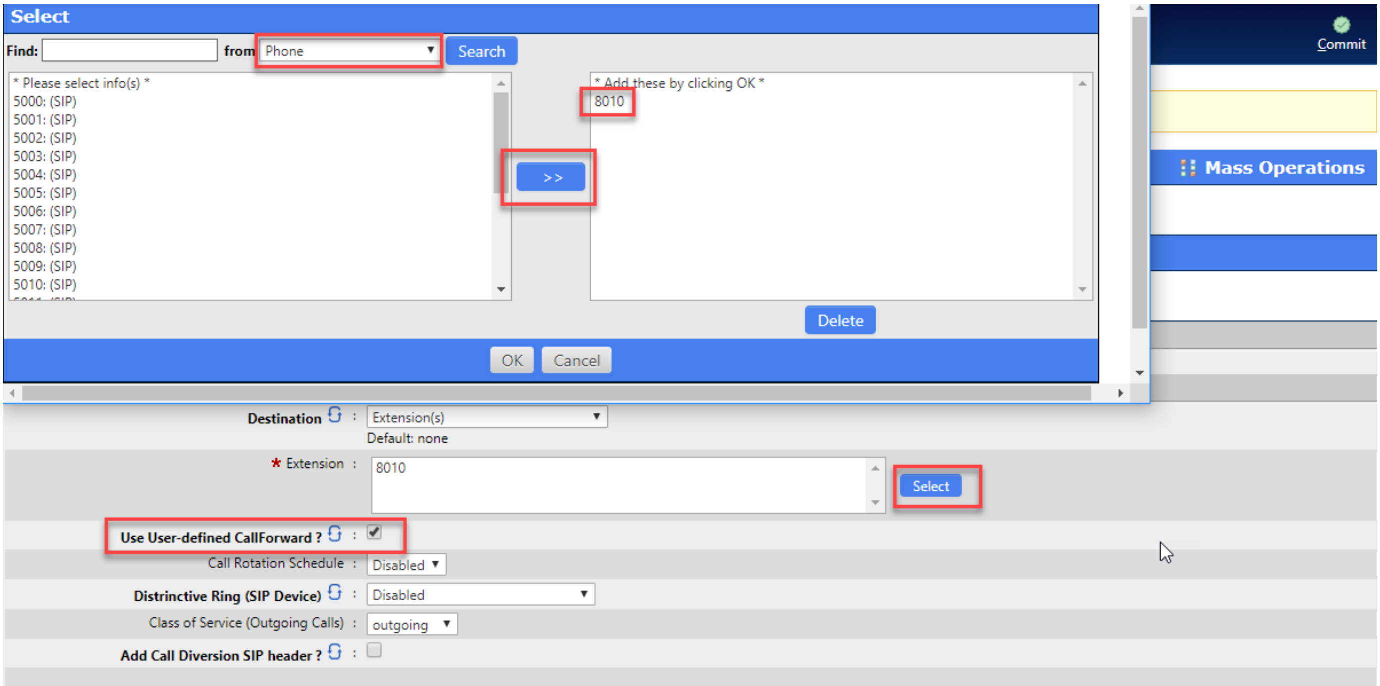
General

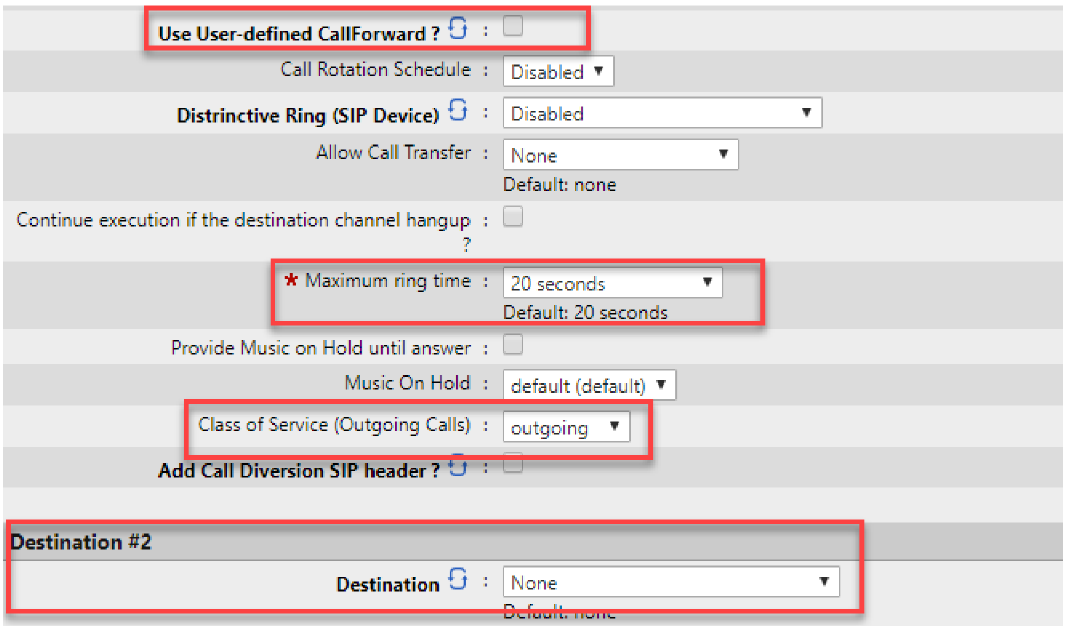
- This is an example of SIP trunk DNIS configuration
- The Extension (DNIS) field is configured to match 5000 patterns are also supported like: 5XXX
- There is fixed length requirement to the DNIS field, DNIS matches are matched from right to left
- Outgoing Lines, Extensions, Applications, Incoming Lines are unique objects so there is no conflict when there are matching prefix patterns. Example: if an extension’s leading digit is a 9 and the Outgoing Line assigned in the CoS leading digit is a 9 and the DNIS leading digit is also a 9 there is no dial plan conflict.
- The trunk is selected from the drop list, in this example ‘gateway (SIP) (Global)’
- Once the DNIS and Trunk(s) are configured click on the Destination tab



