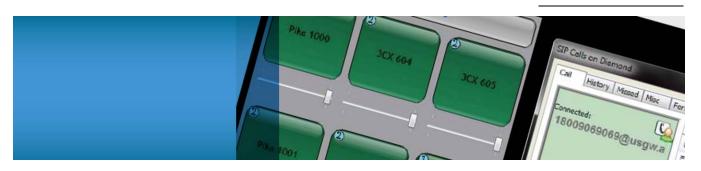
# SIP Quick Reference Guide



# **Supported SIP (Session Initiated Protocol) Features**

## **SIP Basic**

Call Hold Places the current call on hold or takes a call off hold.

Call Waiting Allows an audible indicator is heard when a call is received from a third party while the

console is currently engaged in.a call.

## **SIP Enhanced**

Blind Transfer Transfers a caller without announcing the transferred party to the recipient.

Consultative Transfer Transfers a caller and allows the console operator to announce the call.

Call Forwarding Unconditional Call Forwarding: routes incoming calls directly to voicemail.

Conditional Forwarding: routes incoming calls to voicemail after the call is unanswered for a

pre-defined number of rings.

Busy Call Forwarding: routes incoming calls to voicemail if the line is busy.

3-Way Call

Conferencing A conference bridge can be established for up to three (3) calls.

Call Conference Multiple outgoing calls can be bridged into a conference call.

Do Not Disturb Allows all incoming calls to be routed to voicemail.

Mute Mic Allows the microphone to be muted during a call.

Crosspatch Radio PTT Allows the console operator to interface radios via SIP, to be included in the interoperable

conferences.

Recording in

Network Recorder Allows C-Soft to record via echo packets.

Software Security Key

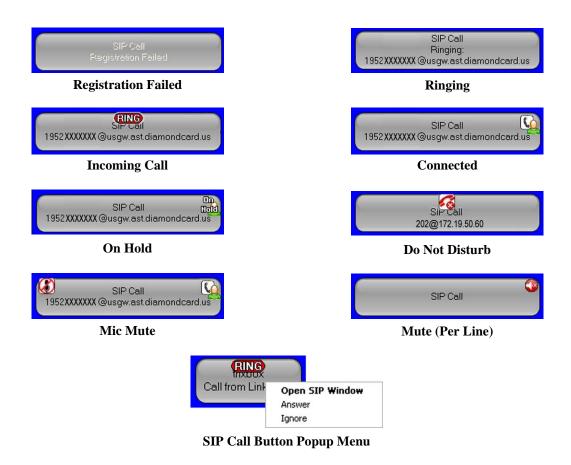
(ADHB-4 Licensing Allows password input for SIP validation on an ADHB-4 (Advanced Digital Headset Box),

eliminating the need for hardware security keys (i.e. dongles).

LIT000539000 Revision A 10/2010

# **SIP Call Buttons with Status Indicators**

The following status indicators appear on the SIP control button in C-Soft.



# **SIP Calls Window**

The **SIP Calls** window is used to manage SIP calls, view call history and view missed calls.

**NOTE:** If using only SIP Basic features, buttons for SIP Enhanced features on the Call, History and Missed pages are disabled.



# **Call Page**

When a SIP Call Control button is clicked from the C-Soft console, the SIP Calls window opens to the Call page.

The **Call** page is used to place incoming and outgoing SIP phone calls. When licensed or SIP Enhanced, this window is also used to lace hold, initiate 3-way calling, mute the microphone, and activate do not disturb.



# **SIP Call Display**

The **SIP Calls display** is used to enter a phone number to call. It also provides status messages and icons to indicate the state of the associated SIP line.

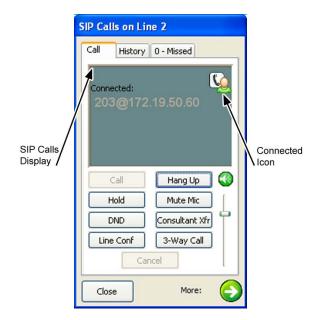
The background color can be specified by the user.

