Technical Description of

New Feature

C-Soft: SIP Enhancement

Radio Dispatch Engineering

TELEX®

RADIO DISPATCH PRODUCTS

Revision A

Revision Date: 6/22/2010

Table of Contents

Revision History

Revision	Date	Author	Description
A	06/22/10	James Dicke	Draft

C-Soft Software Additions and Modifications

This document describes the additions and modifications made to the C-Soft software since the writing of C-Soft Software Version 4.200 Manual. These changes include modifications to the layout of existing dialog windows and altering how existing features function as well as adding new UI buttons and setup dialogs.

1.1 Manual Fixes

Page 117:

Click a **SIP Call** button.

The server connection process begins if the connections is successful, a success message appears in the console status bar, see Figure 60.

Change to:

Start C-Soft. If the connection is successful, a success message appears in the console status bar, see Figure 60.

Page 117:

When the connection to the SIP server fails, the call must be removed from the Global Call History window and C-Soft Runtime restarted before attempting to connect again.

This is incorrect. First of all, there is no way to remove an entry from the Global Call History window. Instead, the user just needs to restart C-Soft to attempt to connect again. Typically the connection fails when they have configured something incorrectly, so the user will likely need to make modifications in C-Soft designer.

1.2 Manual Changes

Page 110:

Specifically note that the paragraph is still accurate and does not need to be changed.:

The SIP phone line option is available on C-Soft dongles that carry 24 or more lines. Two (2) SIP lines are available on dongles that carry from 24–50 lines. Six (6) SIP lines are available on dongles that carry 50 or more lines.

This licensing method is still in effect. The SIP Enhancement license is an additional option, and does not supersede this method.

Add after paragraph starting with “Configuring SIP in your console…” the following:

In addition, a SIP Enhancement license is available that allows access to additional SIP features. These features include hold, do not disturb, 3-way-calling, microphone mute, line conference, consultant transfer, blind transfer, missed call history, call waiting, and conditional forwarding. The SIP Enhancement license is available for two or six lines.

Page 249:

Make Outgoing Call Page. When the SIP Call Control button is clicked in C-Soft Runtime, the SIP Calls window appears to the **Make Outgoing Call** page. See Figure 142.

It is now called simply the Call page.

Remove "The line must be idle in order to access the SIP Calls window." This is no longer true. Regardless of line state, pressing the SIP Call Control button always opens up the SIP Call Control Window.

From pages 249 to 251, add/integrate the new information contained in the remainder of this document.

C-Soft SIP Enhancement Overview

C-Soft SIP Enhancement takes the basic SIP functionality already existing in C-Soft and adds a number of additional features to allow more flexibility and usefulness in regards to SIP telephony. These new features include hold, do not disturb, 3-way-calling, microphone mute, line conference, consultant transfer, blind transfer, missed call history, call waiting, and conditional call forwarding.

SIP Enhancement licenses are available for two or six lines. The license can be applied by either purchasing an additional SIP Enhancement USB dongle or by purchasing a license code for the ADHB-4. If a user has not purchased the SIP Enhancement license and has been using the SIP telephony, all basic features from previous versions of C-Soft will still be available. These basic features include placing calls, receiving calls, crosspatching, outbound DTMF, per line call history, contact list, auto-answer incoming calls, stun and proxy server, auto adjustments, silence detection, and network recording or monitoring.

GUI Updates

1.2.1 SIP Call Control Window

The SIP Call Control window contains all SIP Enhancement phone controls. The SIP Call Control window contains a number of tabs that provide access to a variety of useful tools. The Call page contains important call operation controls. The History page displays a history of all SIP calls on the corresponding SIP line. The Missed Calls page displays a list of all missed calls. The Miscellaneous page contains information pertaining to C-Soft's line's connection and registration with the SIP Server. Finally, the Forwarding page contains controls to set up different conditions and numbers for call forwarding.