Destination

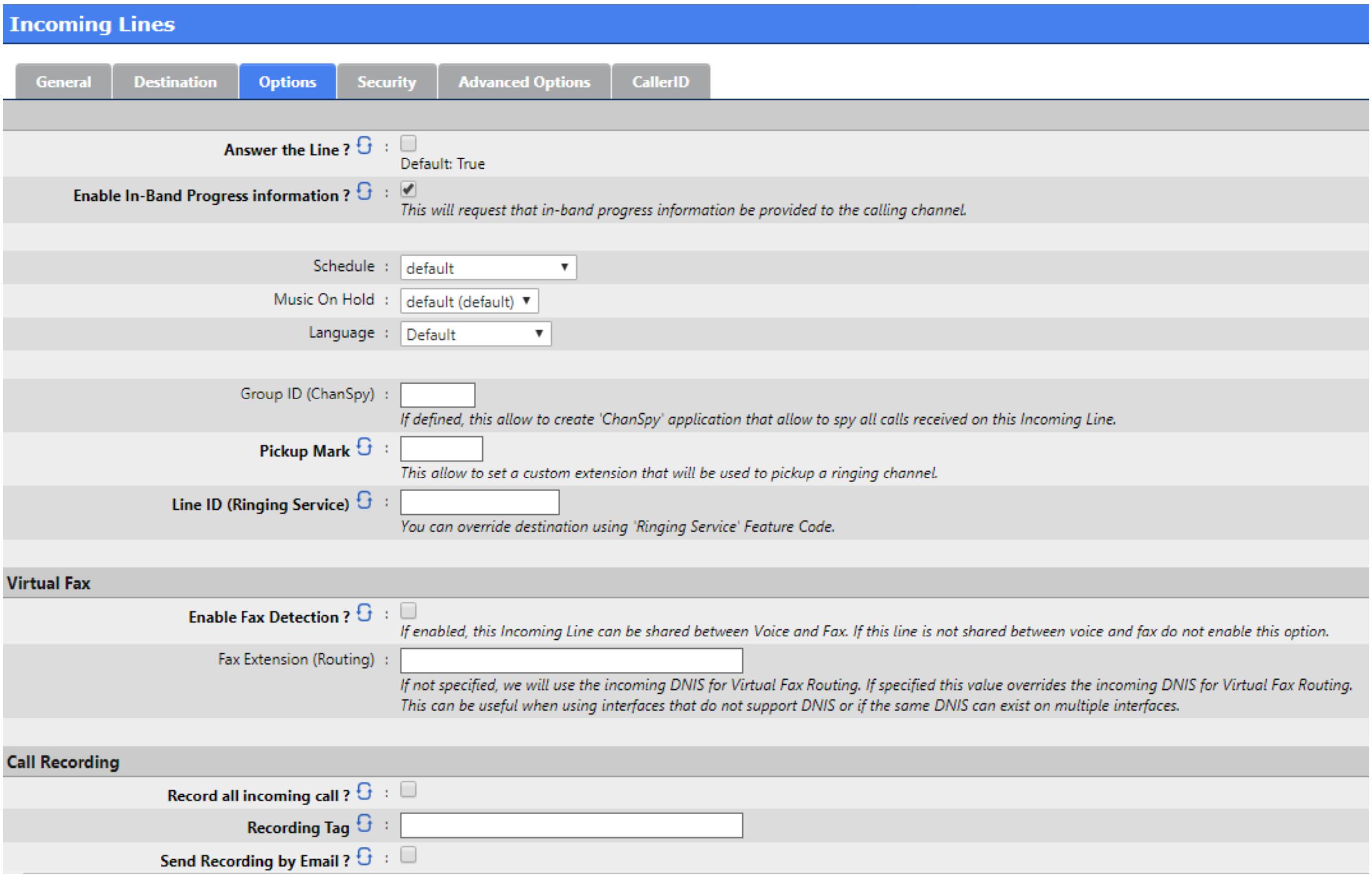
- Use the drop list to select a Destination type in this example Extension(s)
- Use the Select button to choose one or more Extensions from using the Select tool.
- To choose Internal Extensions Use the Phone drop list selector and to enter an External Number choose that option from the drop list
- In this example Phone is selected and Extension 8010 is added from the left column to the right
- The ‘Use User-defined CallForward’ option is enabled in this example because we want the User Options in Extension 8010 to be applied to this Incoming Line. If special considerations are required then do not enable this option.
- If the ‘Use User-defined CallForward’ option is disabled you must configure additional CoS details and configure Destination 2 which is invoked after the ‘Maximum ring time’ value in Destination 1. If Destination 2 is not defined the Maximum ring time will terminate with a hangup.
- Click on the Options tab when done





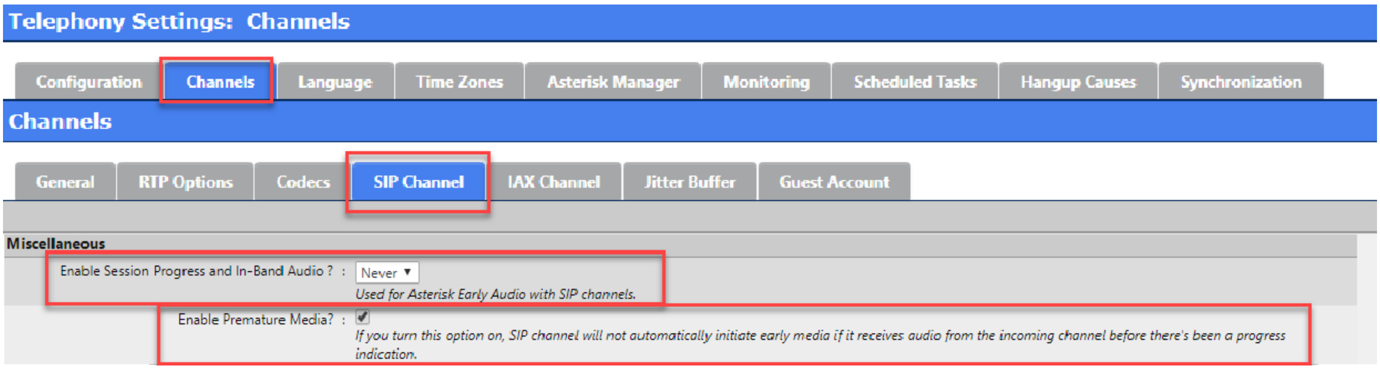
Options

- There are many Incoming Line options and advanced usage is dependent on the Carrier and Trunk type .
- In this example In - Band Progress messages are enabled on the SIP trunk which is very common among SIP ITSP's . Before this option can function there are SIP Channel pre - requisites to configure
- Virtual Fax and Call Recording Options can be configured if needed.
- Click on the Security tab



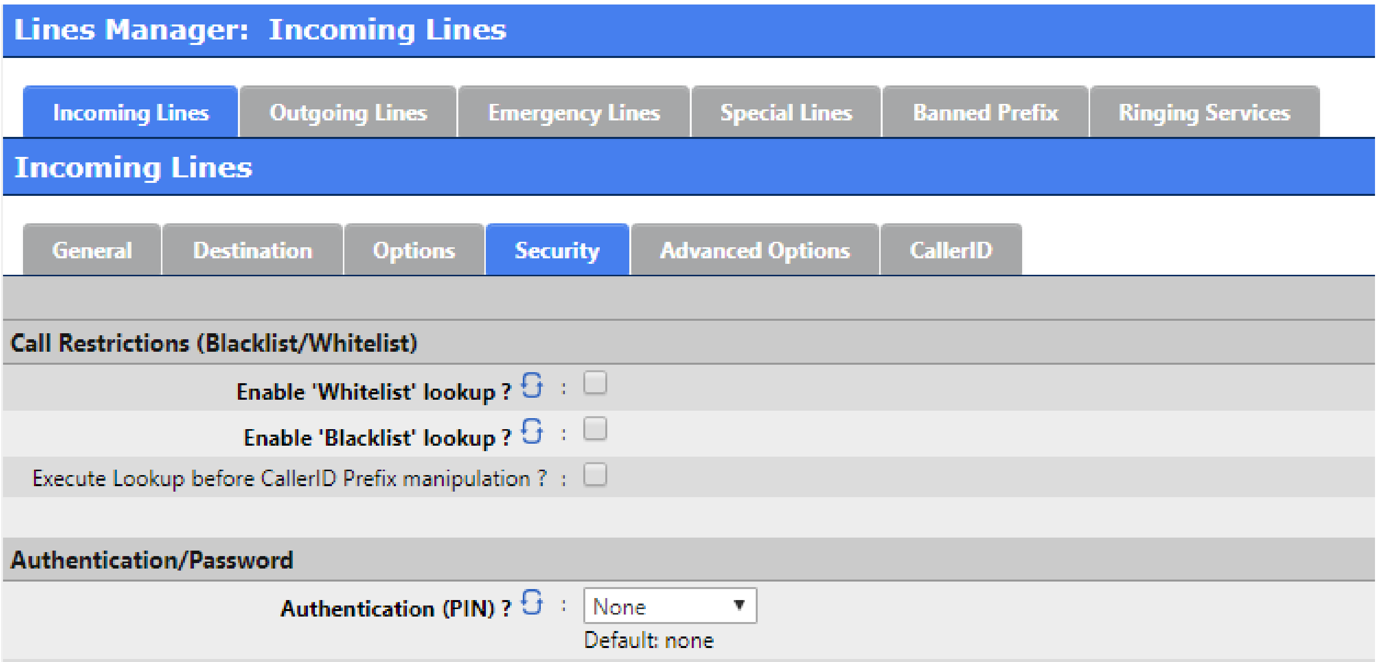
SIP In-Band Progress Pre-requisites

- Enable Session Progress and In - Band Audio must be set to Never
- Enable Premature Media must be enabled
- These are Global options and will effect all SIP VoIP Interfaces.



