



A Subsidiary of
Comtech Telecommunications Corp.

The background of the lower half of the page is a dark blue field with a network diagram. It consists of numerous small blue dots connected by thin white lines, forming a complex web. Several circular icons are integrated into the network: a location pin, a smartphone, a document with a checkmark, and a telephone handset. On the left side, there is a large, stylized 'S' inside a circle, which is part of a larger circular graphic with concentric rings and radial lines.

EMERGENCY SERVICES PLATFORM GUARDIAN USER GUIDE

Release 19.1

Confidential Proprietary Information

This document and the products and/or services described herein are the property of Solacom, a subsidiary of Comtech Telecommunications Corp. Disclosure in any form is prohibited, except as authorized.

C2214114-30 SR1

Copyright © 2019

Contents

Contents

- Legal ix
- Support and services..... x
- Safety summary xii
- 1 **About this document 1**
 - 1.1 Conventions..... 2
 - 1.2 References 2
- 2 **What’s new 3**
- 3 **Getting started 4**
 - 3.1 What is a Guardian answering position? 5
 - 3.2 Logging in and logging out..... 6
 - 3.2.1 Roaming account 7
 - 3.2.2 Alternate server..... 8
 - 3.2.3 Exit Guardian..... 10
 - 3.2.3.1 Disconnect calls before logging out..... 11
 - 3.3 Application basics..... 12
 - 3.3.1 Menu bar 13
 - 3.3.2 ALI tools 13
 - 3.3.3 Status bar..... 14
 - 3.3.4 Phone connection window 15
 - 3.3.4.1 One click is all it takes..... 16
 - 3.3.4.2 Context-activated..... 16
 - 3.3.4.3 Call alert while in other application..... 16

4

Areas of the interface	17
4.1 NG9-1-1 info.....	18
4.1.1 Emergency calls.....	19
4.1.2 Administrative calls.....	20
4.2 Caller information	21
4.2.1 Display ALI after call release.....	22
4.2.2 ALI plus PIDF-LO.....	23
4.2.3 Additional caller information	25
4.3 Phone connection.....	26
4.3.1 Automatic call expansion	28
4.3.1.1 Mute, hold, and privacy	29
4.3.1.2 Exclusive hold.....	32
4.4 Disconnect reasons	33
4.5 Make busy	34
4.5.1 Make busy reasons	36
4.5.2 Busy out warning.....	36
4.6 Direct access	37
4.6.1 Usage behavior	38
4.6.2 Transfers only.....	39
4.7 Line appearances	40
4.8 Position status.....	43
4.9 Smart pad	45
4.9.1 DTMF awareness.....	47
4.10 Dial pad.....	47
4.10.1 DTMF pad.....	49
4.11 Contacts.....	51
4.12 Ring group assignments	53
4.13 Call history.....	55
4.13.0.1 My calls.....	60
4.13.0.2 Specific day filter	61
4.13.0.3 ANI Filter	61
4.14 Telecommunicator call history.....	62
4.15 Redial list.....	66
4.16 Tenant status	69
4.16.1 Tenant status bar	69
4.16.2 Tenant status buttons.....	72
4.16.3 Current emergency/admin calls	72
4.16.3.1 Call monitoring.....	74
4.16.4 Emergency/admin calls on hold	76
4.16.5 Waiting emergency/admin calls	78
4.16.6 Abandoned emergency calls.....	80
4.16.6.1 Abandoned call replacement.....	80
4.16.7 Active telecommunicators	82

4.16.7.1	Telecommunicator monitoring	84
4.17	Tenant dashboard	86
4.18	Procedural messages.....	88
4.19	Ticker message bar	89
4.20	Local text conversations.....	90
4.21	Websites.....	91
4.22	Instant replay recorder	92

5	Emergency calls	96
5.1	Answering calls	97
5.1.1	Automatic hold on answer.....	97
5.2	Automatically answered calls	97
5.2.1	Multiple automatically answered calls	98
5.2.1.1	Wrap up	98
5.2.2	Automatic greeting.....	99
5.3	Abandoned emergency calls.....	101
5.3.1	Queued abandoned calls	101
5.3.2	Disconnected calls	103
5.4	Conference/transfer calls.....	104
5.4.1	Selective transfer agency.....	104
5.4.1.1	STA transfers	107
5.4.1.2	SIP REFER.....	109
5.4.2	Fixed transfer codes	111
5.4.2.1	FTC transfers.....	112
5.4.2.2	SIP REFER.....	114
5.4.3	Manual multiple-party conference.....	116
5.4.4	Direct access.....	120
5.4.4.1	Alternate phone number	121
5.4.4.2	Conference mode.....	121
5.4.4.3	Alternative behavior	122
5.4.4.4	Quick transfer mode.....	123
5.4.5	Contacts	124
5.4.5.1	Alternate phone number	124
5.4.6	Manually dialing	125
5.4.6.1	Smart pad.....	125
5.4.7	Automated answering system selections.....	127
5.4.8	Conference another telecommunicator	128
5.4.9	Joining calls.....	130
5.5	Releasing calls	131
5.6	Working with the ALI.....	132
5.6.1	Re-query the ALI.....	133
5.6.2	Manual ALI Retrieval	133

5.6.3	Making ALI corrections	135
5.6.3.1	Email ALI corrections	137
5.6.3.2	Print ALI	137
5.6.4	Supplemental ALI notes	138
5.6.4.1	Adding note on conferenced call	140
5.6.5	Call notes	141
5.7	Caller history	146

6 Administrative calls 148

6.1	Answering calls	149
6.1.1	Automatic hold on answer	149
6.1.2	Automatic call answer	149
6.2	Conference/transfer calls	150
6.2.1	Direct access buttons	150
6.2.1.1	Alternate phone number	151
6.2.1.2	Conference mode	151
6.2.1.3	Alternative behavior	152
6.2.1.4	Quick transfer mode	153
6.2.2	Contacts	153
6.2.2.1	Alternate phone number	153
6.2.3	Manual multiple-party conference	155
6.2.3.1	Smart pad	155
6.2.3.2	Dial pad	157
6.2.4	Automated answering system selections	159
6.2.5	Conference another telecommunicator	160
6.2.6	Joining two calls	162
6.3	Releasing calls	163
6.4	Placing call	164

7 Enhanced admin calls 166

7.1	Answering calls	167
7.1.1	Automatic hold on answer	167
7.1.2	Automatic call answer	167
7.2	Conference/transfer calls	168
7.2.1	SIP REFER	168
7.2.2	Direct access buttons	169
7.2.2.1	Conference mode	170
7.2.2.2	Alternative behavior	172
7.2.2.3	Quick transfer mode	173
7.2.3	Contacts	173
7.2.4	Manual multiple-party conference	175
7.2.4.1	Smart pad	175

	7.2.4.2	DTMF pad.....	177
	7.2.5	Automated answering system selection.....	180
	7.2.6	Conference another telecommunicator	180
	7.2.7	Joining two calls.....	183
7.3		Releasing calls	183
7.4		Placing call.....	184

8 Text conversations 186

8.1		Text conversations.....	187
	8.1.1	Customize font and button size	188
	8.1.2	Sessions list.....	190
8.2		Text-to-911	192
	8.2.1	Mute, hold, and privacy	193
	8.2.2	Conference call on same Guardian system	195
	8.2.3	Conference call to different system	197
	8.2.4	Conference text-to-911 call.....	199
	8.2.5	Disconnected text-to-911 call	200
8.3		Hearing impaired calls	202
	8.3.0.1	Voice carry over mode	203
	8.3.0.2	Hearing carry over mode.....	204
	8.3.1	Conference hearing impaired calls.....	205
	8.3.1.1	Conference T/C to hearing impaired call	205
	8.3.2	DTMF tone detection	206
8.4		Instant messaging	207
	8.4.1	Status	210
	8.4.2	Personal message.....	211
	8.4.3	Transfer files	213

9 User tools..... 216

9.1		Audio source.....	217
9.2		Change password	217
	9.2.1	Password rules.....	218
9.3		Color themes	219
9.4		PAC II Settings.....	220
	9.4.1	Auxiliary audio selection.....	220
	9.4.2	Push-to-signal	221
	9.4.2.1	PTT mode.....	222
	9.4.2.2	PTM mode.....	222
9.5		Personal contacts.....	224
	9.5.1	Alternate phone number.....	225
9.6		Personal greetings.....	228
9.7		Preferences.....	231

9.8	Share IRR.....	233
9.8.1	Send recordings.....	233
9.8.1.1	Search recordings.....	234
9.8.1.2	Recipient list.....	236
9.8.2	Receive recordings.....	239
9.9	Volume.....	241
9.9.1	Volume meter.....	243
10	Guardian alarm notification	246
10.1	Server switchover.....	247
10.1.1	Both servers operational.....	247
10.1.2	Communication failure on primary server.....	247
10.1.3	Primary server failure.....	248
10.1.4	Deliberate switchover.....	249
10.2	Tenant status alarms.....	249
10.3	Ticker message alarms.....	250
10.4	Notifications.....	251
10.5	Audible alarms.....	253
10.6	PSAP make busy warning.....	254
10.6.1	Log in when busied out.....	254
10.7	ALI status.....	256
A	Acronyms.....	257
B	TDD abbreviations	260
C	Icon glossary	269
D	Release history	272
	Release 19.0.1.....	272
	Release 19.0.....	272
	Release 18.1.....	273
	Release 18.0.....	273
	Release 17.3.....	273
	Release 17.2.....	274
	Release 17.1.....	274
	Release 17.0.....	274
	Release 16.1.....	275
	Release 16.0.....	275
	Release 15.1.....	277

Release 15.0 277

Release 14.1 277

Release 14.0 278

Release 13.0 278

Release 12.2.2 279

Release 12.2 279

Release 12.1 280

Release 12.0 281

Release 11.0.0.1 282

Release 11.0 283

Release 10.2 284

Release 10.1 285

Release 10.0 286

IX **Index 288**

Legal

The information furnished in this document by Solacom, a subsidiary of Comtech Telecommunications Corp. (hereinafter referred to as Solacom), is believed to be accurate. Solacom makes no warranties, expressed or implied, regarding the information contained herein, and assumes no liability for errors or omissions.

Solacom assumes no liability otherwise arising from the application or use of any such information or product for any infringement of patents or other intellectual property rights owned by others that may result from such application or use.

Neither the supply of this information or the purchase of product conveys any license, either expressed or implied, under patents or other intellectual property rights owned by Solacom or licensed from third parties by Solacom, whatsoever.

Purchasers of product are also hereby notified that the use of this information or product in certain ways or in combination with Solacom or non-Solacom furnished goods or services may infringe patents or other intellectual property rights.

This document and related products contain valuable trade secrets and proprietary information belonging to Solacom. No part of this document may be reproduced, stored in a retrieval system, or transmitted in any form or by any means without prior written permission from Solacom. Reverse engineering, decompiling and disassembling are explicitly prohibited.

The product specifications and information contained in this document are subject to change by Solacom without notice.

Support and services

Support and services

The Solacom Customer Support & Services Center is staffed by qualified service professionals to provide the support, service, and know-how you need to seamlessly operate and manage your mission-critical communications infrastructure.

The Center offers a complete range of support and services, including:

- Answering queries and providing information.
- Helping to diagnose and troubleshoot incidents.
- Dispatching qualified field support personnel for training, installation, configuration, QoS audits, personalized requirements, and more.

You may request support and service online or by telephone:

- Web: services.solacom.com
- Email: services@solacom.com
- North America toll free: 1-888-solacom (1-888-765-2266)
- Telephone anywhere: +1-819-205-8100

Normal business hours are Monday to Friday from 09:00 to 17:00 EST.

For emergency or urgent requests, 7 days per week, 24 hours per day, use the telephone numbers above and follow the voice prompts.

Commenting on the documentation

To provide corrections and suggestions on the documentation, send your comments by email to documentation@solacom.com. When possible, please include specific information about the document for which you are commenting; for example, this document part number (Document ID: C2214114-30 SR1) and section name or number.

Safety summary

Safety summary

The following general safety precautions must be observed during all phases of operation, service, and repair of this product. Failure to comply with these precautions or with specific warnings elsewhere in this manual violates safety standards of design, manufacture, and intended use of the product. Solacom assumes no liability for the customer's failure to comply with these requirements.

Ground the equipment

To minimize shock hazard, the equipment switch must be connected to an electrical ground. The equipment has a three-conductor AC power cable. This power cable must be plugged into an approved three-contact electrical outlet with the grounding wire (green) firmly connected to an electrical ground at the power outlet. The power cables meet International Electrotechnical Commission (IEC) safety standards.

Keep away from live circuits

Operating personnel must not remove modules or otherwise tamper with the equipment switch or related components. Component replacement and internal adjustments must be made by qualified maintenance personnel. Do not replace components with the power cable connected unless the component is hot-swappable. To avoid injuries, always disconnect power and discharge circuits before removing equipment shelves or making major modifications.

Do not service or adjust alone

Do not attempt major component replacement, internal service, or adjustment unless another person, capable of rendering first aid and resuscitation, is present.

Replacement of modules

Replacement of modules should be performed by qualified maintenance personnel (those who have been trained and certified by Solacom).

Electrostatic discharge (ESD)

This product contains components that can be damaged by electrostatic discharge. You must take precautions when handling modules. As a minimum precaution, use grounded wrist straps at all times when handling modules.

1 About this document

This user guide describes the typical tasks carried out by a user of the Guardian client application stationed at a Guardian answering position. Guardian users may be PSAP telecommunicators, supervisors, or administrators. This document is of interest to you if you are using the application to train as, or in the normal day-to-day activities of, a PSAP telecommunicator. The guide describes the features of the Guardian client application, its interface, and call handling.

1.1 Conventions

Throughout this document, the terms “user” and “telecommunicator” are used to refer to an individual interacting with the Guardian answering position in a configured role or layout, regardless if their function may be a telecommunicator, agent, or dispatcher. The terms “administrator” and “supervisor” refer to anyone with additional application privileges. The term “you” can refer to any user of the application (telecommunicator, administrator, etc.), and should be considered in context with the intended audience of each section of the document.

Emergency calls refer to calls originating from emergency 9-1-1 trunks and to AdminEmergency calls which arrive on administrative phone lines but are treated as emergency calls when they are processed by the Guardian system.

Administrative calls refer to non-emergency calls on phone lines originating from a PBX, central office, or another user. The terms “administrative” and “admin” are used interchangeably throughout the user guide to refer to these calls.

The terms “trunks” refer to incoming emergency circuits and “lines” refer to administrative phone lines for incoming and outgoing calls. The term “position” is used to refer to the Guardian answering position running the Guardian client application.

The Guardian application supports both touchscreen and mouse-keyboard interactions. To avoid confusion, a decision was made to consistently describe the tasks in this document as involving the “clicking” of items on the screen. Some useful alternative methods, such as clicking with the right mouse button or using the numeric keypad on a keyboard, are also mentioned occasionally. If your interface comprises only a touchscreen device, all clicking actions should be simply interpreted as pressing.

1.2 References

Some common industry acronyms are used throughout this document. For reference, *Appendix A, “Acronyms”* contains a list of the acronyms used and their expanded form.

For detailed information describing the features and configuration of Guardian, see the *Guardian Administration Guide*.

For information on configuring the Guardian hardware and software, see the *Guardian Position Setup Guide* and the *ESP Configuration and Maintenance Guide*.

2 What's new

This describes the new features with improved performance introduced in Release 19.1 to meet your growing needs. For a description of new and changed features in previous releases of Guardian, refer to *Appendix D, "Release history"*.

SHARE IRR RECORDINGS BETWEEN POSITIONS

Provide the ability to share an IRR recording, including metadata, from a Guardian position to one or more answering positions within the same tenant. Only users with the required privilege(s) can send and/or receive recordings. A notification is sent to the Notifications window, from which the recording can be played and metadata viewed, to notify a user a recording has been received.

ALERT TONE FOR NEW TEXT MESSAGE

A zip tone can be configured to play a short tone notifying a user that a new message has been received in a text session when the user's focus is not in the Text Conversation window.

ALERT TONE WHEN TENANT THRESHOLD IS TRIGGERED

An alarm tone can be configured to play notifying a user that either a duration or count tenant threshold alarm has been triggered. The alert can be configured to play to a specific role or role group. The alert can be configured per threshold setting.

3 Getting started

This chapter gets you started with an introduction to Guardian, its interface and common features, and logging in and out of the Guardian client application. Although the graphical user interface (GUI) is completely customizable by your administrator there are features common to a call handling application.

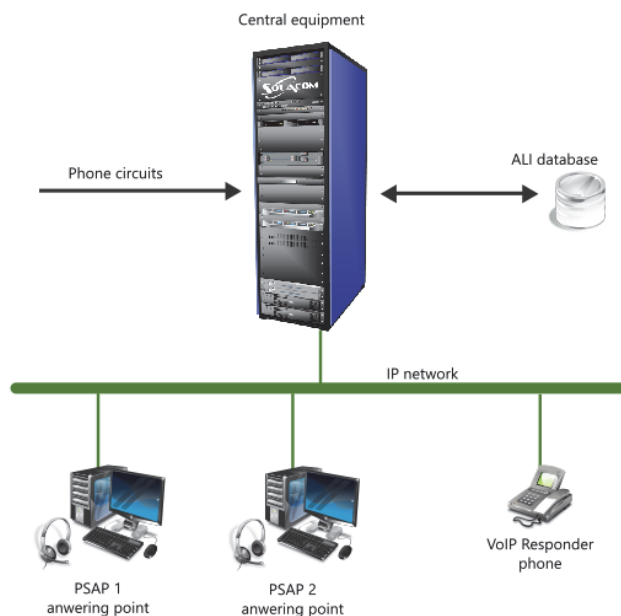
3.1 What is a Guardian answering position?

The Guardian application is a next generation 9-1-1 (NG9-1-1) call taking application for public safety answering point (PSAP) solutions. Next generation 9-1-1 is an updated 9-1-1 service infrastructure to address the growing wireless mobile population as specified by the National Emergency Number Association (NENA).

In a Guardian system, users are stationed at answering positions running the Guardian client application on desktop computers, which communicate with the central equipment (CE) and other answering positions via an IP network. Using an IP network allows positions to be located remotely from the central equipment and even allows users to log in at another PSAP hosted on the same CE and receive the layout and call handling capabilities of their own PSAP.

Figure 3.1 depicts a basic system with some optional external connections including an ALI database and VoIP Responder phone.

Figure 3.1 Guardian system



3.2 Logging in and logging out

Logging in to the system from a position authenticates your account, assigns your role, starts the logging of events, and loads your current role's assigned layout.

Depending on the system configuration, the log in dialog box may display different information. For example, a log in dialog box for a system with multiple servers (redundant) displays an **Options** button whereas a log in dialog box for a system with one server (simplex), does not display this parameter.

Log in at your PSAP

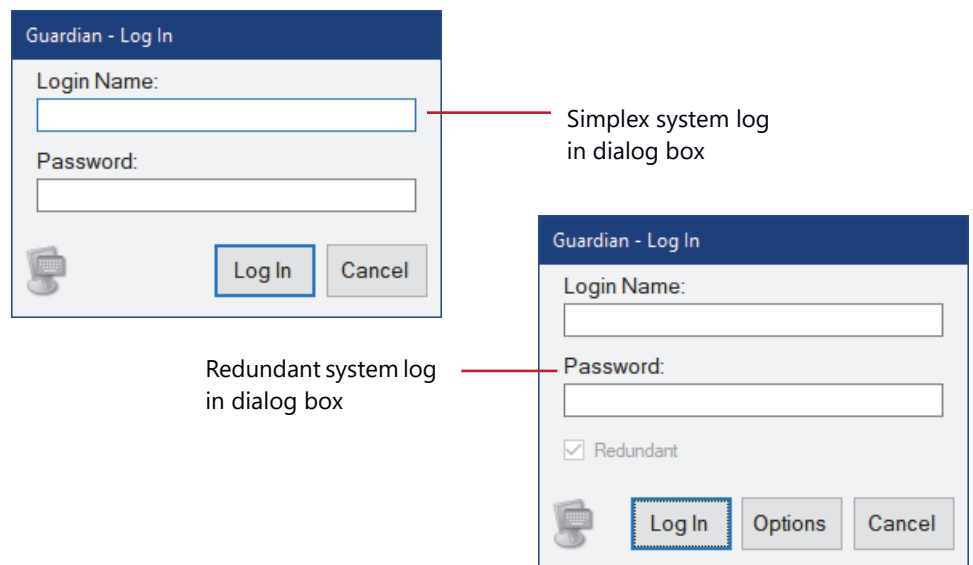
Step 1 Double-click the **Guardian** icon on the desktop.

Step 2 In the **Guardian - Log In** dialog box, enter your **Login Name** and **Password**.

More info: Both entries are case sensitive.

Step outcome: Depending on the equipment of your Guardian system, either the simplex dialog box or the redundant dialog box is displayed on your screen.

Example:



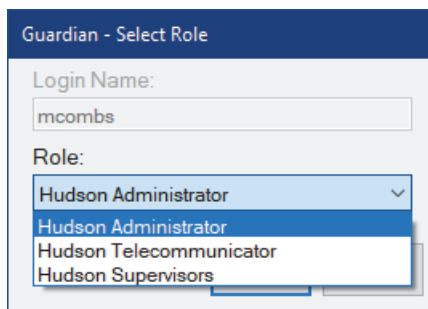
More info: If your system is configured with centralized login, you are authenticated by the network login mechanism and do not need to provide a login name or password.

Step 3 Click **Log In**.

Step outcome: If your account is not assigned to multiple roles, the main Guardian application window is displayed.

Step outcome: If your account is assigned to multiple roles, the **Guardian - Select Role** dialog box is displayed and you need to select the role to use for the current session.

Example:

**Step 4** In the **Role** list, select the role you want to use for the current session.**Step 5** Click **Log In**.

3.2.1 Roaming account

When logging in with an account configured for roaming, you can log in to a roaming enabled position from any PSAP in the system while maintaining the layout and call handling capabilities configured for your own PSAP - this includes any outbound call and call back prefixes configured for your PSAP.

Log in using roaming account

Step 1 Double-click the **Guardian** icon on the desktop.**Step 2** In the **Guardian - Log In** dialog box, enter your **Login Name** and **Password**.

More info: Both entries are case sensitive.

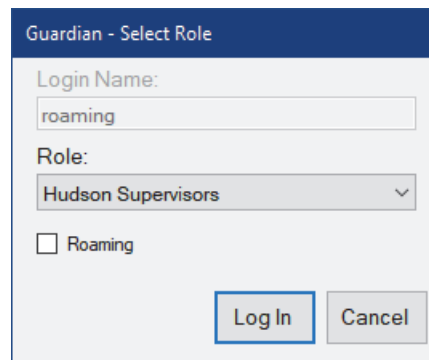
Step 3

Click **Log In**.

More info: The **Roaming** option is visible.

Step outcome: If your user account is assigned to multiple roles, the **Guardian - Select Role** dialog box is displayed and you need to select the role to use for the current session.

Example:



Guardian - Select Role

Login Name:
roaming

Role:
Hudson Supervisors

☐ Roaming

Log In Cancel

Step 4

Select the **Roaming** option.

More info: Your administrator may have configured the system to have the **Roaming** option preselected for your role. In this case skip step 4.

Step 5

Click **Log In**.

Note: The **Roaming** option is displayed when roaming is configured for both the user account and position.

3.2.2 Alternate server

From time to time it is necessary for system administrators to upgrade server applications or perform system maintenance. The system administrators may request you to log into another server. The following procedure is used only when requested by your system administrator as logging into a server other than the default server is service-affecting.

Only use this procedure when instructed.

Log in to alternate server

Step 1 Double-click the **Guardian** icon on the desktop.

Info If your system is configured with centralized login, you are not presented with the Options button. Contact your system administrator to disable centralized login.

Step 2 In the **Guardian - Log In** dialog box, enter your **Login Name** and **Password**.

More info: Both entries are case sensitive.

Step 3 Click **Options**.

Step outcome: The **Log In Options** dialog box is displayed.

Example:

The dialog box titled "Log In Options" has a blue header. It contains two radio buttons: "Redundant" (selected) and "Force Simplex". Below "Force Simplex" is a group box containing two radio buttons: "Primary" (selected) and "Secondary". To the right of these are IP addresses: "192.168.28.103:50077" for Primary and "192.168.28.203:50077" for Secondary. At the bottom are "OK" and "Cancel" buttons.

Step 4 Select **Force Simplex**.

Step outcome: The primary and secondary options are displayed with the IP address of each server.

Step 5 Select the server as directed by your system administrator.

Example:

The dialog box titled "Log In Options" has a blue header. It contains two radio buttons: "Redundant" and "Force Simplex" (selected). Below "Force Simplex" is a group box containing two radio buttons: "Primary" and "Secondary" (selected). To the right of these are IP addresses: "192.168.28.103:50077" for Primary and "192.168.28.203:50077" for Secondary. At the bottom are "OK" and "Cancel" buttons.

Step 6 Click **OK**.

Step outcome: You are returned to the **Guardian - Log In** dialog box.

Step 7 Click **Log In**.

More info: If your account is assigned to multiple roles, you are presented with the **Guardian - Select Role** dialog box and need to select the role to use for the current session.

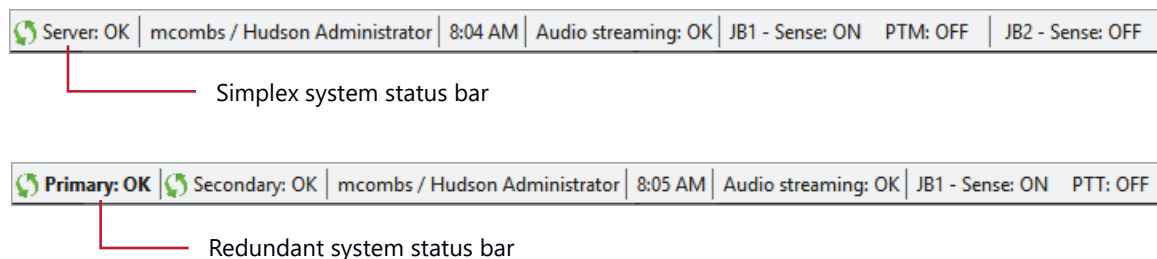
Step 8 In the **Role** list, select the role you want to use for the current session.

Step 9 Click **Log In**.

Confirm logged in server

When you have completed the log in process, the status bar at the bottom of the application window indicates to which server the position is connected, as shown in Figure 3.2.

Figure 3.2 Simplex and redundant server status bars



The server name that appears in **bold** is the server that your position is connected to currently.

3.2.3 Exit Guardian

Depending on your layout, there may be more than one method to exit Guardian:

- 1 A log out entry is included on the **File** menu.
- 2 Your administrator placed the **Log Out** button in your layout.

3 You may have a **Close** button in the upper-right corner of the Guardian window.

It is recommended to use either option 1 or 2 listed above and avoid option 3, when possible.

Log out

Step 1 On the **File** menu, select **Log Out**.

More info: If available, click the **Log Out** button in the layout.

Step 2 In the **Guardian Logout** confirmation dialog box, click **Yes**.

More info: If you have made layout changes in the application window, you may want to save these settings before logging out.

Info

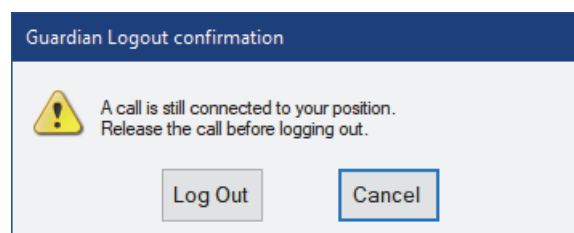
If your role allows you to make changes to the layout, save the changes before logging out or the changes are lost.

3.2.3.1 Disconnect calls before logging out

Always verify that there are no calls connected to your position before logging out (look for any entries in the **Phone Connection** window). Depending on the activity at your PSAP, it is possible that a call placed on hold was forgotten or possibly your 9-1-1 caller abandoned the call. These calls are rerouted, per your PSAP call flow, if you were to log out while they are still connected.

To prevent these scenarios from occurring, a different confirmation dialog box is displayed when you choose to log out as shown in Figure 3.3.

Figure 3.3 Guardian logout confirmation window

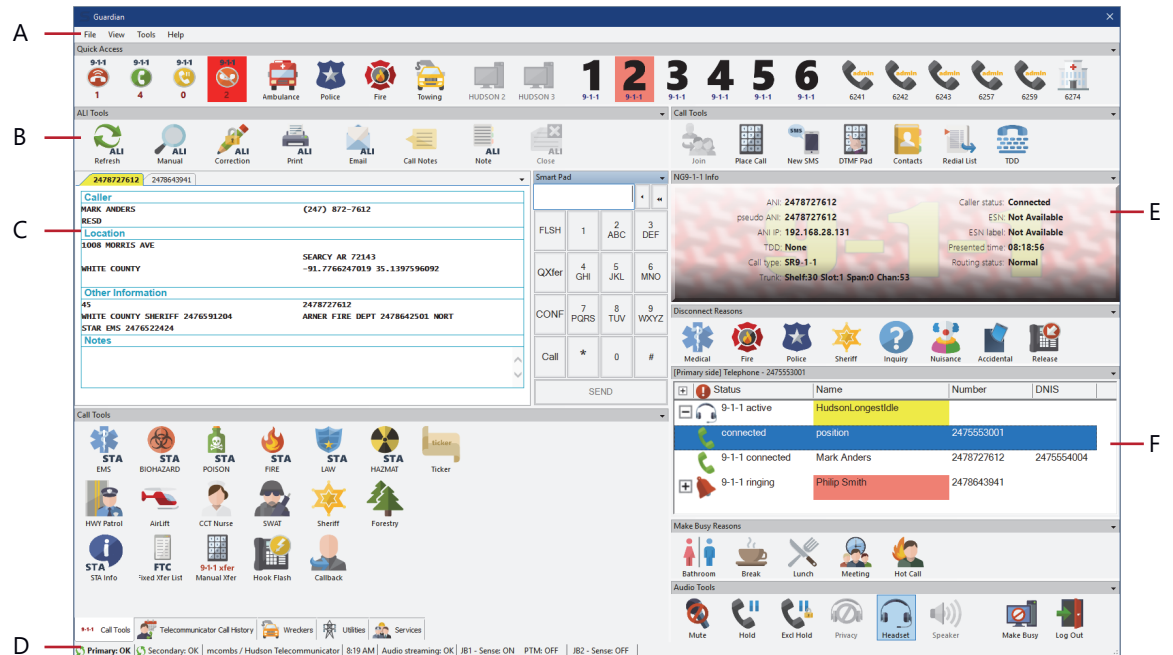


To log out, click **Cancel**, release any connected calls, and choose to log out again.