Several buttons located at the bottom of the SIP Call Dialog are common to all pages:

Figure 1: SIP Call Control Window

Pop out/Pop in– Expands or shrinks the SIP Call Control Window. Depending on which tab is selected, additional information is displayed in the additional area. On the Call page, the Contacts List or keypad are displayed. On the History and Missed Call History pages, additional table data is shown. In addition, while expanded, the Misc and Forward tabs are added.

Close – Closes the SIP Call Control window. When the SIP Call Control window is closed, any active calls will remain open.

1.2.2 SIP Call Control : Call Page

The SIP Call Control window allows the user to access all SIP Enhancement features, as well as provide feedback of currently connected SIP calls. If the SIP Enhancement feature has not been purchased, the Hold, Mic Mute, DND , Transfer, Line Conference, 3-Way Call, and Cancel buttons will be disabled.

A list of each button and some basic information is included below.

Call – Allows the user to place an outgoing call, or to answer an incoming call.

Hangup – Disconnects the current call. If multiple parties are connected through the dialog (via consultant transfer, 3-way-calling, or call waiting), the Hang Up button will display a pop up menu to select which party (or all parties) to disconnect.

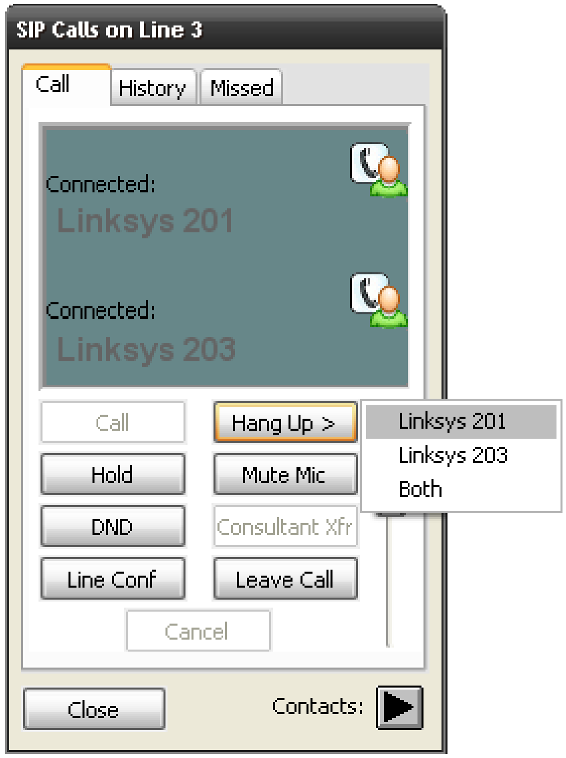

Figure 2: SIP Call Control Window: Hang Up Menu

Hold – Places or takes the current call on or off hold. While on hold, the connected call will neither send or receive any audio from C-Soft. Note that this functionality is different from the Hold button accessible from the main console window, which only stops outgoing audio. Additional hold options can be configured on the SIP Server. If multiple parties are connected through the dialog, the hold button will place all calls on hold. More specifically, if in a 3-way-call, all parties will be placed on or taken off of hold. If talking to the consulting party during a consultant transfer or speaking to one party during a call waiting call, the hold button will affect the active participant. See the Operations section for more information on Hold behavior during these operations. The Hold button will only be available if SIP Enhancement license has been acquired.

Mute Mic – Mutes the microphone and stops any microphone audio from reaching any connected party. The Mic Mute button will only be available if SIP Enhancement license has been acquired.

DND – Enables or disables Do Not Disturb mode. While DND mode is enabled, any incoming call is automatically receives a message stating that the number is unavailable. The specific audio message can be configured by the SIP Server. While DND mode is enabled, DND icons will appear on the Call Page and on the associated SIP button. The DND button will only be available if SIP Enhancement license has been acquired.