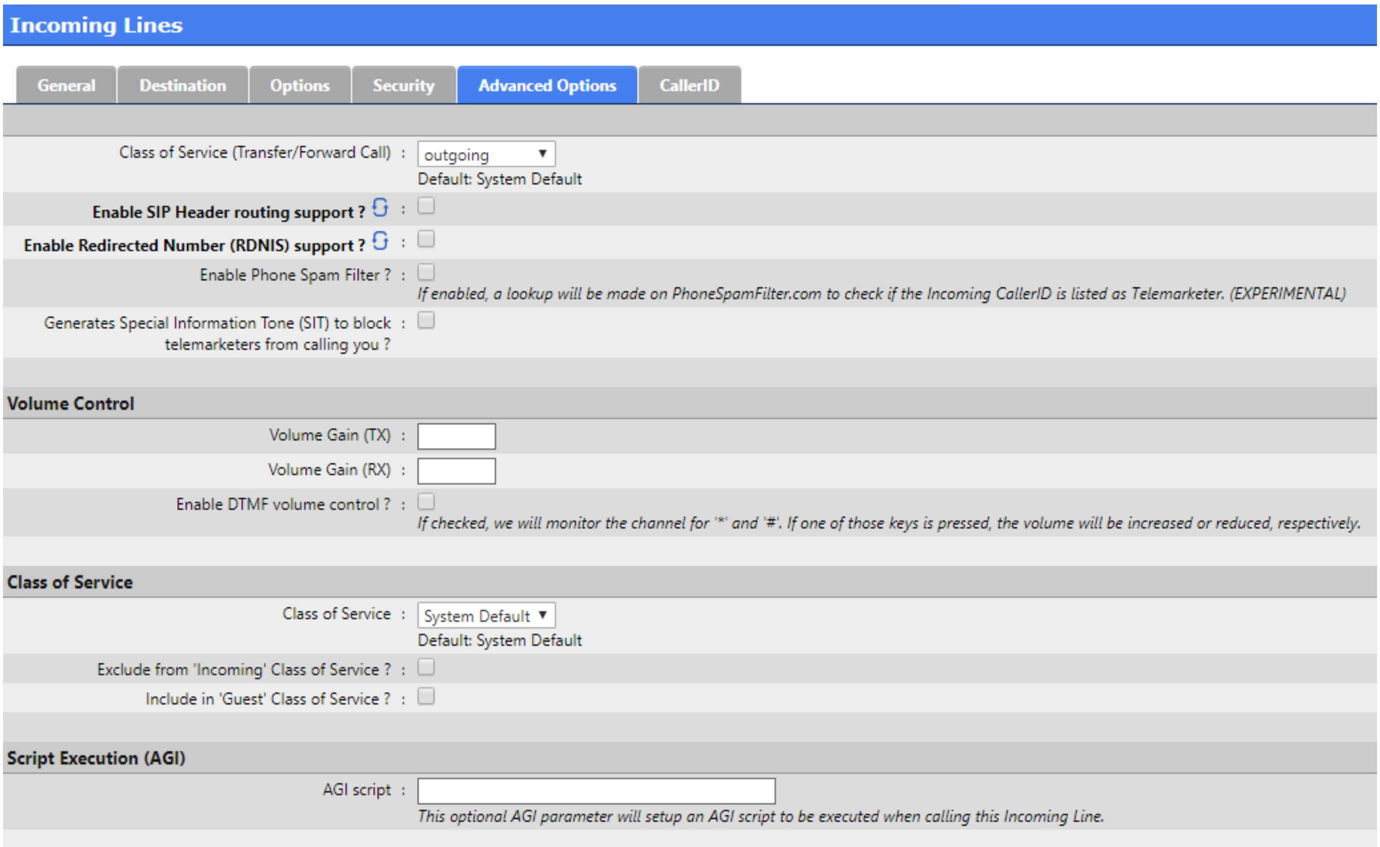
Security

- Advanced Security options may be optional configured
- Click on Advanced Options



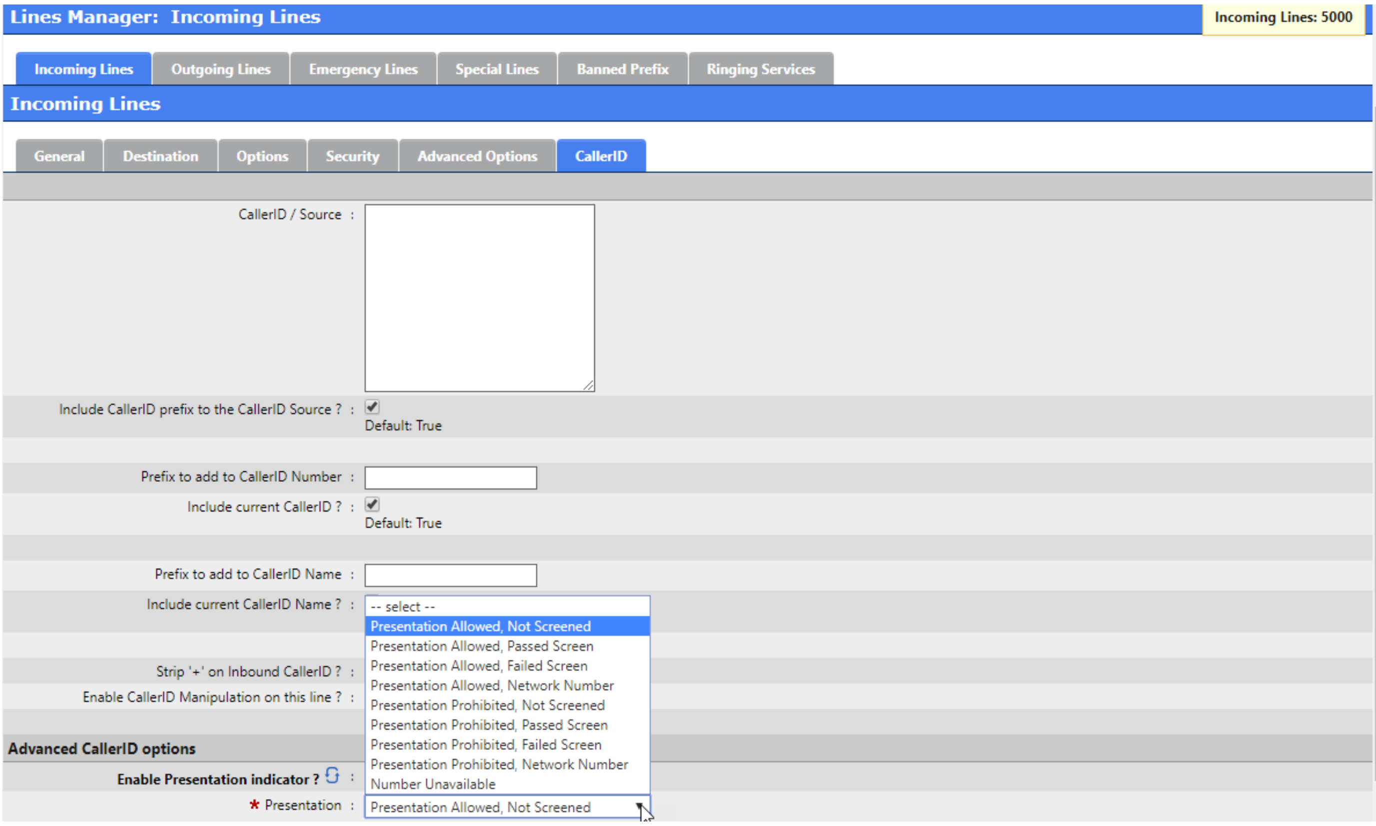
Advanced Options

- Class of Service (Transfer/Forward Call) will apply specific Class of Service security considerations to any call which is transferred or forwarded by an Extension or Auto Attendant after being answered by this Incoming Line
- Class of Service selection should be normally be left on System Default which does not expose the Incoming Line to any advanced Feature Codes or Outgoing Lines assigned to another Class of Service. The System Default matches the Source Interface's CoS only with assigned Incoming Lines and any Incoming Call will be rejected if there is no matching Incoming Line object. Using another Class of Service should only be considered in rare use cases.
- Click on CallerIDwhen done