3.3 Application basics

Guardian is a highly customizable application. Although your PSAP layout is likely to be customized, the interface features in Figure 3.4 are common to most call taking layouts.

Figure 3.4 Sample Guardian layout



Interface features:

A. Menu bar **B.** ALI tools **C.** Caller Information window **D.** Status bar **E.** NG9-1-1 Info window (Answer button) **F.** Phone Connection window

Assigned role privileges determine the windows and other resources that are available to each user, and how they are laid out in the application space.

With full privileges assigned, each window can be resized by dragging the edges, moved by dragging the title bar, floated, docked, and hidden. Toolbars can also be moved to any location, but it is recommended to keep them within the window they are suited to. The layout can then be saved to the server by user and/or by role. Refer to the *Guardian Administration Guide* for details on layout configuration.

The columns in the **Phone Connection** and **Contacts** windows can be resized without special privileges.

3.3.1 Menu bar

The Guardian menu contains four entries: **File**, **View**, **Tools**, and **Help**. Throughout this guide, when you are required to access a feature from one of the four menus, you will see the menu entry identified as:

Menu: <menu_name> > <feature>

The menu entries provide access to all the features available to your current role.

Tip: If you close a window that was docked in your layout, you can always re-open the window by launching the feature from the **View** menu.

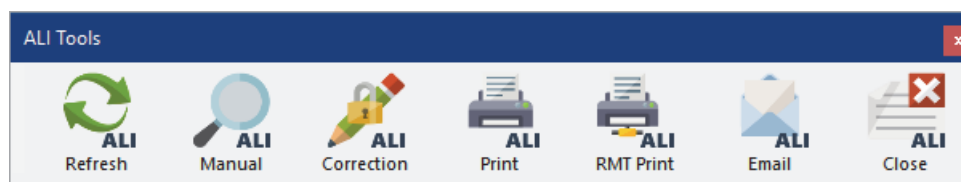
3.3.2 ALI tools

The **ALI Tools** window (see Figure 3.5) contains a direct access toolbar with a set of buttons that you can use to interact with the ALI displayed in the active tab of the **Caller Information** window.

If **ALI Tools** is not visible in your layout, launch it from the **View** menu as a floating window.

Menu: **View** > **ALI Tools**

Figure 3.5 Available buttons in the ALI Tools window



The following table describes the ALI related features that can be placed as buttons on the **ALI Tools** toolbar. The toolbar can be customized to display a subset of these buttons.

Table 3.1 ALI Tools action buttons

Button	Description
Close	Allows you to manually remove the ALI for the last call if your role is configured to display the ALI for the last call released.
Correction	Allows you to write ALI corrections right on the screen. Corrections can be printed or sent via email to a configured recipient.
Email	Allows you to send ALI corrections to a configured recipient via an email.
Manual	Allows you to manually query the ALI provider with a valid 10-digit ANI.
Print	Prints the ALI displayed in the Caller Information window to a (local) printer available to the answering position.
Refresh	Re-queries the ALI provider with the ANI to determine if more current information is available to display in the Caller Information window. Your administrator can configure a time period in-between refresh attempts. The button remains disabled during this time period.
RMT Print	Prints the ALI displayed in the Caller Information window to a remote printer connected to the Solacom servers.

3.3.3 Status bar

The status bar (see Figure 3.6) located at the bottom of the Guardian window contains information on the following:

- Health of the application server(s)
- User name and role that you are currently logged in with position's current time
- Status of Guardian call control interface (GCCl) if enabled
- Health of the audio connection between the position and the server
- Position busied out status
- Status of audio sources from position audio control (PAC) module if connected to the position

- Status of automatic greetings if enabled

Figure 3.6 Sample status bar

Server: OK	mcombs / Hudson Telecommunicator	8:39 AM	GCCI: DOWN	Audio streaming: OK	Busied Out	JB1 - Sense: ON	PTT: OFF	JB2 - Sense: OFF	PT
Server health	User name and role		GCCI status	Audio health	Busied out status			PAC I/II status	

3.3.4 Phone connection window

The **Phone Connection** window is one of the most important windows of Guardian and is further explained in *Phone connection* on page 26. The phone number of your position is displayed in the title as shown in Figure 3.7.

If **Phone Connection** is not visible in your layout, launch it from the **View** menu as a floating window.

Menu: View > Phone

Figure 3.7 Entry for active emergency call in the Phone Connection window

[Primary side] Telephone - 2475553001			
Status	Name	Number	DNIS
9-1-1 active	HudsonLongestIdle		
connected	position	2475553001	
9-1-1 connected	Mark Anders	2478727612	2475554004

3.3.4.1 One click is all it takes

The Guardian interface operates on single-click actions. Since the interface can be utilized with touchscreen monitors, the interface was designed to operate with single clicks. When interacting with buttons in your layout remember to click the buttons once only.

Note: Double-clicking the buttons may lead to unexpected results.

3.3.4.2 Context-activated

Buttons and features found in your layout are designed to be context-activated. The button or feature only becomes enabled when the action is possible. For example, the **Release** button remains disabled if there is no active call to release.

3.3.4.3 Call alert while in other application

If your administrator has enabled the Taskbar Alert Indicator feature, a visual alert is presented when a new incoming call arrives at your position and you have moved your focus to an application other than Guardian. The **Guardian** icon on the Windows status bar changes color and flashes until focus is returned to Guardian. In addition, the title bar of the Guardian application flashes until it is in focus.

The alert stops when focus is returned to the Guardian application or the call is answered.

4 Areas of the interface

This chapter describes the various feature windows available in the Guardian interface and the information provided by each feature. In addition, it describes the interactions available within each window.

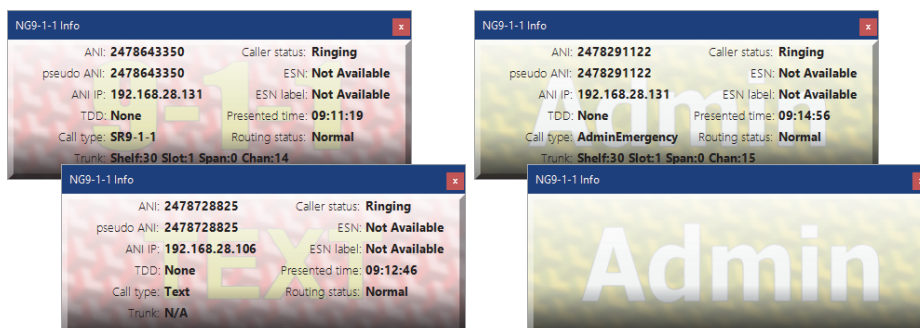
4.1 NG9-1-1 info

The **NG9-1-1 Info** window identifies the type of incoming call (emergency, text-to-911, AdminEmergency, or administrative), as shown in Figure 4.1, and behaves as an answer button, allowing you to answer an incoming call with a single-click on the window. In addition, for emergency calls, it displays the ANI of the caller and related incoming 9-1-1 call information.

If **NG9-1-1 Info** is not visible in your layout, launch it from the **View** menu as a floating window.

Menu: View > NG9-1-1 Info

Figure 4.1 Available NG9-1-1 Info windows



Ringing emergency, text-to-911, AdminEmergency, and administrative calls as they appear on NG9-1-1 Info

The **NG9-1-1 Info** window also behaves as a position status indicator. The window indicates if the position is idle, busied out, ringing, active, or in wrap-up mode via the background color and the text displayed.

4.1.1 Emergency calls

The Guardian system distinguishes between three types of emergency calls:

- 1 9-1-1
- 2 Text-to-911
- 3 AdminEmergency

A 9-1-1 call is an emergency call that has been directly routed from a central office, VoIP or wireless provider, or via a selective router (SR) on dedicated emergency phone trunks.

A text-to-911 calls is an emergency call that has been sent via SMS from a caller using a mobile phone.

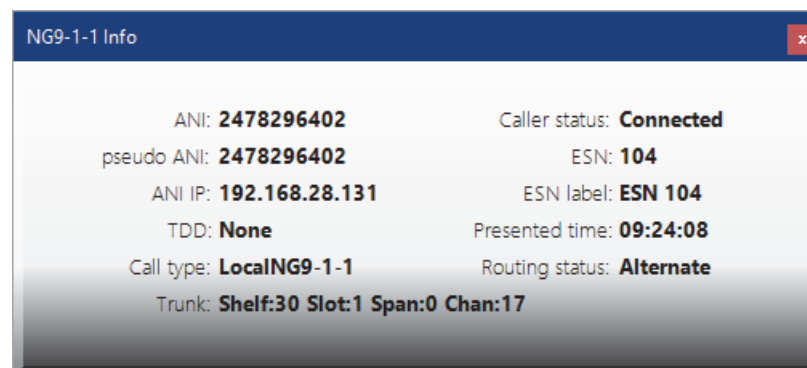
An AdminEmergency call is an incoming administrative call that has been configured to behave as an emergency call once it is received by the Guardian system and can be processed as a regular emergency call with all the same behaviors.

- For example, if your PSAP receives frequent emergency call transfers from a local agency (law, fire, security, and others), handling of these calls can be expedited by having the call treated as AdminEmergency since ALI is included and caller information is preserved if the call becomes disconnected.
- The availability of AdminEmergency configuration is dependent on jurisdictional laws and may not be available for your PSAP.

Note: Throughout this guide, both types of voice calls are referred to as emergency.

The **NG9-1-1 Info** window displays the ANI, pANI, and other information derived from the caller's phone number, see Figure 4.2.

Figure 4.2 Sample NG9-1-1 Info window for emergency call



Note: The columns displayed in the **NG9-1-1 Info** window may have been customized by your administrator and may not be exactly as shown in Figure 4.2.

The **Caller Status** states whether the call is one of the following:

- Ringing
- Abandoned
- Connected

The **Routing Status** states whether the call is:

- Normal – designated for your PSAP
- Overflow – a previous route from a Solacom SR was busy or congested
- Alternate – a previous route failed or was busy
- Transfer – the call was answered and transferred to your PSAP

Info If you are on an active call, the NG9-1-1 Info window only displays information about the incoming call after you answer the call.

If NG9-1-1 information is not included for your role, the window does not display information about the incoming call and only indicates the type of call. The window still behaves as an **Answer** button.

4.1.2 Administrative calls

The **NG9-1-1 Info** window flashes “Admin” when an administrative call arrives at your position. However, no information is derived from the caller’s phone number for administrative calls.

4.2 Caller information

The **Caller Information** window displays one or more forms containing caller information retrieved from the ALI or PIDF-LO databases as each incoming emergency call is received or answered. Each call is associated with its own tab; the tab corresponding to the active call is highlighted in yellow. The tabs can be arranged within **Caller Information**, but the window itself cannot be moved.

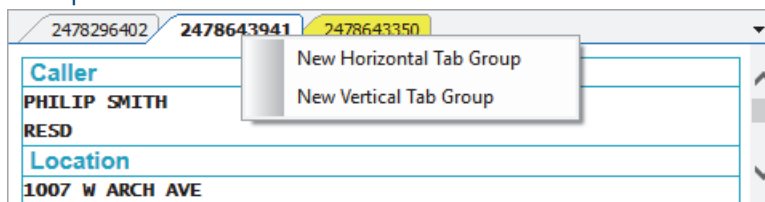
The appearance of the ALI or PIDF-LO may have been customized by your administrator but it includes common data such as:

- Automatic number identification (ANI) of the caller
- Callback phone number
- Address information for non-wireless calls
- Emergency service number (ESN) and selective transfer agencies (STA)
- Latitude and longitude for wireless calls

Arrange ALI tabs

Step 1 Right-click a tab and click **New Horizontal Tab Group** or **New Vertical Tab Group**.

Example:

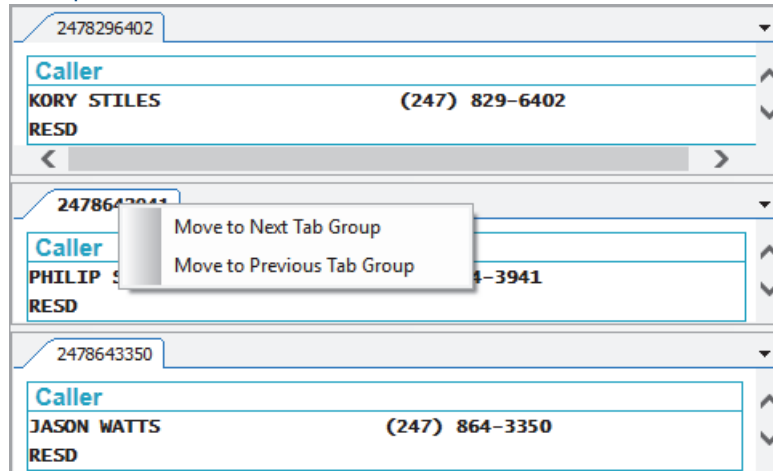


Step outcome: The form is tiled into a new tab group.

Step 2

Right-click a tab and click **Move to Next Tab Group** or **Move to Previous Tab Group**.

Example:



Step outcome: The forms are re-combined into an existing tab group.

More info: Combining all the tab groups results in a full-window form.

Step 3

Drag a tab over another tab to rearrange the order within a tab group.

Info

The right-click context menu is only available when more than one emergency call has been answered and is only applicable for the duration of the calls.

4.2.1 Display ALI after call release

The information for the last call answered at your position may be configured to remain displayed in the **Caller Information** window after you release the call. It remains displayed on your screen until you answer a new call when it is replaced with the ALI for your current caller. If configured, your administrator may have also included the **Close** ALI toolbar button in your layout to remove the ALI for the last call on your screen. If the **Close** button is not included in your layout, the ALI from the last call remains displayed until it is replaced with the ALI from a new call you answered - it is not replaced with the ALI from a call that was ringing at your position but which you did not answer.

Remove ALI data from last call

Your administrator must have included the **Close** ALI button in your layout.

Step Click **Close**.

Result of this task: The ALI from the last call is removed and the **Caller Information** window is cleared.

4.2.2 ALI plus PIDF-LO

If your PSAP supports PIDF-LO data, your system administrator may modify your ALI presentation to display the PIDF-LO data. PIDF-LO data contains only civic addresses or geodetic information in relation to your call.

Since PIDF-LO data contains less information than standard ALI data, both values are presented on your screen when a call arrives or is answered. The ALI plus PIDF-LO format includes the legacy ALI data in addition to extra buttons that allow you to view the PIDF-LO data in a separate screen.

Display ALI data

Step Click **ALI** to view legacy ALI data.

Example:

ALI	Location	Notes
Caller Carol Polks RESD		
Location 100 E EDGEWOOD DR DURH NC 27704		
Other Information 057 (247) 238-3257 LAW 247-374-9214 FIRE 247-568-4597 EMS Durham Station 4		
Notes 		

Display PIDF-LO data

Step

Click **Location** to view i3 PIDF-LO data.

Example:

ALI	Location	Notes
Caller		
Carol Polks		2472383257
Location		
100 E EDGEWOOD DR		DURH NC 27704

Display call notes

Step

Click **Notes** to view any call notes that were added to the call.

Example:

ALI	Location	Notes
Notes		
+++ mittle:Start 2014-06-03 14:59:43 +++ Suspect not found at confirmed address. --- mittle:End 2014-06-03 14:59:55 ---		
+++ mittle:Start 2014-06-03 15:00:00 +++ Asked caller to confirm given address. --- mittle:End 2014-06-03 15:00:19 ---		

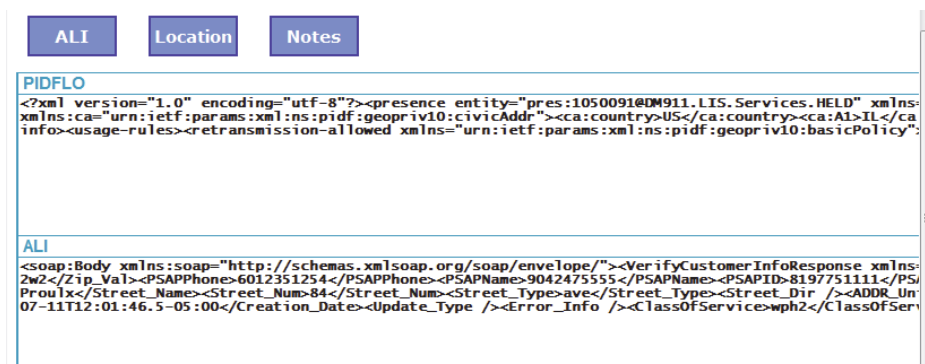
Display raw data

Your system administrator may have chosen to provide you access to view the RAW data message for the ALI and PIDF-LO.

Step

Click **Raw** from either the ALI or location screens.

Example:



4.2.3 Additional caller information

If your PSAP is equipped, you may receive additional caller information when processing wireless calls. The additional caller information is automatically displayed in the ALI presentation as shown in Figure 4.3. The additional information includes more accurate coordinates and the radius for the location of the wireless device.

Figure 4.3 Additional caller information

2475116001	
Caller	
Wagner Mobility	(247) 578-9612
WPH2	
Location	
Tower 357	HUDSONVILLE AS
	35.9969 -78.9017 UNCEP 5.0 CONF 89.5
	SECTOR 14 EAST QUADRANT
Other Information	
104	2475116001
QUERY CALLER FOR LOCATION	QUERY CALLER FOR LOCATION
QUERY CALLER FOR LOCATION	
Additional Caller Location	
35.996084 -78.900923	21.6
Notes	

Additional caller information includes coordinates and the radius

The additional caller information includes more precise X and Y coordinates for the location of the wireless device compared to the coordinates received with the ALI, and includes the radius.

If no additional caller information is available for the calling device, the additional caller location area remains blank.

4.3 Phone connection

The **Phone Connection** window lists all the current phone connections at your position and displays the participants on each call. The window displays active calls, abandoned calls, calls on hold, and ringing calls. From the window you can place calls on hold, mute a participant, place a participant on privacy, join one or more calls, release participants, or release the entire call.

If **Phone Connection** is not visible in your layout, launch it from the **View** menu as a floating window.

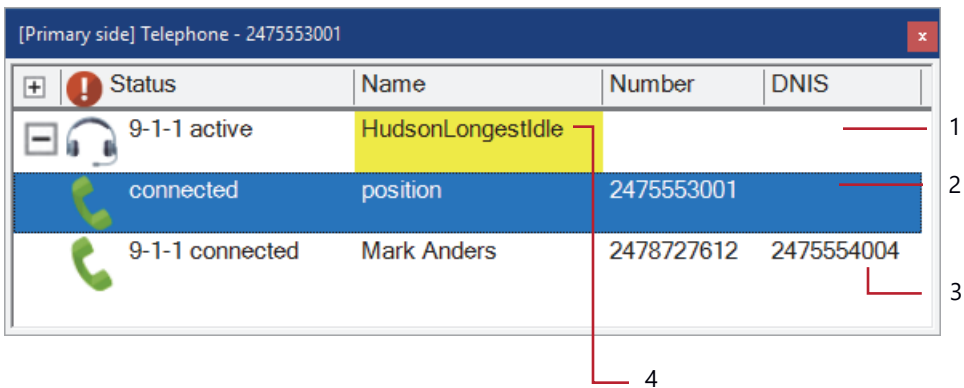
Menu: View > Phone

The unique feature of the **Phone Connection** window is that all calls display an entry for each participant on the call. Displaying each participant with its own entry allows you to better control the call. For instance, you can select which participant to mute, which participant to place on privacy, or which participant to release from the call.

Every call always has a minimum of three entries (or lines). Using Figure 4.4 as an example:

- 1 This line represents the entire call from the current telecommunicator’s perspective. It is always the topmost entry of the call. Any action you perform while this entry is selected affects all parties on the call. You can expand/collapse the view of the call by clicking the plus/minus sign (+/-) at the beginning of this line.
- 2 This line represents the telecommunicator. It is always labeled “position.” By default, this entry always has the focus (indicated by the blue highlight) until you move the focus to another participant.
- 3 This line represents the caller (incoming call) or callee (outgoing call).

Figure 4.4 Phone Connection window with active call



1. Complete call 2. Telecommunicator 3. Caller/Callee 4. Ring group name

The **Phone Connection** window always displays the position’s phone number in the title. This phone number is also listed in the *Number* column for the “position” entry. If received, the caller’s name and phone number appear on their line in the *Name* and *Number* columns respectively.

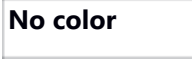
When an incoming call is routed through a ring group, the name of the ring group is displayed in the *Name* column for the top entry at all positions that receive the ringing call. This feature allows the system configuration to identify calls to the telecommunicators. For example, if an intake telecommunicator receives this call, assesses the call, and transfers the call to the “Fire” ring group, the telecommunicators within the fire ring group can easily

identify the call as a fire related call since the ring group name is set to "Fire". The last column, labeled *DNIS*, represents the phone number of the ring group this call is routed through.

In addition to the ring group name identifying emergency calls from administrative calls, you can see that *Status* includes text identifying the call as well as a unique icon. For a complete listing of status icons displayed in the **Phone Connection** window, refer to *Appendix C, "Icon glossary"*.

The background color of the *Name* column for the top entry indicates the current state of the call at your position.

Table 4.1 Color definitions for call status in Phone Connection

Color	Definition
 Red	Ringing incoming call
 Yellow	Current active call
 No color	Call is not active - telecommunicator is on hold or muted

4.3.1 Automatic call expansion

The system includes the ability for the call entries in the **Phone Connection** window to remain collapsed even after you answer the call. If you do not need to select individual call participants or conference third parties to your calls frequently, your administrator may have enabled this feature. When you answer, the call is not expanded automatically in the **Phone Connection** window as shown in Figure 4.5.

Figure 4.5 Answered call with automatic call expansion disabled

[Primary side] Telephone - 2475553001				
+	Status	Name	Number	DNIS
+	9-1-1 active	HudsonSequential	2478643941	
		Caller's ANI		
		Ring group name or callerID		

- The *Name* column displays the ring group or line appearance name.
- The *Number* column displays the caller's phone number received with the call.

If you need to select a participant on the call, you can expand the call by clicking on the plus sign (+) at the beginning of the line as shown in Figure 4.6.

Figure 4.6 Answered call after call is expanded

[Primary side] Telephone - 2475553001				
+	Status	Name	Number	DNIS
-	9-1-1 active	HudsonSequential		
	connected	position	2475553001	
	9-1-1 connected	Philip Smith	2478643941	2475554005

By default the focus remains on the "position" entry even when the call is not expanded in the **Phone Connection** window. This allows you to quickly mute yourself, place the call on hold, release the call, or release only yourself after conferencing in a third party (explained in the following chapters.)

4.3.1.1 Mute, hold, and privacy

- Placing a call participant on mute disconnects the microphone of the participant so that other call parties cannot hear them.

- Placing a call participant on privacy disconnects the earphones of the participant so that they cannot hear the remaining parties. The telecommunicator cannot be placed on privacy and the button remains disabled while the “position” entry is selected.
- Placing a participant on hold disconnects the microphone and earphones of the participant.

To place the entire call (all participants) in one of the three states, select the top line before applying the action.

Info

If automatic call expansion is disabled, you must expand the call and select the top line to place the entire call in one of the three states.

All three buttons act as toggle buttons: clicking the button enables/disables the action.

Mute participant

Step 1 Select the participant to mute.

Step 2 Click **Mute**.

Remove mute

Step 3 Ensure the same participant entry is selected.

Step 4 Click **Mute**.

Mute with automatic call expansion disabled

Step 1 Click **Mute**.

[More info](#): Only the telecommunicator’s entry is muted.

Remove mute

Step 2 Click **Mute**.

Place on privacy

- Step 1** Select the participant to place on privacy.
- Step 2** Click **Privacy**.

Remove privacy

- Step 3** Ensure the same participant entry is selected.
- Step 4** Click **Privacy**.

Place privacy with automatic call expansion disabled

[More info](#): Privacy is disabled while the focus is on the "position" entry.

- Step 1** Expand the call.
- Step 2** Select the participant to place on privacy.
- Step 3** Click **Privacy**.

Remove privacy

- Step 4** Ensure the same participant is selected.
- Step 5** Click **Privacy**.

Place on hold

[More info](#): When only two participants are on the call, both parties are placed on hold - you do not need to select a participant entry if the "position" entry is selected.

- Step 1** Select the participant to place on hold.
- Step 2** Click **Hold**.

Remove from hold

- Step 3** Ensure the same participant entry is selected.
- Step 4** Click **Hold**.

Place on hold with automatic call expansion disabled

More info: When only two participants are on the call, both parties are placed on hold. If more than two participants are on the call, only the telecommunicator is placed on hold.

Step 1 Click **Hold**.

Remove from hold

Step 2 Click **Hold**.

Info

If two telecommunicators are on hold with a 9-1-1 caller that abandons the call, only the telecommunicator who clicks hold is made available. The second telecommunicator is not automatically removed from hold in case they are active on another call.

4.3.1.2 Exclusive hold

Placing a call on exclusive hold disconnects the microphone and earphones of the participants as with regular hold. However, the call is “owned” by the telecommunicator who placed it on exclusive hold. No other Guardian telecommunicator can retrieve the call from hold.

The call is listed in the appropriate current calls and calls on hold tenant status windows. However, the **Answer call** button is disabled for all telecommunicators except the telecommunicator who placed the call on exclusive hold.

If line appearances are configured on the system, a call placed on exclusive hold is displayed in a different color from regular hold (default light orange.) Only the “owning” telecommunicator can retrieve the call from hold.

Placing any participant on exclusive hold places all call participants on exclusive hold. Additionally, all entries on the call must be in a “connected” state for the exclusive hold feature to be enabled.

Since all parties on the call are placed on exclusive hold, conferencing a third party does not automatically conference the party into the call. Instead, a new call is made to the third party and you must join the two calls manually. (Exclusive hold is automatically removed when the two calls are joined.) See the procedures for emergency and administrative calls in the respective chapters.

The **Excl Hold** button acts as a toggle button: clicking the button enables/disables the action.

Place on exclusive hold

More info: Exclusive hold is not affected by the setting of automatic call expansion. All participants are placed on exclusive hold automatically.

Step 1 Click **Excl Hold**.

Step outcome: All participants of the call are placed on exclusive hold.

Remove from exclusive hold

Step 2 Click **Excl Hold**.

4.4 Disconnect reasons

Releasing calls on the Guardian system depends on which entry is selected in the **Phone Connection** window before you click the **Release** button. In addition, your administrator may have configured disconnect reason buttons and included them in your layout. These buttons behave like the **Release** button when clicked while also recording the reason for the call. A sample of custom disconnect reason buttons is shown in Figure 4.7. The reason a call was disconnected is visible in the **Call History** and **Telecommunicator Call History** windows.

Figure 4.7 Sample disconnect reason buttons



Disconnect reasons and the images used for the buttons are customizable

No reason is recorded when a call or call participant is released using the **Release** button; if you need to document the reason for a call, use the disconnect reason buttons instead of the **Release** button. You can use disconnect reason buttons even if automatic call expansion is not enabled in the **Phone Connection** window.

Your administrator is able to create as many disconnect reasons as required and apply custom images to the buttons.

4.5 Make busy

The Make Busy feature blocks incoming calls from being routed to your position, whether directly dialed or distributed via a ring group or line appearance, while still allowing you to continue any current calls and make outgoing calls.

When in a busied out state, the **NG9-1-1 Info** window changes color according to your administrator's selection and displays "Busied Out" as shown in Figure 4.8. The position status button for your position also shows your busied out state to other telecommunications via the color of the button. Guardian's status bar on your position displays "Busied Out" in red.

Figure 4.8 NG9-1-1 Info window while in busied out mode



Your administrator can configure which calls are blocked or allowed to ring at your position while you are busied out. For example, calls dialed directly via the position phone number may be allowed while calls via a ring group or line appearance are blocked from ringing at your position while you are busied out. For the configured types of calls, your administrator may also give you the ability to reject incoming calls when not busied out. For example, if calls via a ring group are blocked when you are busied out, you can also reject an incoming call from a ring group while you are not busied out.

You can also busy out your position while you are on an active call, for example, in the case of a long "hot call" or while you are idle, for example, taking a break.

Prevent incoming calls with make busy

Step 1 Click **Make Busy**.

Step outcome: Your position is set to busied out.

Return to enabled mode

More info: The **Make Busy** button behaves as a toggle.

Step 2 Click it again to remove your position from a busied out state.

Prevent incoming calls by unplugging headset

More info: Unplugging your headset/handset causes the position to automatically go into a busied out state.

Step 1 Depending on your current state, disconnecting your headset results in the following:

- If you and another telecommunicator are plugged into the same PAC II unit, the position is not busied out since the other telecommunicator is available.
- If your position has an active call, the position is busied out. Audio from your caller is sent to your position's external speakers until you replug your headset.
- If your position is in a wrap-up state, busied out is applied and the wrap-up time is canceled.

Step 2 Plugging in your headset cancels the busied out state.

Reject incoming call

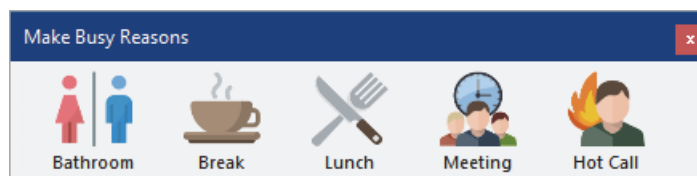
Step While a call is ringing at your position, click **Make Busy**.

Result of this task: The call stops ringing at your position and your position is set to the busied out state.