Transfer – Forwards the current call to another party in one of two methods: Blind or Consultant. In order to select which type of transfer to perform, select either “Blind Transfer” or “Consultant Transfer” from the menu accessed by right clicking the transfer button. The Transfer button will only be available if SIP Enhancement license has been acquired.More information about Blind Transfer and Consultant Transfer operations are included in Operation Notes: Blind Transfer Consultant Transfer.

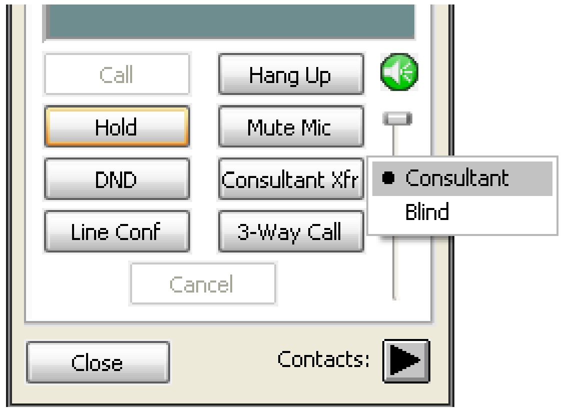

Figure 3: SIP Call Control Window: Transfer Menu

Line Conference – Places any connected calls in a Line Conference, which allows calls from different lines to be able to communicate with each other. This operation is similar to a crosspatch between two or more SIP calls. While in a line conference, the SIP Call Control's border will turn black. The Line Conference button will only be available if SIP Enhancement license has been acquired. More information about Line Conferencing is included in Operation Notes : Line Conference

3-Way-Call – Allows the user to place an call while one is already in place, and allow communication between all call participants. The 3-Way-Call button will only be available if SIP Enhancement license has been acquired. More information about 3-Way-Calling is included in Operation Notes : 3-Way Call.

Cancel – Used to abort specific operations, and restore a call to its previous state. The cancel button is only enabled during Blind Transfer, Consultant Transfer, and 3-Way Calling. In addition, during a call waiting call, the cancel button changes to a “Switch” button. When the switch button is pressed, the active call is switched between the two calls. The inactive call is automatically placed on hold.

Mute/Unmute – Mutes or unmutes the current call. If the Volume Bar is set to its lowest level, the mute button will automatically change to reflect the muted state.

Volume Bar – Sets the volume on the current call's incoming audio.

Contact List – Used to create a phone directory of contacts. When a contact is clicked, the contact's phone number will be automatically entered in the phone number entry text box for easy calling.

When the contact list is right-clicked, the contact list context menu will appear allowing access to the primary functions of the contact list. The context menu contains the following items: “Call”, “Add New...”, “Edit...”, “Delete”, “Import...”, and “Export...”.

- To call a contact, first select a contact from the list. Right click the contact and select “Call” from the context menu, or simply double click the contact.

- To add a new contact, right click the contact list and select “Add New...” from the context menu to open to Contact Window. When finished with entering information, press OK. The new contact will appear in the contact list.

- To edit a contact, first select a contact from the list. Right click the contact and select “Edit...” from the context menu to open the Contact Window. When finished with editing the contact, press OK.