CallerID

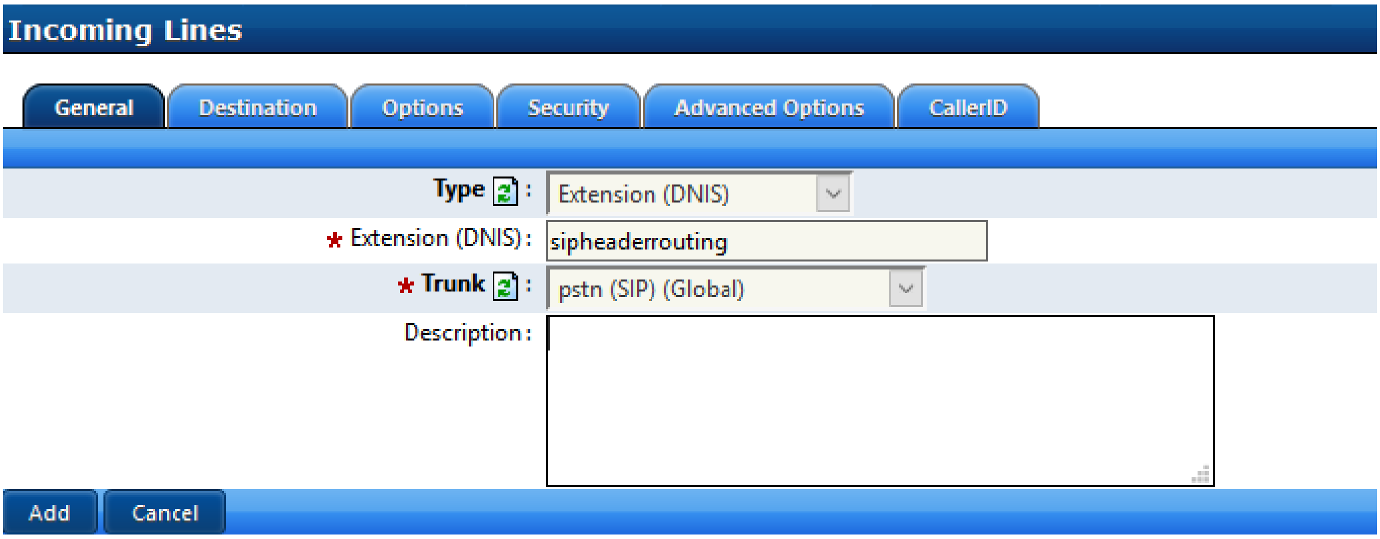
- CallerID/Source text box allows multiple CallerIDfilters to be associated with this Incoming Line which can be used for specialized routing.
- A numeric prefix can be added to the incoming call display which will be passed to the ringing phone. This can be useful if some additional dialing prefix is required to initiate an Outgoing call from the phone's CallerIDhistory
- A Name prefix can be added to the incoming call display of the ringing phone. This can be useful to distinguish inbound calls for each customer and answer the phone with the correct greeting response.
- Advanced CallerID options can be selected to comply with https://tools.ietf.org/html/rfc3325
- Click on Add when done

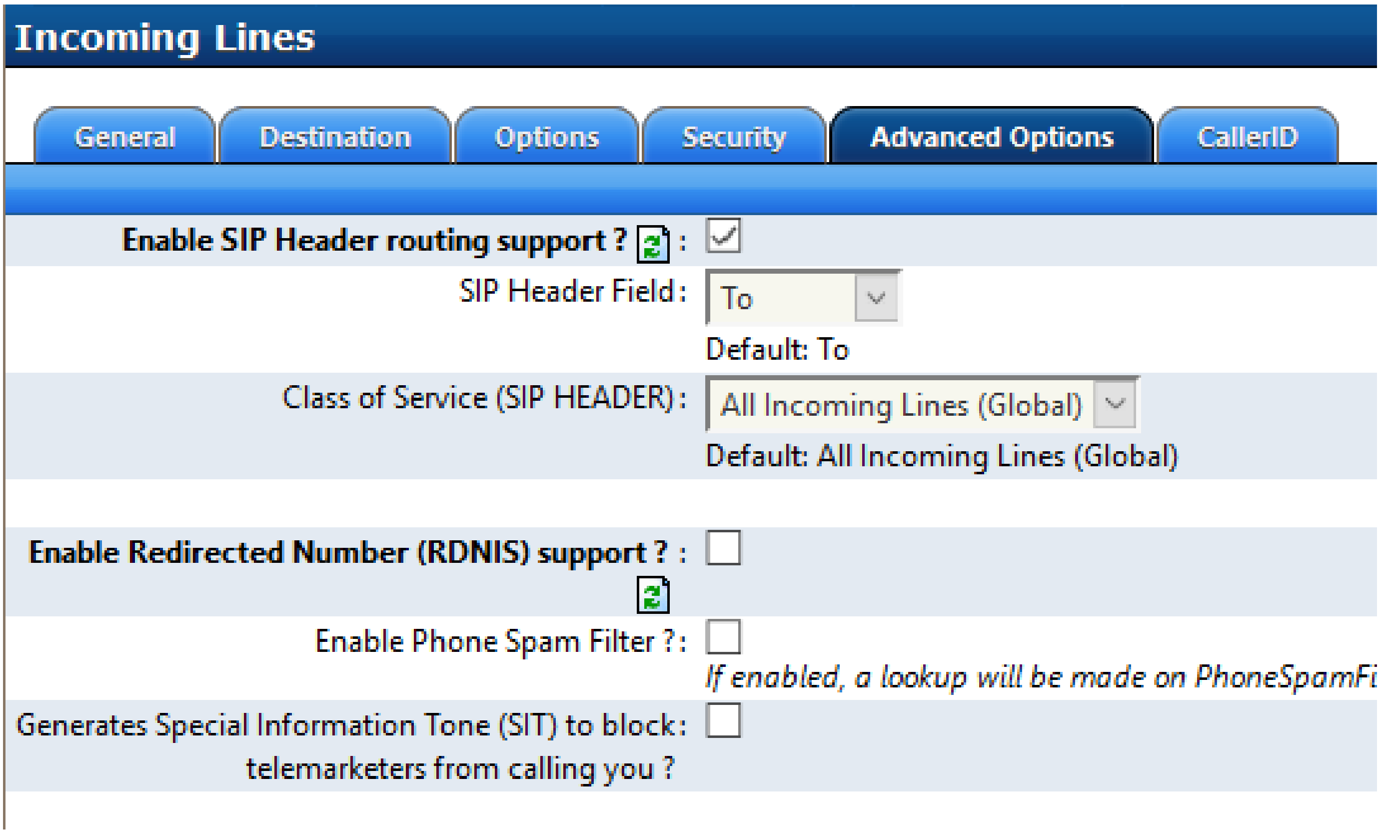


SIP To Header Routing

- In some cases the SIP Carrier will require you to route DNIS based on the SIP To Header
- The INVITE will look something like this :
    - [2016-12-0116:36:41]INVITEsip:sipheaderrouting@1.1.1.1:5060SIP/2.0
    - [2016-12-0116:36:41]Via:SIP/2.0/UDP1.1.1.1:5060;branch=z9hG4bKie9fu1207o25grh3mop0.1
    - [2016-12-0116:36:41]From:<sip:9055551234@youritsp.com;user=phone>;tag=150378187-1480628201110-
    - [2016-12-0116:36:41]To:"CompanyABC"<sip:4165551234@youritsp.com>
- Follow these steps to enable support for SIP To Header Routing

1. Create a New Incoming Line using the VoIP Interface you created for your ITSP
2. In the Extension (DNIS) field enter the text ‘sipheaderrouting’
3. Click on the Advanced Options tab
4. Check the enable option for ‘Enable SIP Header routing support? [x]
5. Choose the SIP Header Field drop selection = ‘To’
6. Choose the Class of Service (SIP HEADER): drop selection to ‘All Incoming Lines (Global)’. NOTE that by choosing this option all Incoming Lines and their Destinations will now use SIP To Header Routing for the chosen VoIP Interface/Trunk.
7. Add this Incoming Line
8. Commit