To enter a phone number to call, do the following:

- 1. Ensure the **SIP Calls** window is open.
- 2. Using the keypad pane or keyboard keypad, enter the **phone number** to call.

  The digits appear in the SIP display.
- 3. Click Call.ORWhile the SIP display has focus, press Enter.

**NOTE:** When the call is connected, the Connected icon appears in the display.



#### **Call/Answer Button**

The Call/Answer button is used to place or answer a call.

If a contact is selected from the Contacts pane, or a phone number is entered in the SIP Calls display, the Call button is enabled.

If a call is being received while the SIP Calls window is open, a ring icon appears on the SIP Call display and the Answer button is enabled.

If a call is received while the line is engaged in another call, the Answer button is enabled.

To answer a call, do the following:

> Click **Answer**.

The incoming call is connected.

To **ignore an incoming call**, do the following:

> Click Ignore.

**NOTE:** If the line is already engaged in a call, the existing call is placed on hold and the Answer button changes to Switch.

To **return to a call on hold while on another call**, do the following:

Click Switch. The current caller is placed on hold and the other caller is taken off hold.



# **Call/Answer Button (continued)**

To answer and merge a waiting call with the current call, do the following:

- 1. While already engaged in a call, click **Answer**. *The current call is placed on hold and the incoming call is answered.*
- 2. Click 3-Way Call.

The on-hold call is taken off hold, and joined with the other call. The two (2) callers and console operator are connected in one (1) conversation. The 3-way Call button label changes to Leave Call.

To place an outgoing call, do the following:

- Using the keypad pane or keyboard, enter the phone number.
- 2. Click Call.

The call is placed to the phone number entered in step 1. OR

Press Enter.

The call is placed to the phone number entered.

**NOTE:** To disconnect from the call, click **Hang Up**.



# Hang Up/Hang Up>/Ignore Button

The **Hang Up/Hang Up>/Ignore** buttons is used to disconnect the current SIP call or ignore an incoming call.During a call, the Hang Up button is enabled.

If multiple parties are connected via consultant transfer, 3-way calling, or call waiting, the Hang Up > button appears.

Click the Hang Up> button to open a popup menu containing all currently connected calls. While an incoming call is ringing, the Hang Up button's text changes to Ignore.

To disengage a 2-way call, do the following:

> Click the **Hangup** button.

The call is disconnected and the connected icon disappears.

To disengage a call with multiple participants, do the following:

1. Click **Hang Up** >.

A popup menu appears.

2. Select an individual **call** to disconnect from.

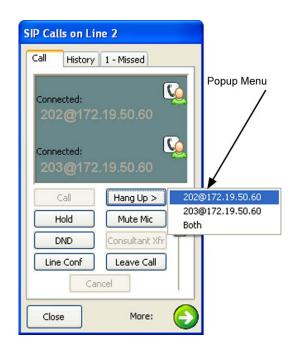
OR

Select both to disconnect both calls.

To **ignore an incoming call,** do the following:

Click Ignore.

The call is ignored and the caller information is added to the list on the Missed page.



# **Hold Button (SIP Enhanced only)**

The **Hold** button places the current call on hold. During the hold, C-Soft does not send or receive audio.

If multiple parties are connected at one (1) time, the hold button places all calls on hold or takes all calls off hold.

If the console operator is talking to the consulting party during a consultant transfer or speaking to one (1) party during a call waiting event, clicking the Hold button places the active participants on hold

#### NOTE:

- Hold options can be configured on the SIP server.
- The SIP hold button stops both incoming and outgoing audio, whereas the console hold button only stops the outgoing audio.

# To place a call on hold, do the following:

> While in a SIP call, click **Hold**.

The call is placed on hold and the On Hold icon appears in the SIP Call display.

## To take a call off hold, do the following:

> Click **Hold**.