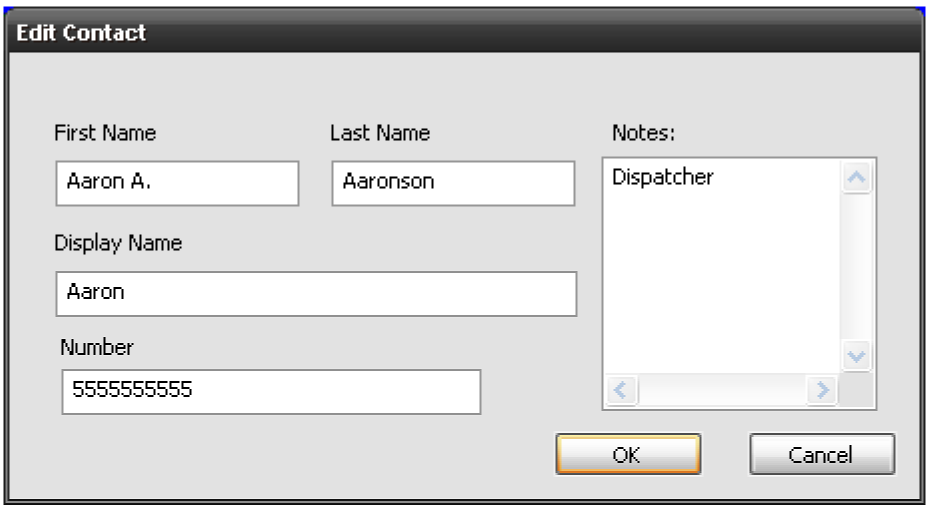

Figure 4: Edit Contact Window

- To remove a contact, first select a contact from the list. Right click the contact and select “Remove” from the context menu.

- To import Contacts from a Comma Separated Values file (.CSV), right click the contact list and select “Import...” from the context menu. An Open File dialog will appear. Select the desired file from the Open File dialog and press OK. Contacts from the specified CSV file will be imported into the contact list.

- To export Contacts to a Comma Separated Values file (.CSV), right click the contact list and select “Export...” from the context menu. A Save File dialog will appear. Specify the the desired location and press OK. All contacts in the list will be exported to the specified CSV file.

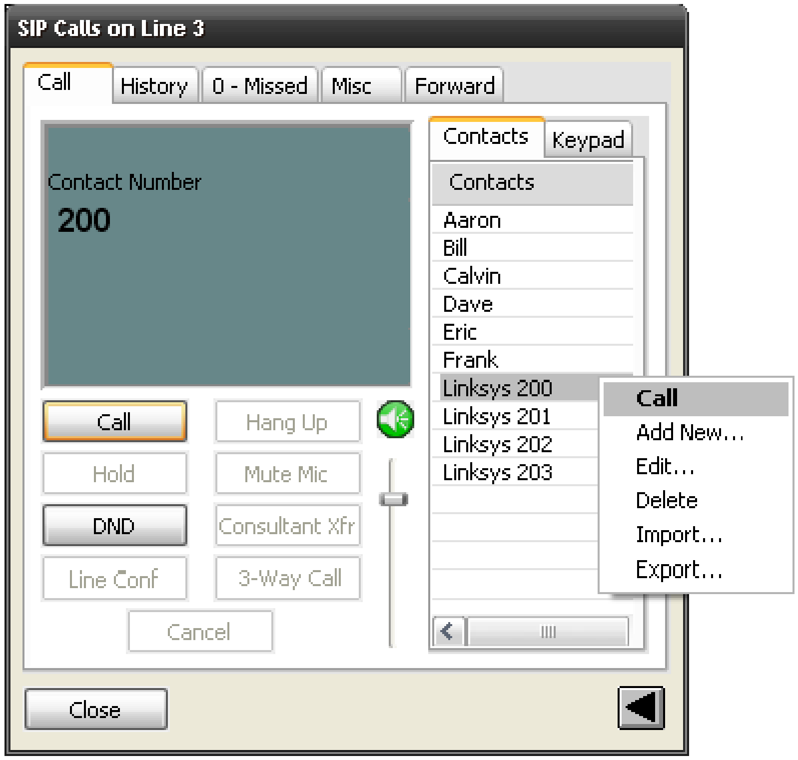

Figure 5: SIP Call Control Window: Contact Context Menu

Keypad – Provides graphical access to a DTMF keypad, as well as a dedicated voicemail button. DTMF digits can also be input using the keyboard. While in a call, the keypad generates DTMF tones. While not in a call, the keypad will insert digits to the phone number prompt text box.