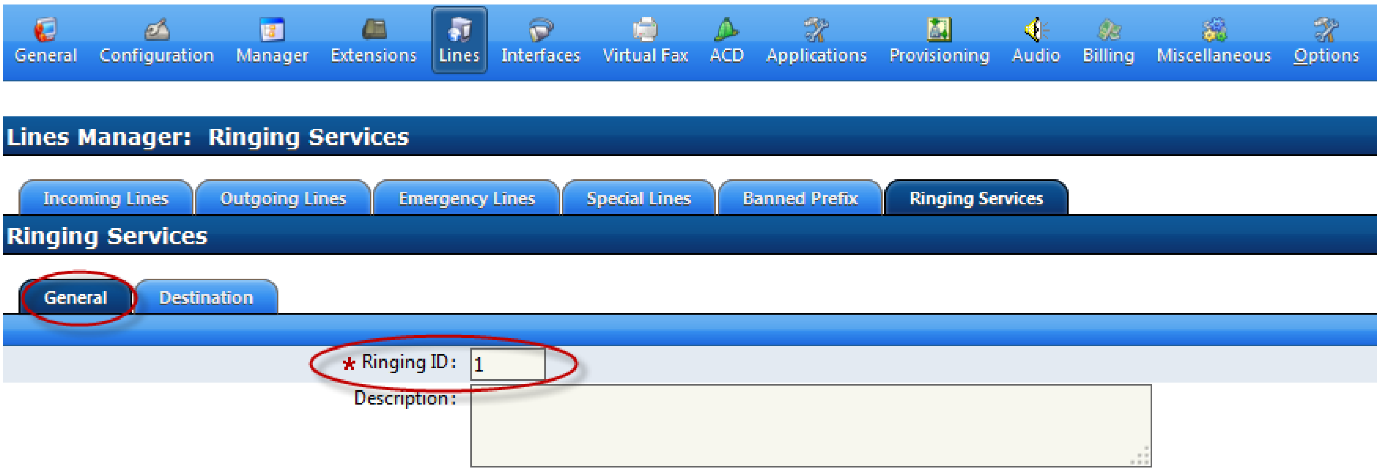
Ringing Services

Sometimes it is necessary to override a schedule or incoming line rule and route incoming calls to a preset destination. This is often referred to as Night Service but has many uses.

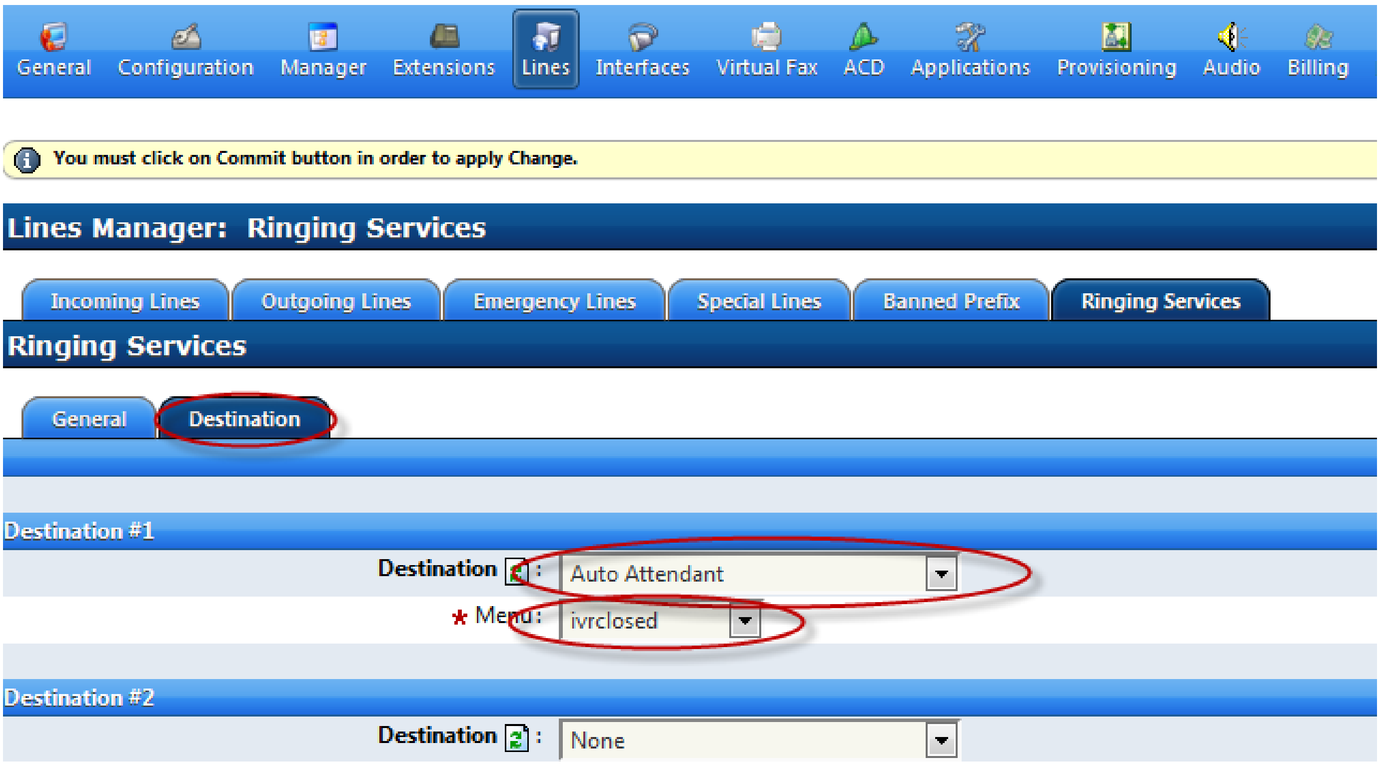
Examples:

- Receptionist normally answers calls but wants to forward to someone else so they can take a break or change a shift.
- An emergency situation requires that lines are forwarded to an emergency greeting or business closed greeting or routed to an external phone number such as an answering service.
- The system uses an automated schedule but needs to close early and enable an IVR menu.
- The system does not use automated schedules an wants to manually toggle routes during the day between open and closed hours. Operator during open hours Vs. IVR menu during closed hours.
- SCOPSERV does not put any restrictions on the number of Ringing Services you can create.

- The first step is to add a new Ringing Service
- Lines>Ringing Services>Add a new Ringing Service
- Give each Ringing Service a unique numerical Ringing ID value
- Then click on the Destination tab



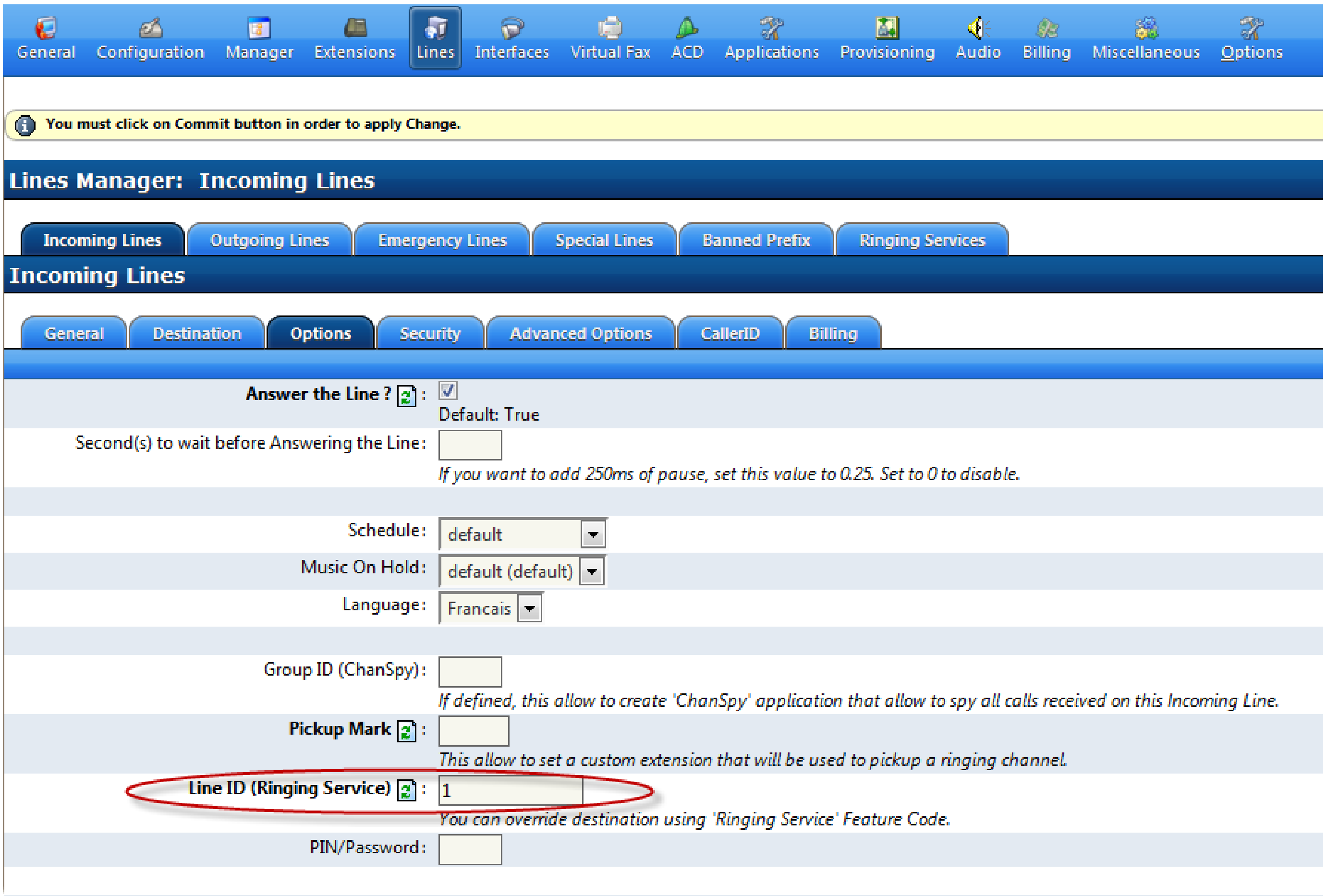
- Choose your preferred destination type which in this example is an after hours IVR menu.
- In this example the Menu is an existing IVR Menu called ivrclosed.
- Click on Add when your are done.



