

SCOPTTEL IP PBX Software - Managing Incoming Lines

Contents

SCOPTTEL IP PBX Software - Managing Incoming Lines	1
Incoming Lines	1
Background Information	1
Add a new Incoming Line	2
General	2
Destination	3
Options	3
SIP In-Band Progress Pre-requisites	4
Security	4
Advanced Options	5
CallerID	5
SIP To Header Routing	6
Ringing Services	7
IMPORTANT	12
USAGE	12
NOTES	12
Ringing Services BLF	12

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]
“_INCOMING_DNIS=211”) in new stack

Set(“SIP/gateway-00000086”,

- “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”.

The screenshot shows the Asterisk Management GUI. At the top, there is a navigation bar with tabs: General, Configuration, Manager, Extensions, Lines (selected), Interfaces, Virtual Fax, ACD, Applications, Provisioning, Audio, Miscellaneous, and Commit. Below the navigation bar, a yellow message box states: "You must click on Commit button in order to apply Change." The main content area is titled "Lines Manager: Incoming Lines" and includes a "Mass Operations" link. There are tabs for Incoming Lines, Outgoing Lines, Emergency Lines, Special Lines, Banned Prefix, and Ringing Services. The "Incoming Lines" tab is active, showing a list of lines. A red box highlights the "+ Add a new Incoming Line" button. The table below shows the following data:

Extension	Trunk	Forward To	Schedule	Priority	Tenant	✓	✗
5000	gateway (SIP) (Global)	ext_8010/8010	default		default	✓	✗
5001	gateway (SIP) (Global)	phone:5001/5001: (SIP)	No Schedule (Always)		default	✓	✗
5002	gateway (SIP) (Global)	ext_8020/8020 ext_8011/8011	No Schedule (Always)		default	✓	✗
X.	pstn (SIP) (Global)	line	default		default	✓	✗
X.	gateway (SIP) (Global)	line	default		default	✓	✗

At the bottom of the table, there are controls for "Action: - select an action -", "Filter: All", and "Columns to display: Select".

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

The screenshot shows the configuration page for a new Incoming Line. The page is titled "Lines Manager: Incoming Lines" and has tabs for Incoming Lines, Outgoing Lines, Emergency Lines, Special Lines, Banned Prefix, and Ringing Services. The "Incoming Lines" tab is active, and the "General" sub-tab is selected. The configuration fields are as follows:

- Type: Extension (DNIS)
- * Extension (DNIS): 5000
- * Trunk: gateway (SIP) (Global)
- Description: (empty text area)

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

The screenshot shows the 'Select' dialog box with the following details:

- Find:** [] from **Phone** Search
- Left List:** * Please select info(s) *
5000: (SIP)
5001: (SIP)
5002: (SIP)
5003: (SIP)
5004: (SIP)
5005: (SIP)
5006: (SIP)
5007: (SIP)
5008: (SIP)
5009: (SIP)
5010: (SIP)
- Right List:** * Add these by clicking OK *
8010
- Buttons:** >>, Delete, OK, Cancel

Below the dialog, the configuration fields are:

- Destination:** Extension(s) [] Default: none
- * Extension:** 8010 [] Select
- Use User-defined CallForward?**
- Call Rotation Schedule:** Disabled []
- Distinctive Ring (SIP Device):** Disabled []
- Class of Service (Outgoing Calls):** outgoing []
- Add Call Diversion SIP header?**

The screenshot shows the 'Options' tab configuration with the following details:

- Use User-defined CallForward?**
- Call Rotation Schedule:** Disabled []
- Distinctive Ring (SIP Device):** Disabled []
- Allow Call Transfer:** None [] Default: none
- Continue execution if the destination channel hangup:** ?
- * Maximum ring time:** 20 seconds [] Default: 20 seconds
- Provide Music on Hold until answer:**
- Music On Hold:** default (default) []
- Class of Service (Outgoing Calls):** outgoing []
- Add Call Diversion SIP header?**
- Destination #2:** Destination [] None [] Default: none

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

Incoming Lines

General Destination **Options** Security Advanced Options CallerID

Answer the Line ? : Default: True

Enable In-Band Progress information ? : This will request that in-band progress information be provided to the calling channel.

Schedule :

Music On Hold :

Language :

Group ID (ChanSpy) : : If defined, this allow to create 'ChanSpy' application that allow to spy all calls received on this Incoming Line.

Pickup Mark : This allow to set a custom extension that will be used to pickup a ringing channel.