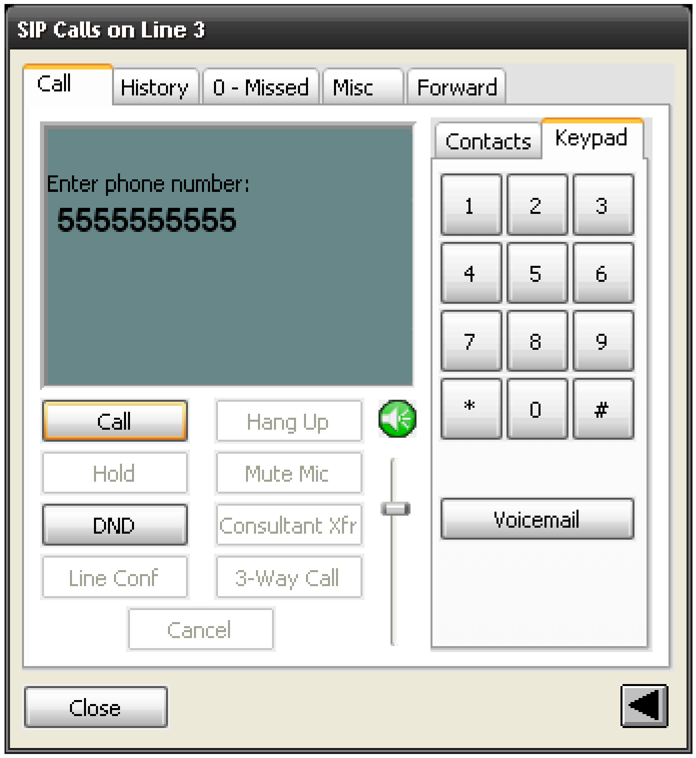

Figure 6: SIP Call Control Window: Keypad

Voicemail – Acts as a shortcut to a voicemail number. To configure and call a voicemail number, first right click on the Voicemail button and select Configure from the context menu. The Edit Contact window will appear. When finished with entering the voicemail number, press OK. Once the voicemail number is set up, clicking the button again will call the specified number.

1.2.3 SIP Call Control : Call History Page

The SIP call history page lists recent incoming and outgoing calls. The history list displays a description, time stamp, and phone number of each call. Incoming calls are colored red, while outgoing calls are blue as shown by the color-coded legend. To place a call directly from the history page, select an entry from the list and press the Call button.