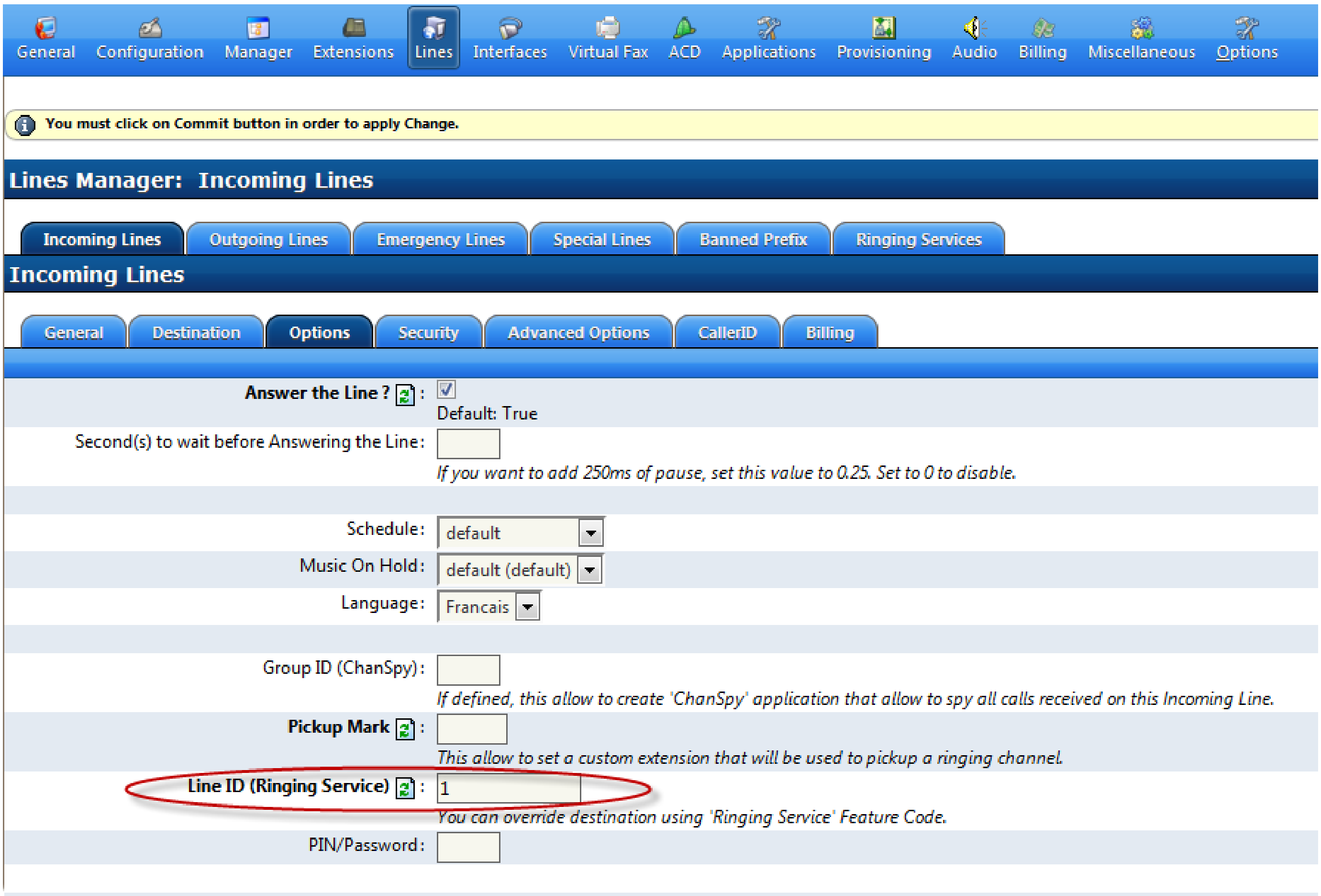
- Incoming Lines>Edit or Add a new Incoming Line.

- In this example we are editing an existing Incoming Line.
- Click the Edit button and navigate to the Options tab.

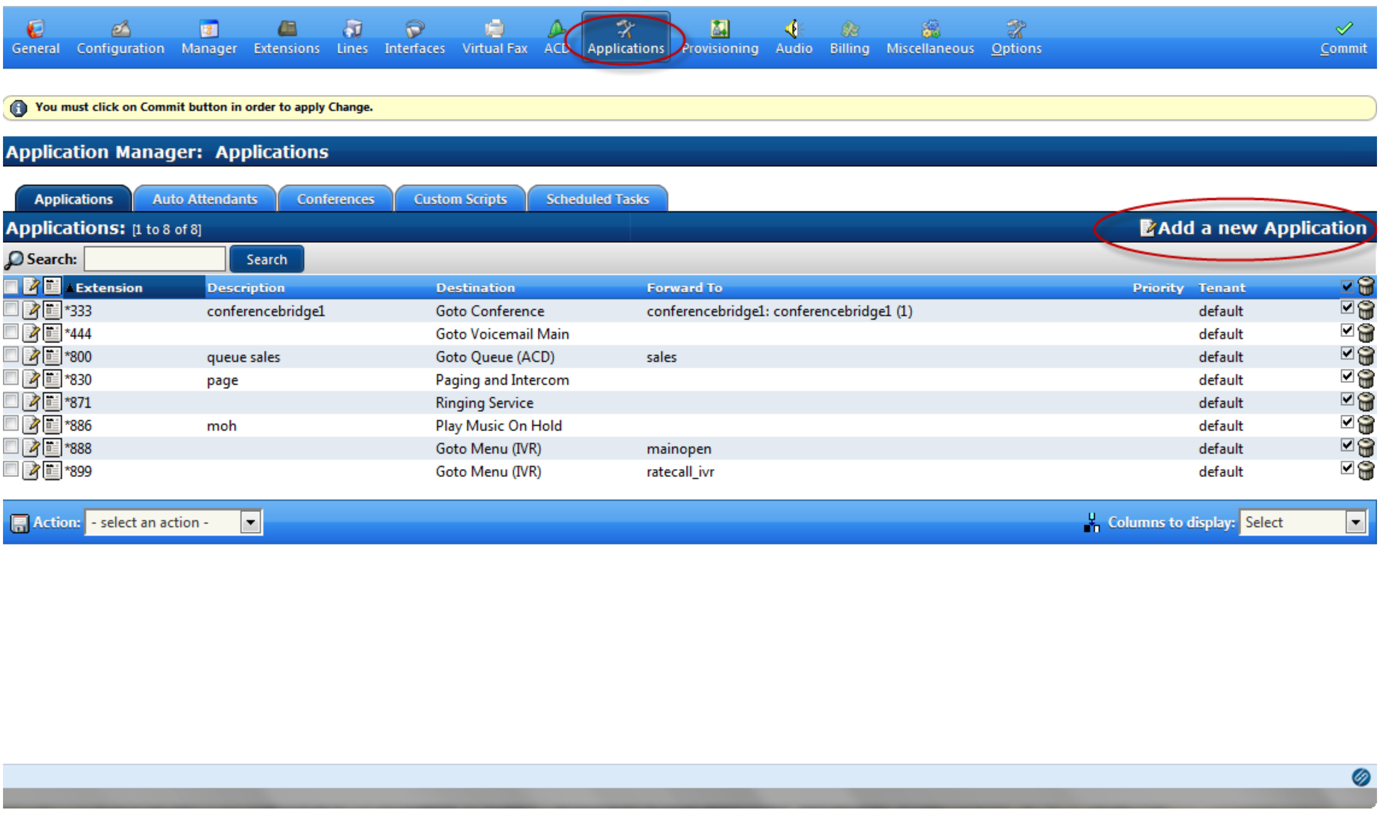
- Put the Ringing Service ID value you defined on the Ringing Service object earlier into the Line ID (Ringing Service) text field.
- Save or Add the new Incoming Line.
- The PIN/Password is optional but recommended to control user access.