Line ID (Ringing Service) : You can override destination using 'Ringing Service' Feature Code.

Virtual Fax

Enable Fax Detection ? : If enabled, this Incoming Line can be shared between Voice and Fax. If this line is not shared between voice and fax do not enable this option.

Fax Extension (Routing) : : If not specified, we will use the incoming DNIS for Virtual Fax Routing. If specified this value overrides the incoming DNIS for Virtual Fax Routing. This can be useful when using interfaces that do not support DNIS or if the same DNIS can exist on multiple interfaces.

Call Recording

Record all incoming call ? :

Recording Tag :

Send Recording by Email ? :

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.

Telephony Settings: Channels

Configuration **Channels** Language Time Zones Asterisk Manager Monitoring Scheduled Tasks Hangup Causes Synchronization

Channels

General RTP Options Codecs **SIP Channel** IAX Channel Jitter Buffer Guest Account

Miscellaneous

Enable Session Progress and In-Band Audio ? : : Used for Asterisk Early Audio with SIP channels.

Enable Premature Media? : : If you turn this option on, SIP channel will not automatically initiate early media if it receives audio from the incoming channel before there's been a progress indication.

Security

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

Lines Manager: Incoming Lines

Incoming Lines

Outgoing Lines

Emergency Lines

Special Lines

Banned Prefix

Ringing Services

Incoming Lines

General

Destination


Options


Security

Advanced Options

CallerID


Call Restrictions (Blacklist/Whitelist)

Enable 'Whitelist' lookup ?  :

Enable 'Blacklist' lookup ?  :

Execute Lookup before CallerID Prefix manipulation ? :

Authentication/Password

Authentication (PIN) ?  :

Default: none

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 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 CallerID when done

Incoming Lines

General

Destination


Options


Security

Advanced Options

CallerID

Class of Service (Transfer/Forward Call) :
Default: System Default

Enable SIP Header routing support ?  :

Enable Redirected Number (RDNIS) support ?  :

Enable Phone Spam Filter ? :

If enabled, a lookup will be made on PhoneSpamFilter.com to check if the Incoming CallerID is listed as Telemarketer. (EXPERIMENTAL)

Generates Special Information Tone (SIT) to block telemarketers from calling you ? :

Volume Control

Volume Gain (TX) :

Volume Gain (RX) :

Enable DTMF volume control ? :

If checked, we will monitor the channel for '' and '#'. If one of those keys is pressed, the volume will be increased or reduced, respectively.*

Class of Service

Class of Service :
Default: System Default

Exclude from 'Incoming' Class of Service ? :

Include in 'Guest' Class of Service ? :

Script Execution (AGI)

AGI script :

This optional AGI parameter will setup an AGI script to be executed when calling this Incoming Line.

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

Lines Manager: Incoming Lines Incoming Lines: 5000

[Incoming Lines](#)
[Outgoing Lines](#)
[Emergency Lines](#)
[Special Lines](#)
[Banned Prefix](#)
[Ringing Services](#)

Incoming Lines

[General](#)
[Destination](#)
[Options](#)
[Security](#)
[Advanced Options](#)
[CallerID](#)

CallerID / Source :

Include CallerID prefix to the CallerID Source ? : Default: True

Prefix to add to CallerID Number :

Include current CallerID ? : Default: True

Prefix to add to CallerID Name :

Include current CallerID Name ? :

Strip '+' on Inbound CallerID ? :

Enable CallerID Manipulation on this line ? :

Advanced CallerID options

Enable Presentation indicator ?

* Presentation :

-- select --
 Presentation Allowed, Not Screened
 Presentation Allowed, Passed Screen
 Presentation Allowed, Failed Screen
 Presentation Allowed, Network Number
 Presentation Prohibited, Not Screened
 Presentation Prohibited, Passed Screen
 Presentation Prohibited, Failed Screen
 Presentation Prohibited, Network Number
 Number Unavailable


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

Incoming Lines

General Destination Options Security Advanced Options CallerID

Type  : Extension (DNIS) v

* Extension (DNIS): sipheaderrouting


* Trunk  : pstn (SIP) (Global) v

Description:

Add Cancel


Incoming Lines

General Destination Options Security Advanced Options CallerID

Enable SIP Header routing support ?  :

SIP Header Field: To v
Default: To

Class of Service (SIP HEADER): All Incoming Lines (Global) v
Default: All Incoming Lines (Global)

Enable Redirected Number (RDNIS) support ? : 