The call is taken off hold.

# **Mute Mic Button (SIP Enhanced only)**

The **Mute Mic** button is used to stop microphone audio from reaching any connected party.

## To mute the microphone, do the following:

> While engaged in a SIP call, click **Mute Mic**.

The button changes to Unmute Mic, the Mute Mic icon appears in the SIP Call display, and microphone audio is not sent.

## To unmute the microphone, do the following:

> Click Unmute Mic.





# **DND/Disable Button (SIP Enhanced only)**

The **DND** (Do Not Disturb) /**Disable DND** button is used to activate Do Not Disturb mode. While active, callers automatically receive a message stating the number is unavailable. While DND is active, DND icons appear on the Call page and the associated SIP button.

**NOTE:** The specific DND message is configured on the SIP server.

To activate DND, do the following:

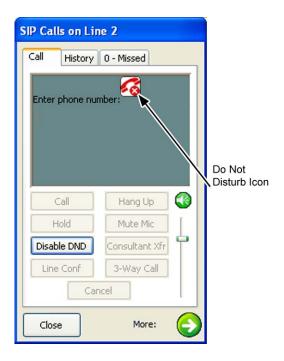
> Click **DND**.

The button label changes to Disable DND and the DND icon appears in the SIP Call display.

**NOTE:** Calls received while in Do Not Disturb mode are logged on the Missed page.

To **disable DND**, do the following:

> Click **Disable DND**.



# **Consultant Xfr/Blind Xfr Button (SIP Enhanced only)**

The **Consultant Xfr/Blind Xfr** button is used to blind transfer a call or transfer with consultation.

A blind transfer is a call transferred to a second party without announcing the call to the receiver.

A consultant transfer allows the console operator to announce the transferred call to the receiver.

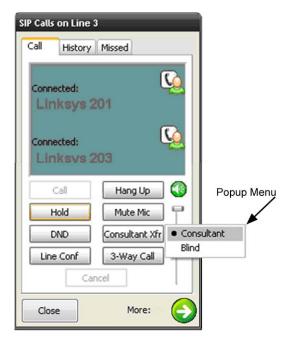
To **toggle the button's function**, do the following:

- 1. Right-click the **button**. *A popup menu appears*.
- **2.** Select the **function** you want to use from the menu.

To abort the transfer function at any time, do the following:

> Click Cancel.

The incoming call remains connected.



# **Consultant Xfr/Blind Xfr Button (continued)**

#### To make a consultant transfer, do the following:

 From the Call page for the line, if the button label is Consultant Xfr, go to step 2 OR

Right-click Blind Xfr.

a. Select Consultant Xfr.

The button label and function changed to Consultant Xfr.

## 2. Click Consultant Xfr.

Enter transfer destination appears and the SIP Call display is ready to accept a transfer destination.

**3.** Using the Keypad pane, keyboard, or the Contacts list, enter the new **destination number** in the SIP Calls window.

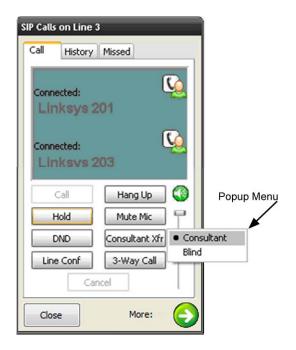
#### 4. Click Consultant Xfr

The current call is placed on hold and a call is placed to the destination number. When the call is answered, the Hang Up button changes to Hang Up>.

- **5.** Announce the caller to the **second party**.
- 6. Click Consultant Xfr.

The first and second call are now connected.

**7.** Click **Hang Up** to disconnect from the call. *Both callers are disconnected from the console.* 



## To make a blind transfer, do the following:

1. From the Call page for the line, if the button label is Blind Xfr, go to **step 2**.

OR

Right-click Consultant Xfr.

A popup menu appears.

a. Select Blind Xfr.

The button changes to Blind Xfr.