4.5.1 Make busy reasons

Your system may be configured with make busy reasons that you use to indicate the reason you are placing your answering position into a busied out mode; for example, break, lunch, hot call, etc. Your administrator can configure and customize make busy reason buttons that are suited for your PSAP. The behavior of the make busy buttons is the same as the regular **Make Busy** button.

Figure 4.9 Make busy reason buttons



Prevent incoming calls with make busy reason

Step Click a make busy reason button.

Result of this task: Your position is set to busied out. All the make busy reason buttons are disabled except for the reason selected and the regular **Make Busy** button.

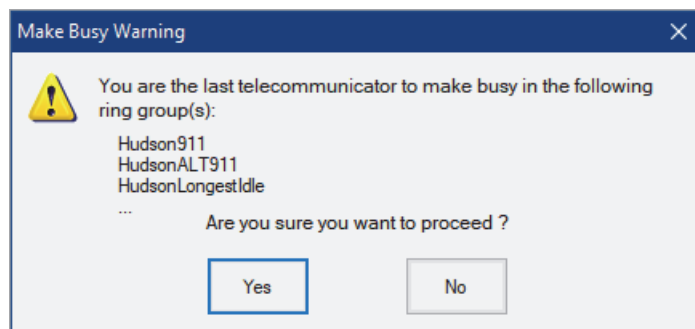
Return to enabled mode

More info: The make busy reason and **Make Busy** buttons behave as a toggle.

Step Click either to remove your position from a busied out state.

4.5.2 Busy out warning

The system can be configured to alert you when you are the last member of a ring group and you attempt to busy out your position. A message as shown in Figure 4.10 is displayed on your screen indicating the ring groups that have no active telecommunicator other than yourself:

Figure 4.10 Make busy warning

- 1 if you click **Yes**, your position is busied out and any incoming calls to the ring groups listed are rerouted or rejected depending on system configuration.
- 2 If you click **No**, your position is not busied out and your answering position is available for any incoming calls to the ring groups listed.

Your administrator may also configure an audible zip tone in addition to the visual indicator to notify you when you are the last member of a ring group to busy out.

4.6 Direct access

Direct access (DA) contacts are toolbars with buttons representing entries from your contact list as created by your administrator. The buttons on the toolbar provide quick single-click access to call a contact, conference a contact to your active call, or transfer your call to the contact.

The toolbars may be located anywhere in your layout but typically are placed inside a direct access group creating a window effect with a descriptive title as shown in Figure 4.11.

Figure 4.11 Sample direct access toolbar within a direct access group named EMS

If you close a direct access group and want to retrieve it, you can find all direct access groups by navigating to the **View** menu.

Menu: View > Direct Access

Note: When clicking a contacts DA button without an active call, the corresponding phone number from the contact list is dialed.

The buttons on a contacts DA toolbar are also sometimes referred to as speed dial buttons. However, direct access buttons are much more than speed dial buttons. Depending on the configuration of the buttons and the line services available to your PSAP, the buttons have the intelligence to dial differing phone numbers when clicked while on an active call. For example, clicking a contact DA button while on an emergency call may result in a three-way call, while clicking the same contact DA button on an administrative call may result in a new long distance call.

4.6.1 Usage behavior

The behavior of the buttons on a contact DA toolbar while on active call is configurable by your administrator. Your administrator has the ability to assign a universal behavior for all direct access toolbars and can also apply a different behavior per individual button on a DA toolbar.

Three behaviors are available for the direct access buttons:

- 1** New call mode
 - Place your active call on hold and make a new call to the contact.
- 2** Conference mode
 - Conference the contact to your active call.

3 Quick transfer mode

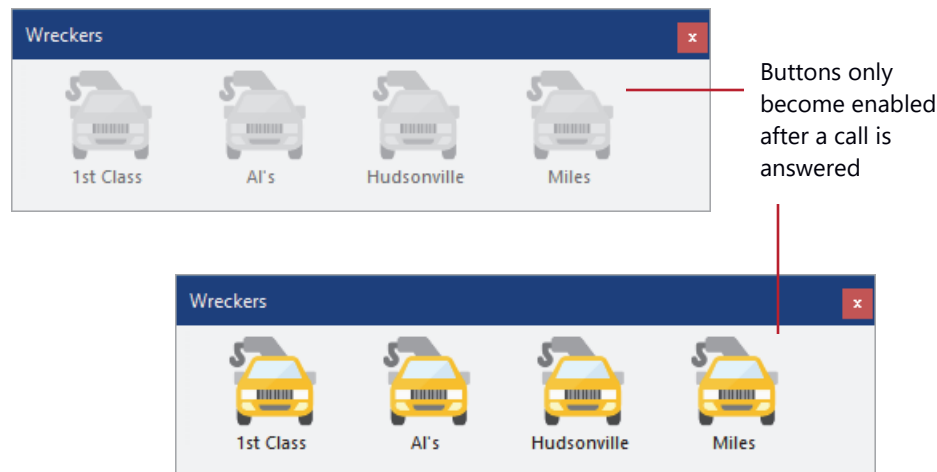
- Transfer your active call to the contact and release you from the call when the contact answers.

If a universal behavior was selected for all direct access toolbars and certain direct access buttons have custom behaviors, the custom behavior takes precedence over the universal behavior. For example, using a DA contact button on an active call usually results in the contact being conferenced to your active call. However, buttons to utility companies are set to quick transfer mode so you are released from the call automatically when the utility contact answers.

4.6.2 Transfers only

Your system may be configured with direct access buttons that are only enabled when you are on an active call that can be used for transfers. These buttons are easily identified as they remain disabled when you are not on a call. They cannot be used to make a new outgoing call except if they were configured for new call mode and you are on an active call.

Figure 4.12 DA buttons configured for transfers only



Buttons remain disabled until you answer a call and return to a disabled state after you release the call

4.7 Line appearances

Line appearances are a special direct access toolbar that represent the phone circuits (phone lines) available at your PSAP. Each individual button on a line appearance toolbar represents one specific phone circuit. Toolbars can be created for emergency circuits, administrative circuits, or any combination of both. The availability of line appearance toolbars is dependent on the type of line service supplied to your PSAP and may not be available for emergency circuits, administrative circuits, or both.

Figure 4.13 depicts a line appearance toolbar representing six 9-1-1 circuits and the states of the lines via the default color of the buttons.

Figure 4.13 9-1-1 line appearance toolbar

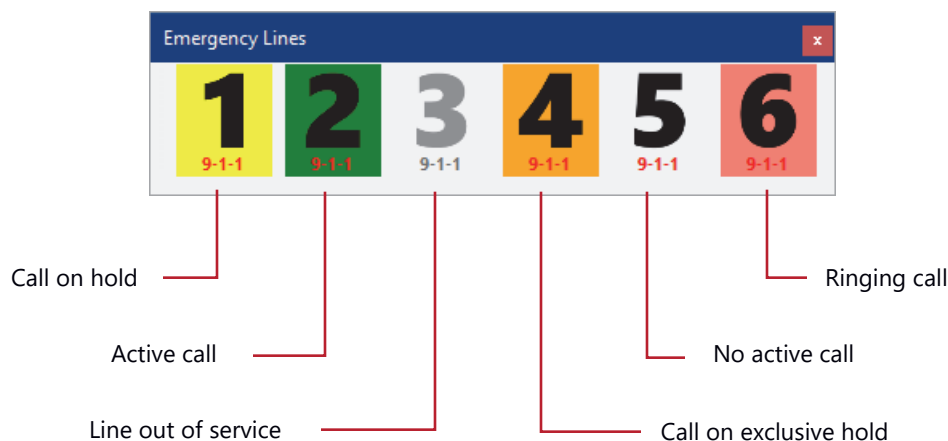
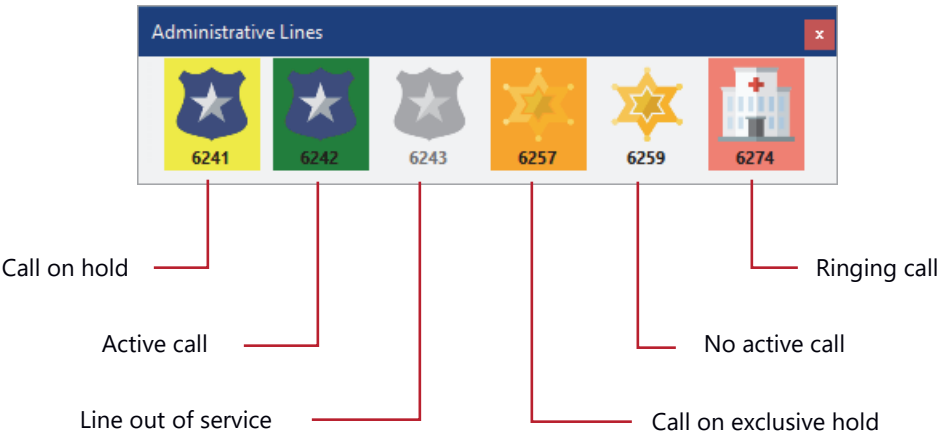


Figure 4.14 depicts a line appearance toolbar of six administrative circuits and the states of the lines via the default color of the buttons.

Figure 4.14 Administrative line appearance toolbar



If your PSAP is equipped with line services that can be mapped to individual line appearance buttons on a toolbar, your administrator can configure the buttons to allow you to dial out on a specific line, allow you to dial out automatically on a specific line using a preset phone number (ringdown), or allow incoming calls only.

Table 4.2 Default colors for line appearance buttons

Color	Description
Enabled	The line is in service but no call is present on the line.
Red	The line has an incoming call that hasn't been answered.
Green	The line is in use with an active call.
Yellow	The line is on hold.
Orange	The line is on exclusive hold.
Disabled	The line is out of service.

Info Be aware, your administrator may have customized the various state colors for your PSAP.

Answer incoming call

Step Single-click the line appearance button with an incoming call (default red).

Result of this task: The call is answered and the button color changes at all positions displaying the button to notify other telecommunicators the call has been answered.

Retrieve call from hold

Step 1 Single-click the line appearance button when the call has been placed on hold (default yellow).
Step outcome: When you retrieve the call, you have two-way audio with the caller while the original telecommunicator remains on the call in a hold state.

Release original telecommunicator

Step 2 Select the entry for the original telecommunicator in the **Phone Connection** window.

Step 3 Click **Release**.

Step outcome: The original telecommunicator is released from the call and you remain connected to the caller.

Place call back on hold

More info: If you retrieved the wrong line appearance call from hold or need to place the call back on hold for another reason:

Step 4 Ensure the "position" entry is selected.

Step 5 Click **Release**.

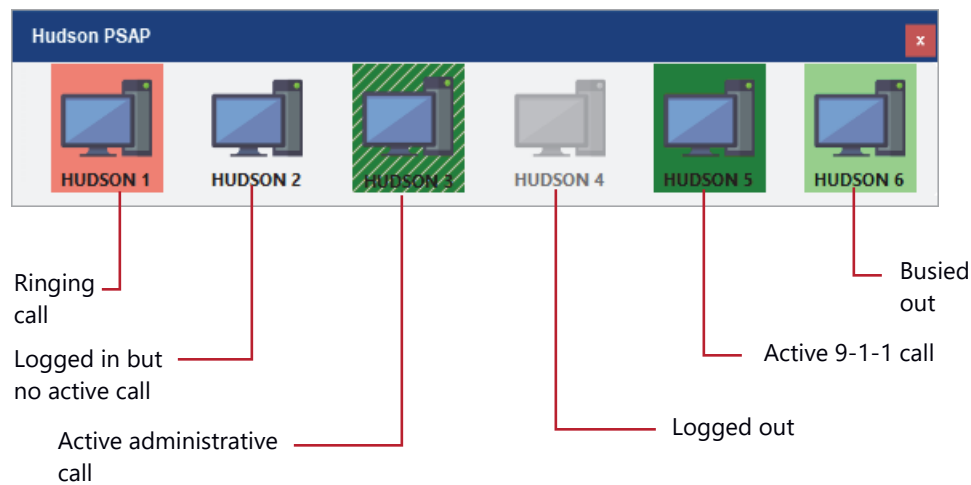
Result of this task: You are released from the call while the caller and original telecommunicator are placed back on hold.

4.8 Position status

Position status is a special DA toolbar that represents the telecommunicators logged in at positions in your PSAP (for special configurations, buttons can be configured for positions in multiple PSAPs.) Each individual button on a position status toolbar represents one specific position. Position status toolbars are not inherently included with a Guardian system and your administrator may have decided to not include them for your PSAP.

Position status buttons indicate the current state of the position by means of the button color, as shown in Figure 4.15.

Figure 4.15 Default position status toolbar colors



The background color of the button indicates the state of the position.

The position status toolbar allows you to quickly view the state of all positions in your PSAP. The colors associated to the various states are customizable by your administrator and may not appear as displayed above.

Tip: Be aware that you do not see a button representing your own position.

If your PSAP is equipped with a position status toolbar, you can quickly call another telecommunicator, conference a telecommunicator into your active call, or transfer a call to another telecommunicator. The behavior of the position status button is configurable as with all direct access toolbars, see *Usage behavior* on page 38. Consult your administrator to determine the behavior applied to the buttons.

Call position

Step While not on an active call, single-click the button representing the position you want to call.

Result of this task: A new call is made to the selected position.

Conference telecommunicator to your active call if conference mode enabled

Step 1 While on an active call, single-click the button representing the position you want to conference.

Result of this task: When the telecommunicator at the position answers, they are automatically conferenced to your active call.

Transfer call

Step 2 Click **Release**.

Result of this task: You are released from the call while the caller remains connected with the other telecommunicator.

Transfer call to another position if conference mode enabled

Step 1 While on an active call, single-click the button representing the position you want to transfer your active call to.

Step 2 When you hear the ring tone in your headset or see a new ringing entry in the **Phone Connection** window, click **Release**.

Result of this task: You are released from the call and the call rings at the position you selected - the caller hears a ring tone until the telecommunicator at the position answers.

Transfer call to another position if quick transfer mode enabled

Step While on an active call, single-click the button representing the position you want to conference.

Result of this task: When the telecommunicator at the position answers, you are released from the call automatically while the caller remains connected with the other telecommunicator.

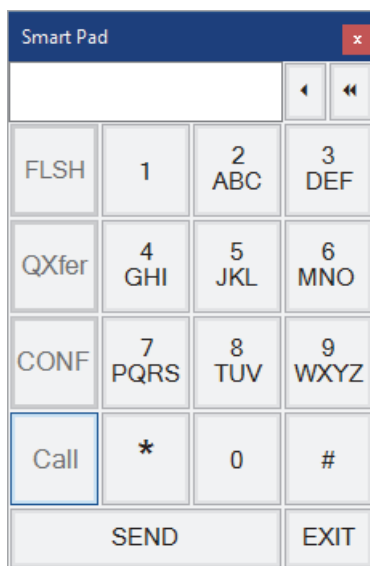
4.9 Smart pad

The Smart Pad combines the functionality of Guardian's dial pad and DTMF pad into a single dialing tool. From the **Smart Pad** window you can choose to make a call, conference a party to your active call, quick transfer your active call to another party, or navigate through interactive menus on an interactive voice response (IVR) call. You can click the **Smart Pad** button to open or activate the **Smart Pad**.

If **Smart Pad** is not visible in your layout, launch it from the **View** menu as a floating window or click the **Smart Pad** button if included in your layout.

Menu: View > Smart Pad

Figure 4.16 Smart Pad window



The following button commands are available from the **Smart Pad** window. The hook flash, quick transfer, and conference buttons are only enabled when you have an active call.

	Click FLSH when you want to conference another participant to an active call using hook flash.
	Click QXfer to transfer your active call to the phone number dialed while automatically releasing yourself from the call when the party answers.
	Click CONF to call the phone number and have the party automatically conferenced into your active call.
	Click Call while on an active call to make a new call to the phone number entered followed by SEND .
	Click EXIT to close the Smart Pad window when it is in floating mode.
	Click SEND after you enter a phone number to call the number.
	Click Erase to remove the last number you entered.
	Click Clear to erase all the numbers you entered.

Note: You must click **Call**, **QXfer**, or **CONF** before entering the phone number.

Once the **Smart Pad** window is in focus, you have the option to use your mouse to click on the numbers or use the numeric keypad on your keyboard. Clicking the **Enter** key on your numeric keypad automatically activates the send action.

4.9.1 DTMF awareness

The **Smart Pad** is intelligent as it remains aware of which participant on the call is connected from outside Guardian. It also keeps track of the last participant to be added to a call. So if you want to hook flash a third party to your call, you do not need to first select your caller's entry in the **Phone Connection** window. Instead, just click the **FLSH** button and continue. Or if you conference a third party to an active call and they have an IVR that answers the call which requires your input, you do not need to first select the third party's entry. Instead, just click the required number in the **Smart Pad** window. The system is aware that the third party entry was the last outside source to be added to the call and sends the digits to the third party's line. If you release the third party from the call, **Smart Pad** sends any DTMF digits entered to the caller's line automatically.

In addition, **Smart Pad** is aware when the participants on the call are from an internal source such as another Guardian telecommunicator or a Responder phone. In this case, sending DTMF tones and hook flash capability are disabled.

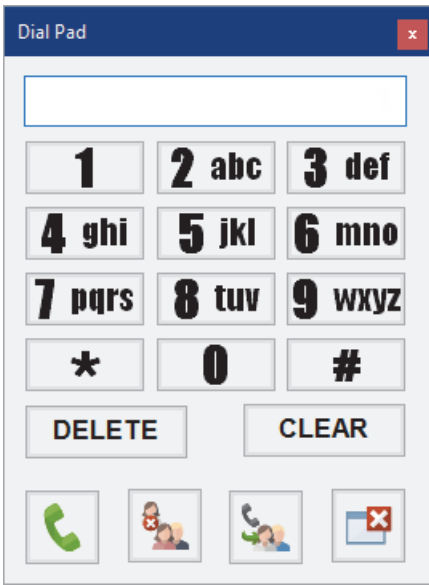
4.10 Dial pad

The **Dial Pad** is used to manually dial phone numbers. From the **Dial Pad** window you can make a call, conference a party to your active call, or quick transfer your active call to another party. The difference from the **Smart Pad** is that the **Dial Pad** does not support DTMF tones. Instead, you need to activate the **DTMF Pad** in addition to the **Dial Pad**.

If **Dial Pad** is not visible in your layout, launch it from the **View** menu as a floating window or click the **Place Call** button if it is included in your layout.

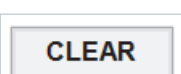
Menu: View > Dial Pad

Figure 4.17 Dial Pad window



The following button commands are available from the **Dial Pad** window. The **Quick transfer** and **Conference** buttons are only enabled when you have an active call.

	Enter a phone number and click Call to place a call to the number entered.
	Click Quick transfer to transfer your active call to the phone number dialed while automatically releasing yourself from the call when the party answers.
	Click Conference to call the entered phone number and have the party automatically joined to your active call.
	Click Close to close the Dial Pad window when it is in floating mode.

	Click DELETE to erase the last number you entered in the dial pad.
	Click CLEAR to erase all the numbers you entered in the dial pad.

Note: You must enter the phone number first for **Call**, **Quick transfer**, and **Conference** to become enabled.

Once the **Dial Pad** window is in focus, you have the option to use your mouse to click on the numbers or use the numeric keypad on your keyboard. Clicking the **Enter** key on your numeric keypad automatically activates the call action.

4.10.1 DTMF pad

If your role is configured to use the **Dial Pad**, you need to access the **DTMF Pad** when you want to make a hook flash call or need to send digits to an IVR. The **DTMF Pad** window is designed specifically to send DTMF tones to the line of the participant selected in the **Phone Connection** window.

You can access the **DTMF Pad** window by clicking the **DTMF Pad** button for both emergency and administrative calls, or the **Manual Xfer** button for emergency calls only.

The **DTMF Pad** button is only enabled for an active call and when an entry other than "position" is selected in the **Phone Connection** window. The DTMF tones are sent to the party selected in the **Phone Connection** window.

The **Manual Xfer** button is only enabled with an active emergency call - the tones are sent automatically to the incoming call party.

Figure 4.18 DTMF Pad window



The following button commands are available from the **DTMF Pad** window. The **Quick transfer** and **Conference** buttons are only enabled when you have an active call.

Note: You must select an entry other than “position” in the **Phone Connection** window to access the **DTMF Pad**.

	Click Hook Flash to activate three-way calling.
	Click CLEAR to erase the numbers in the pad’s display.
	Click Close to close the DTMF Pad window.

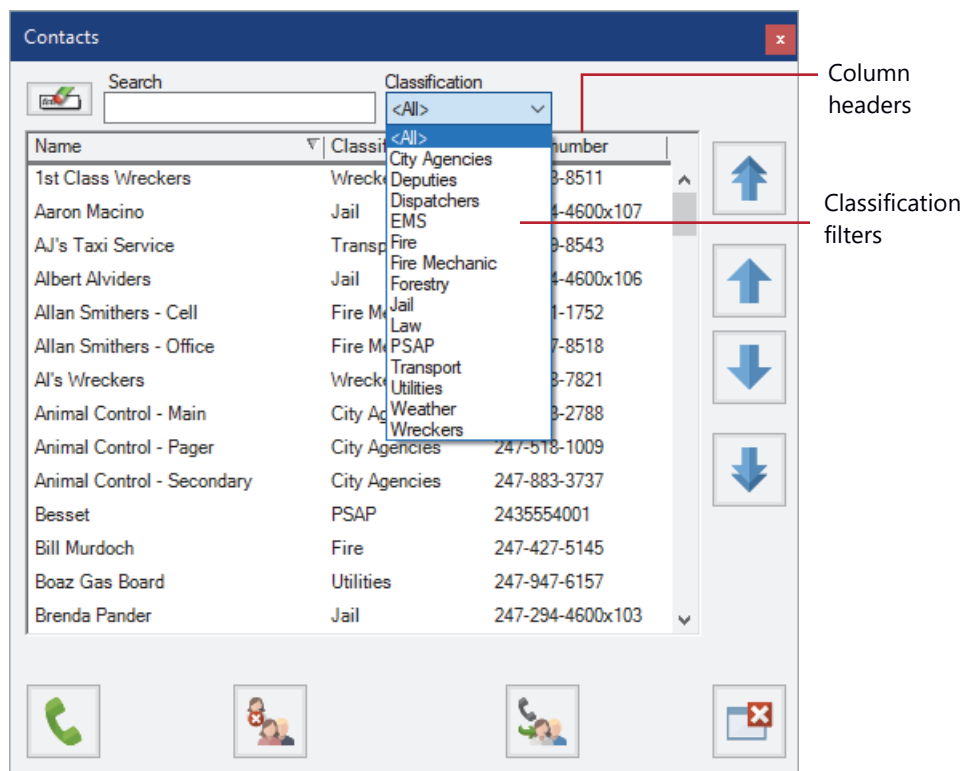
4.11 Contacts

The **Contacts** window contains a combined list of phone numbers from public and personal contacts. The list can be searched, filtered, and sorted to help locate contacts. Click the **Contacts** button to open or activate the **Contacts** window.

If the **Contacts** window is not visible in your layout, launch it from the **View** menu as a floating window.

Menu: View > Contacts

Figure 4.19 Contacts window



The columns visible in the Contacts window may be customized by your administrator and may not appear as shown.

You can use **Search** to find entries based on a name or phone number. The list filters matching results as you type in characters. You can also filter the list of results using the **Classification** list.

You can sort the entries by *Name*, *Classification*, or *Phone number* by clicking on the column header. By default, the entries are sorted by *Name*.

Contacts can be configured with an alternate phone number. This phone number is dialed when the primary phone number fails or a busy notification is received after dialing the primary phone number.

The following commands are available from the **Contacts** window. The **Quick transfer** and **Conference** buttons are only enabled when you have an active call.

	Select an entry and click Call to place a call to the phone number selected.
	Click Quick transfer to transfer your active call to the entry selected while automatically releasing yourself from the call when the party answers.
	Click Conference to call the selected entry and have the party automatically joined to your current active call.
	Click Clear Search to remove the results from a previous search.
	Click Page up to navigate upwards one page.
	Click Scroll up to navigate to the entry above the current selection.
	Click Scroll down to navigate to the entry below the current selection.
	Click Page down to navigate downwards one page.
	Click Close to close the Contacts window when it is in floating mode.

4.12 Ring group assignments

Incoming calls are routed to your position using a feature known as ring groups. Your administrator may have numerous ring groups configured for your PSAP. For example, emergency calls, administrative calls, calls coming from alarm companies, calls for the police department, or calls for the sheriff's office can all be configured to route to your PSAP using separate ring groups. Your administrator may have given you access to answer calls from all the ring groups configured at your PSAP; however, they may not want calls from all the ring groups to ring at your position at all times. Instead, your administrator may want you to receive calls from a specific ring group only when requested by your supervisor to assist with call volume.

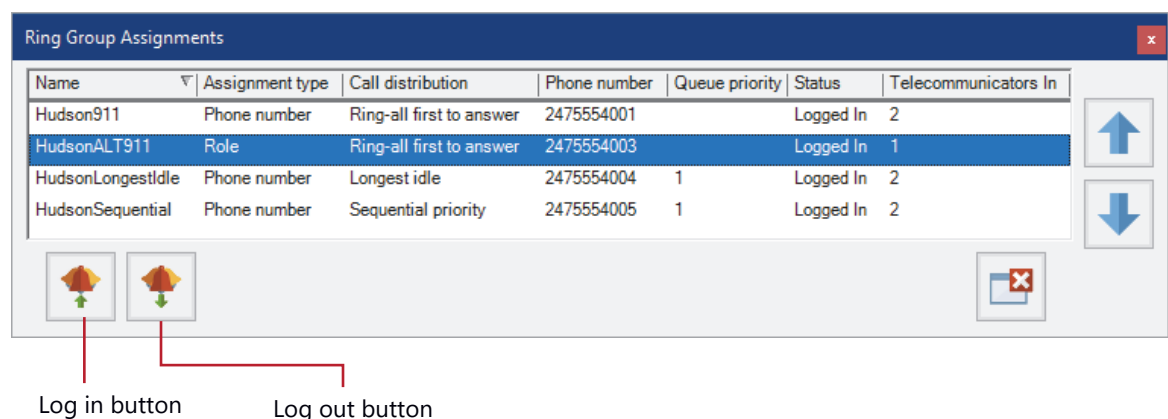
Guardian has a feature named Ring Group Assignments which allows you to view the ring groups configured for your PSAP and allows you to log in and/or log out of specific ring groups as shown in Figure 4.20.

If the **Ring Group Assignments** window is not visible in your layout, launch it from the **View** menu as a floating window.

Menu: View > Ring Group Assignments

If your role is not assigned the privilege to view ring group assignments, you do not see an entry for ring group assignments under the **View** menu.

Figure 4.20 Ring Groups Assignment window



The columns visible in the **Ring Group Assignments** window are configurable and your administrator may have chosen to remove columns or reordered them from the default presentation as displayed in Figure 3.26.

Log in to a ring group

More info: Only applicable if your role includes the privilege to log in/out of ring groups and you are a valid member of the ring group.

Step 1 In the **Ring Group Assignments** window, select a ring group.

More info: If you are logged out of the ring group, the **Log in** button is enabled.

Step 2 Click **Log in**.

Result of this task: The status changes to logged in on the ring group assignments window and your position is added to the selected ring group. You receive calls routed to the ring group following the selected call distribution for the ring group.

Log out of a ring group

Note: When the *Prohibit last member logout* privilege is enabled, you cannot log out of a ring group if you are the last user logged in.

More info: Only applicable if your role includes the privilege to log in/out of ring groups.

Step 1 In the **Ring Group Assignments** window, select a ring group.

More info: If you are logged into the ring group, the **Log out** button is enabled.

Step 2 Click **Log out**.

Result of this task: Your position is removed from the selected ring group and the following occurs:

- **Log out** is disabled.
- **Log in** is enabled.
- Your position does not receive calls routed to the ring group.
- Active calls remain at your position until released.
- Calls in the queue are handled per configuration settings.

4.13 Call history

The **Call History** window displays historical information pertaining to incoming emergency calls received at positions in your tenant group. (A Guardian tenant group represents a PSAP). The window displays a scrollable and sortable list of information on each call, provides access to ALI and text conversations, and allows you to call back an entry.

If **Call History** is not visible in your layout, launch it from the **View** menu as a floating window.

Menu: View > Call History

The columns and buttons visible in the **Call History** window may have been modified by your administrator and may vary from Figure 4.21. If your role has the assigned privileges, refer to the *Guardian Administration Guide* to modify the column appearance.

Figure 4.21 Call History window

Start time	Media type	Callback #	Call type	ALI display
2019-01-31 11:48:09 AM	Voice	2478724785	Emergency	RESD 470 HORSESHOE LAKE RD BALD KNOB
2018-12-17 1:31:29 PM	Voice	2478293652	Emergency	WRLS
2018-12-17 1:30:34 PM	Text	2478728825	Emergency	RESD 901 W VINE AVE SEARCY
2018-12-17 1:29:50 PM	Voice	2478293652	Emergency	WRLS
2018-12-17 1:29:22 PM	Voice	2478724785	Emergency	RESD 470 HORSESHOE LAKE RD BALD KNOB
2018-12-17 1:29:07 PM	Voice	2478643941	Emergency	RESD 1007 W ARCH AVE SEARCY
2018-12-17 12:27:31 PM	Voice	2478643941	Emergency	RESD 1007 W ARCH AVE SEARCY
2018-12-17 11:37:22 AM	Voice	2478649612	Emergency	WPH2 PANGBURN
2018-12-17 9:49:40 AM	Voice	2478649612	Emergency	WPH2 PANGBURN

The following commands are available from the **Call History** window.

	Select an entry and click Show ALI info to display the last queried ALI for the selected call.
	Click Callback to make a call using the "Call back #" of the selected entry.
	When a text-to-911 or TDD call is selected, click Display conversation to view the text messages from the call.
	Click Text callback to make a SMS callback using the "Call back #" of the selected entry for wireless and text calls.
	Select an entry and click Call notes to display any notes for the call. You can edit an existing note or add a new note to the call.
	Click My calls to display only the calls which have your user name as the <i>Telecommunicator</i> .
	Click Lock/Unlock to freeze the contents of the window at the current page so you have the ability to precisely select the entry required.
	Click Remove to delete the selected entry. You are presented with a confirmation dialog box to confirm the deletion. This action removes the call from Call History window of all user's.
	Click Scroll up to navigate to the entry above the current selection.
	Click Scroll down to navigate to the entry below the current selection.
	Click Filter on/off to remove an applied filtered.