- Put the Ringing Service ID value you defined on the Ringing Service object earlier into the Line ID (Ringing Service) text field.
- Save or Add the new Incoming Line.

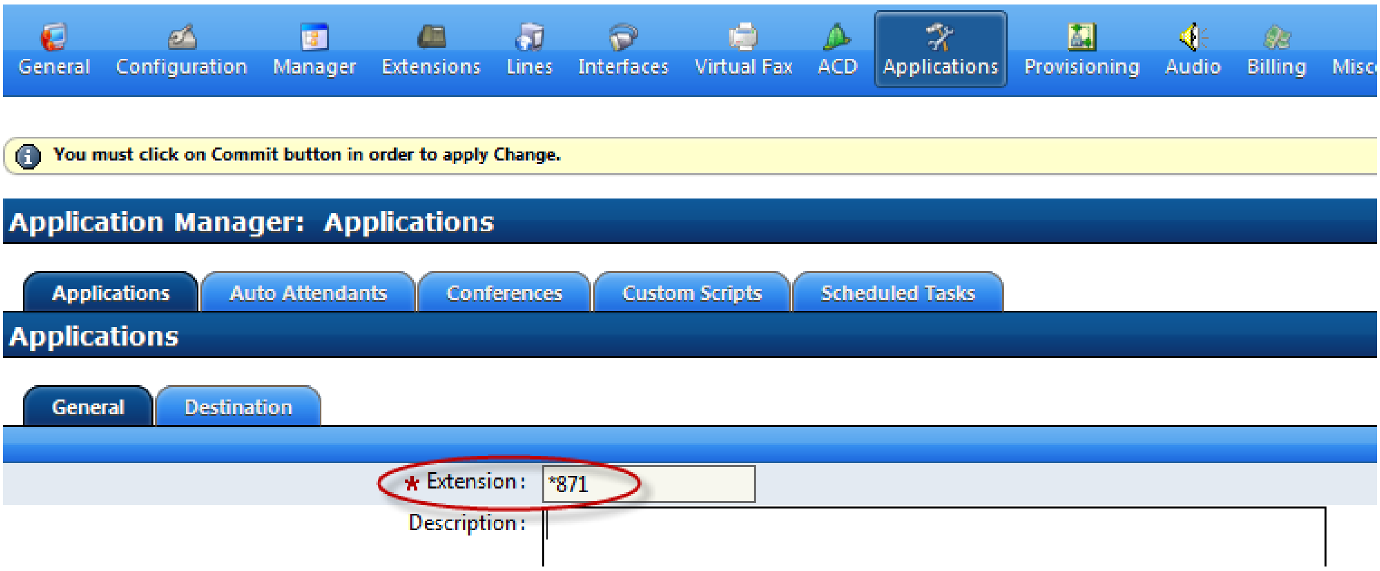


- Navigate to the Applications section.
- Click on Add a new Application.

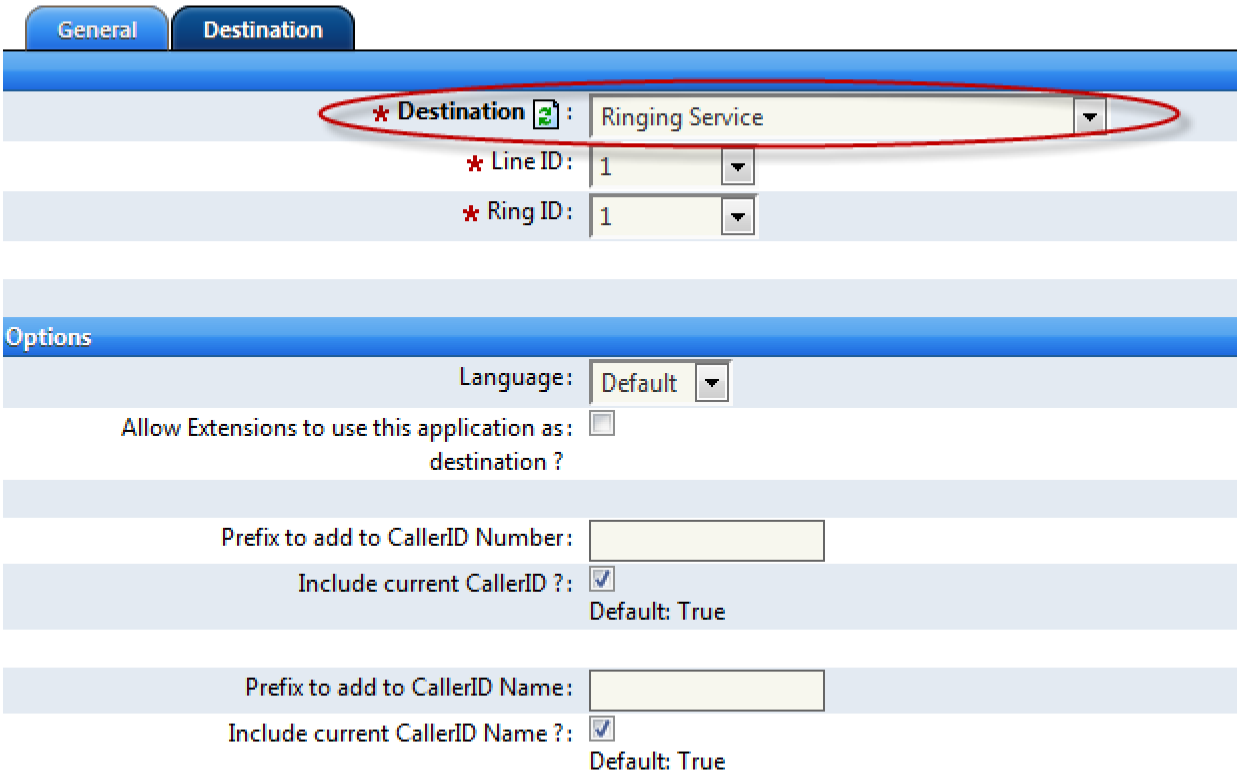


- Give this Application a number you would like to use as a feature code and make sure this code does not conflict with any other extension number, Feature Code or Outgoing Line string.
- This 3 digit code starting with *87 dedicates *87 for Ringing Services codes and allows Ringing Groups 1-9 to be controlled.

- Click on the Destination tab.



- Choose Ringing Service from the Destination pull down.
- Enter the Line ID value you specified earlier on the Incoming Line Manager.
- Enter the Ring ID to the Ringing Service you specified on the Ringing Services Manager.
- These settings allow you to control Multiple Incoming Lines and override the Incoming Line Destination with the Ring ID you define in this section.
- Click Add when done.