#### 2. Click Blind Xfr.

Enter the transfer destination appears in the display and the SIP Call display is ready to accept a number.

**3.** Using the Keypad pane, keyboard, or Contacts list enter the **destination number**.

The phone number appears at the bottom of the display.

## 4. Click Blind Xfr.

The original call is transferred to the destination number and immediately disconnected from the console.



# **Line Conf Button (SIP Enhanced only)**

The **Line Conf** button is used to join two (2) SIP phone calls already in progress into a line conference. A line already involved in a 3-way call can be included in the line conference. During a line conference, the SIP Calls window border changes to black.

Due to processing demands, a maximum of six (6) remote parties (plus the console) can be mixed together at one (1) time.

To place two (2) lines into a line conference. do the following:

- 1. Engage **two** (2) **SIP calls** on two (2) separate lines.
- 2. From the first line's call window, click **Line Conf**. *The window's border turns black*.
- From the second line's call window, click Line Conf.

The second line's window turns black. Both lines are now in communication with each other.



# 3-Way Call Button (SIP Enhanced only)

The **3-Way Call** button is used to establish a call in which three (3) participants are engaged.

To **establish a 3-way call**, do the following:

- 1. Click 3-way Call.

  Enter New Destination appears in the display.
- **2.** Using the Keypad pane, keyboard, or Contact list enter the **new destination number**. *The number appears at the bottom of the display.*
- 3. Click 3-way Call.

The current call is placed on hold and a call is placed to the new destination number. The Hang Up button changes to Hang Up>.

4. Click 3-Way Call.

The first call is taken off hold and joined in the current call. Both parties and the console operator are connected. The 3-Way Call button's label changes to Leave Call.



## **Cancel/Switch Button**

The **Cancel/Switch** button is used to cancel an operation or switch to another call.

If the button is active and *Cancel* appears on it, it can be used to cancel a blind transfer, consultant transfer, 3-way call or restore a call to its previous stare.

If the button is active and *Switch* appears on it, it can be used to switch from the current call to a call-waiting call.

To cancel a transfer or 3-way call operation before it is completed, do the following:

> Click Cancel.

The operation is canceled, the current caller remains on the line, and the Cancel button is grayed-out.

To **switch to a waiting call while already in a call**, do the following:

> Click Switch.

The current call is placed on hold, the waiting call becomes active.

**NOTE:** Click **Switch** to switch back to the other call.

#### Mute Icon

The **Mute** icon is used to instantly mute the console audio. Clicking the volume icon mutes the currently selected line.

To mute the selected lines, do the following:

> Click Mute.

The volume slider moves to the bottom of the volume bar. The speaker icon changes to a red mute icon and audio is turned off.

**NOTE:** Clicking the **Mute** button restores the volume slider to its previous value.

The **Volume** slider is used to adjust the speaker volume.

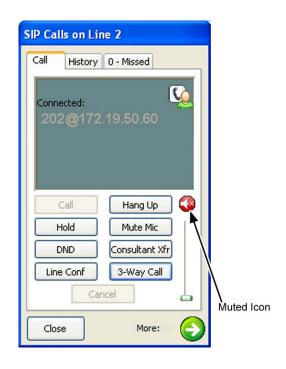
To decrease the volume, do the following:

> Slide the **volume slider** down.

To increase the volume, do the following:

> Slide the **volume slider** up.





# **Expand Button**

The **Expand** button is used to maximize the SIP Calls window. When the Expand button is clicked, a Misc tab, Forward tab, and additional features appear on the Call, History and Missed pages.



The **Collapse** button is used to minimize the SIP Calls window.



#### **Contacts Pane**

The **Contacts** pane is used to view the list of contacts within the SIP system. This list is managed using a popup menu.