	Click Set filter to apply a filter to the entries.
	Click Close to close the Call History window when it is in floating mode.

View ALI from call history

Step 1 Select a call from the list.

Step 2 Click **Show ALI info**.

Step outcome: The last ALI retrieved while the call was active is displayed in the **Caller Information** window.

Close ALI

Step 3 Click **Show ALI info** again or select another entry to close the ALI displayed.

ALI corrections from call history

Step 1 Select a call from the list.

Step 2 Click **Show ALI info**.

Step outcome: The last ALI retrieved while the call was active is displayed in the **Caller Information** window.

Step 3 Click **Correction**.

Step outcome: All fields in the displayed ALI can be edited.

Step 4 Replace existing data with correct data.

Print corrections

Step 5 Click **Print**.

Step outcome: The ALI is printed to your default Windows printer.

Email corrections

Step 6 Click **Email**.

Step outcome: The ALI corrections are sent to the configured recipient.

Close ALI

- Step 7** Click **Show ALI info** again or select another entry to close the ALI displayed.

Print ALI from call history

- Step 1** Select a call from the list.
- Step 2** Click **Show ALI info**.
Step outcome: The last ALI retrieved while the call was active is displayed in the **Caller Information** window.
- Step 3** Click **Print**.
Step outcome: The ALI is printed to your default Windows printer.

Close ALI

- Step 4** Click **Show ALI info** again or select another entry to close the ALI displayed.

View conversation from call history

- Step 1** Select a text call from the list.
More info: The **Display conversation** button becomes enabled when a text-to-911 or TDD call is selected.
- Step 2** Click **Display conversation**.

Result of this task: The conversation from the call is displayed in the **Text Conversation** window.

Call back from call history

- Step 1** Select a call from the list.
- Step 2** Click **Callback**.

Result of this task: The ALI for the entry is displayed in the **Caller Information** window and a call is made to callback phone number displayed in the ALI.

Text back from call history

More info: Only wireless and text calls can be sent a SMS message. Identify the call from the *Media type* column or via the COS if included in the *ALI display* column.

Step 1 Select a call from the list.

Step 2 Click **Text callback**.

Step outcome: The **Text Conversation** window is launched in SMS mode to the callback phone number displayed in the ALI.

Step 3 Enter the text of your message.

Step 4 Click **Send**.

Prevent updates to window

Step 1 Click **Lock/Unlock**.

More info: This prevents the **Call History** window from being updated with new entries and freezes the contents of the window so you have the ability to precisely select the entry required.

Step 2 Click **Lock/Unlock** again to allow updates to the **Call History** window.

Delete entry from call history

Step 1 Select a call from the list.

Step 2 Click **Remove**.

Step 3 In the **Confirm Call History Delete** dialog box, click **Yes**.

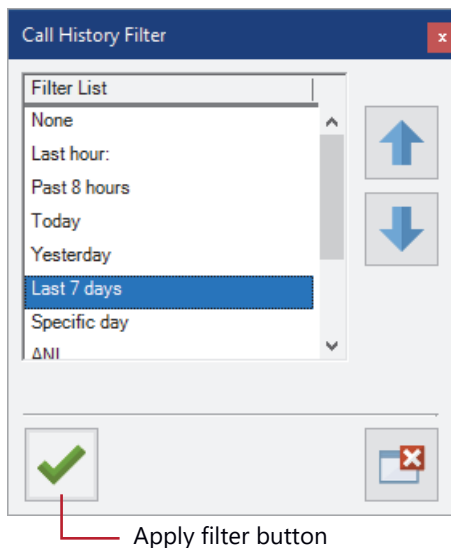
Result of this task: The entry is removed from the call history list. Deleting an entry removes it for all users in the tenant group.

Apply filter to call history

Step 1 Click **Set filter**.

Step outcome: The **Call History Filter** dialog box is displayed.

Example:



Step 2 Select the required filter and click **Apply filter**.

Step outcome: The list in the **Call History** window is filtered to match the selection you specified.

Step 3 Click **Close** in the **Call History Filter** window.

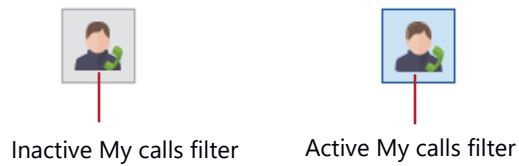
More info: The filter remains in effect until you click **Filter on/off** or log out of Guardian.

4.13.0.1 My calls

When you click the **My calls** button, the list is filtered to display only the calls where your user name is listed as the "Telecommunicator".

Be aware, that the *Telecommunicator* column displays the last telecommunicator to join the call. This means that calls that you answered and subsequently conferenced in another telecommunicator are not listed when the My Calls filter is applied.

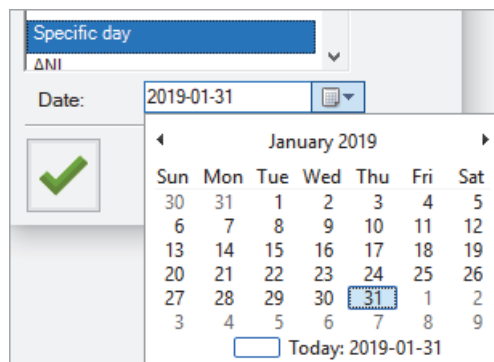
My calls behaves as a toggle. If you click the button once, the filter is applied. When you click it again, the filter is removed. Figure 4.22 shows the two states of the button.

Figure 4.22 My calls button states

The My Calls option does not impact the Set Filter option. If My Calls is active, you can also apply any of the filters to your call list.

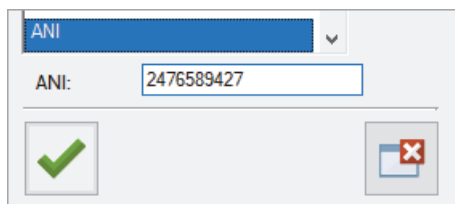
4.13.0.2 Specific day filter

When you select the Specific Day filter option, either enter in the date manually or click the calendar icon to select the date from a calendar view as seen in Figure 4.23.

Figure 4.23 Specific day filter date selection

4.13.0.3 ANI Filter

When you select the ANI filter option, enter in a 10-digit land-line phone number as shown in Figure 4.24.

Figure 4.24 ANI filter phone number entryThe screenshot shows a web interface for entering an ANI (Automatic Number Identification) filter. At the top, there is a dropdown menu with 'ANI' selected. Below it, a text input field is labeled 'ANI:' and contains the number '2476589427'. At the bottom of the form, there are two buttons: a green checkmark icon on the left and a red 'X' icon on the right.

4.14 Telecommunicator call history

The **Telecommunicator Call History** window is very similar to the **Call History** window except it only displays calls in which you participated. The window displays historical information pertaining to incoming emergency calls that you answered and is maintained until you log out of Guardian. The window displays a scrollable and sortable list of information on each call, provides access to ALI and text conversations, and allows you to callback an entry.

If your role is not assigned access to the **Telecommunicator Call History** window, you probably have access to tenant call history instead.

If **Telecommunicator Call History** is not visible in your layout, launch it from the **View** menu as a floating window.

Menu: View > Telecommunicator Call History

The columns and buttons visible in the **Telecommunicator Call History** window may have been modified by your administrator and may vary from Figure 4.25. If your role has the assigned privileges, refer to the *Guardian Administration Guide* to modify the column appearance.

Figure 4.25 Telecommunicator Call History window

Telecommunicator Call History					
Start time	Media type	Callback #	Call type	ALI display	Disconnect reason
2019-02-04 8:26:21 AM	Text	2478728825	Emergency	RESD 901 W VINE AVE SEARCY	Police
2019-02-04 8:25:48 AM	Voice	2472689574	Emergency	RESD 320 JAYBIRD LN SEARCY	Sheriff
2019-02-04 8:25:00 AM	Text	2478728825	Emergency	RESD 901 W VINE AVE SEARCY	Accidental
2019-02-04 8:24:40 AM	Voice	2478643350	Emergency	RESD 1104 N LEMON ST BEEBE	Inquiry
2019-02-04 8:24:20 AM	Voice	2478291851	AdminEmergency	BUSN 1006 W CENTER AVE SEARCY	Police
2019-02-04 8:24:12 AM	Voice	2478649612	Emergency	WPH2 PANGBURN	Accidental
2019-02-04 8:17:06 AM	Text	2478728825	Emergency	RESD 901 W VINE AVE SEARCY	Medical
2019-02-04 8:15:40 AM	Voice	2478643941	Emergency	RESD 1007 W ARCH AVE SEARCY	Medical
2019-02-04 8:15:31 AM	Voice	2478643941	Emergency	RESD 1007 W ARCH AVE SEARCY	Fire
2019-02-04 7:52:51 AM	Voice	2478724785	Emergency	RESD 470 HORSESHOE LAKE RD BALD KNOB	Police

The following commands are available from the **Telecommunicator Call History** window.

	Select an entry and click Show ALI info to display the last queried ALI for the selected call.
	Click Callback to make a call using the "Call back #" of the selected entry.
	When a text-to-911 or TDD call is selected, click Display conversation to view the text messages from the call.
	Click Text callback to make a SMS callback using the "Call back #" of the selected entry for wireless and text calls.
	Select an entry and click Call notes to display any notes for the call. You can edit an existing note or add a new note to the call.
	Click Scroll up to navigate to the entry above the current selection.

	Click Scroll down to navigate to the entry below the current selection.
	Click Close to close the Call History window when it is in floating mode.

View ALI from T/C call history

Step 1 Select a call from the list.

Step 2 Click **Show ALI info**.

Step outcome: The last ALI retrieved while the call was active is displayed in the **Caller Information** window.

Close ALI

Step 3 Click **Show ALI info** again or select another entry to close the ALI displayed.

ALI corrections from T/C call history

Step 1 Select a call from the list.

Step 2 Click **Show ALI info**.

Step outcome: The last ALI retrieved while the call was active is displayed in the **Caller Information** window.

Step 3 Click **Correction**.

More info: All fields in the displayed ALI can be edited.

Step 4 Replace existing data with correct data.

Print corrections

Step 5 Click **Print**.

Step outcome: The ALI is printed to your default Windows printer.

Email corrections

Step 6 Click **Email**.

Step outcome: The ALI corrections are sent to the configured recipient.

Print ALI from T/C call history

Step 1 Select a call from the list.

Step 2 Click **Show ALI info**.

Step outcome: The last ALI retrieved while the call was active is displayed in the **Caller Information** window.

Step 3 Click **Print**.

Step outcome: The ALI is printed to your default Windows printer.

Close ALI

Step 4 Click **Show ALI info** again or select another entry to close the ALI displayed.

View conversation from T/C call history

Step 1 Select a text call from the list.

More info: The **Display conversation** button becomes enabled when a text-to-911 or TDD call is selected.

Step 2 Click **Display conversation**.

Result of this task: The conversation from the call is displayed in the **Text Conversation** window.

Call back from T/C call history

Step 1 Select a call from the list.

Step 2 Click **Callback**.

Step outcome: The ALI for the entry is displayed in the **Caller Information** window and a call is made to the callback number displayed in the ALI.

Text back from T/C call history

More info: Only wireless and text calls can be sent a SMS message. Identify the call from the *Media type* column or via the COS if included in the *ALI display* column.

Step 1 Select a call from the list.

Step 2 Click **Text callback**.

Step outcome: The **Text Conversation** window is launched in SMS mode to the callback number displayed in the ALI.

Step 3 Enter the text of your message.

Step 4 Click **Send**.

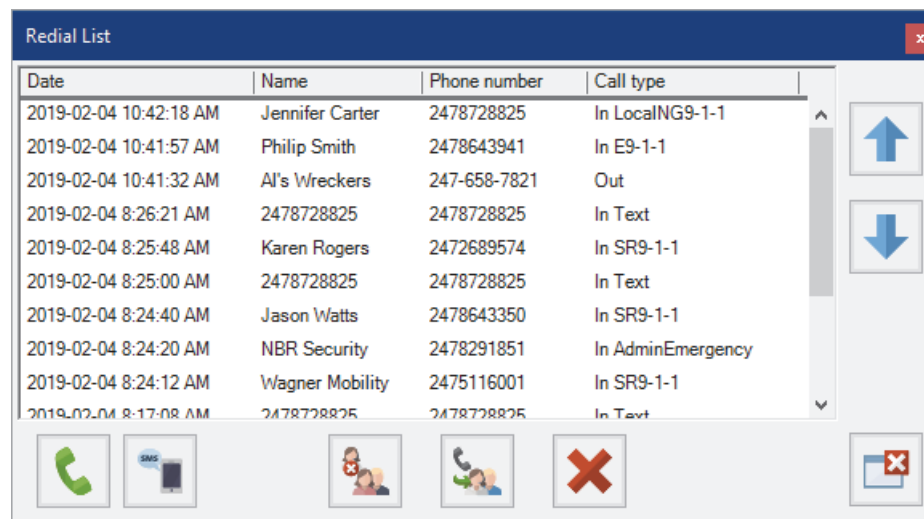
4.15 Redial list

The Redial List feature displays the most recently dialed and received phone numbers at your position. Unlike call history, the redial list displays both emergency and administrative calls that arrived at or were made from your position. The list can contain a maximum of 150 entries and is maintained by your account. The list is cleared when you log out of Guardian. Click the **Redial List** button to open the **Redial List** window.

If **Redial List** is not visible in your layout, launch it from the **View** menu as a floating window.

Menu: View > Redial

Figure 4.26 Redial List window



The screenshot shows a 'Redial List' window with a table of call records. The table has four columns: Date, Name, Phone number, and Call type. The records are sorted by date in descending order. To the right of the table are two large blue arrows for scrolling. At the bottom of the window are several icons: a green phone handset, a blue speech bubble with a phone handset, a group of people icon, a green phone handset with a speech bubble, a red 'X' icon, and a blue document icon with a red 'X'.

Date	Name	Phone number	Call type
2019-02-04 10:42:18 AM	Jennifer Carter	2478728825	In LocalNG9-1-1
2019-02-04 10:41:57 AM	Philip Smith	2478643941	In E9-1-1
2019-02-04 10:41:32 AM	Al's Wreckers	247-658-7821	Out
2019-02-04 8:26:21 AM	2478728825	2478728825	In Text
2019-02-04 8:25:48 AM	Karen Rogers	2472689574	In SR9-1-1
2019-02-04 8:25:00 AM	2478728825	2478728825	In Text
2019-02-04 8:24:40 AM	Jason Watts	2478643350	In SR9-1-1
2019-02-04 8:24:20 AM	NBR Security	2478291851	In AdminEmergency
2019-02-04 8:24:12 AM	Wagner Mobility	2475116001	In SR9-1-1
2019-02-04 8:17:08 AM	2478728825	2478728825	In Text

You can sort entries by *Date*, *Name*, *Phone number*, or *Call type* by clicking on the column header. By default, the entries are sorted by *Date*.

Call type indicates whether the call was incoming, outgoing, emergency, or administrative:

- In - inbound administrative call
- In E9-1-1 - inbound 9-1-1 call
- In SR9-1-1 - inbound 9-1-1 call from a selective router
- In LocalNG9-1-1 - inbound next generation 9-1-1 call from a Solacom selective router
- In NG9-1-1 - inbound next generation 9-1-1 from a selective router
- In AdminEmergency - inbound administrative call configured as AdminEmergency
- Out - outbound call

The following commands are available from the **Redial List** window. However, the **Quick transfer** and **Conference** buttons are only enabled when you have an active call.

	Select an entry and click Call to place a call to the phone number selected.
	Select an entry and click Text callback to place a SMS call to the party selected. Only applicable to wireless and text calls.
	Click Quick transfer to transfer your active call to the entry selected while automatically releasing yourself from the call when the party answers.
	Click Conference to call the selected entry and have the party automatically joined to your current active call.
	Click Clear list to remove all entries from your list.
	Click Scroll up to navigate to the entry above the current selection.
	Click Scroll down to navigate to the entry below the current selection.
	Click Close to close the Contacts window when it is in floating mode.

4.16 Tenant status

A tenant in Guardian is typically configured to represent a single PSAP. The tenant status windows provide live monitoring of resources for emergency calls and administrative calls for your tenant group (PSAP).

If the tenant status windows are not visible in your layout, launch them from the **View** menu as floating windows.

Menu: View > Tenant Status

Your role must include the privileges to view these windows. There are additional privileges that allow you to answer or monitor calls, and reset statistics (see the *Guardian Administration Guide*). If your role does not include these privileges, the command buttons for these actions are not visible.

Each tenant status window provides a live status view that you can lock (freeze the display) in order to more precisely select one of the lines.

Prevent updates in tenant status window

- Step 1** Click **Lock/Unlock**.
- Step 2** Click **Lock/Unlock** again or execute another action to unlock the display and resume live updates.

4.16.1 Tenant status bar

The tenant status bar displays live status information on the number of current calls, calls on hold, calls waiting to be answered, and users that are logged in. The phone numbers or ring groups that are used in determining the counts must be part of the same tenant group.

View the tenant status bar

Step 1 On the **View** menu, select **Tenant Status**.

Step 2 Select **Emergency** or **Admin**.

- Select **Emergency Calls Status**.

Step outcome: The **Emergency Calls Status** window is displayed.

Example:

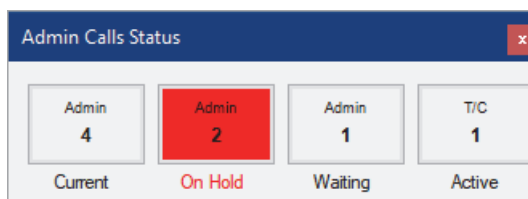


More info: If alarm thresholds are configured and have been triggered, the button is highlighted (red by default.)

- Select **Admin Calls Status**.

Step outcome: The **Admin Calls Status** window is displayed.

Example:



More info: If alarm thresholds are configured and have been triggered, the button is highlighted (red by default.)

Clicking the individual buttons in the status window opens individual windows that provide detailed status information and the ability to perform various actions. The windows can also be opened at any time from the **View** menu.

Table 4.3 provides a description of each window that is launched when the corresponding button in the status bar is clicked.

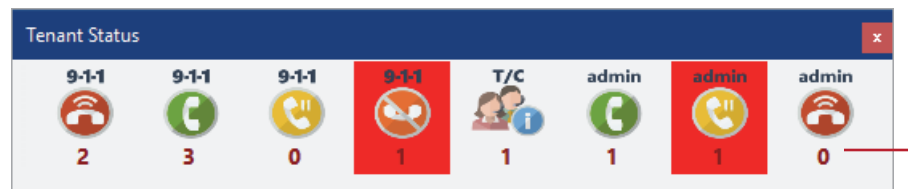
Table 4.3 Tenant call status buttons

Call type	Description
Current	Emergency: Displays the number of in progress emergency or AdminEmergency calls, including calls that are ringing, answered, on hold, or abandoned. Admin: Displays the number of in progress administrative calls including calls that are ringing, answered, or on hold.
On Hold	Emergency: Displays the number of emergency or AdminEmergency calls currently on hold. Admin: Displays the number of administrative calls currently on hold.
Waiting	Emergency: Displays the number of emergency or AdminEmergency calls currently ringing or queued but not yet answered. Admin: Displays the number of administrative calls currently ringing or queued but not yet answered.
Abandoned	Emergency: Displays the number of emergency or AdminEmergency calls that have been abandoned by the caller. The caller has hung up but the calls are still in the system. Admin: Not available
Active	Emergency and Admin: Displays the number of currently logged in telecommunicators whose position phone numbers are assigned to the current tenant group. This number includes telecommunicators whose position is busied out or who may be logged out of all ring groups.

4.16.2 Tenant status buttons

Tenant status buttons allow your administrator to customize which buttons are visible in your layout. Instead of including both of the tenant status bars, your administrator can choose to include only the relevant tenant status buttons for your PSAP in the layout as shown in Figure 4.27.

Figure 4.27 Tenant status buttons



Label displays number of calls in the state represented by the button

Tenant status buttons for emergency and administrative status

The buttons behave the same as the buttons found on the status bars; for example, they change color when an alarm is triggered and they launch the corresponding tenant status window when clicked. The label for each button displays the number of calls matching the indicated state. Your administrator can customize the images for each button and may not include all buttons on the toolbar.

4.16.3 Current emergency/admin calls

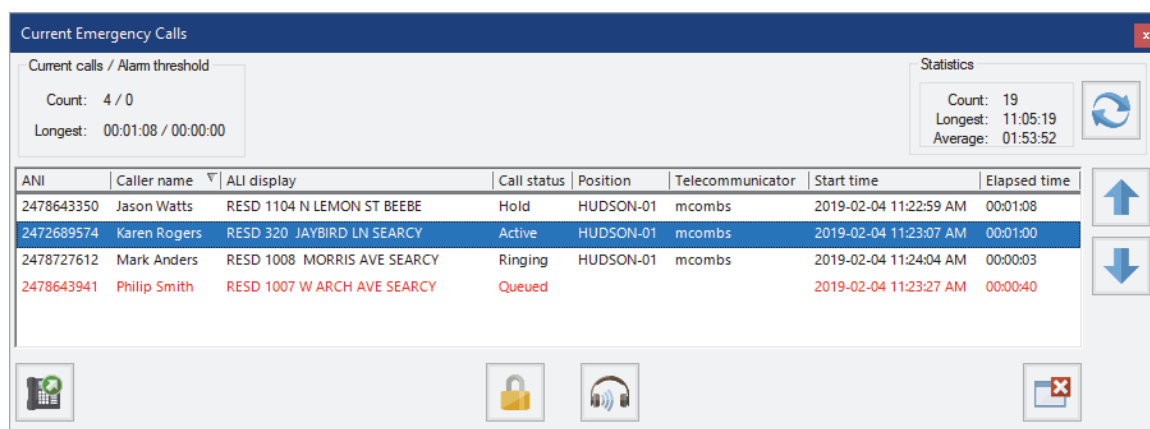
The **Current Emergency Calls** window displays a scrollable and sortable list of all the in progress emergency and AdminEmergency calls, and allows you to answer, retrieve, or join a call, and to monitor the voice communication of an ongoing call. The **Current Admin Calls** window displays a scrollable and sortable list of all the in progress administrative calls, and allows you to answer, retrieve, or join a call, and to monitor the voice communication of an ongoing call.

Understanding how multiple calls for multiple tenants are displayed in your tenant status window can help you to organize your call handling. The status of a call in your tenant window reflects what is happening to calls in your tenant group. For example, if a telecommunicator in tenant group A makes a call to tenant group B, the call is displayed as active in the status window for tenant group A and ringing in the status window for tenant group B.

The **Current Emergency Calls** window is displayed in Figure 4.28. Call counts, alarm threshold settings, and statistics are available at the top of the **Current Emergency Calls** and **Current Admin Calls** windows.

The columns visible in the window may have been customized by your administrator and may not appear exactly as shown.

Figure 4.28 Current Emergency Calls window



The following commands are available from the current emergency/admin calls window.

	Select an entry and click Answer call to answer/retrieve the selected call. Answer call can retrieve a call from any of the six call states with the exception of exclusive hold if you were not the original telecommunicator. Visibility of the button is dependent on the privileges assigned to your role.
	Click Lock/Unlock to freeze the contents of the window at the current page so you have the ability to precisely select the entry required.
	Click Monitor call to conference into the selected call in a monitoring state: you have one-way audio only and are not able to communicate on the call. Visibility of the button is dependent on the privileges assigned to your role.
	Click Reset to clear the statistics values. Visibility of the button is dependent on the privileges assigned to your role.

	Click Scroll up to navigate to the entry above the current selection.
	Click Scroll down to navigate to the entry below the current selection.
	Click Close to close the current calls window when it is in floating mode.

Participate in call

Step 1 Select a call from the current calls window.

Step 2 Click **Answer call**.

Result of this task: If the call status is:

- Ringing or queued - you answer the call
- Hold - you retrieve the call with the original user remaining on hold
- Active - you conference into the call

Info

Answering a call from the tenant status window before the call is displayed in your Phone Connection window prevents the call from being part of normal ring group distribution. As a result, wrap up time, if set, is not applied when you release the call.

4.16.3.1 Call monitoring

Call monitoring allows users to listen in on a call answered by a telecommunicator within the same tenant group. Any number of users can monitor the same call. The monitoring user can listen in on any call available from the **Current Emergency Calls** or **Current Admin Calls** window.

The monitoring user is "invisible" to the call participants; there is no audible or visual indication to the parties being monitored. As the monitoring user, you can easily choose to participate (two-way audio) in the call by barging into the monitored call. When you barge

into the monitored call, an audible indicator (one beep) is played and an entry for your position is added to the **Phone Connection** window.

If you are monitoring an emergency call, the location information is displayed on your screen. Additionally, the complete audio state of the call is presented at your position - you are aware of all of the voice control actions taken on the call; mute, privacy, hold, transfer/conference to other parties. Call monitoring continues as parties are added or removed from the call until the call is terminated. Call monitoring can be stopped at any time.

For text calls the complete conversation is available in the **Text Conversation** window at your position.

Monitor call

Step 1 Select a call from the current calls window.

Step 2 Click **Monitor call**.

Step outcome: The call is added to your **Phone Connection** window, displaying a "Monitoring" status and the ALI is also available in the **NG9-1-1 Info** window for emergency calls. However, other parties on the call are not aware of your presence on the call either visually or audibly.

Stop monitoring

Step 3 Ensure the "position" entry is selected.

- Click **Release**.

Step outcome: You are removed from the call.

- Click a disconnect reason button if available.

Step outcome: You are removed from the call and the disconnect reason is displayed in the **Telecommunicator Call History** window at your position only. The disconnect reason in the **Call History** window displays the reason the active telecommunicator selects when they end the call.

Info

If you rejoin the call in monitoring mode, the disconnect reason in the Telecommunicator Call History window is updated if you select a different disconnect reason to stop monitoring the call.

Barge into monitored call

More info: While in call monitoring mode, you can barge into the currently monitored call.

Step 1 Ensure the monitored call is selected in the current calls window.

Step 2 Click **Answer call** in the current calls window.

Result of this task: Two-way audio becomes enabled and your position is added to the call in the **Phone Connection** window.

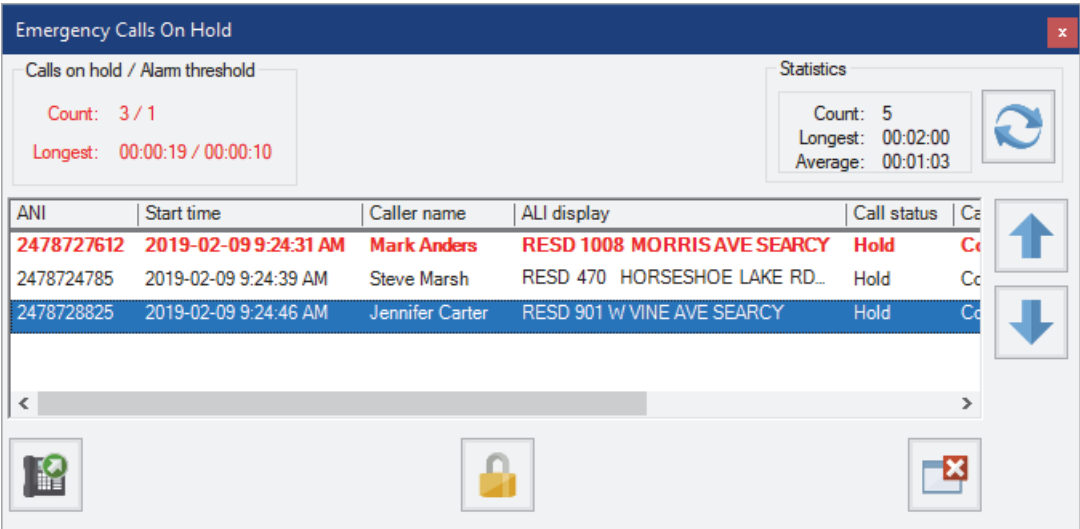
4.16.4 Emergency/admin calls on hold

The **Emergency Calls On Hold** window displays a scrollable and sortable list of all the in progress emergency and AdminEmergency calls in which the caller is in a hold or exclusive hold state, and allows you to retrieve a call. You cannot retrieve a call from exclusive hold if you were not the telecommunicator to place it on exclusive hold. The **Admin Calls On Hold** window displays the same information for administrative calls.

The **Emergency Calls on Hold** window is shown in Figure 4.29. Call counts, alarm threshold settings, and statistics are available at the top of the **Emergency Calls On Hold** and **Admin Calls On Hold** windows.

The columns visible in the window may have been customized by your administrator and may not appear exactly as shown.

Figure 4.29 Emergency Calls On Hold window



The following commands are available from the emergency/admin calls on hold window.

	Select an entry and click Answer call to retrieve the selected call. Visibility of the button is dependent on the privileges assigned to your role.
	Click Lock/Unlock to freeze the contents of the window at the current page so you have the ability to precisely select the entry required.
	Click Reset to clear the statistics values. Visibility of the button is dependent on the privileges assigned to your role.
	Click Scroll up to navigate to the entry above the current selection.
	Click Scroll down to navigate to the entry below the current selection.
	Click Close to close the current calls window when it is in float-ing mode.

Retrieve call from hold

Step 1 Select a call and click **Answer call**.

Step outcome: The original telecommunicator remains on hold and you are connected with the caller.

Release the original telecommunicator

Step 2 Select the entry for the original telecommunicator in the **Phone Connection** window.

Step 3 Click **Release**.

Result of this task: The original telecommunicator is released from the call and you remain connected to the other party.

4.16.5 Waiting emergency/admin calls

The **Waiting Emergency Calls** window displays a scrollable and sortable list of all the emergency and AdminEmergency calls that are currently ringing or queued while the **Waiting Admin Calls** window displays the same information for administrative calls.