- Choose Ringing Service from the Destination pull down.
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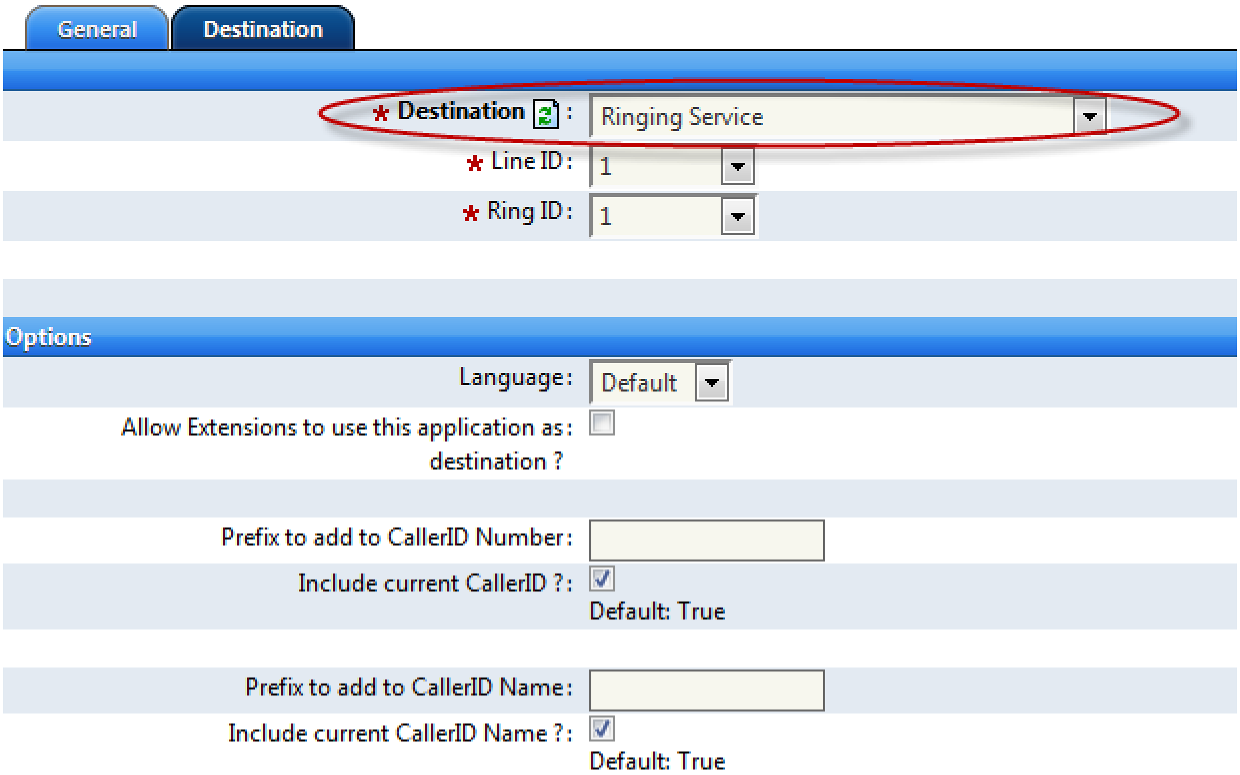
- Enter the Ring ID to the Ringing Service you specified on the Ringing Services Manager.
- These settings allow you to control Multiple Incoming Lines and override the Incoming Line Destination with the Ring ID you define in this section.
- Click Add when done.
- Commit changes to activate the feature.



IMPORTANT

Ensure you allow this Ringing Service in the Class of Service you assign to the extension(s) used to dial the code.

USAGE

- Dial the code as in example *871 once to enable the Ringing Service and override the default schedule or configured schedules.
- Dial the code again to disable the Ringing Service and allow the default schedule or configured schedules to control Incoming Line routes.

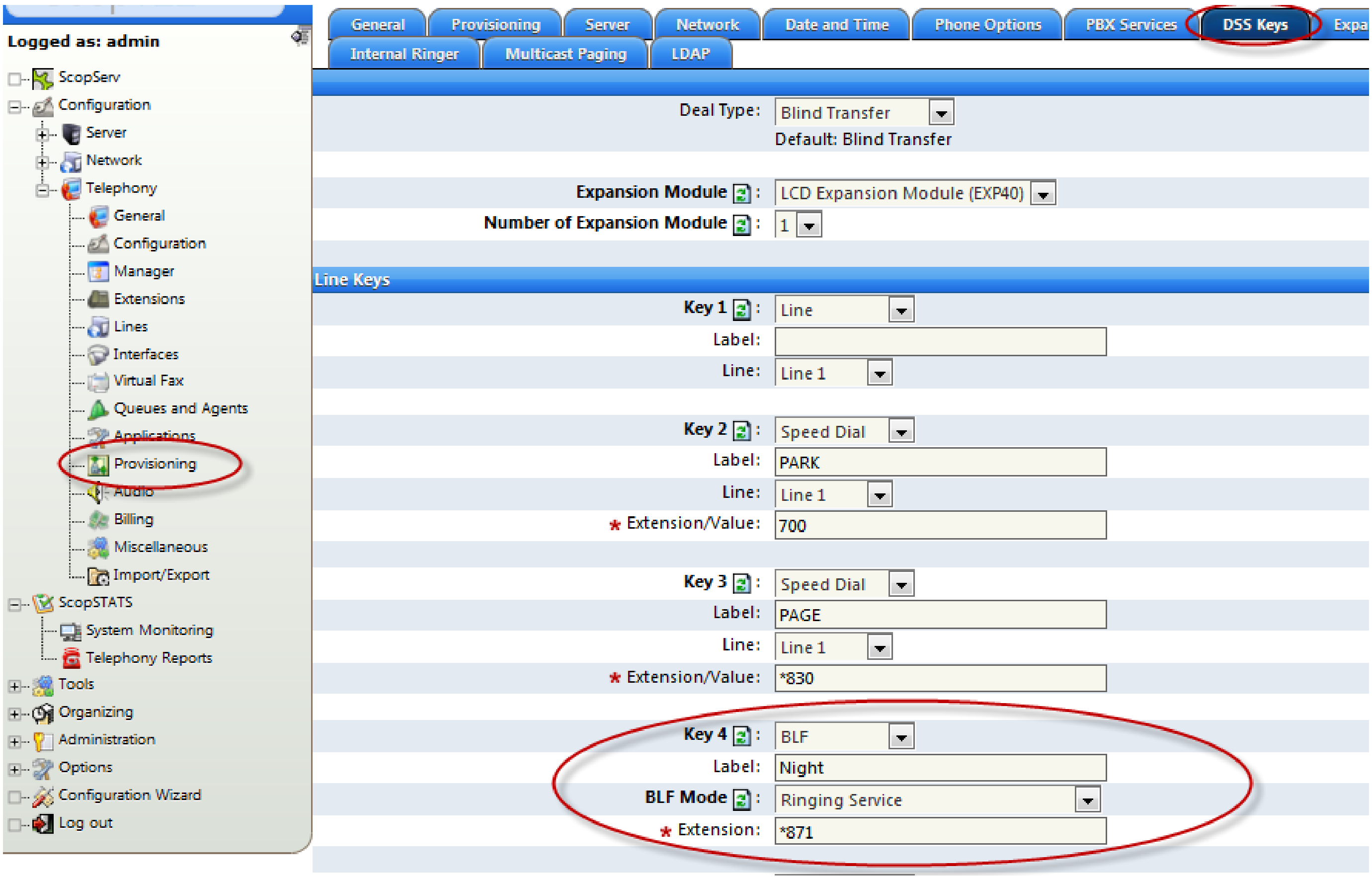
NOTES

- Some phones will allow you to configure a BLF DSS Key to monitor the on off status of the Ringing Service.
- The BLF value would be in the format ringservice_<tenant>_<application>. Example: ringservice_default_*871.

Ringing Services BLF

Ringing Services BLF on Yealink DSS Key

- It is very easy to set up a Ringing Service Key on a Yealink phone.
- Using the previous steps to build an application *871 to enable Ringing Service ID 1.
- Navigate to Provisioning>Yealink>DSS Keys and configure as in this example.
- Once the phone is rebooted you can use the DSS Key to enable and disable the Ringing Service.
- The BLF key will light **Red** when in Use.



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Extension: *871