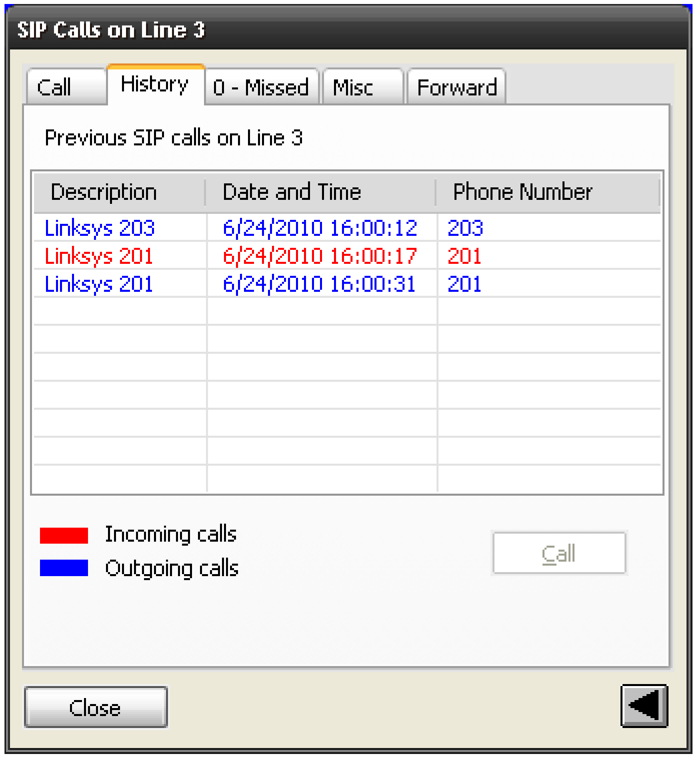

Figure 7: SIP Call Control Window: History Page

1.2.4 SIP Call Control : Missed Call History Page

While the SIP call history page keeps track of successfully connected calls, the Missed Call History page logs unanswered, ignored, forwarded, or denied due to Do Not Disturb. The Missed Calls History list contains a description, time stamp, and phone number of each missed call. Unanswered calls are colored blue, ignored calls are colored purple, calls denied due to DND are colored red, and forwarded calls are colored green as shown by the color-coded legend. To place a call directly from the missed call history page, select an entry from the list and press the Call button. To clear the list of all missed calls, press the Clear List Button. The Missed Call History page will only be available if SIP Enhancement license has been acquired.

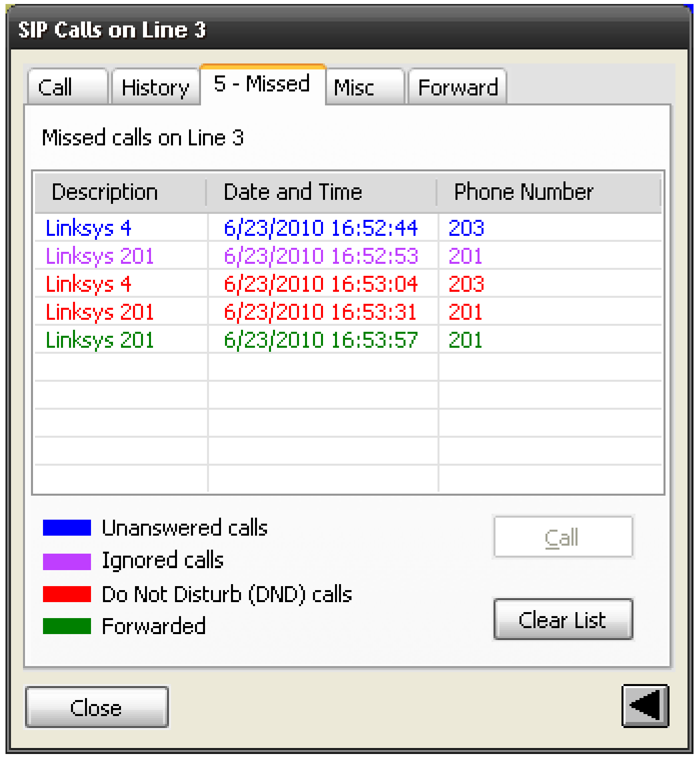

Figure 8: SIP Call Control Window: Missed Call History Page

1.2.5 SIP Call Control : Miscellaneous Page

The Miscellaneous page simply shows information pertaining to the connection to the SIP server including domain, protocol, and registration name.