The **Waiting Emergency Calls** window is shown in Figure 4.30. Call counts, alarm threshold, and statistics are available at the top of the both windows.

The columns visible in the window may have been customized by your administrator and may not appear exactly as shown.

Figure 4.30 Waiting Emergency Calls window

Waiting Emergency Calls							
Total waiting / Alarm threshold		Queued / Alarm threshold		Statistics			
Count: 4 / 0		Count: 0 / 0		Total waiting		Queued	
Longest: 00:00:30 / 00:00:30		Longest: 00:00:00 / 00:00:59		Count: 35		Count: 7	
				Longest: 03:08:39		Longest: 03:08:39	
				Average: 00:06:54		Average: 00:31:55	
Start time	ANI	Caller name	ALI display	Call status	Caller status	Elapsed time	
2019-02-04 3:05:09 PM	2478643941	Philip Smith	RESD 1007 W ARCH AVE SEARCY	Ringing	Connected	00:00:30	
2019-02-04 3:05:13 PM	2478643350	Jason Watts	RESD 1104 N LEMON ST BEEBE	Ringing	Connected	00:00:26	
2019-02-04 3:05:17 PM	2478641244	Laurie Dunbar	RESD 1196 FAIRVIEW RD ALBION	Ringing	Connected	00:00:22	
2019-02-04 3:05:25 PM	2475113120	Commcin Cellular	WRLS	Ringing	Connected	00:00:13	

The following commands are available from the waiting emergency/admin calls windows.

	Select an entry and click Answer call to answer the selected call. Visibility of the button is dependent on the privileges assigned to your role.
	Click Lock/Unlock to freeze the contents of the window at the current page so you have the ability to precisely select the entry required.
	Click Reset to clear the statistics values. Visibility of the button is dependent on the privileges assigned to your role.
	Click Scroll up to navigate to the entry above the current selection.
	Click Scroll down to navigate to the entry below the current selection.
	Click Close to close the current calls window when it is in floating mode.

Answer ringing call

- Step 1** Select an entry.
- Step 2** Click **Answer call**.

Info

Answering a call from the tenant status window before the call is displayed in your Phone Connection window, prevents the call from being part of normal ring group call distribution, as a result wrap-up, if set, is not applied to the call.

4.16.6 Abandoned emergency calls

Clicking **Abandoned** on the tenant status bar opens the **Abandoned Emergency Calls** window. This window displays a scrollable and sortable list of all the emergency and AdminEmergency calls that have been abandoned (caller hung up) before the calls were answered. These calls are maintained in the Guardian system and remain either ringing or queued, though they have not been answered.

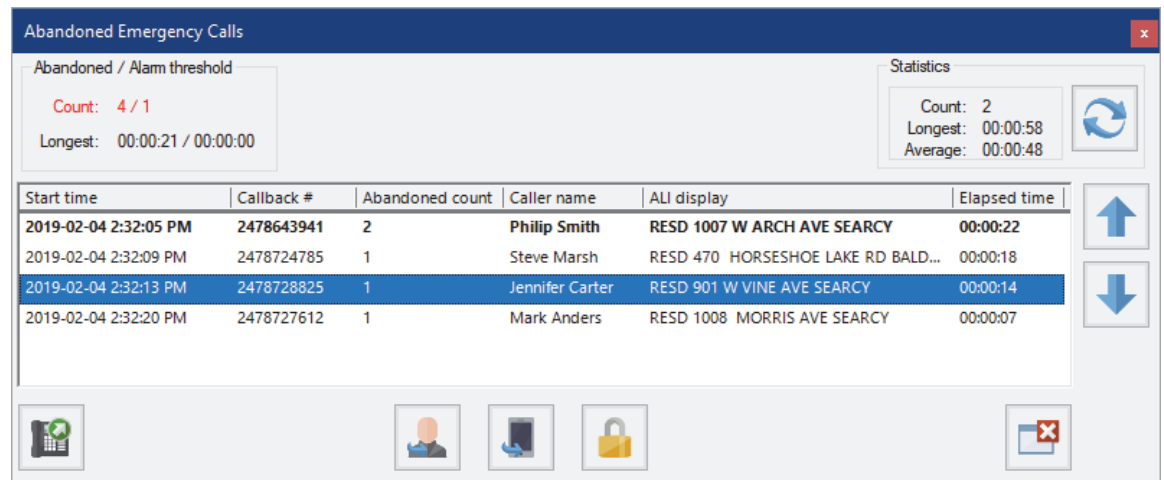
Abandoned Emergency Calls is shown in Figure 4.31. Call counts, alarm threshold, and statistics are available at the top of the **Abandoned Emergency Calls** window.

4.16.6.1 Abandoned call replacement

If your PSAP is configured for Abandoned Call Replacement, calls made from the same phone number are condensed into one entry in the window. The *Abandoned count* column displays the number of times a caller from the same phone number has called and hung up. The *Callback #* column displays the number received from the ALI response used to identify calls from the same source. If a callback number is not available from the ALI, the ANI received with the call is used to identify the call.

The columns visible in the abandoned emergency calls window may have been customized by your administrator and may not appear exactly as shown.

Figure 4.31 Abandoned Emergency Calls window



The following commands are available from the **Abandoned Emergency Calls** window.

	Select an entry and click Answer call to answer the selected call. Visibility of the button is dependent on the privileges assigned to your role.
	Select an entry and click Answer and call back to retrieve the call and make an outgoing call to the ANI of the selected call. Visibility of the button is dependent on the privileges assigned to your role.
	Select an entry and click Answer and text back to retrieve the call and launch the Text Conversation window to send a SMS message to the callback number of the call (applicable for incoming wireless and text calls.) Visibility of the button is dependent on the privileges assigned to your role.
	Click Lock/Unlock to freeze the contents of the window at the current page so you have the ability to precisely select the entry required.
	Click Reset to clear the statistics values. Visibility of the button is dependent on the privileges assigned to your role.
	Click Scroll up to navigate to the entry above the current selection.
	Click Scroll down to navigate to the entry below the current selection.
	Click Close to close the current calls window when it is in float-ing mode.

Answer abandoned call

Step Select a call and click **Answer call**.

Result of this task: The call is presented in the **Phone Connection** window and the ALI is displayed, but the caller is not reconnected.

Answer call then call back

Step Select a call and then click **Answer and call back**.

Result of this task: This is the equivalent of answering an abandoned call from the **Phone Connection** window and then clicking **Callback** in the layout.

Answer call and text back

Step 1 Select a call and then click **Answer and text back**.

Step outcome: The call is answered at your position, the ALI for the call is displayed, and the **Text Conversations** window is launched in SMS mode.

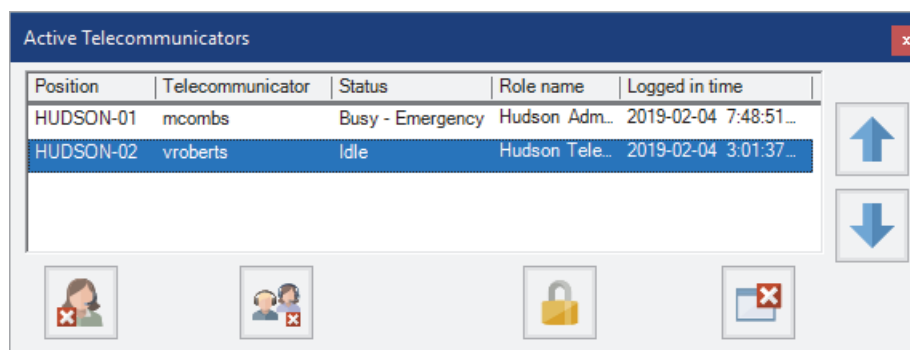
Step 2 Enter the text of your message.

Step 3 Click **Send**.

4.16.7 Active telecommunicators

Clicking **Active** on the tenant status bar opens the **Active Telecommunicators** window. This window displays a scrollable and sortable list of all users currently logged in to the tenant group as shown in Figure 4.32.

Figure 4.32 Active Telecommunicators window



The following commands are available from the **Active Telecommunicators** window.

	Select an entry and click Force log out to log out the selected user from Guardian. Visibility of the button is dependent on the privileges assigned to your role.
	Select an entry and click Telecommunicator monitoring to enable automatic monitoring of all calls answered by the selected user. Visibility of the button is dependent on the privileges assigned to your role.
	Click Telecommunicator monitoring to disable current monitoring of the specified user. Visibility of the button is dependent on the privileges assigned to your role.
	Click Lock/Unlock to freeze the contents of the window at the current page so you have the ability to precisely select the entry required.
	Click Scroll up to navigate to the entry above the current selection.
	Click Scroll down to navigate to the entry below the current selection.
	Click Close to close the current calls window when it is in floating mode.

Force log out

- Step 1** Select an entry from the **Active Telecommunicators** window.
- Step 2** Click **Force log out**.

Result of this task: The selected telecommunicator is logged out of Guardian at their position. Any active calls at the position are rerouted.

4.16.7.1 Telecommunicator monitoring

Telecommunicator Monitoring allows a user to automatically listen in on all calls answered by the specified telecommunicator as in a trainer/trainee scenario. The monitoring user (trainer) can log into any position belonging to the same tenant group as the monitored telecommunicator (trainee) to monitor their calls - they do not need to be connected to the same PAC II unit.

The monitoring user is automatically placed in a "listen" state - they can hear the conversation but their audio is muted. The monitoring user can easily choose to participate (two-way audio) in the call by barging into the monitored call.

The monitoring user cannot affect the participants on the call (hold, mute, privacy); the only action available to the trainer is to place themselves on hold. If they release themselves from an active call, they are automatically reconnected.

The trainer can make outgoing calls while in telecommunicator monitoring mode - they are placed on hold on any currently monitored call.

Enable telecommunicator monitoring

Step 1 Select an entry from the **Active Telecommunicators** window.

Step 2 Click **Telecommunicator monitoring**.

Result of this task: You are placed in busied out - monitoring mode. You are connected to all calls answered by the selected telecommunicator in a monitoring state.

Barge into monitored call

Step While in telecommunicator monitoring mode, you can barge into the currently monitored call via one of these two methods:

- Click **Answer Call** in the layout.

Step outcome: Two-way audio becomes enabled and you are removed from telecommunicator monitoring mode to busied out mode. If you release yourself from the call, your position is busied out and telecommunicator monitoring is disabled. After releasing yourself from the call, your position

is busied out mode. Click **Make Busy** to become available for incoming calls.

- Click **Answer call** from a current calls tenant status window.

Step outcome: Two-way audio becomes enabled and you remain in telecommunicator monitoring mode. If you release yourself from the call, you are reconnected to the call and telecommunicator monitoring is enabled.

Disable telecommunicator monitoring

Step Click **Telecommunicator monitoring**.

Result of this task: Your position is removed from busied out - monitoring mode and telecommunicator monitoring are stopped. Finally your position is placed in busied out mode. Calls are routed to your position as specified by the busied out privileges assigned to your role. Click **Make Busy** to become available for incoming calls.

Escape telecommunicator monitoring

More info: If you, as a monitor, need to remove yourself from telecommunicator monitoring mode quickly and become available for incoming calls, the easiest procedure is the following.

Step 1 Click **Make Busy**.

Step 2 Ensure your entry is still selected in the **Phone Connection** window.

Step 3 Click **Release**.

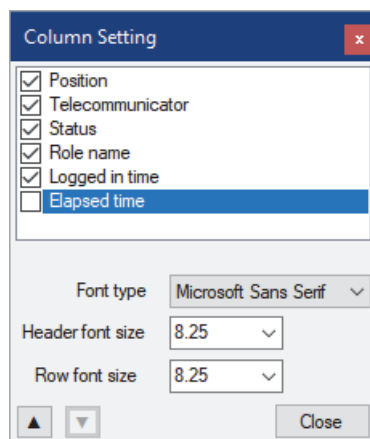
View elapsed time in current state

Step 1

Right-click the column header.

Step outcome: The **Column Setting** dialog box is displayed.

Example:



Step 2

Select **Elapsed time**.

More info: Elapsed time tracks the time the telecommunicator has been in their current state. The time is reset whenever the telecommunicator changes their state.

Step 3

Click **Close**.

Result of this task: The **Elapsed time** column is added to the **Active Telecommunicators** window.

When done, remember to: Save the layout to include the column permanently.

4.17 Tenant dashboard

The Tenant Dashboard feature provides real-time statistical information about emergency calls answered within the tenant group and any defined subtenants.

The **Tenant Dashboard** window consists of three distinct metrics: Grade of Service, Calling Activity, and Position Call Status. You can view all three statistical components in one window, **Tenant Dashboard**, or you can view each metric in its own window.

Menu: View > Tenant Dashboard > Tenant Dashboard/Grade Of Service/Calling Activity/Position Call Status

Figure 4.33 Sample Tenant Dashboard window



The data, color, and columns are customizable by your administrator and may not appear exactly as shown.

The three statistical metrics provide the following data:

- 1 Grade of Service — chart to display the answer rate for two defined time thresholds for 15 minute and 24 hour intervals

- 2 Call Activity — tabular view of the number of calls answered per subtenant and average answer time and occupancy percentage
- 3 Position Call Status — chart displaying the status of positions per subtenant

4.18 Procedural messages

The Procedural Messages feature assists you in following your PSAP protocols. Phrases are displayed within the **Procedural Messages** window when certain application features are launched or certain actions executed.

Messages are displayed once you answer a call and are added to the window for the duration of the call. The newest message is added to the bottom of the list and appears in bold as shown in Figure 4.34. As you launch a feature or execute an action, any messages configured for the event are added to the window. If an event was configured with more than one message, the messages appear simultaneously in the window, each on their own line, in the order they were created by your administrator.

Figure 4.34 Procedural Messages window

Procedural Messages	
Time	Message
3:05:44 PM	Emergency Police Fire EMS?
3:05:44 PM	Verify location of emergency.
3:37:46 PM	Has the poisoned party vomited?
3:37:46 PM	Has the poisoned party injected anything after the poison? Milk? Water?
3:37:46 PM	Is the poisoned party conscious?
3:37:46 PM	Is the poisoned party breathing steadily?

The window is cleared of messages when the call is released.

If the **Procedural Messages** window is not docked in your layout, your administrator may have included the **Messages** button in your layout. If a message is added to the window, the button changes color as shown in Figure 4.35. Clicking on the button launches the **Procedural Messages** window.

Figure 4.35 Procedural Messages button

4.19 Ticker message bar

The **Ticker Message Bar** displays user configurable or preset text that scrolls horizontally, right to left, on the Guardian position interface. The text consists of one or multiple messages displayed on a single text line with a slider bar to increase or decrease the scrolling speed.

If **Ticker Message Bar** is not visible in your layout, launch it from the **View** menu as a floating window.

Menu: View > Ticker Message Bar

Alternately, your administrator may have included the **Ticker** button in your layout. When displayed, the **Ticker** button notifies you of incoming messages by means of the background color which is customizable by your administrator. The background color (default red) indicates one or more messages are active. When there are no active messages the button's default color is displayed.

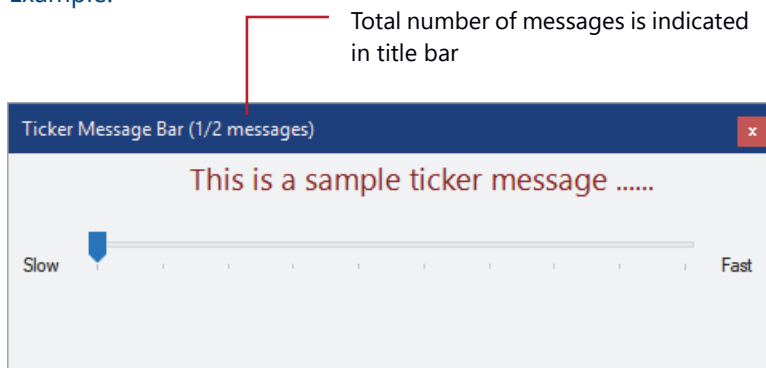
Figure 4.36 Ticker button

View ticker message bar

- On the **View** menu, select **Ticker Message Bar**.
- If available, click **Ticker** in the layout.

Step outcome: The **Ticker Message Bar** opens.

Example:



Change message scrolling speed

Step 1 Drag the slider to the left to decrease scrolling speed

Step 2 Drag the slider to the right to increase scrolling speed.

4.20 Local text conversations

The **Local Text Conversations** window displays the written text of all past 9-1-1 text conversations and instant messages with callers that occurred on the current position, regardless of user account. It also allows you to print the conversations. The messages are retained until they are manually removed by an administrator from the individual positions.

If **Local Text Conversations** is not visible in your layout, launch it from the **View** menu as a floating window.

Menu: **View > Local Text Conversations**

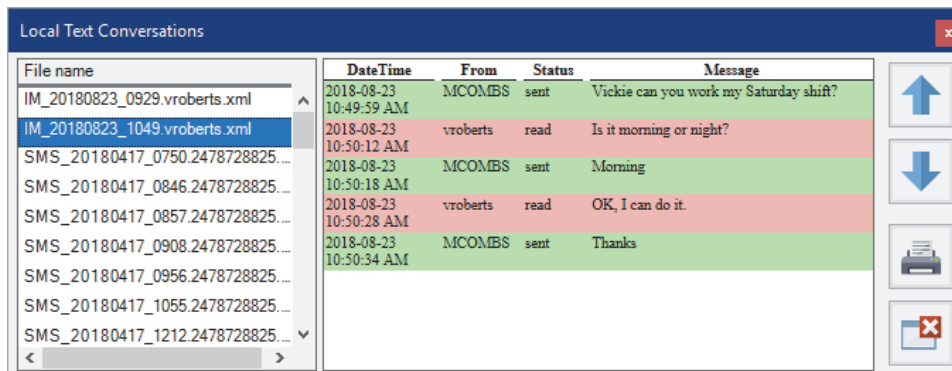
View local text conversations

Step 1 Launch **Local Text Conversations** by either of these two methods:

- On the **View** menu, select **Local Text Conversations**.
- If available, click **Conversation** in the layout.

Step outcome: The **Local Text Conversations** window opens.

Example:



More info: IM, SMS, and TDD conversations that have occurred on your current position are listed in the left panel of the window.

Step 2 Select an entry in the left panel.

Step outcome: The written conversation is displayed in the right panel of the window. The incoming and responding messages are distinguished by color.

Step 3 To print the conversation, click **Print**.

Step 4 Click **Close** to exit **Local Text Conversations**.

4.21 Websites

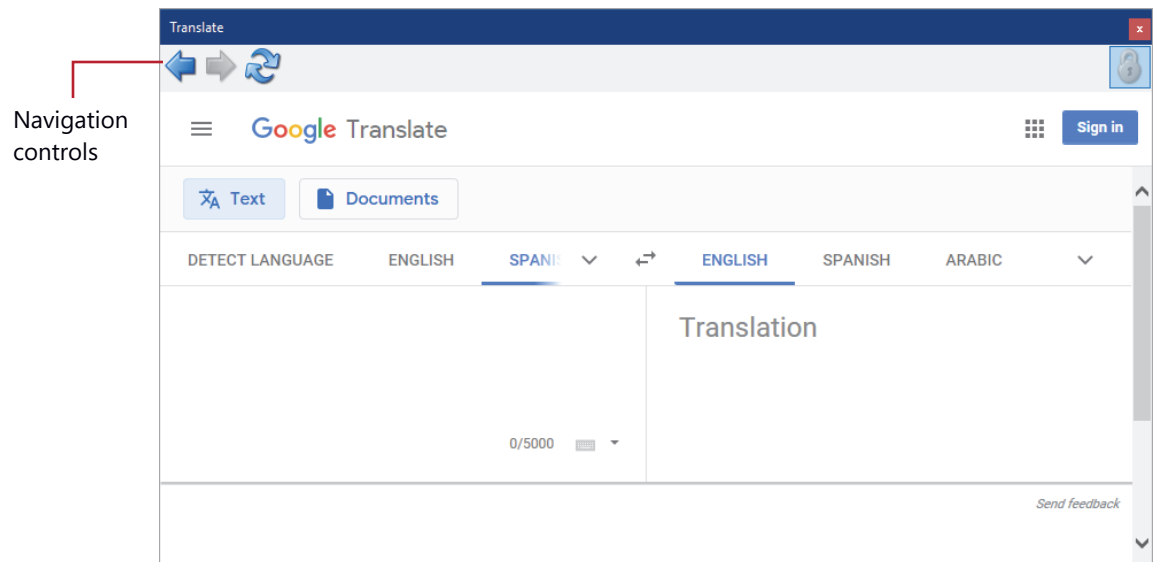
Your administrator may have configured website links that you can access from Guardian. These links are to specific websites and do not offer the ability to surf the Internet from your Guardian position.

You can access all website links configured for your role on the **View** menu.

Menu: **View > Websites**

If you select an entry from the list, a browser window opens in Guardian and the linked website is displayed. Your administrator may configure navigation controls for the website allowing you to navigate backwards/forwards from web pages and refresh the web page as shown in Figure 4.37.

Figure 4.37 Website browser window



Launch website

- Step 1** On the **View** menu, click **Websites**.
Step outcome: A list of selectable websites appears.
- Step 2** Select a website from the list.

Result of this task: A browser window opens displaying the website.

More info: Depending on the navigation privileges, you can use the **Back**, **Forward**, and **Refresh** buttons.

4.22 Instant replay recorder

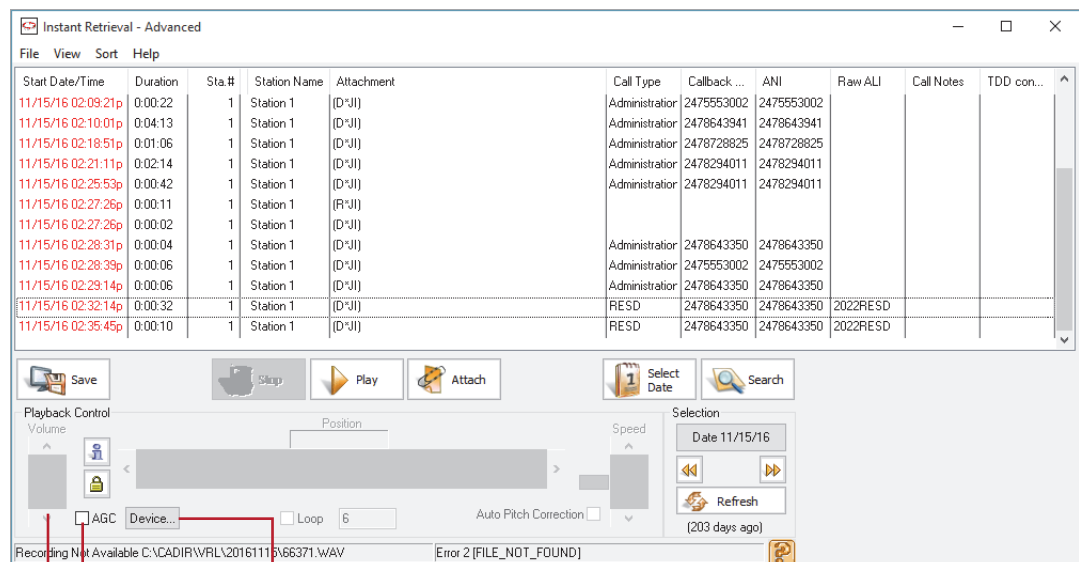
Guardian systems have two recording options: local recording and centralized recording.

Local recording records the audio of all calls answered at the Guardian position. This provides you with immediate access to replay any portion of a past conversation. Each call is stored as an individual file and an automation task handles deletion of old files as needed, so there is no administration required to manage the recordings at the positions. The default retention period of recordings is 20 days but this period may be customized by your administrator.

Centralized recording is a more comprehensive solution. It provides complete log recording capabilities of the audio, CDR, ALI, and TDD conversations from all positions to the server.

If your position is configured with local recording capabilities, the **IRR** (Instant Replay Recorder) button is included in your layout. This button launches a voice recording playback application that gives you access to all the voice communications that passed through your position as shown in Figure 4.38. If your position also features integrated radio communication, radio conversations are also recorded and available from the same instant retrieval window.

Figure 4.38 Instant Retrieval window



Playback volume control

AGC

Device selector

Play recording

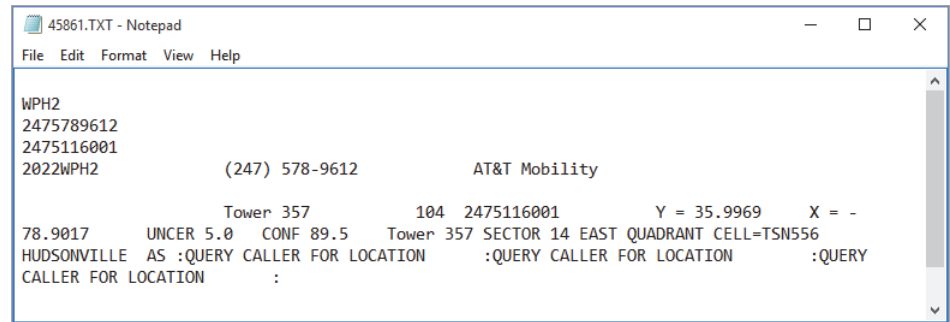
- Step 1** Click **IRR** in the layout.
Step outcome: The **Instant Retrieval - Advanced** window opens. All recordings from today, regardless of the telecommunicator logged in, are displayed.
- Step 2** Select the recording you want to play.
Step outcome: The recording starts to play in your headset automatically.
- Step 3** If you want to play a recording from a previous date, click **Select Date**.
Step outcome: The **Choose New Date** dialog box opens.
- Step 4** Click the date of the recording you are interested in accessing.
Step outcome: **Instant Retrieval - Advanced** displays the recordings from the date you selected.
- Step 5** Select the recording you want to play.
Step outcome: The recording starts to play in your headset automatically.
- Step 6** You can increase/decrease the volume of the recording playback using the volume slider in the **Instant Retrieval - Advanced** window.
More info: If your recording sounds "scratchy", ensure the **AGC** (Automatic Gain Control) option is not enabled.
- Step 7** If you would prefer to have the recording play back through your position's external speakers, click **Device**.
- Select the entry representing your speakers from the list.
 - Click **OK**.

View RAW ALI for emergency call

Step 1 Select the call and click **Attach**.

Step outcome: The received ALI for the call is displayed in a Microsoft Notepad window.

Example:



The screenshot shows a Notepad window titled "45861.TXT - Notepad". The text inside the window is as follows:

```
WPH2
2475789612
2475116001
2022WPH2          (247) 578-9612          AT&T Mobility

                Tower 357          104 2475116001          Y = 35.9969          X = -
78.9017          UNCR 5.0  CONF 89.5  Tower 357 SECTOR 14 EAST QUADRANT CELL=TSN556
HUDSONVILLE AS :QUERY CALLER FOR LOCATION          :QUERY CALLER FOR LOCATION          :QUERY
CALLER FOR LOCATION          :
```

Step 2 Close the **Instant Retrieval - Advanced** window when you are finished listening to recordings.

5 Emergency calls

This chapter describes the procedures to answer and process emergency voice calls by a user of the Guardian client application stationed at an answering position. Guardian users may be PSAP telecommunicators, supervisors, or administrators.

Due to the variety of phone line services available the most common and simple methods are documented.

5.1 Answering calls

When an emergency call arrives, a ring tone may be heard in either the headset/handset or from the desktop speakers at your position depending on system configuration. A new entry is added to the **Phone Connection** window and **NG9-1-1 Info** flashes "9-1-1". Your administrator may also have opted to display the ALI data on your screen prior to answering; otherwise, it is displayed once the call is answered.

Answer emergency call

- | | |
|-------------|--|
| Step | Perform one of the following actions to answer the call: <ul style="list-style-type: none">• Click the flashing NG9-1-1 Info window.• Click the Answer button.• Double-click the entry in the Phone Connection window.• If available, click the line appearance button.• Select the call in the Waiting Emergency Calls window and click Answer call.• If a keyboard shortcut is configured for your role, enter the keys.• If a keypad is configured for your position, click Answer on the keypad. |
|-------------|--|

5.1.1 Automatic hold on answer

Any active calls are placed on hold automatically when you answer a new call; there is no need to place the existing call on hold first.

5.2 Automatically answered calls

The system may be configured to automatically answer incoming calls at your position. If this is the case, when a call arrives at your position, the system plays a short tone in your headset just before the call is answered to notify you that you are connected with a caller. All remaining call handling functionality remains the same as for manually answered calls.

5.2.1 Multiple automatically answered calls

When all telecommunicators are on active calls, new incoming calls are queued until a telecommunicator is available. If both emergency and administrative calls are in queue, emergency calls take precedence and are automatically answered in order of arrival. Queued administrative calls are only automatically answered when no emergency calls remain in the queue. If multiple queues are configured, the calls are answered in order of arrival following the priority of the queues.

Answer call out of sequence

More info: Use this procedure when automatic call answer is enabled but you need to answer a specific call that is not necessarily the next in the queue.

Step 1 Click **Make Busy**.

More info: This prevents the next call in queue being answered automatically at your position.

Step 2 Select either of the following options for the current active call:

- Release current call.
- Place current call on hold.

Step 3 Open either of the following tenant status windows depending on the call you want to answer:

- **Waiting Emergency Calls**
- **Abandoned Emergency Calls**

Step 4 Select a call from the list.

Step 5 Click **Answer call** in the tenant status window.

5.2.1.1 Wrap up

Typically, a wrap up time is configured for after the call is released. The wrap up time is a short period, starting as soon as the current call is released, in which no other calls are routed to your position to give you breathing space between automatically answered calls.

The **NG9-1-1 Info** window notifies you when you are in wrap up mode as shown in Figure 5.1.

Figure 5.1 NG9-1-1 Info displays wrap up status

Your administrator may configure the wrap up time to a fixed time period such as five seconds or they may assign you the ability to manually remove yourself from wrap up mode when you are ready.

End wrap up mode

Step

While in wrap up mode, click **Make Busy**.

Step outcome: The **NG9-1-1 Info** window status changes to indicate you are no longer in wrap up mode.