Enable Phone Spam Filter ? :
If enabled, a lookup will be made on PhoneSpamFi

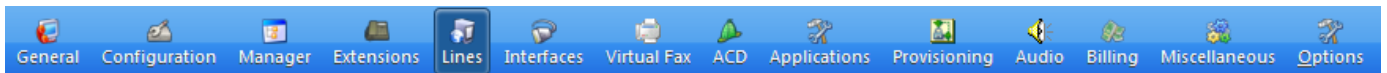
Generates Special Information Tone (SIT) to block telemarketers from calling you ? :

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 and 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



Lines Manager: Ringing Services

Incoming Lines Outgoing Lines Emergency Lines Special Lines Banned Prefix **Ringing Services**

Ringing Services

General Destination

* Ringing ID: 1

Description:

- 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 you are done.



You must click on Commit button in order to apply Change.

Lines Manager: Ringing Services

Incoming Lines Outgoing Lines Emergency Lines Special Lines Banned Prefix **Ringing Services**

Ringing Services

General **Destination**

Destination #1

Destination: Auto Attendant

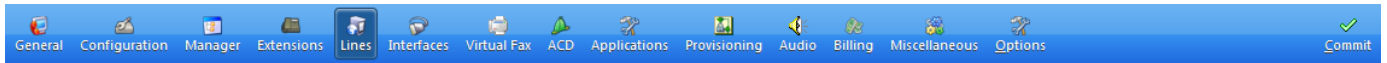
* Menu: ivrclosed

Destination #2

Destination: None

- 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.



You must click on Commit button in order to apply Change.

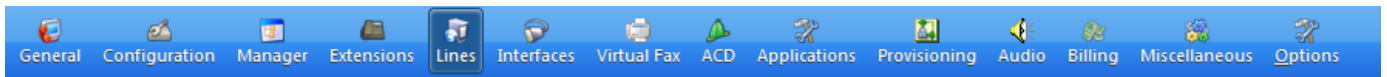
Lines Manager: Incoming Lines

Incoming Lines: 1 to 5 of 5

[Add a new Incoming Line](#)

Extension	Description	Trunk	Line ID	Destination	Forward To	Schedule	Priority	Tenant	
6XXX	gateway (SIP) (Global)			Auto Attendant	None	default		default	<input checked="" type="checkbox"/>
6500	gateway (SIP) (Global)				mainopen	default		default	<input checked="" type="checkbox"/>
6666	gateway (SIP) (Global)			Extension(s)	6002: Extension 6002 (SIP) 6004: Yealink (SIP)	default		default	<input checked="" type="checkbox"/>
5555555555	gateway (SIP) (Global)			Extension(s)	0: Dial 0 (RINGGROUP)	default		default	<input checked="" type="checkbox"/>
9055551212	gateway (SIP) (Global)	1		Queue (ACD)	sales	default		default	<input checked="" type="checkbox"/>

- 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.



You must click on Commit button in order to apply Change.

Lines Manager: Incoming Lines

Incoming Lines

General Destination **Options** Security Advanced Options CallerID Billing

Answer the Line? Default: True

Second(s) to wait before Answering the Line:
If you want to add 250ms of pause, set this value to 0.25. Set to 0 to disable.

Schedule:

Music On Hold:

Language:

Group ID (ChanSpy):
If defined, this allow to create 'ChanSpy' application that allow to spy all calls received on this Incoming Line.

Pickup Mark:
This allow to set a custom extension that will be used to pickup a ringing channel.

Line ID (Ringing Service):
You can override destination using 'Ringing Service' Feature Code.

PIN/Password:

- 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.

General Configuration Manager Extensions **Lines** Interfaces Virtual Fax ACD Applications Provisioning Audio Billing Miscellaneous Options

You must click on Commit button in order to apply Change.

Lines Manager: Incoming Lines

Incoming Lines Outgoing Lines Emergency Lines Special Lines Banned Prefix Ringing Services

Incoming Lines

General Destination **Options** Security Advanced Options CallerID Billing

Answer the Line ? Default: True

Second(s) to wait before Answering the Line: If you want to add 250ms of pause, set this value to 0.25. Set to 0 to disable.

Schedule:

Music On Hold:

Language:

Group ID (ChanSpy): If defined, this allow to create 'ChanSpy' application that allow to spy all calls received on this Incoming Line.

Pickup Mark This allow to set a custom extension that will be used to pickup a ringing channel.