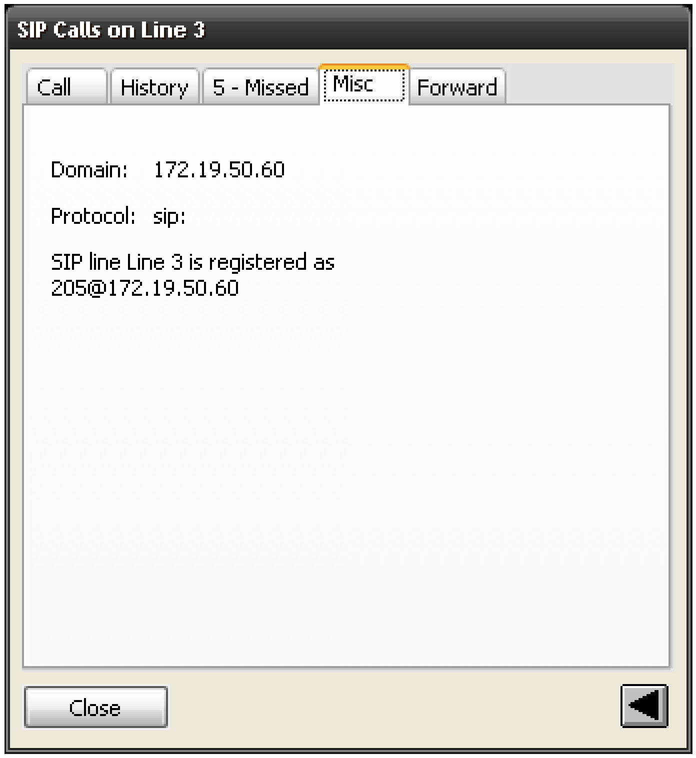

Figure 9: SIP Call Control Window: Miscellaneous Page

1.2.6 SIP Call Control : Forward Page

The Forward page provides access to call forwarding options. The three types of available forwarding are All, While Busy, and Unanswered. “Forward All” will automatically forward all incoming calls to the specified destination. “Forward while busy” will forward any incoming call while the phone line is already connected in another call. “Forward Unanswered” will automatically forward an incoming call after it has been ringing for the specified number of seconds.

To set up forwarding, check the checkbox next to the desired forwarding mode. Next enter a destination to forward the call to. Press the Address Book button to open an address book, allowing easier selection of the desired destination. When finished, be sure to press the “Apply” button the activate call forwarding. Note that selecting “Forward All” will disable the other two options, since their conditional trigger is superseded by “Forward All”. The Forward page will only be available if SIP Enhancement license has been acquired.

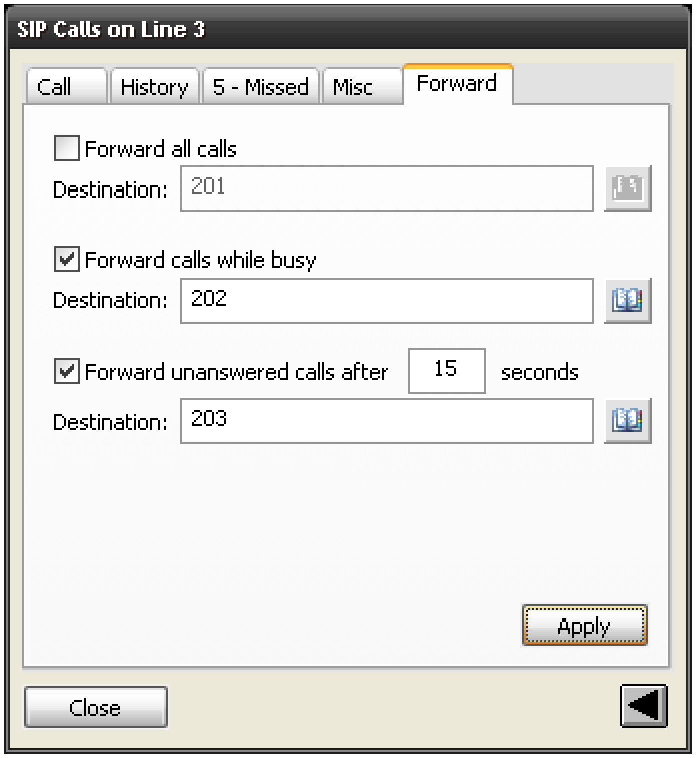

Figure 10: SIP Call Control Window: Call Forwarding Page

Operation Notes

1.2.7 Blind Transfer

A blind transfer immediately transfers the current call to a new number.

To perform a blind transfer, do the following steps while in a call:

1. If the Transfer button does not display “Blind Xfr”, right click the button and select “Blind Xfr” from the context menu. The button's text will change to “Blind Xfr”.

2. Press the Blind Transfer button.

3. Enter a destination number into the SIP Call window. Press the Blind Xfr again to transfer the call.

   The current call is transferred to the specified number.

1.2.8 Consultant Transfer

A consultant transfer puts the current call on hold, and places a call to a new number. When the consultant transfer button is pressed again, the call on hold is transferred to the new number. When finished, C-Soft is disconnected from the transferred call.