5.2.2 Automatic greeting

Your administrator may have an automatic greeting configured for the system. This automatic greeting plays right after the zip tone is heard after the call is answered. The greeting can be heard by your caller and you. You can control the volume of the greeting playing in your headset, stop playing the greeting, or disable the greeting, depending on the privileges assigned by your administrator and if the action buttons are included in your role layout.

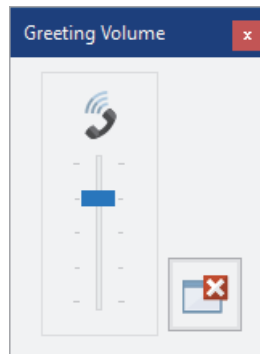
Note: Regardless if a greeting is playing, you always have two-way audio with your caller.

Adjust volume of greeting

Step 1 While the greeting is playing, click the greeting **Volume** button.

Step outcome: The **Greeting Volume** dialog box is displayed.

Example:



Step 2 Move the slider to adjust the volume of the greeting playing in your headset.

More info: The slider does not affect the volume played to the caller.

Step 3 Click **Close**.

More info: The volume adjustment is in effect until you log out or you readjust the volume.

Stop playing greeting

Step While the greeting is playing, click **Stop**.

Result of this task: The greeting ceases to play to your caller and in your headset for this call only. The greeting resumes play for all new incoming calls.

Disable/enable greeting feature

Step 1 To disable the automatic greeting feature, click **Disable**.

Step outcome: The button changes to notify you that the feature is disabled.

More info: The automatic greeting feature is disabled until you log out or until you click **Enable** to re-enable the feature.

Step 2 To re-enable the automatic greeting feature, click **Enable**.

5.3 Abandoned emergency calls

An emergency call that is released before a telecommunicator can answer the call becomes an abandoned emergency call. These calls are not released by the Guardian system. In fact, they continue to ring at the positions until a telecommunicator answers the call. You can identify an abandoned call by viewing the caller status in the **NG9-1-1 Info** window.

Note: Exclusive hold is not available when using the callback button for an abandoned call.

Answer abandoned call

- Step 1** Answer the ringing call like a regular emergency call.
- Step 2** Once the call is answered, click **Callback**.

Answer abandoned call and text back

- Step 1** Answer the ringing call like a regular emergency call.
- Step 2** Once the call is answered, click **Text callback**.
Step outcome: The **Text Conversation** window is displayed in SMS mode.
- Step 3** Enter the text of your message and click **Send**.

5.3.1 Queued abandoned calls

Instead of ringing at the positions, the system may be configured to queue abandoned emergency calls. If this is the case, whenever a call is abandoned by the caller, it is removed from your **Phone Connection** window and is placed in a special queue awaiting handling by an available user. Abandoned calls that are assigned to the queue are accessible from the **Abandoned Emergency Calls** window.

Note: Abandoned calls are not placed in the queue when your role has automatic call answer enabled.

Answer abandoned call from queue

Step 1 Select an entry in the **Abandoned Emergency Calls** window.

Retrieve ALI data first

Step 2 Click **Answer call**.

Step outcome: The call is answered at your position and the ALI for the call is displayed. An automatic callback to the caller is not made.

Step 3 Click **Callback** in your layout.

Call back immediately

Step 4 Click **Answer and call back**.

Step outcome: The call is answered at your position, the ALI for the call is displayed, and an automatic callback is made to the caller.

Answer abandoned call from queue and text back

Step 1 Select an entry in the **Abandoned Emergency Calls** window.

Retrieve ALI data first

Step 2 Click **Answer call**.

Step outcome: The call is answered at your position and the ALI for the call is displayed. An automatic callback to the caller is not made.

Step 3 Click **Text callback** in your layout.

Text back immediately

Step 4 Click **Answer and text back**.

Step outcome: The call is answered at your position, the ALI for the call is displayed, and the **Text Conversations** window is launched in SMS mode.

Step 5 Enter the text of your message and click **Send**.

5.3.2 Disconnected calls

When an emergency caller hangs up after the call is answered by a telecommunicator, or the call is lost for a technical reason, it is not treated as an abandoned call and placed in the queue.

In this scenario, the call becomes a disconnected call and remains displayed in your **Phone Connection** window. The status for the caller's entry changes to "9-1-1 disconnected" and you hear a short double-beep tone in your headset/handset.

Call back disconnected caller

- Step 1** Ensure the disconnected call is selected in the **Phone Connection** window (regardless which line has focus.)
- Step 2** Click **Callback** in your layout.

Result of this task: The callback phone number from the ALI, if available, or the ANI displayed in the **NG9-1-1 Info** window is used to dial back the caller.

Text back disconnected caller

- Step 1** Ensure the disconnected call is selected in the **Phone Connection** window (regardless which line has focus.)
- Step 2** Click **Text callback** in your layout.
Step outcome: The **Text Conversation** window is launched in SMS mode to the callback number of the call.
- Step 3** Enter the text of your message and click **Send**.

Release unavailable callback

- Step 1** Ensure the disconnected call is selected in the **Phone Connection** window (regardless which line has focus.)
- Step 2** Click **Callback** in your layout.
Step outcome: The callback phone number from the ALI, if available, or the ANI displayed in the **NG9-1-1 Info** window is used to dial back the caller.

- Step 3** If you receive a busy signal, unavailable message, or the callback cannot be completed for some other reason:
- a** Select the entry for the callback in the **Phone Connection** window.
 - b** Click **Release**.

Result of this task: The callback attempt is released but you still have the original disconnected call displayed in your **Phone Connection** window. You can reattempt a callback as directed in the previous procedure.

Release disconnected call after call back

More info: After making a callback for a disconnected call, it is recommended to release the entire call.

- Step 1** Click the topmost line of the call in the **Phone Connection** window.
- Step 2** Click **Release**.

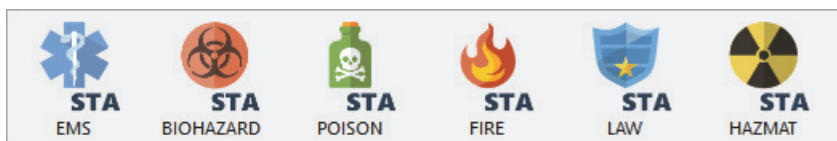
Result of this task: The entire call is released and cleared from your **Phone Connection** window.

5.4 Conference/transfer calls

Although you may normally answer and handle an incoming call yourself, there are many instances when you need to conference another party to the active call or you may need to transfer the call to another emergency service. The Guardian system supports selective transfers, fixed transfers, or manually dialed phone numbers.

5.4.1 Selective transfer agency

Selective transfer agency (STA) buttons are special dial commands that assist you to conference/transfer a 9-1-1 call to another emergency service without having to worry about the specific geographical or jurisdictional location of the 9-1-1 caller. The availability of STA functionality is directly correlated to the equipment of the central office supporting your PSAP. Figure 5.2 shows typical selective transfer agencies; however, more or less may be configured for your PSAP.

Figure 5.2 STA toolbar

If STA functionality is available, the phone numbers are assigned to the STA buttons dynamically with every incoming 9-1-1 call. If STA functionality is not available, and STA buttons are included in your layout, the buttons remain disabled during the call.

You can see the assignments for the active 9-1-1 call in the **NG9-1-1 STA Info** window. It displays a list of selective transfer agencies populated from the geographic or jurisdictional information received with the incoming call.

Conference via STA buttons

Step 1 Answer the incoming emergency call.

Step 2 Click the STA button for the agency required.

[More info:](#) A new entry is not displayed in the **Phone Connection** window.

Result of this task: A call is made to the agency. You and your caller hear the ringing to the other agency. When the agency answers, they are automatically conferenced into the call.

Release STA participant

Step 1 Ensure the emergency call is selected in the **Phone Connection** window.

Step 2 Click **Hook Flash**.

Result of this task: The STA participant conferenced into the call is released while you remain connected to the original emergency caller.

Transfer via STA buttons

Step 1 Answer the incoming emergency call.

Step 2 Click the STA button for the agency required.

More info: A new entry is not displayed in the **Phone Connection** window.

Step outcome: A call is made to the agency. You and your caller hear the ringing to the other agency. When the agency answers, they are automatically conferenced into the call.

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the STA party remain connected.

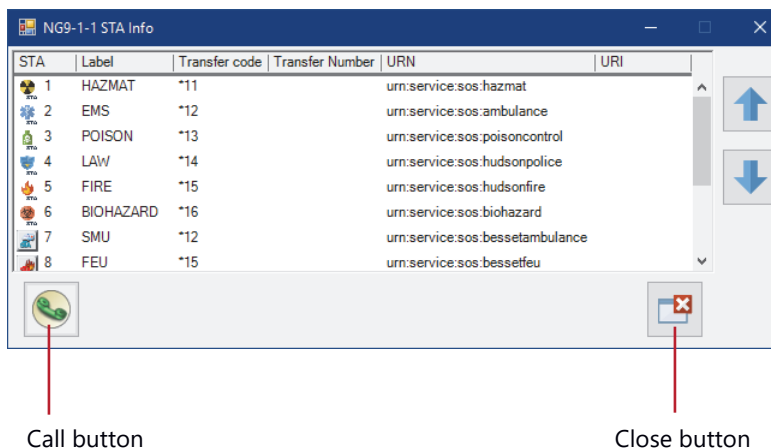
Using STA Info window

Step 1 Answer the incoming emergency call.

Step 2 Click **STA info**.

Step outcome: The **NG9-1-1 STA Info** window is displayed.

Example:



Step 3 From **NG9-1-1 STA Info**, select the agency to call.

Step 4 Click **Call**.

More info: A new entry is not displayed in the **Phone Connection** window.

Step outcome: A call is made to the agency. You and your caller hear the ringing to the other agency. When the agency answers, they are automatically conferenced into the call.

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the STA party remain connected.

5.4.1.1 STA transfers

Typically, when you have a selective transfer agency conferenced into your active 9-1-1 call, you do not see an entry for the agency in your **Phone Connection** window. However, if your system is connected to an Emergency Services IP network (ESInet), the selective transfer agency has its own entry added to the **Phone Connection** window. You are able to interact with the entry as with a regular conference party; i.e., mute, hold, and privacy is enabled.

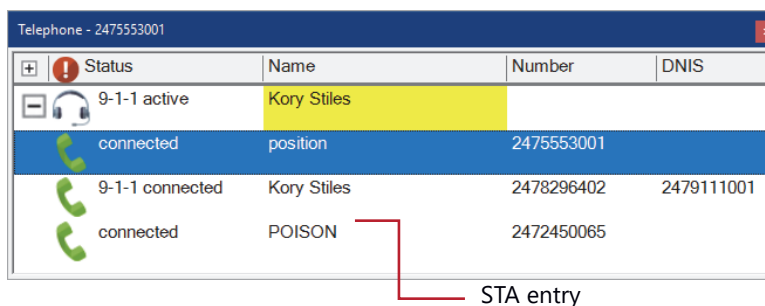
Conference via STA buttons

Step 1 Answer the incoming emergency call.

Step 2 Click the STA button for the agency required.

Step outcome: A new entry is displayed in the **Phone Connection** window.

Example:



Status	Name	Number	DNIS
9-1-1 active	Kory Stiles		
connected	position	2475553001	
9-1-1 connected	Kory Stiles	2478296402	2479111001
connected	POISON	2472450065	

STA entry

Result of this task: A call is made to the agency. You and your caller hear the ringing to the other agency. When the agency answers, they are automatically conferenced into the call.

Release STA Participant

- Step 1** Select the selective transfer agency entry in the **Phone Connection** window.
- Step 2** Click **Release**.

Result of this task: The conferenced STA participant is released while you remain connected to the emergency caller.

Transfer via STA buttons

- Step 1** Answer the incoming emergency call.
- Step 2** Click the STA button for the agency required.
Step outcome: A new entry is displayed in the **Phone Connection** window. A call is made to the agency. You and your caller hear the ringing to the other agency. When the agency answers, they are automatically conferenced into the call.
- Step 3** Ensure the “position” entry is selected in the **Phone Connection** window.
- Click **Release**.
 - Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the STA party remain connected.

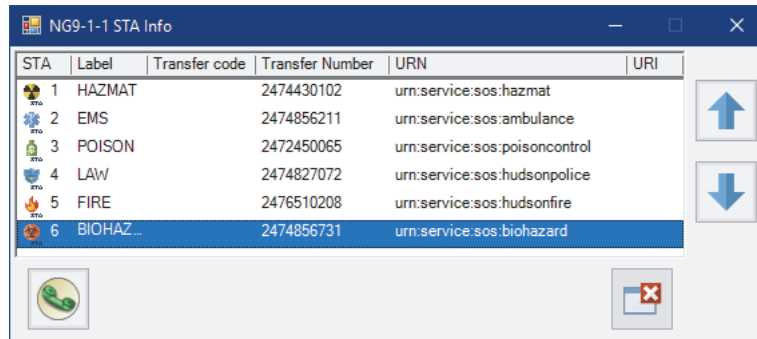
STA Info window

Step 1 Answer the incoming emergency call.

Step 2 Click **STA info**.

Step outcome: The **NG9-1-1 STA Info** window is displayed.

Example:



Step 3 From **NG9-1-1 STA Info**, select the agency to call.

Step 4 Click **Call**.

Step outcome: A new entry is displayed in the **Phone Connection** window. A call is made to the agency. You and your caller hear the ringing to the other agency. When the agency answers, they are automatically conferenced into the call.

Step 5 Ensure the "position" entry is selected in the **Phone Connection** window.

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the STA party remain connected.

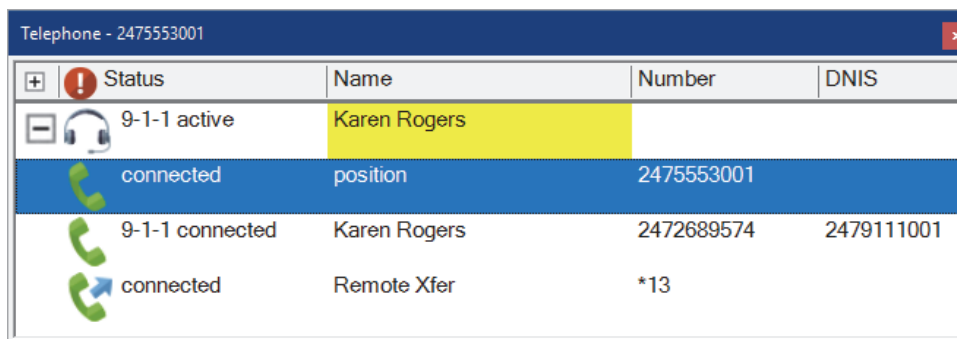
5.4.1.2 SIP REFER

Typically, when you have a selective transfer agency conferenced into your active 9-1-1 call, you do not see a visible entry for the agency in your **Phone Connection** window. However, when the phone company managing the selective transfer agency uses a special method known as SIP REFER, you see an entry in the **Phone Connection** window as shown in Figure 5.3.

The *Name* column is set to "Remote Xfer" for the party and the status notifies you if a connection is made. Another difference: SIP REFER allows you to disconnect the selective

transfer agency by selecting their entry and clicking **Release** rather than using **Hook Flash** as with the standard method.

Figure 5.3 Call with SIP REFER transfer



Status	Name	Number	DNIS
9-1-1 active	Karen Rogers		
connected	position	2475553001	
9-1-1 connected	Karen Rogers	2472689574	2479111001
connected	Remote Xfer	*13	

The SIP REFER participant has their Name set to "Remote Xfer" and the icon is different from a regular connected participant.

Conference via STA buttons with SIP REFER

- Step 1** Answer the incoming emergency call.
- Step 2** Click the STA button for the agency required.
Step outcome: A new entry is displayed in the **Phone Connection** window.

Result of this task: A call is made to the agency. You and your caller hear the ringing to the other agency. When the agency answers, they are automatically conferenced into the call.

Release STA participant with SIP REFER

- Step 1** Select the "Remote Xfer" entry in the **Phone Connection** window.
- Step 2** Click **Release**.

Result of this task: The STA participant conferenced onto the call is released while you remain connected to the original emergency caller.

Transfer via STA buttons with SIP REFER

Step 1 Answer the incoming emergency call.

Step 2 Click the STA button for the agency required.

Step outcome: A new entry is displayed in the **Phone Connection** window. A call is made to the agency. You and your caller hear the ringing to the other agency. When the agency answers, they are automatically conferenced into the call.

Step 3 Ensure the “position” entry is selected in the **Phone Connection** window.

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the STA party remain connected.

5.4.2 Fixed transfer codes

Fixed transfer codes (FTC) are similar to STA except they are configured with preset phone numbers for emergency services, neighboring PSAPs, state police, and more, as shown in Figure 5.4. The availability of fixed transfer codes is dependent on your PSAP administrator and central office.

Figure 5.4 FTC toolbar



You can view the available list of fixed transfer codes in the **Fixed Xfer List** window. It displays a list of fixed transfer codes for your PSAP and is available for all incoming emergency calls.

Conference via FTC buttons

Step 1 Answer the incoming emergency call.

Step 2 Click the FTC button for the party required.

Step outcome: A new entry is not displayed in the **Phone Connection** window.

Result of this task: A call is made to the agency. You and your caller hear the ringing to the other agency. When the agency answers, they are automatically conferenced into the call.

Release FTC participant

Step 1 Select the incoming emergency call entry in the **Phone Connection** window.

Step 2 Click **Hook Flash**.

Result of this task: The FTC participant conferenced into the call is released while you remain connected to the original emergency caller.

Transfer via FTC buttons

Step 1 Answer the incoming emergency call.

Step 2 Click the FTC button for the party required.

More info: A new entry is not displayed in the **Phone Connection window**.

Step outcome: A call is made to the agency. You and your caller hear the ringing to the other agency. When the agency answers, they are automatically conferenced into the call.

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the FTC party remain connected.

5.4.2.1 FTC transfers

Typically, when you have a FTC agency conferenced into your active emergency call, you do not see an entry for the agency in your **Phone Connection** window. However, if your

system is connected to an ESInet, the fixed transfer code agency has its own entry added to the **Phone Connection** window. You are able to interact with the entry as with a regular conference party; i.e., mute, hold, and privacy is enabled.

Conference via FTC buttons

Step 1 Answer the incoming emergency call.

Step 2 Click the FTC button for the party required.

Step outcome: A new entry is displayed in the **Phone Connection** window.

Example:

Status	Name	Number	DNIS
9-1-1 active	HudsonLongestIdle		
connected	position	2475553001	
9-1-1 connected	Jason Watts	2478643350	2475554004
privacy	DeWalt Trauma Center	2477859622	

FTC party's entry

FTC entry on an ESInet can be interacted with such as placed on privacy

Result of this task: A call is made to the agency. You and your caller hear the ringing to the other agency. When the agency answers, they are automatically conferenced into the call.

Release FTC participant

Step 1 Select the FTC entry in the **Phone Connection** window.

Step 2 Click **Release**.

Result of this task: The FTC participant conferenced into the call is released while you remain connected to the original emergency caller.

Transfer using FTC buttons

Step 1 Answer the incoming emergency call.

Step 2 Click the FTC button for the party required.

Step outcome: A new entry is displayed in the **Phone Connection** window. A call is made to the agency. You and your caller hear the ringing to the other agency. When the agency answers, they are automatically conferenced into the call.

- Click **Release**.
- Click a disconnect reason button if available.

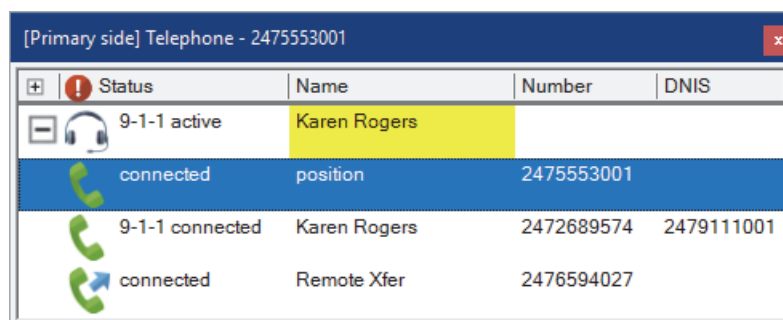
Result of this task: You are released from the call while your caller and the FTC party remain connected.

5.4.2.2 SIP REFER

Typically, when you have a FTC agency conferenced into your active emergency call, you do not see an entry for the agency in your **Phone Connection** window. However, when the phone company managing the fixed transfer code agency uses a special method known as SIP REFER, you see an entry in the **Phone Connection** window as shown in Figure 4.9.

The *Name* column is set to "Remote Xfer" for the entry and the status notifies you if a connection is made. Another difference: SIP REFER allows you to disconnect the fixed transfer code party by selecting their entry and clicking **Release** rather than using **Hook Flash** as with the standard method.

Figure 5.5 SIP REFER FTC transfer



The screenshot shows a window titled "[Primary side] Telephone - 2475553001". It contains a table with columns: Status, Name, Number, and DNIS. The table lists three participants: "9-1-1 active" (Karen Rogers), "9-1-1 connected" (Karen Rogers), and "connected" (Remote Xfer). The "Remote Xfer" entry is highlighted with a blue background and a green arrow icon, distinguishing it from the other entries which have a green phone icon.

Status	Name	Number	DNIS
9-1-1 active	Karen Rogers		
connected	position	2475553001	
9-1-1 connected	Karen Rogers	2472689574	2479111001
connected	Remote Xfer	2476594027	

The SIP REFER participant has their Name set to "Remote Xfer" and the icon is different from a regular connected participant.

Conference via FTC buttons with SIP REFER

Step 1 Answer the incoming emergency call.

Step 2 Click the FTC button for the agency required.

More info: A new entry is displayed in the **Phone Connection** window.

Result of this task: A call is made to the agency. You and your caller hear the ringing to the other agency. When the agency answers, they are automatically conferenced into the call.

Release FTC participant with SIP REFER

Step 1 Select the "Remote Xfer" entry in the **Phone Connection** window.

Step 2 Click **Release**.

Result of this task: The FTC participant conferenced into the call is released while you remain connected to the original emergency caller.

Transfer via FTC buttons with SIP REFER

Step 1 Answer the incoming emergency call.

Step 2 Click the FTC button for the party required.

More info: A new entry is displayed in the **Phone Connection** window.

Step outcome: A call is made to the agency. You and your caller hear the ringing to the other agency. When the agency answers, they are automatically conferenced into the call.

Step 3 Ensure the "position" entry is selected in the **Phone Connection** window.

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the FTC party remain connected.

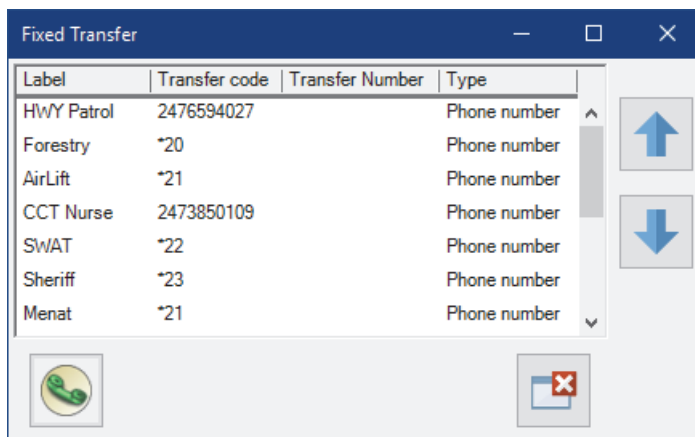
Using Fixed Xfer List

Step 1 Answer the incoming emergency call.

Step 2 Click **Fixed Xfer List**.

Step outcome: The **Fixed Transfer** window is displayed.

Example:



Step 3 From **Fixed Transfer**, select the agency to call.

Step 4 Click **Call**.

More info: A new entry is not displayed in the **Phone Connection** window. A call is made to the agency. You and your caller hear the ringing to the other agency. When the agency answers, they are automatically conferenced into the call.

Step 5 Click **Close**.

Transfer call

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the FTC party remain connected.

5.4.3 Manual multiple-party conference

Most 9-1-1 phone lines support 3-way conferencing; however, you should verify with your administrator if the ability is available in your PSAP.

Conference via smart pad

Step 1 Answer the incoming emergency call.

Step 2 Click **Smart Pad** or **Manual Xfer** if the window is not included in your layout.

Step outcome: The **Smart Pad** window is displayed in DTMF mode.

Example:

Smart Pad			
[Text Input Field]			[Navigation Buttons]
FLSH	1 ABC	2 DEF	
QXfer	4 GHI	5 JKL	6 MNO
CONF	7 PQRS	8 TUV	9 WXYZ
Call	*	0	#
SEND			EXIT

Step 3 From **Smart Pad**, click **FLSH**.

More info: Wait until you hear a dial tone in your headset.

Step 4 Dial the number of the third party required.

Step 5 If you make a mistake dialing the phone number or receive a busy signal:

- a** Click the **FLSH** button to end the attempted conference and start again from Step 3.

Step outcome: A call is made to the third party. You and your caller hear the ringing to the other party. When the party answers, they are automatically conferenced into the call.

Transfer call

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the third party remain connected.

Conference using smart pad with SIP REFER

- Step 1** Answer the incoming emergency call.
- Step 2** Click **Smart Pad** or **Manual Xfer** if the window is not included in your layout.
- Step 3** From **Smart Pad**, click **FLSH**.
More info: Wait until you hear a dial tone in your headset.
- Step 4** Dial the number of the third party required.
- Step 5** If you make a mistake dialing the phone number:
a Select the third party entry in the **Phone Connection** window, click **Release**, and start again from Step 3.

Step outcome: A call is made to the third party. You and your caller hear the ringing to the other party. When the party answers, they are automatically conferenced into the call.

Transfer call

- Step 6** Ensure the "position" entry is selected in the **Phone Connection** window.
- Click **Release**.
 - Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the third party remain connected.

Conference using DTMF pad

Step 1 Answering the incoming emergency call.

Step 2 Click **Manual Xfer**.

Step outcome: The **DTMF Pad** window is displayed.

Example:



Step 3 From **DTMF Pad**, click **Hook Flash**.

More info: Wait until you hear a dial tone in your headset.

Step 4 Dial the number of the third party required.

Step 5 If you make a mistake dialing the phone number:

- a** Click **Hook Flash** to end the attempted conference and start again from Step 3.

Step outcome: A call is made to the third party. You and your caller hear the ringing to the other party. When the party answers, they are automatically conferenced into the call.

Transfer call

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the third party remain connected.

Conference using DTMF pad with SIP REFER

- Step 1** Answer the incoming emergency call.
- Step 2** Click **Manual Xfer**.
- Step 3** From **DTMF Pad**, click **Hook Flash**.
More info: Wait until you hear a dial tone in your headset.
- Step 4** Dial the number of the third party required.
- Step 5** If you make a mistake dialing the phone number:
a Select the third party entry in the **Phone Connection** window, click **Release**, and start again from Step 3.
- Step outcome:* A call is made to the third party. You and your caller hear the ringing to the other party. When the party answers, they are automatically conferenced into the call.

Transfer call

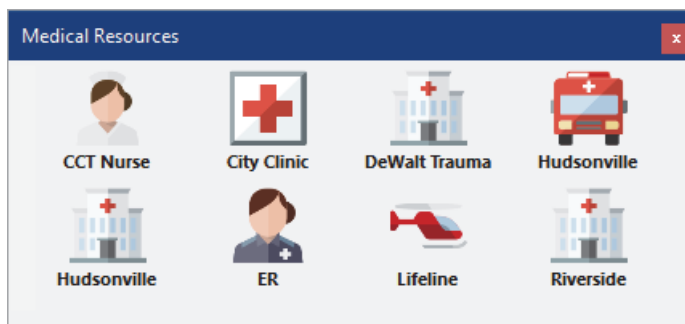
- Step 6** Ensure the "position" entry is selected in the **Phone Connection** window.
- Click **Release**.
 - Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the STA party remain connected.

5.4.4 Direct access

Direct access buttons are single-click commands to phone numbers existing in your contacts list. They may be placed directly in your layout or may be placed in windows as shown in Figure 5.6.

Figure 5.6 Direct access toolbars in custom “Medical Resources” window



Direct access toolbars can be customized to hide the image or label, and display custom fonts

5.4.4.1 Alternate phone number

An alternate phone number may be configured for the direct access party. The alternate phone number is dialed automatically if a failure occurs when the primary phone number is dialed or if a busy notification is received from the primary phone number. If the alternate phone number is dialed, a new entry for the alternate number is added to the call in the **Phone Connection** window.

Info

Dialing alternate phone numbers when a busy notification is received is available for SIP, SS7, and ISDN circuits only.

5.4.4.2 Conference mode

As explained in Chapter 3, the behavior of each DA button can be customized by your administrator. The procedures describing the three behaviors is explained below; consult your administrator to determine which has been implemented in your PSAP.

Conference using direct access buttons

Step 1 Answer the incoming emergency call.

Step 2 Click the direct access button for the party required.

Step outcome: A call is made to the direct access party. You and your caller hear the ringing. When the party answers, they are automatically conferenced into the call.

Transfer call

More info: If you would prefer to transfer the call rather than create a conference, follow the procedure above. You can release yourself from the call either while it is ringing or after the contact answers.

Step 3 Ensure the "position" entry is selected in the **Phone Connection** window.

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the direct access contact remain connected.

Info

If you placed your original caller on exclusive hold, the direct access party is not automatically conferenced into your call. Follow the [Two-step direct access conference](#) procedure below.

5.4.4.3 Alternative behavior

Typically the system is configured to create a conference with the direct access contact you click. However, your administrator may have chosen to not allow an automatic conference. If your system is not configured to create a conference, follow the two-step conference procedure.

Two-step direct access conference

- Step 1** Answer the incoming emergency call.
- Step 2** Click the direct access button for the party required.
Step outcome: Your current emergency call is placed on hold and a new call is made to the direct access contact.
- Step 3** Wait for the direct access contact to answer the call.
- Step 4** Click **Join**.
Step outcome: The direct access contact call and your emergency call are conferenced into a single call.

Transfer call

More info: If you would prefer to transfer the call rather than create a conference, follow the procedure above until Step 3.