**TIP:** The Contacts pane appears on the Call page when the Expand button is clicked.

To **place a call to a contact in the list**, do the following:

1. From the Contacts pane, select a **contact**. *The contact's number appears in the SIP call display.* 

# 2. Click Call.

A call is placed to the selected contact The default action (the bolded item in the menu e.g. Call is performed and the call is placed. OR

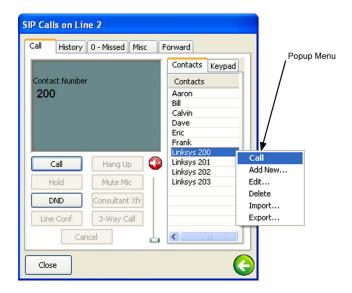
Double-click the contact.

The default action (the bold item in the popup menu, i.e. Call) is performed and the call is placed.

OR

Right-click the **contact** and select **Call** from the menu.

A call is placed to the selected contact.

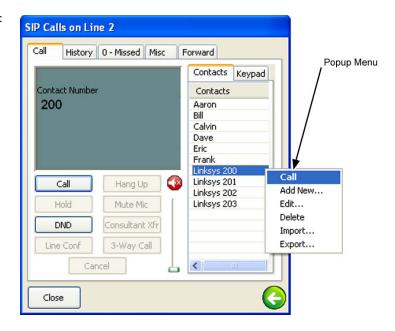


#### Contacts—Add New

# To add a new contact to the list, do the following:

- 1. Right-click the **contact** list. *A popup menu appears*.
- 2. Click **Add New**.

  The Edit Contact window appears.
- 3. In the First Name field, enter the **first** name of the contact.
- 4. In the Last Name field, enter the **last** name of the contact.
- 5. In the Display Name field enter a **user-recognizable name** for the contact.
- 6. In the Number field, enter the **phone number** for the contact.
- 7. On the Notes field, enter appropriate comments for the contact
- 8. Click **OK**The contact is added to the contacts list.



# Contacts—Edit

# To **edit a contact in the list**, do the following:

- 1. Right-click the **contact** to edit. *A popup menu appears*.
- 2. Click Edit.

  The Edit Contact window appears.
- 3. Make **changes** to the appropriate fields.
- 4. Click **OK**.

The changes to the contact information are saved.

#### Contacts—Delete

To **delete an entry from the contacts list**, do the following:

**1.** From the list, right-click a **contact** to delete.

The contact list popout menu appears.

2. Click Delete.

The contact is removed from the list.

## Contacts—Import

To import a contact, do the following:

- 1. Right-click the **contact list**. *The contact list popup menu appears*.
- 2. Click **Import**. *The Import window appears*.
- **3.** Select a .csv file to import.
- 4. Click OK.

All contacts contained in the .csv file are added to the list.

# Contacts—Export

To **export a contact**, do the following:

- 1. Right-click the **contacts list**. *A popup menu appears*.
- 2. Click Export.

The Export window appears.

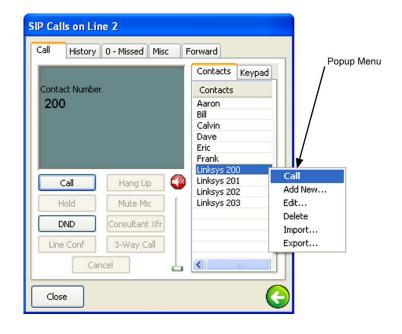
**3.** From the Name drop down menu, select a **name**.

OR

In the Name field, enter a **name** for the .csv file.

4. Click OK.

All contacts are exported to the selected .csv file.



# **Keypad Pane**

The **Keypad** pane is used to enter DTMF digits and access voicemail messages.

- If the console is engaged in a call, the keypad does not generate DTMF tones, the digits are inserted in the SIP display.
- If the console is not engaged in a call the keypad generates DTMF tones.

## NOTE:

- These digits can also be entered using the keyboard.
- While in a call, the keypad generates DTMF tones.
   When not is a call, the keypad inserts the digits in the SIP call display.