To perform a consultant transfer, do the following steps while in a call:

1. If the Transfer button does not display “Consultant Xfr”, right click the button and select “ Consultant Xfr” from the context menu. The button's text will change to “ Consultant Xfr”.

2. Press the Consultant Transfer button.

3. Enter a new destination number into the SIP Call window. Press the Consultant Xfr to place the call.

   The current call is put on hold, and a call is placed to the new number. When the call is answered, the following changes also occur:

   - *The Hang Up button changes its text to “Hang Up >”. When pressed, a popup menu appears to choose which party to hang up on.*

   - *When pressed, the Hold button places the secondary call on or off hold.*

4. When finished talking to the second party, press the Consultant Transfer button again.

   The on-hold call is transferred to the new number.

1.2.9 3-Way Call

A three way call puts the current call on hold, and places a call to a new number. When the 3-Way call button is pressed again, the call on hold joins the current conversation.

To perform a 3-Way Call, do the following steps while in a call:

1. Press the 3-Way Call button.

2. Enter a new destination number into the SIP Call window. Press the 3-Way Call button to place the call.

   The current call is put on hold, and a call is placed to the new number. When the call is answered, the following changes also occur:

   - *The Hang Up button changes its text to “Hang Up >”. When pressed, a popup menu appears to choose which party to hang up on.*

3. When desired, press the 3-Way Call button again.

   The on-hold call is taken off hold, and joined with the new call. Both parties can communicate with each other. The following changes also occur:

- *When pressed, the Hold button places both parties on or off hold.*
- *The 3-Way Call Button's text changes to "Leave Call".*

4. When finished, end the call in one of the following ways:
    - Hang up on one or both of the parties using the Hang Up button. This will hang up on all parties, disconnecting all participating calls.
    - Leave the 3-Way Call by pressing the "Leave Call" button. This will leave the primary and secondary parties in a call with each other.

1.2.10 Line Conference

Line conference is used to join two different SIP phone calls into the same conversation. Line Conferencing is very similar to a crosspatch. Once two calls are established, the two lines can be line-conferenced together so that both connected parties will be able to communicate with each other. In addition, a line that is already in a 3-Way call can be line-conferenced, and all 3-way call parties will be able to communicate with any other line-conferenced lines.

Due to processing demands, a limitation on the number of lines that can be concurrently mixed together via 3-way calling and line conferencing was placed. A maximum of six remote parties can be mixed together at one time. This means that six line-conferenced lines (6 calls * 1 remote party/call = 6), three line-conferenced 3-way calls (3 3-way calls * 2 remote parties per 3-way call = 6), or any combination of normal calls and 3-way calls are allowed.

To put two lines into a line conference, do the following steps:

1. Establish two calls on two different lines.
2. On each line, press the Line Conf button.

The borders on both SIP Call Control dialogs turn black. Both calls should be able to communicate with each other.

1.2.11 Call Waiting

If a line is currently in use, an incoming call on the same line will be treated as a call waiting line. If desired, the user can choose to answer the incoming call. The previous call is placed on hold. After accepting the incoming call, the can also choose to merge the calls together, effectively making a 3-way call between both parties.

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

1. While already in a call, receive an additional incoming call.

2. Press the “Answer” button.
The current call is put on hold, and the incoming call is answered. The following changes also occur:

- *The Hang Up button changes its text to “Hang Up >”. When pressed, a popup menu appears to choose which party to hang up on.*
- *The Cancel button text is changed to “Switch”.*

3. If desired, switch between the primary and secondary calls using the Switch button. The primary and secondary calls will toggle between connected/on hold and on hold/connected.
4. Press the 3-Way Call button.

The on-hold call is taken off hold, and joined with the other call. Both parties can communicate with each other, effectively forming a 3-way call.

The following changes also occur:

- *When pressed, the Hold button places both parties on or off hold.*

The 3-Way Call Button's text changes to “Leave Call”. See 3-Way call for more information on this operation.