- Step 5** Ensure the “position” entry is selected in the **Phone Connection** window.
- Click **Release**.
 - Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the direct access contact remain connected.

5.4.4.4 Quick transfer mode

The Guardian system provides an option to transfer a call to another party without remaining on the original call yet ensuring the third party answers the call. The Quick Transfer feature is available with direct access buttons, as well as contacts and manually dialing as explained later.

Quick transfer via direct access buttons

- Step 1** Answer the incoming emergency call.
- Step 2** Click the direct access button for the party you require.

Result of this task: You and your original caller hear the ring to the third party. When the party answers, you are automatically released from the call.

5.4.5 Contacts

Contacts is a combined list of public and personal phone numbers presented as an electronic phone book.

5.4.5.1 Alternate phone number

An alternate phone number may be configured for the direct access party. The alternate phone number is dialed automatically if a failure occurs when the primary phone number is dialed or if a busy signal is received from the primary phone number. If the alternate phone number is dialed, a new entry for the alternate number is added to the call in the **Phone Connection** window.

Info Dialing alternate phone numbers when a busy notification is received is available for SIP, SS7, and ISDN circuits only.

Conference via contacts

- Step 1** Answer the incoming emergency call.
- Step 2** Click **Contacts**.
- Step 3** Find the entry for the required party and select it.
- Step 4** Click **Conference**.

Step outcome: A call is made to the third party. You and your caller hear the ringing. When the party answers, they are automatically conferenced into the call.

Transfer call

- Step 5** Ensure the "position" entry is selected in the **Phone Connection** window.
 - Click **Release**.
 - Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the direct access party remain connected.

Quick transfer via contacts

- Step 1** Answer the incoming emergency call.
- Step 2** Click **Contacts**.
- Step 3** Select the entry for the required party.
- Step 4** Click **Quick transfer**.

Result of this task: A call is made to the third party. You and your caller hear the ringing. When the party answers, you are automatically released from the call.

5.4.6 Manually dialing

If you want to conference a party that is not in your contacts, you can manually dial their phone number.

5.4.6.1 Smart pad

To facilitate the transition for existing Guardian PSAPs, the buttons from the previous version of Guardian still function although they launch the **Smart Pad** window in the appropriate mode. For example, if you have the **DTMF Pad** button in your layout but the smart pad behavior has been enabled, you can continue to use the **DTMF Pad** button - it launches the **Smart Pad** window in DTMF mode.

Consult with your administrator to determine if smart pad behavior has been enabled for your role.

Manual conference via smart pad

- Step 1** Answer the incoming emergency call.

Without conference button in layout

- a** Click **Smart Pad** if the window is not included in your layout.
- b** From **Smart Pad**, click **CONF**.

Step outcome: **Smart Pad** is placed in conference mode.

With conference button in layout

- c Click **Conference**.

Step outcome: **Smart Pad** is placed in conference mode.

Step 2 Enter the phone number to call.

Step 3 Click **SEND**.

Step outcome: A call is made to the third party. You and your caller hear the ringing. When the party answers, they are automatically conferenced into the call.

Transfer call

Step 4 Ensure the "position" entry is selected in the **Phone Connection** window.

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the third party remain connected.

Manual conference via dial pad

Step 1 Answer the incoming emergency call.

Step 2 Click **Place Call**.

Step outcome: The **Dial Pad** window is displayed.

Step 3 From the dial pad, enter the phone number to call.

Step 4 Click **Conference**.

Step outcome: A call is made to the third party. You and your caller hear the ringing. When the party answers, they are automatically conferenced into the call.

Transfer call

Step 5 Ensure the "position" entry is selected in the **Phone Connection** window.

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the third party remain connected.

Manual quick transfer via smart pad

Step 1 Answer the incoming emergency call.

Without QXfer button in layout

- a** Click **Smart Pad** if the window is not included in your layout.
- b** From **Smart Pad**, click **QXfer**.

Step outcome: **Smart Pad** is placed in quick transfer mode.

With QXfer button in layout

- c** Click **QXfer**.

Step outcome: **Smart Pad** is placed in quick transfer mode.

Step 2 Enter the phone number to call.

Step 3 Click **SEND**.

Result of this task: A call is made to the third party. You and your caller hear the ringing. When the party answers, you are automatically released from the call.

Manual quick transfer via dial pad

Step 1 Answer the incoming emergency call.

Step 2 Click **Place Call**.

Step outcome: The **Dial Pad** window is displayed.

Step 3 Enter the phone number to call.

Step 4 Click **Quick transfer**.

Result of this task: A call is made to the third party. You and your caller hear the ringing. When the party answers, you are automatically released from the call.

5.4.7 Automated answering system selections

The third party you conference to your emergency call may have an automated answering system requiring input from yourself such as a password or a menu selection to specify you are calling about an emergency.

Entering data via smart pad

Step 1 When you hear the automated message asking for input, click whichever of these three buttons are included in your layout: **Smart Pad**, **Manual Xfer**, or **DTMF Pad**.

Step outcome: The **Smart Pad** window is displayed in DTMF mode.

Step 2 From **Smart Pad**, click the numbers required.

Step outcome: The numbers entered are sent to the receiving party immediately.

Step 3 Click **Exit** when you do not need to enter any further data.

Result of this task: The **Smart Pad** window is closed if it is floating mode.

Entering data via DTMF pad

Step 1 When you hear the automated message asking for your input, click **Manual Xfer**.

Step outcome: The **DTMF Pad** window is displayed.

Step 2 From **DTMF Pad**, click the numbers required.

Step outcome: The numbers entered are sent to the receiving party immediately.

Step 3 Click **Close** when you do not need to enter any further data.

5.4.8 Conference another telecommunicator

You may need to conference another telecommunicator from your PSAP to your active call. To simplify the procedure, your administrator may have configured position status buttons in your role layout. These buttons are similar to direct access buttons with the additional ability to notify you of the status of the other positions in your PSAP via their color. *Position status* on page 43 explains the various states that are reflected via the position status buttons.

Conference telecommunicator with status buttons

Step 1 Answer the incoming emergency call.

Step 2 Click the button representing the telecommunicator's position you want to join your call.

Step outcome: When the second telecommunicator answers the call, they are automatically conferenced to your call.

Transfer call

Step 3 Ensure the "position" entry is selected in the **Phone Connection** window.

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the second telecommunicator remain connected.

Conference telecommunicator using Smart Pad

Step 1 Answer the incoming emergency call.

Without conference button in layout

Step 2 Click **Smart Pad** if the window is not included in your layout.

Step 3 From **Smart Pad**, click **CONF**.

Step outcome: The **Smart Pad** is placed in conference mode.

With conference button in layout

Step 4 Click **Conference**.

Step outcome: The **Smart Pad** is displayed in conference mode.

Step 5 Type the phone number of the other telecommunicator's position.

Step 6 Click **SEND**.

Step 7 Click **EXIT**.

Step outcome: When the second telecommunicator answers the call, they are automatically conferenced to your call.

Transfer call

- Step 8** Ensure the “position” entry is selected in the **Phone Connection** window.
- Click **Release**.
 - Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the second telecommunicator remain connected.

Conference telecommunicator using Dial Pad

- Step 1** Answer the incoming emergency call.
- Step 2** Click **Place Call**.
- Step 3** In **Dial Pad**, type the phone number of the other telecommunicator’s position.
- Step 4** Click **Conference**.
- Step 5** Click **Close**.

Step outcome: When the second telecommunicator answers the call, they are automatically conferenced to your call.

Transfer call

- Step 6** Ensure the “position” entry is selected in the **Phone Connection** window.
- Click **Release**.
 - Click a disconnect reason button if available.

Result of this task: You are released from the call while your caller and the second telecommunicator remain connected.

5.4.9 Joining calls

If you have two or more calls answered at your position you can join calls together into a conference call. The calls to join do not need to be adjacent in the **Phone Connection** window; whichever call on hold you select is conferenced when you click the **Join** button. You can join more than two calls into a single conference call if needed.

At least one call must be active in order to join calls.

Join calls

Step 1 Select the call to be conferenced into your active call from the **Phone Connection** window.

Step 2 Click **Join**.

Result of this task: The selected call on hold is conferenced into your current active call.

5.5 Releasing calls

Releasing calls on the Guardian system depends on which entry is selected in the **Phone Connection** window before you click **Release**. The following procedures explain how to release the required party from an emergency call.

Release the entire call

Step In the **Phone Connection** window, select the top line of the call to be released.

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: All parties on the call are released and the call is cleared from your **Phone Connection** window.

Release yourself

Step In the **Phone Connection** window, ensure the "position" entry is selected.

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: You are released from the call and the call is cleared from your **Phone Connection** window but the other parties remain on the call.

Release specific party

Step 1 In the **Phone Connection** window, select the entry for the party to be released.

Step 2 Click **Release**.

Result of this task: You remain on the call and only the selected party is released.

Release party from 3-way call

More info: If you used DTMF mode to conference a third party to your emergency call, you can release the third party by using **Hook Flash**.

Step 1 Select the incoming emergency call entry.

Step 2 Click **Hook Flash**.

Result of this task: You remain connected with the original incoming caller.

Release yourself from 3-way call

More info: If you used DTMF mode to conference a third party to your emergency call, you can release yourself from the call by clicking **Release**.

- Click **Release**.
- Click a disconnect reason button if available.

Result of this task: The conference parties on the emergency call remain connected.

5.6 Working with the ALI

Automatic Location Information (ALI) consists of read-only records that provide you with the necessary details about a caller's phone service and location. A record is usually retrieved from an ALI database, formatted according to your current role, and displayed in the **Caller Information** window either when an emergency call is answered, or immediately upon receiving an emergency call, as this option can be configured by your administrator. If there are multiple active calls, the ALI for each caller is displayed in separate tabs. No information is retrieved when calls are subsequently transferred.

The buttons in the **ALI Tools** window are used for most operations associated with the ALI displayed for the active emergency call. You can also see a continuous list of the last 20 generated ALI-related events in the **ALI Status** window, which is available from the **View** menu.

5.6.1 Re-query the ALI

Once the ALI is retrieved and displayed, it is cached in the system and available for transferring to other telecommunicators. However, you may still need to refresh (re-query) the ALI if, for example, the text was corrupted in transmission, or the location of a wireless caller changes during the progress of the call. You may also need to query the database manually if, for example, the caller who placed the emergency call is calling on behalf of someone else, or an administrator has configured incoming emergency calls to not display the ALI automatically.

Refresh the displayed ALI

Step With the ALI displayed in the selected tab of the **Caller Information** window, click **Refresh**.

Result of this task: The ALI database is re-queried and any updated information is displayed.

5.6.2 Manual ALI Retrieval

The Manual ALI Retrieval feature allows you to query the ALI database for a 10-digit phone number. Only land-line phone numbers including an area code can be queried.

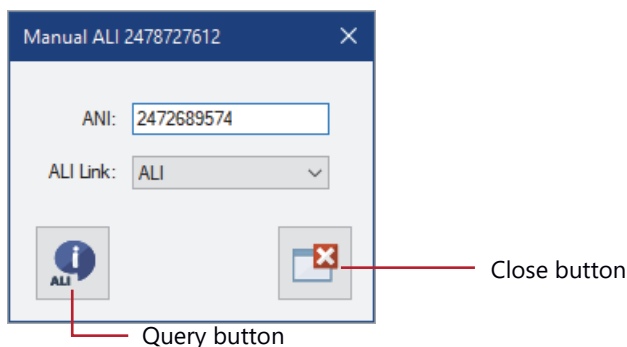
Note: Depending on your role's assigned privileges, the **Manual** button may be enabled only when you have an active emergency call.

Retrieve manual ALI

Step 1 Click **Manual**.

Step 2 In the **Manual ALI Query** dialog box, enter the phone number to query.

Example:



The image shows a dialog box titled "Manual ALI 2478727612". It contains two input fields: "ANI:" with the value "2472689574" and "ALI Link:" with a dropdown menu showing "ALI". At the bottom left is a button with an information icon and the text "ALI" (labeled "Query button" with a red line). At the bottom right is a button with a red 'X' icon (labeled "Close button" with a red line).

Step 3 Select the ALI link to query if more than one is available.

Step 4 Click **Query**.

Step outcome: When you execute a query with an emergency call, the retrieved information replaces the current data in the **Caller Information** window (the tab for the call lists two phone numbers: the original ANI and the manually revised ANI).

Example:

2478727612/2472689574	
Caller	
KAREN ROGERS RESO	(247) 268-9574
Location	
320 JAYBIRD LN WHITE COUNTY	SEARCY AR 72143 -91.8601479529 35.2453026377
Other Information	
41 WHITE COUNTY SHERIFF 2476591204 STAR EMS 2476522424	2472689574 ENTER HILL FIRE 2476542311 NORT

Tip: To retrieve the original ALI data, enter the original ANI in the **Manual ALI Query** window and click **Query**. Be aware, that this action results in a new query with potentially updated information.

Step 5 Click **Close** in the **Manual ALI Query** dialog box.

More info: If you execute the query without an active emergency call, click **Close** to clear all the information.

5.6.3 Making ALI corrections

If, in the course of an emergency call, you determine that some of the ALI is incorrect or out of date, you can make corrections directly on the screen. Corrections are not sent back to the ALI provider, so re-querying, for example, would again retrieve the previous information, but they can be printed for record management and emailed for further processing.

Correct ALI data

Step 1 Click **Correction** to unlock the ALI fields.

Step 2 Type into any of the fields to make necessary corrections.

Step 3 Click **Correction** again to lock the ALI.

More info: Release 13 introduced the ability to print both the original ALI data and corrections on the same printout. However, if this feature is not enabled on your system:

- Click **Print** before Step 1 to print a copy of the original ALI and click **Print** after making the corrections to print a copy of the corrected ALI.

Info

When you send the corrected ALI to another telecommunicator or remote printer, the original ALI is sent. You must send the updated ALI information using the Call Notes feature.

Correct ALI data after call ends

With tenant call history enabled

Step 1 Click **Call History**.

Step outcome: The **Call History** window is displayed.

Step 2 Click **My calls**.

Step outcome: The list is filtered to display only the calls in which your user name is in the *Telecommunicator* field making it easier to find your call.

With telecommunicator call history enabled**Step 3** Click **T/C History**.

Step outcome: The **Telecommunicator Call History** window is displayed.

Step 4 Select the call for which you want to modify ALI data.**Step 5** Click **Show ALI info**.

Step outcome: The ALI for the call is displayed in the **Caller Information** window.

Step 6 Click **Correction**.**Step 7** Type into any of the fields to make necessary corrections.**Step 8** Click **Correction** again to lock the ALI.

Step outcome: The corrections are displayed in red in the ALI presentation.

Example:

2478727612	
Caller	
MARK ANDERS	(247) 872-7612
RESID	
Location	
1008 MORRIS AVE	
-> 1006 Morris Ave	
SEARCY AR 72143	
WHITE COUNTY	-91.7766247019 35.1397596092
Other Information	
45	2478727612
WHITE COUNTY SHERIFF 2476591204	ARNER FIRE DEPT 2478642501 NORT
STAR EMS 2476522424	
Notes	
Corrections made by mcombs	

Step 9 Click **Print**.

Step outcome: The ALI data with the corrections displayed in red along with your user account, date, and time is sent to the printer.

Step 10 Click **Show ALI info**.

Step outcome: The ALI data is removed from the **Caller Information** window.

Step 11 Click **Close** in the history window.

Step outcome: The **Call History** or **Telecommunicator Call History** window is removed.

Caution

The corrections you make are not saved to the ALI data visible from Call History or Telecommunicator Call History after release the call. Ensure you print the corrections before removing the ALI data from your screen.

5.6.3.1 Email ALI corrections

After entering any corrections into the ALI using the ALI Correction feature, you can send the corrections via email to a designated recipient.

Email ALI corrections

Step Click **Email**.

Result of this task: The ALI corrections are sent to the designated recipient as a HTML attachment.

5.6.3.2 Print ALI

With the ALI displayed in the selected tab of **Caller Information**, you can print the information locally for your reference, or send it to a remote site via a dial-up or serial connection, where it can be printed or displayed on a terminal.

Print ALI locally

Step Click **Print**.

Result of this task: The ALI is printed to your default Windows printer.

Print ALI remotely

Step 1 Click **RMT print**.

Step 2 In the **RMT Print** window, select a printer.

a For dial-up printers, enter the phone number or select a contact.

Step outcome: The list of available dial-up contacts is populated from the contacts list.

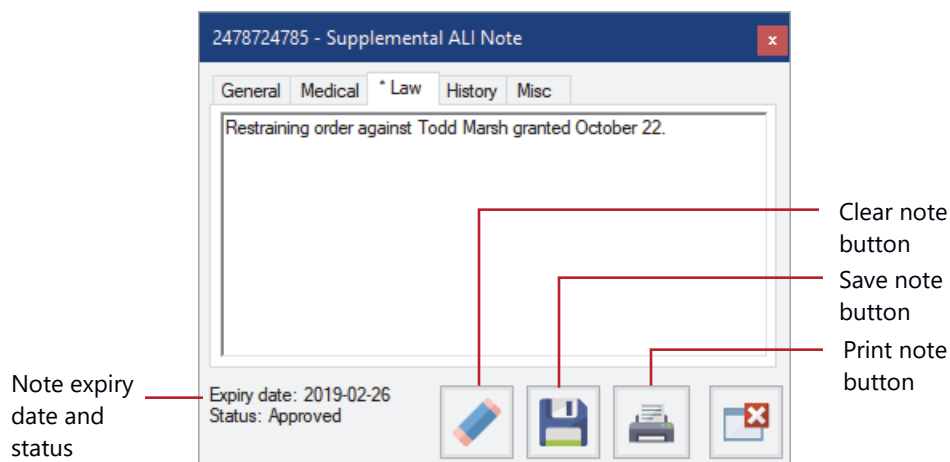
Step 3 Click **ALI print**.

Step 4 Click **Close**.

5.6.4 Supplemental ALI notes

While on an active emergency call, you can add an ALI note which is associated to the call-back phone number of the caller. Upon review and approval of the note by an administrator, this note is viewable for all calls from the same phone number. Existing ALI notes on your system are available for review whenever you answer an emergency call from the phone number.

Figure 5.7 Approved Supplemental AL note window



View existing note

- Step 1** Answer the incoming emergency call.
- More info:** If the phone number has an existing supplemental ALI note, the **Note** button in your layout changes color (default green but your administrator may customize the color).
- Step 2** Click **Note**.
- Step outcome:** The **Supplemental ALI Note** window is displayed. The existing data is displayed in the window as well as the expiry date of the note and the status of the note.

Modify existing ALI note

- Step 3** To edit the existing data or add more data to the note:
- a** Click **Clear note** to erase the existing note data.
 - b** Use the backspace key to erase part of the existing text.
 - c** Enter additional text in the window.
- Step 4** Click **Save note** when done.
- Step outcome:** The edited note is added to the management queue awaiting approval.
- Step 5** Click **Close**.

Info While the note is awaiting approval in the management queue, only the user to make the modification has access to the new data. The original data is not visible to other users until the new data is approved.

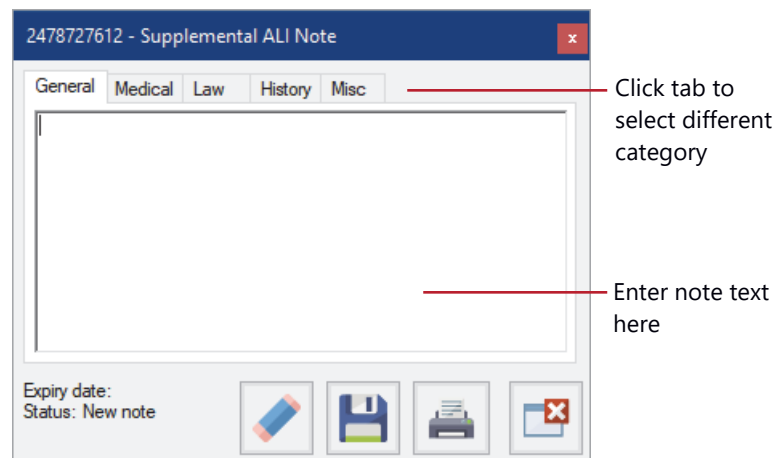
Add supplemental ALI note

Step 1 Answer incoming emergency call.

Step 2 Click **Note**.

Step outcome: The **Supplemental ALI Note** window is displayed.

Example:



Step 3 If additional tabs have been configured by your administrator:

a Click the appropriate tab to add the note.

b Enter the text of the note.

Step 4 Click **Save note**.

Step outcome: The note is added to the management queue awaiting approval.

Step 5 Click **Close**.

5.6.4.1 Adding note on conferenced call

It is possible to add or edit a supplemental ALI note while two or more Guardian telecommunicators are conferenced into the same call. However, the conferenced telecommunicators do not have access to view the new note data until the changes are approved. Since all supplemental ALI notes must be approved by a user with access to the **Supplemental ALI Manager**, the changes made by each user are placed in the supplemental ALI note management queue. New and edited ALI notes are not placed in the management queue until the emergency call is released.

Note: If more than one conferenced telecommunicator edits/creates a note for a conferenced call, only the comments added by the last telecommunicator to release the call are submitted to the management queue.

If you need to add a note to a conferenced call that is viewable by all currently conferenced users, you must use the Call Notes feature described below.

Print supplemental ALI note

- Step 1** Answer the incoming emergency call.
[More info:](#) If the phone number has an existing supplemental ALI note, the **Note** button in your layout turns green.
- Step 2** Click **Note**.
[Step outcome:](#) The **Supplemental ALI Note** window is displayed.
- Step 3** Click **Print note**.

Result of this task: The note is printed to the default printer configured for your position.

5.6.5 Call notes

You can take notes while on a call to record an event or status of significance. Notes are associated with a single call, so all telecommunicators who have the call active on their position, or any telecommunicator to whom a call is transferred, can add a note to the incident. Notes can be added collaboratively back and forth (like chat) by telecommunicators conferenced into the call, and can continue even after the emergency caller has been released, if the call is still active. Guardian stores the notes temporarily with the call history, if set by your administrator. Call notes can be:

- viewed in **Caller Information** during an active call
- accessed from **Call History** after the call is terminated
- automatically sent with the call detail record
- sent to your CAD system

You can edit an existing note after the call has ended by retrieving the call from one of the call history windows. You can only edit messages that you added to the call when more than one telecommunicator added notes to the call.

Your assigned role must include the privilege in order to enter call notes.

Add note to active call

Step 1 Click **Call Notes**.

Step outcome: The **Call Notes** window is displayed.

Example:

Call Notes

Add note:

2019-02-07 17:05:04

Remaining characters: 1000

Elapsed time: 00:00:31

Note list:

Telecommunicator	Time	Note text
------------------	------	-----------

Add note button

Close button

Step 2 Type your message in the **Call Notes** window.

Step 3 Click **Add note**.

Step outcome: The **Call Notes** window closes automatically and the message is added to the *Notes* section of the **Caller Information** window with the date and time stamp.

Example:

2478727612	
Caller	
MARK ANDERS	(247) 872-7612
RESO	
Location	
1008 MORRIS AVE	SEARCY AR 72143
WHITE COUNTY	-91.7766247019 35.1397596092
Other Information	
45	2478727612
WHITE COUNTY SHERIFF 2476591204	ARNER FIRE DEPT 2478642501 NORT
STAR EMS 2476522424	
Notes	
+++ mcombs:Start 2019-02-07 17:05:04 +++ OK, I'll dispatch them to marker 42 in case they've moved on. --- mcombs:End 2019-02-07 17:09:04 ---	

Call note text

Step 4 You can repeat steps 1 through 3 to add more notes.

Add notes to call between telecommunicators

- Step 1** Answer the incoming emergency call.
- Step 2** Conference in a second telecommunicator.
- Step 3** Click **Call Notes** to add a call note.
- Step 4** Type your message in the **Call Notes** window.

Step 5 Click **Add note**.

Step outcome: The **Call Notes** window closes automatically and the message is added to the *Notes* section of **Caller Information** on both telecommunicator's positions with the date and time stamps.

Example:

Telecommunicator	Time	Note text
vroberts	2018-11-07 10:51:43	Did you understand the location he stated?
mcombs	2018-11-07 10:52:13	Sounding like they were just south of marker 43.
vroberts	2018-11-07 10:52:55	Ok, I'll dispatch them to marker 42 in case they've moved on.
vroberts	2018-11-07 10:53:08	cw

Existing notes are listed at the bottom of the window and identified by the author

More info: The second telecommunicator can repeat Steps 3 through 5 from their position to add a call note.

Step 6 Repeat Steps 3 through 5 to add more notes to the call.

Add note after call ends

Step 1 Launch either the **Call History** or **Telecommunicator Call History** window.

Step 2 Select the call in the window.

Step 3 Click **Call Notes** in the history window.

Step outcome: The **Call Notes** window is displayed.

Step 4 Enter the text of the note in the upper portion of the window.

Step 5 Click **Add note**.

Result of this task: The **Call Notes** window closes and the note is added to the call. The note, your login name, and the date you added the note is visible when the call is retrieved

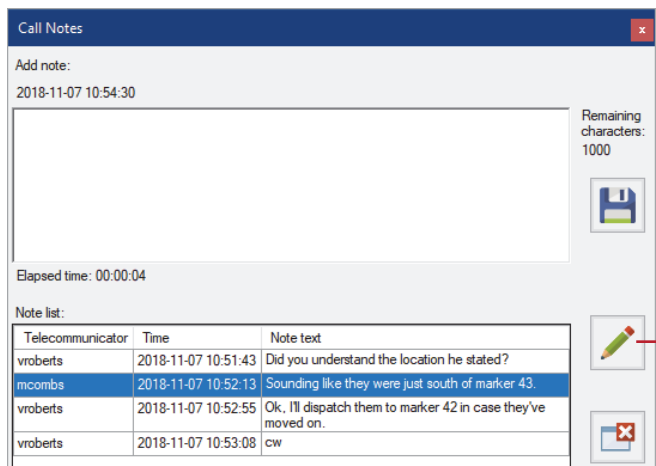
from the call history window. If your system is configured to send the call note to MIS, the call note is saved to the MIS database.

Edit call note during call

Step 1 Click **Call Notes** to open the **Call Notes** window.

Step 2 Select your note from the bottom of the window.

Example:



The screenshot shows the 'Call Notes' window. At the top, there's a header bar with the title 'Call Notes'. Below it, the 'Add note:' section displays the timestamp '2018-11-07 10:54:30' and a large text area for entering notes. To the right of the text area, it says 'Remaining characters: 1000'. Below the text area, the 'Elapsed time: 00:00:04' is shown. At the bottom, there is a 'Note list' table with three columns: 'Telecommunicator', 'Time', and 'Note text'. The table contains four rows of data. A red arrow points to a pencil icon in the right sidebar, labeled 'Edit button'.

Telecommunicator	Time	Note text
vroberts	2018-11-07 10:51:43	Did you understand the location he stated?
mcombs	2018-11-07 10:52:13	Sounding like they were just south of marker 43.
vroberts	2018-11-07 10:52:55	Ok, I'll dispatch them to marker 42 in case they've moved on.
vroberts	2018-11-07 10:53:08	cw

Step 3 Click **Edit Note**.

Step outcome: The existing note text is displayed in the *Add note* section at the top.

More info: You can only edit a note that you authored. The **Edit note** button remains disabled if you try to select a note authored by another user.

Step 4 Change the text of the note in the *Add note* section.

Step 5 Click **Add Note**.

Result of this task: The **Call Notes** window closes and the note is updated with the new data.

Edit note after call ended

Step 1 Launch either the **Call History** or **Telecommunicator Call History** window.

Step 2 Select the call in the window.

Step 3 Click **Call Notes** in the history window.

Step outcome: The **Call Notes** window is displayed.

Step 4 Select the note to edit.

Step 5 Click **Edit note**.

Step outcome: The existing note is displayed in the *Add note* section at the top.

Step 6 Change the text of the note in *Add note* section.

Example:

The screenshot shows the 'Call Notes' window. At the top, it says 'Edit note:' followed by the timestamp '2018-12-17 09:52:38'. Below this is a text area containing the text 'Sounds like he said they were just south of mile 43.' To the right of the text area is a 'Remaining characters: 948' indicator. Below the text area is an 'Elapsed time: 00:11:29' label. At the bottom, there is a 'Note list' table with three columns: 'Telecommunicator', 'Time', and 'Note text'. The table contains three rows of data. A red arrow points to a 'Cancel edit button' icon (a red circle with a diagonal line) located to the right of the table.

Telecommunicator	Time	Note text
vroberts	2018-12-17 09:50:08	Did you understand the location he stated?
mcombs	2018-12-17 09:50:36	Sounding like they were just south of mile 43.
vroberts	2018-12-17 09:51:21	OK, I'll dispatch them to mile 42 in case they've moved on.

Step 7 If you change your mind, click **Cancel edit**.

Step 8 Click **Add Note**.

Result of this task: The **Call Notes** window closes and the note is updated with the new data.

5.7 Caller history

While on an active emergency call, you can view the number of previous calls to your PSAP from the current phone number of the caller.

Caller history is only available for emergency calls. To accommodate the ANI and pANI for wireless calls, the system scans the received ALI response for the callback phone number (ANI) and compares this number to the entries in the call history. For sites that utilize PIDF-LO only, the system compares the entries in the call history with the ANI of the current call. The length of time calls are retained in the call history is configurable by your administrator.

and may include all calls since the PSAP became active, calls from the last year only, calls from the last month only, etc.

The **Caller History** button only becomes enabled after the call has been answered and an ALI/PIDF-LO was received. If you place the call on hold, the **Caller History** window closes automatically and the button becomes disabled. Once the call is removed from hold, the button becomes enabled again.

View caller history

Step 1 Answer incoming emergency call.

Step outcome: If the callback phone number of the call is found in the call history database, the **Caller History** button becomes enabled in your layout.

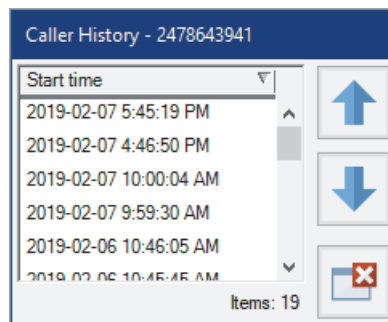
Example:



Step 2 Click **Caller History**.