To access the keypad, do the following:

- 1. Click the **Expand** button.
- **2.** Click the **Keypad** tab. *The Keypad pane opens.*

## **CLR Button**

The **CLR** (Clear) button is used to clear the information from the SIP Call display.

NOTE: The SIP Call display entry field must be enabled.

## <-- Button

The <-- button is used to delete the last entered character from the SIP Call display.

**NOTE:** The SIP Call display entry field must be enabled.

# Flash Button

The **Flash** button is used to send a hook flash signal.

**NOTE:** This feature is only available during a SIP call.



# **Voicemail Button (SIP Enhanced only)**

The **Voicemail** button is used to access a specific voicemail number. When the voicemail notification icon appears in the SIP Calls display, voicemails are waiting. The number indicated to the right of the *New* text, (e.g. 4), is the quantity of voicemail messages in queue.

**NOTE:** The SIP server must support this feature.

To **configure a specific voicemail number for the button**, do the following:

- Right-click Voicemail and select Configure from the popup menu.
   The Edit Contact window appears.
- **2.** In the Number field, enter a **phone number** for the button.
- 3. Click **OK**.

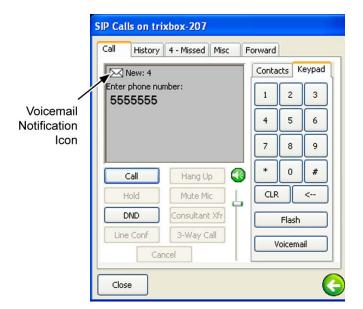
  The voicemail number is stored.

To call the stored voicemail number, do the following:

> Click Voicemail.

A call is placed to the phone number.

The **Voicemail** notification icon indicates voicemail messages are available. The number next to the icons indicates the number of voicemails saved.



# **History Page**

The **History** page is used to view a list of calls received during the current C-Soft session.

**NOTE:** To maximize the SIP Calls window, click **Expand**.

## **Description Column**

The **Description** column displays the call's number and url or the alias.

#### Date and Time Column

The **Date and Time** column displays the date and time the call occurred.

## History Legend

The **History** Legend displays the color format used for incoming and outgoing calls.

#### Call Button

The **Call** button is used to place a call to the selected phone number. If no call is selected from the list, the Call button is grayed-out.

**TIP:** The Call button appears on the History page when the Expand button is clicked.

To call a contact in the History list, do the following:

- 1. Select the **phone number** to place a call to.
- 2. Click Call.

OR

From the entry's popup menu, select **Call**.

Double-click the entry.

The call is placed. The SIP Call window Call Button.



# **Missed Page**

The Missed page is used to view calls received, but not answered on the selected SIP line.

#### Call Button

The **Call** button is used to place a call to the selected phone number. If no call is selected from the list, the Call button is disabled.

TIP: The Call button appears when the Missed

page is maximized. Click the Expand button to maximize the window.

# **Description Column**

The **Description** column displays the call's number and url or the alias.

#### Date and Time

The **Date and Time** column displays the date and time the call occurred

## Missed Calls Legend

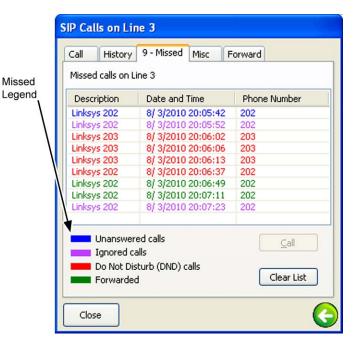
The Missed Calls legend display the color format used for unanswered, ignored, do not disturb, and forwarded calls.

#### Call Button

The Call button is used to place a call to the selected phone number. If no call is selected from the list, the Call button is disabled.

TIP: The Call button appears when the Missed

page is maximized. Click the Expand button to maximize the window.



# **Missed Page (continued**

#### Phone Number Column

The **Phone Number** column displays the phone number the call originated from.

This column is visible when the Expand button is clicked to maximize the window.