Line ID (Ringing Service) You can override destination using 'Ringing Service' Feature Code.

PIN/Password:

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

General Configuration Manager Extensions Lines Interfaces Virtual Fax ACD **Applications** Provisioning Audio Billing Miscellaneous Options Commit

You must click on Commit button in order to apply Change.

Application Manager: Applications

Applications Auto Attendants Conferences Custom Scripts Scheduled Tasks

Applications: 1 to 8 of 8 **Add a new Application**

Search:

Extension	Description	Destination	Forward To	Priority	Tenant	
*333	conferencebridge1	Goto Conference	conferencebridge1: conferencebridge1 (1)	default	default	<input checked="" type="checkbox"/>
*444		Goto Voicemail Main		default	default	<input checked="" type="checkbox"/>
*800	queue sales	Goto Queue (ACD)	sales	default	default	<input checked="" type="checkbox"/>
*830	page	Paging and Intercom		default	default	<input checked="" type="checkbox"/>
*871		Ringing Service		default	default	<input checked="" type="checkbox"/>
*886	moh	Play Music On Hold		default	default	<input checked="" type="checkbox"/>
*888		Goto Menu (IVR)	mainopen	default	default	<input checked="" type="checkbox"/>
*899		Goto Menu (IVR)	ratecall_ivr	default	default	<input checked="" type="checkbox"/>

Action: Columns to display:

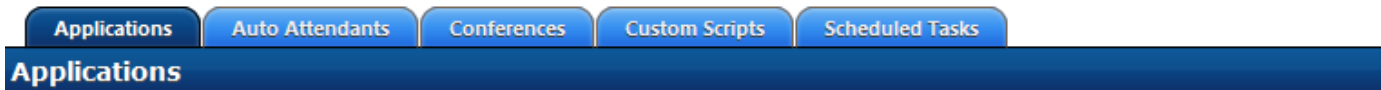
- Give this Application a number you would like to use as a feature code and make sure this code does not conflict with any other extension number, Feature Code or Outgoing Line string.
- This 3 digit code starting with *87 dedicates *87 for Ringing Services codes and allows Ringing Groups 1-9 to be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

- Click on the Destination tab.



You must click on Commit button in order to apply Change.

Application Manager: Applications



Applications

General **Destination**

* Extension: *871

Description:

- 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.

General **Destination**

* Destination: Ringing Service

* Line ID: 1

* Ring ID: 1

Options

Language: Default

Allow Extensions to use this application as destination?

Prefix to add to CallerID Number:


Include current CallerID? Default: True

Prefix to add to CallerID Name:

Include current CallerID Name? Default: True

- 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.
- Commit changes to activate the feature.

General Destination

* Destination  : Ringing Service

* Line ID : 1

* Ring ID : 1

Options

Language : Default

Allow Extensions to use this application as destination ?

Prefix to add to CallerID Number :

Include current CallerID ? Default: True

Prefix to add to CallerID Name :

Include current CallerID Name ? Default: True

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 `ringervice_<tenant>_<application>`. Example: `ringervice_default_*871`.

Ringling Services BLF

Ringling Services BLF on Yealink DSS Key

- It is very easy to set up a Ringling Service Key on a Yealink phone.
- Using the previous steps to build an application *871 to enable Ringling 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 Ringling Service.
- The BLF key will light **Red** when in Use.

Logged as: admin

- ScopServ
- Configuration
 - Server
 - Network
 - Telephony
 - General
 - Configuration
 - Manager
 - Extensions
 - Lines
 - Interfaces
 - Virtual Fax
 - Queues and Agents
 - Applications
 - Provisioning**
 - Audio
 - Billing
 - Miscellaneous
 - Import/Export
- ScopSTATS
 - System Monitoring
 - Telephony Reports
- Tools
- Organizing
- Administration
- Options
- Configuration Wizard
- Log out

General Provisioning Server Network Date and Time Phone Options PBX Services **DSS Keys** Expa

Internal Ringer Multicast Paging LDAP

Deal Type: Blind Transfer
Default: Blind Transfer

Expansion Module: LCD Expansion Module (EXP40)
Number of Expansion Module: 1

Line Keys

Key 1: Line
Label:
Line: Line 1

Key 2: Speed Dial
Label: PARK
Line: Line 1
* Extension/Value: 700

Key 3: Speed Dial
Label: PAGE
Line: Line 1
* Extension/Value: *830

Key 4: BLF
Label: Night
BLF Mode: Ringing Service
* Extension: *871