#### Call Button

The **Call** button is used to place a call to a phone number selected from the list.

This button is visible when the Expand button is clicked to maximize the window.

To place a call from the missed call list, do the following:

- 1. Select a **call** from the list.
- 2. Click Return Call.

OR

Double-click an entry.

OR

Right-click an **entry** and select **Call** from the popup menu.

The call is placed to the selected phone number.

#### Clear List Button

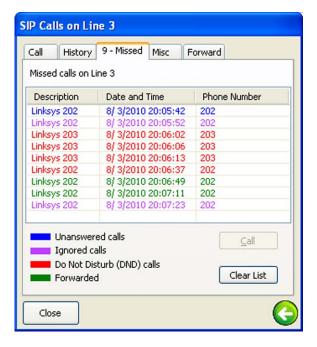
The **Clear List** button is used to clear all calls from the list.

**TIP:** This button is visible when the window is maximized.

To clear all calls from the list, do the following:

> Click Clear List.

The list is cleared.



# Misc Page

The **Misc** page is used to view the following information:

**TIP:** The Misc page, becomes available on the SIP Calls window when the Expand button is clicked,

Domain Displays the register domain of the SIP

line

*Protocol* Displays *sip*: the protocol for the line.

SIP

Registration - Displays the registration information for

the SIP line.



# Forward Page (SIP Enhanced only)

The **Forward** page is used to set up call forwarding options.

**TIP:** The Forward page becomes available in the SIP Calls window when the Expand button is clicked.

The **Forward All Calls** check box indicates all calls received on the line are forwarded to the phone number entered in the Destination field for this check box.

The **Destination** field is used to enter the phone number to forward al incoming calls to.

The **Address Book** button is used to locate a phone number to enter in the Destination fields.

The **Forward call while busy** check box indicates calls received while the line is busy are forwarded to the phone number in the Destination field for this check box.

**NOTE:** This check box can be used in combination with the Forward unanswered calls option.

The **Destination** field is used to enter the phone number to forward incoming calls to when the line is busy.

The **Address Book** button is used to locate a phone number to enter in the Destination field.

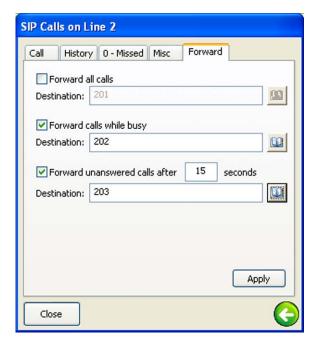
The **Forward unanswered calls after** check box indicates calls left unanswered are forwarded to the phone number entered in the Destination after the amount of time configured in the Forward unanswered all after seconds field.

**NOTE:** This check box can be used in combination with the Forward calls while busy option.

The **Forward unanswered calls after seconds** field is used to enter the amount of time, in seconds, the call is left unanswered before it is forwarded to the phone number in the Destination field.

The **Destination** field is used to enter the phone number to forward unanswered calls after [X] seconds.

The **Address Book** button is used to locate a phone number to enter in the Destination field.



# Forward Page (continued)

# Apply Button

The **Apply** button is used to save changes made to the Forward page.

To set up call forwarding, do the following:

1. From the Forward page, select the **Forward** all calls check box

OR

Select the **Forward calls while busy** check box OR

Select the **Forward unanswered calls** check box

- a. In the Forward unanswered calls after seconds field enter the amount of time in seconds.
- 2. In the Destination field for the selected check box, enter the **phone number** the call is to be forwarded to.

OR

Click the Address Book button.

The address book appears.

- a. Select a phone number to forward calls to.
   The phone number appears in the Destination field.
- c. Click Select.

The window closes and the phone number appears in the selected field.

OR

Click Cancel

The changes are not accepted and the window closes.

3.. Click Apply to accept changes.

The changes are applied.

## Close Button

The Close button is used to close the SIP Calls window.