Step outcome: The **Caller History** window is displayed.

Example:



More info: The total number of calls received from this phone number is displayed at the bottom of the window.

Result of this task: The **Caller History** window closes automatically when you end the call. To close the window before ending the call:

Step 3 Click **Close**.

6 Administrative calls

This chapter describes the common telecommunicator tasks such as answering, conferencing, placing, and releasing administrative calls. This chapter explains the procedures for administrative lines that do not support three-way call.

Due to the various choices in telephony equipment and telephone lines, there may be more than one method to execute the procedures documented in this chapter. For the purpose of this user guide, the most common behavior is documented in the procedures.

6.1 Answering calls

When an administrative call arrives, a ring tone may be heard in either the headset/handset or from the desktop speakers at your position depending on system configuration. A new entry is added to the **Phone Connection** window and the **NG9-1-1 Info** window flashes "Admin". No ALI data is provided with the call.

Answer administrative call

- | | |
|-------------|---|
| Step | Perform one of the following to answer the call: <ul style="list-style-type: none">• Click the flashing NG9-1-1 Info window.• Click the Answer button.• Double-click the entry in the Phone Connection window.• If configured, click the line appearance button.• Select the call in the Waiting Admin Calls window and click Answer call.• If a keyboard shortcut is configured for your role, enter the keys.• If a keypad is configured for your position, click Answer on the keypad. |
|-------------|---|

6.1.1 Automatic hold on answer

Any currently active calls are placed on hold automatically when you answer a new call; there is no need to place the existing call on hold first.

6.1.2 Automatic call answer

The system may be configured to automatically answer incoming calls at your position. If this is the case, when a call arrives at your position, the system plays a short tone in your headset just before the call is answered to notify you that you are connected with a caller. All remaining call handling functionality remains the same as for manually answered calls.

See *Automatically answered calls* on page 97 for procedure to answer a call out of sequence and to adjust the automatic greeting that may be configured with the automatic call answer feature.

6.2 Conference/transfer calls

Although you may normally answer and handle an incoming call yourself, there are many instances when you need to conference another party to your active call or you may need to transfer the call to another party.

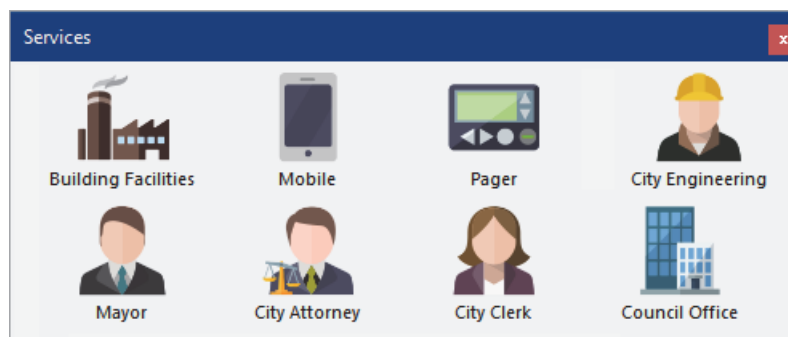
Info

The Guardian system handles all phone transactions as a conference. When you answer an administrative call, you are conferenced with the initial caller; if you need to transfer the call to another party, the Guardian system conferences the new party to the current call and it remains at your discretion whether to remain on the call or release yourself from the call - essentially transferring the call. Understanding this principle will assist you when interacting with the call entries in the Phone Connection window.

6.2.1 Direct access buttons

Direct access buttons are single-click commands to phone numbers existing in your contacts list. They may be placed directly in your layout or may be placed in a window as shown in Figure 6.1.

Figure 6.1 Direct access toolbars within a direct access group window



Direct access buttons located in a window named "Services"

6.2.1.1 Alternate phone number

An alternate phone number may be configured for the direct access party. The alternate phone number is dialed automatically if a failure occurs when the primary phone number is dialed or if a busy notification is received from the primary phone number. If the alternate phone number is dialed, a new entry for the alternate number is added to the call in the **Phone Connection** window.

Info Dialing alternate phone numbers when a busy notification is received is available for SIP, SS7, and ISDN circuits only.

6.2.1.2 Conference mode

As explained in Chapter 3, the behavior of each DA button can be customized by your administrator. The procedures describing the three behaviors is explained below; consult your administrator to determine which has been implemented in your PSAP.

Conference using direct access buttons

- Step 1** Answer the incoming administrative call.
- Step 2** Click the direct access button for the party required.

Result of this task: A call is made to the direct access party. You and your caller hear the ringing. When the party answers, they are automatically conferenced into the call.

Transfer call

- Step 3** If you would prefer to transfer the call rather than create a conference, follow the procedure above.
- Step 4** Ensure the “position” entry is selected in the **Phone Connection** window.
More info: You can choose to release yourself from the call before the direct access party answers once you hear the ring in your headset or see the entry for the direct access party in the **Phone Connection** window.
- Step 5** Click **Release**.

Result of this task: You are released from the call while your caller and the direct access party remain connected.

Info

If you have placed your original caller on exclusive hold, the direct access party is not automatically conferenced into your call. Follow the [Two-step direct access conference](#) procedure below.

6.2.1.3 Alternative behavior

Typically the system is configured to create a conference with the direct access contact you click. However, your administrator may have chosen to not allow an automatic conference. If your system is not configured to create a conference, follow the two-step conference procedure.

Two-step direct access conference

Step 1 Answer the incoming administrative call.

Step 2 Click the direct access button for the party required.

Step outcome: Your current call is placed on hold and a new call is made to the direct access contact.

Step 3 Wait for the direct access contact to answer the call.

Step 4 Click **Join**.

Step outcome: The direct access contact and your administrative call are conferenced into a single call.

Transfer call

Step 5 If you would prefer to transfer the call rather than create a conference, follow the procedure above until Step 4.

Step 6 Ensure the "position" entry is selected in the **Phone Connection** window.

Step 7 Click **Release**.

Result of this task: You are released from the call while your caller and the direct access party remain connected.

Info

If your system is configured for automatic conference when using direct access buttons, you can mimic the two-step direct access conference by ensuring that no call entry is selected in the Phone Connection window. In this scenario, your original caller is placed on hold and a new call is made to the direct access contact. When you are ready to join the two calls, click Join.

6.2.1.4 Quick transfer mode

The Guardian system provides an option to transfer a call to another party without remaining on the original call yet ensuring the third party answers the call. The quick transfer feature is available with direct access buttons, as well as contacts and manually dialing as explained later.

Quick transfer using direct access buttons

- Step 1** Answer the incoming administrative call.
- Step 2** Click the direct access button for the party you require.

Result of this task: You and your original caller hear the ring to the third party. When the party answers, you are automatically released from the call.

6.2.2 Contacts

Contacts is a combined list of public and personal phone numbers presented as an electronic phone book.

6.2.2.1 Alternate phone number

An alternate phone number may be configured for the direct access party. The alternate phone number is dialed automatically if a failure occurs when the primary phone number is dialed or if a busy signal is received from the primary phone number. If the alternate

phone number is dialed, a new entry for the alternate number is added to the call in the **Phone Connection** window.

Info Dialing alternate phone numbers when a busy notification is received is available for SIP, SS7, and ISDN circuits only.

Conference using contacts

- Step 1** Answer the incoming administrative call.
- Step 2** Click **Contacts**.
- Step 3** Find the entry for the required party and select it.
- Step 4** Click **Conference**.

Step outcome: A call is made to the third party. You and your caller hear the ringing. When the party answers, they are automatically conferenced into the call.

Transfer call

- Step 5** Ensure the "position" entry is selected in the **Phone Connection** window.
More info: You can choose to release yourself from the call before the third party answers once you hear the ring in your headset or see the entry for the third party in the **Phone Connection** window.
- Step 6** Click **Release**.

Result of this task: You are released from the call while your caller and the contact remain connected.

Quick transfer using contacts

- Step 1** Answer the incoming administrative call.
- Step 2** Click **Contacts**.
- Step 3** Select the entry for the required party.
- Step 4** Click **Quick transfer**.

Result of this task: A call is made to the third party. You and your caller hear the ringing. When the party answers, you are automatically released from the call.

6.2.3 Manual multiple-party conference

If you want to conference a party which has no direct access button and that is not in your contacts, you can manually dial their phone number.

6.2.3.1 Smart pad

To facilitate the transition for existing Guardian PSAPs, the buttons from the previous version of Guardian still function although they launch the **Smart Pad** in the appropriate mode. For example, if you have the **DTMF Pad** button in your layout but the Smart Pad behavior has been enabled, you can continue to use the **DTMF Pad** button - it launches the **Smart Pad** in DTMF mode.

Consult with your administrator to determine if smart pad behavior has been enabled for your role.

Manual conference using smart pad

Step 1 Answer the incoming administrative call.

Without conference button in layout

- a** Click **Smart Pad** if the window is not included in your layout.
- b** From **Smart Pad**, click **CONF**.

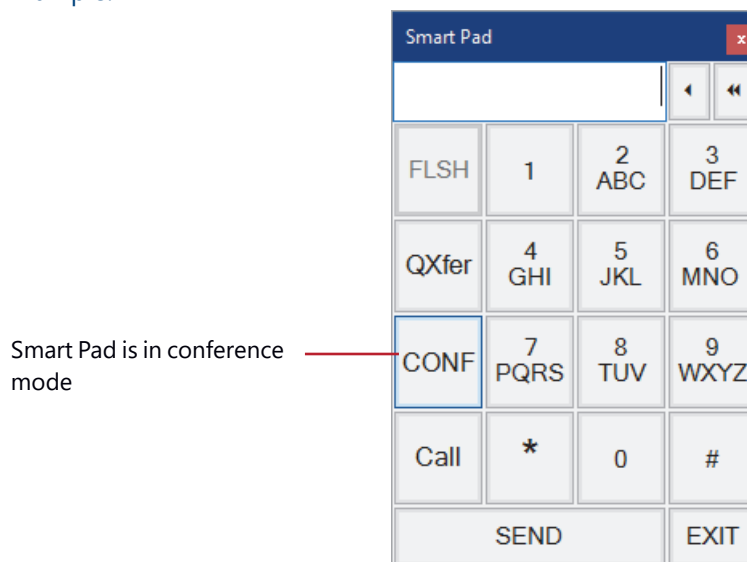
Step outcome: The **Smart Pad** is placed in conference mode.

With conference button in layout

- c Click **Conference**.

Step outcome: **Smart Pad** is displayed in conference mode.

Example:



Step 2 Enter the phone number to call.

Step 3 Click **SEND**.

Step outcome: A call is made to the third party. You and your caller hear the ringing. When the party answers, they are automatically conferenced into the call.

Transfer call

Step 4 Ensure the "position" entry is selected in the **Phone Connection** window.

Step outcome: You can choose to release yourself from the call before the third party answers once you hear the ring in your headset or see the entry for the third party in the **Phone Connection** window.

Step 5 Click **Release**.

Result of this task: You are released from the call while your caller and the third party remain connected.

Manual quick transfer using smart pad

Step 1 Answer the incoming administrative call.

Without QXfer button in layout

- a** Click **Smart Pad** if the window is not included in your layout.
- b** From **Smart Pad**, click **QXfer**.

Step outcome: **Smart Pad** is placed in quick transfer mode.

With QXfer button in layout

- c** Click **QXfer**.

Step outcome: **Smart Pad** is placed in quick transfer mode.

Step 2 Enter the phone number to call.

Step 3 Click **SEND**.

Step outcome: A call is made to the third party. You and your caller hear the ringing. When the party answers, you are automatically released from the call.

6.2.3.2 Dial pad

If **Smart Pad** is not enabled for your role, you can still make a manual conference using the dial pad.

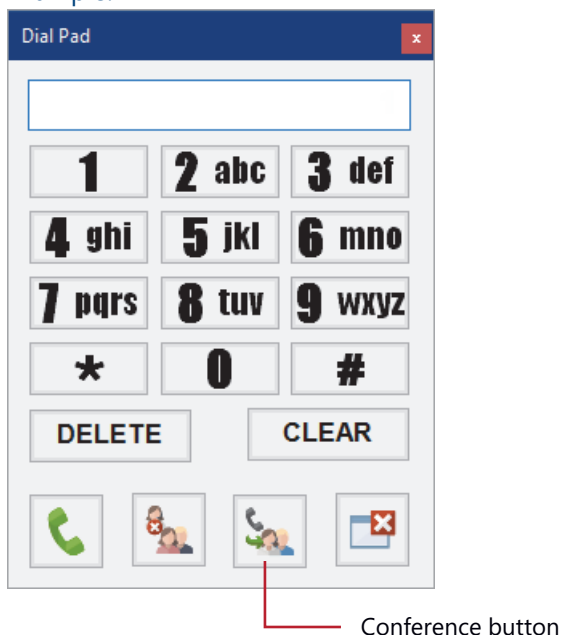
Manual conference via dial pad

Step 1 Answer the incoming administrative call.

Step 2 Click **Place Call**.

Step outcome: The **Dial Pad** window is displayed.

Example:



Step 3 From **Dial Pad**, enter the phone number to call.

Step 4 Click **Conference**.

Step outcome: A call is made to the third party. You and your caller hear the ringing. When the party answers, they are automatically conferenced into the call.

Transfer call

Step 5 Ensure the "position" entry is selected in the **Phone Connection** window.

More info: You can choose to release yourself from the call before the third party answers once you hear the ring in your headset or see the entry for the third party in the **Phone Connection** window.

Step 6 Click **Release**.

Result of this task: You are released from the call while your caller and the contact remain connected.

Manual quick transfer via dial pad

- Step 1** Answer the incoming administrative call.
- Step 2** Click **Place Call**.
Step outcome: The **Dial Pad** window is displayed.
- Step 3** Enter the phone number to call.
- Step 4** Click **Quick transfer**.

Result of this task: A call is made to the third party. You and your caller hear the ringing. When the party answers, you are automatically released from the call.

6.2.4 Automated answering system selections

The third party you conference into your administrative call may have an automated answering system requiring input from you such as a password or a menu selection.

Entering data using smart pad

- Step 1** When you hear the automated message asking for input, click **Smart Pad** or **DTMF Pad**, whichever button is included in your layout.
Step outcome: The **Smart Pad** window is displayed in DTMF mode.
- Step 2** From **Smart Pad**, enter the numbers required.
Step outcome: The numbers entered are sent to the receiving party immediately.
- Step 3** Click **EXIT** when you do not need to enter any further data.
Step outcome: **Smart Pad** is closed when in floating mode.

Entering data using DTMF pad

- Step 1** In the **Phone Connection** window, select the entry for the party requiring input.
- Step 2** When you hear the automated message asking for your input, click **DTMF Pad**.
Step outcome: The **DTMF Pad** window is displayed.

Step 3 From **DTMF Pad**, enter the numbers required.

Step outcome: The numbers entered are sent to the receiving party immediately.

Step 4 Click **Close** when you do not need to enter any further data.

Result of this task: The **DTMF Pad** window is closed.

6.2.5 Conference another telecommunicator

You may need to conference another telecommunicator from your PSAP to your active call. Your administrator may have configured position status buttons in your role layout. These buttons are similar to direct access buttons with the additional ability to notify you of the status of the other positions in your PSAP via their color. Section *Position status* on page 43 explains the various states that are reflected via the operator status buttons.

Conference telecommunicator with status buttons

Step 1 Answer the incoming administrative call.

Step 2 Click the button representing the telecommunicator's position you want to join your call.

Result of this task: When the second telecommunicator answers the call, they are automatically conferenced to your call.

Transfer to telecommunicator with status buttons

Step 1 Answer the incoming administrative call.

Step 2 Click the button representing the telecommunicator's position.

Step 3 As soon as you hear the ring or see an entry for the second telecommunicator in the **Phone Connection** window, click **Release**.

Result of this task: You are released from the call while your caller and the second telecommunicator remain connected.

Transfer with line appearance buttons

More info: The line appearance button turns yellow when the call is answered and placed on hold.

Step 1 Click the yellow line appearance button.

Step outcome: The original telecommunicator is still on the call but in a hold state.

Step 2 Select the original telecommunicator's entry in the **Phone Connection** window.

Step 3 Click **Release**.

Result of this task: You remain on the call with the caller while the original telecommunicator is released from the call.

Conference telecommunicator using Smart Pad

Step 1 Answer the incoming administrative call.

Without conference button in layout

Step 2 Click **Smart Pad** if the window is not included in your layout.

Step 3 From **Smart Pad**, click **CONF**.

Step outcome: **Smart Pad** is placed in conference mode.

With conference button in layout

Step 4 Click **Conference**.

Step outcome: **Smart Pad** is displayed in conference mode.

Step 5 Type the phone number of the other telecommunicator's position.

Step 6 Click **SEND**.

Step 7 Click **EXIT**.

Step outcome: When the second telecommunicator answers the call, they are automatically conferenced to your call.

Transfer call

Step 8 Ensure the "position" entry is selected in the **Phone Connection** window.

Step 9 Click **Release**.

Result of this task: You are released from the call while your caller and the second telecommunicator remain connected.

Conference telecommunicator using Dial Pad

- Step 1** Answer the incoming administrative call.
 - Step 2** Click **Place Call**.
 - Step 3** In **Dial Pad**, type the phone number of the other telecommunicator's position.
 - Step 4** Click **Conference**.
 - Step 5** Click **Close**.
- Step outcome:** When the second telecommunicator answers the call, they are automatically conferenced to your call.

Transfer call

- Step 6** Ensure the "position" entry is selected in the **Phone Connection** window.
- Step 7** Click **Release**.

Result of this task: You are released from the call while your caller and the second telecommunicator remain connected.

6.2.6 Joining two calls

If you have two or more calls answered at your position you can join calls together into a conference call. The calls to join do not need to be adjacent in the **Phone Connection** window; whichever call on hold you select is conferenced when you click **Join**. You can join more than two calls into a single conference call if needed.

At least one call must be active in order to join calls.

Join calls

- Step 1** Select the call to be conferenced into your active call from the **Phone Connection** window.
- Step 2** Click **Join**.

Result of this task: The selected call on hold is conferenced into your active call.

6.3 Releasing calls

Releasing calls on the Guardian system depends on which entry is selected in the **Phone Connection** window before you click **Release**. The following procedures explain how to release the required party from an administrative call.

Release the entire call

Step 1 In the **Phone Connection** window, select the top line of the call to be released.

Step 1 Click **Release**.

Result of this task: All parties on the call are released and the call is cleared from your **Phone Connection** window.

Release yourself

Step 1 In the **Phone Connection** window, ensure the "position" entry is selected.

Step 2 Click **Release**.

Result of this task: You are released from the call and the call is cleared from your **Phone Connection** window but the other parties remain on the call.

Release specific party

Step 1 In the **Phone Connection** window, select the entry for the party to be released.

Step 2 Click **Release**.

Result of this task: You remain on the call and only the selected party is released.

6.4 Placing call

You can make an outgoing call using one of the three procedures documented in this section. All new outgoing calls use your PSAP's administrative phone lines.

Place call using direct access buttons

Step To call a direct access contact, single-click the direct access button.

Result of this task: A call is made to the direct access contact and the call is visible in your **Phone Connection** window.

Place call using contacts

Step 1 Click **Contacts**.

Step outcome: The **Contacts** window is displayed.

Step 2 Select the entry to call.

Step 3 Click **Call**.

Result of this task: A call is made to the entry selected and is visible in your **Phone Connection** window.

Place call using Smart Pad

Step 1 Click **Place Call** or **Smart Pad** (whichever is available in your layout).

Step outcome: The **Smart Pad** window is displayed in call mode.

Step 2 In **Smart Pad**, enter the phone number to call.

Step 3 Click **SEND**.

Result of this task: A call is made to the phone number entered and is visible in your **Phone Connection** window.

Place call using Dial Pad

- Step 1** Click **Place Call**.
Step outcome: The **Dial Pad** window is displayed.
- Step 2** In **Dial Pad**, enter the phone number to call.
- Step 3** Click **Call**.

Result of this task: A call is made to the phone number entered and is visible in your **Phone Connection** window.

7 Enhanced admin calls

This chapter describes the common telecommunicator tasks such as answering, conferencing, placing, and releasing enhanced administrative calls. Enhanced administrative calls originate on administrative lines that support three-way calling such as Centrex lines and calls from a PBX.

Due to the various choices in telephony equipment and telephone lines, there may be more than one method to execute the procedures documented in this chapter. For the purpose of this user guide, the most common behavior is documented in the procedures.

7.1 Answering calls

When a feature-enabled administrative call arrives, a ring tone may be heard in either the headset/handset or from the desktop speakers at your position depending on system configuration. A new entry is added to the **Phone Connection** window and the **NG9-1-1 Info** window flashes "Admin". No ALI data is provided with the call.

Answer administrative call

- | | |
|-------------|---|
| Step | Perform one of the following to answer the call: <ul style="list-style-type: none">• Click the flashing NG9-1-1 Info window.• Click the Answer button.• Double-click the entry in the Phone Connection window.• If configured, click the line appearance button.• Select the call in the Waiting Admin Calls window and click Answer call.• If a keyboard shortcut is configured for your role, enter the keys.• If a keypad is configured for your position, click Answer on the keypad. |
|-------------|---|

7.1.1 Automatic hold on answer

Any currently active calls are placed on hold automatically when you answer a new call; there is no need to place the existing call on hold first.

7.1.2 Automatic call answer

The system may be configured to automatically answer incoming calls at your position. If this is the case, when a call arrives at your position, the system plays a short tone in your headset just before the call is answered to notify you that you are connected with a caller. All remaining call handling functionality remains the same as for manually answered calls.

See *Automatically answered calls* on page 97. for procedure to answer a call out of sequence and to adjust the automatic greeting that may be configured with the automatic call answer feature.

7.2 Conference/transfer calls

Although you may normally answer and handle an incoming call yourself, there are many instances when you need to conference another party to your active call or you may need to transfer the call to another party.

The behavior of incoming enhanced administrative calls is dependent on the phone service provider (far end) of the incoming lines. For example, the far end may place the original caller on hold when you conference in a third party or the far end may create a conference automatically when the third party answers the call. The procedures documented in this chapter describe the most common behavior. However, all combinations of behavior cannot be documented and if in doubt, verify the behavior of your administrative lines with your administrator.

For the sake of simplicity, the term central office is used to identify the far end in the procedures documented in this chapter.

Info

The Guardian system handles all phone transactions as a conference. When you answer an administrative call, you are conferenced with the initial caller; if you need to transfer the call to another party, the Guardian system conferences the new party to the current call and it remains at your discretion whether to remain on the call or release yourself from the call - essentially transferring the call. Understanding this principle will assist you when interacting with the call entries in the Phone Connection window.

7.2.1 SIP REFER

Typically, when you conference a third party into your active administrative call when your phone lines are equipped with multi-party calling such as with a PBX or Centrex lines, you do not see an entry for the party in your **Phone Connection** window. However, when the phone company uses a method known as SIP REFER, an entry in the **Phone Connection** window is displayed as shown in Figure 7.1.

Figure 7.1 SIP REFER conferenced contact

Status	Name	Number	DNIS
active	Jason Watts		
connected	position	2475553001	
connected	Jason Watts	2478643350	2478236243
connected	Remote Xfer	2479476157	

The SIP REFER participant has their Name set to "Remote Xfer" and the icon is different from a regular connected participant.

The entry for a SIP REFER participant has a different icon, the *Name* column is set to "Remote Xfer", and the *Number* column displays the phone number you dialed. Another difference: the SIP REFER method allows you to disconnect the third party by selecting their entry and clicking the **Release** button rather than using **Hook Flash** as with the standard method for multi-party calling.

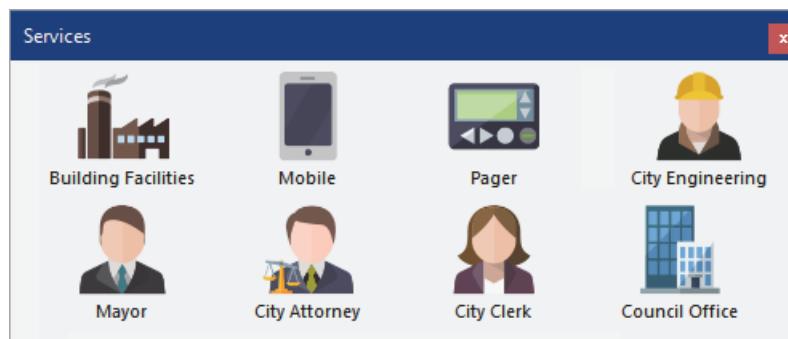
The additional benefit with the SIP REFER method compared to the hook flash method is allowing more than one party to be conferenced onto the call. Although the Guardian hook flash method can support multi-party conferences, typical industry practice allows only one party to be conferenced to the original incoming call with the hook flash method. This limitation is not present with the SIP REFER method.

If you do conference more than one party to the incoming call, you can select which party to release in the **Phone Connection** window.

7.2.2 Direct access buttons

Direct access buttons are single-click commands to phone numbers existing in your contacts list. They may be placed directly in your layout or may be placed in a window as shown in Figure 7.2.

Figure 7.2 Direct access toolbars within a direct access group window



Direct access buttons located in a window named "Services"

7.2.2.1 Conference mode

As explained in Chapter 3, the behavior of each DA button can be customized by your administrator. The procedures describing the three behaviors is explained below; consult your administrator to determine which has been implemented in your PSAP.

Conference using direct access buttons

- Step 1** Answer the incoming administrative call.
- Step 2** Click the direct access button for the party required.
Step outcome: A call is made to the direct access party - you do not see a new entry in the **Phone Connection** window.
- Step 3** Usually, the original caller is placed on hold by the CO while the call is made to the direct access contact. If the original caller is not removed from hold automatically when the direct access contact answers the call:
- a** Click **Hook Flash**.
Step outcome: All parties are joined into the conference call.
- If CO does not place the original caller on hold**
- b** Skip the step above.
Step outcome: All parties are joined into the conference call.

Transfer call

Step 4 If you would prefer to transfer the call rather than create a conference, follow the procedure above, then:

- a** Click **Release**.

Result of this task: You are released from the call while your caller and the direct access party remain connected.

Info

If you placed your original caller on exclusive hold, the direct access party is not automatically conferenced into your call. Follow the [Two-step direct access conference](#) procedure below.

Conference using direct access buttons with SIP REFER

Step 1 Answer the incoming administrative call.

Step 2 Click the direct access button for the party required.

Step outcome: A call is made to the direct access party and a new entry is added to **Phone Connection** window.

Step 3 Typically, the original caller is placed on hold by the CO while the call is made to the direct access contact but is removed from hold automatically when the direct access contact answers the call.

Step outcome: All parties are joined into the conference call.

If CO does not remove the original caller on hold:

- a** Click **Hook Flash**.

Step outcome: All parties are joined into the conference call.

Transfer call

Step 4 If you would prefer to transfer the call rather than create a conference, follow the procedure above, then:

- a** Ensure the "position" entry is selected in the **Phone Connection** window.
- b** Click **Release**.

Result of this task: You are released from the call while your caller and the direct access party remain connected.

7.2.2.2 Alternative behavior

Typically the system is configured to create a conference with the direct access contact you click. However, your administrator may have chosen to not allow an automatic conference. If your system is not configured to create a conference, follow the two-step conference procedure.

Two-step direct access conference

- Step 1** Answer the incoming administrative call.
- Step 2** Click the direct access button for the party required.
Step outcome: Your current call is placed on hold and a new call is made to the direct access contact.
- Step 3** Wait for the direct access contact to answer the call.
- Step 4** Click **Join**.
Step outcome: The direct access contact and your administrative call are conferenced into a single call.

Transfer call

- Step 5** If you prefer to transfer the call rather than create a conference, follow the procedure above, then:
- a** Ensure the “position” entry is selected in the **Phone Connection** window.
 - b** Click **Release**.

Result of this task: You are released from the call while your caller and the direct access party remain connected.

Info

If your system is configured for automatic conference when using direct access buttons, you can mimic the two-step direct access conference by ensuring that no call entry is selected in the Phone Connection window. In this scenario, your original caller is placed on hold and a new call is made to the direct access contact. When you are ready to join the two calls, click Join.

7.2.2.3 Quick transfer mode

The Guardian system provides an option to transfer a call to another party without remaining on the original call yet ensuring the third party answers the call. The quick transfer feature is available with direct access buttons, as well as contacts and manually dialing as explained later.

Quick transfer using direct access buttons

- Step 1** Answer the incoming call administrative call.
- Step 2** Click the direct access button for the party you require.

Result of this task: You and your original caller hear the ring to the third party. When the party answers, you are automatically released from the call.

7.2.3 Contacts

Contacts is a combined list of public and personal phone numbers presented as an electronic phone book.

Conference using contacts

- Step 1** Answer the incoming administrative call.
- Step 2** Click **Contacts**.
- Step 3** Find the entry for the required party and select it.
- Step 4** Click **Conference**.
Step outcome: A call is made to the third party - you do not see a new entry in the **Phone Connection** window.
- Step 5** Usually, the original caller is placed on hold by the CO while the call is made to the contact. If the original caller is not removed from hold automatically when the contact answers the call:
 - a** Click **Hook Flash**.
Step outcome: All parties are joined into the conference call.

If CO does not place the original caller on hold:

- b** Skip the step above.

Step outcome: All parties are joined into the conference call.

Transfer call

Step 6 If you prefer to transfer the call rather than create a conference, follow the procedure above, then:

- a** Click **Release**.

Result of this task: You are released from the call while your caller and the direct access party remain connected.

Conference using contacts with SIP REFER

Step 1 Answer the incoming administrative call.

Step 2 Click **Contacts**.

Step 3 Find the entry for the required party and select it.

Step 4 Click **Conference**.

Step outcome: A call is made to the third party and a new entry is added to **Phone Connection** window.

Step 5 Typically, the original caller is placed on hold by the CO while the call is made to the direct access contact but is removed from hold automatically when the direct access contact answers the call.

Step outcome: All parties are joined into the conference call.

If CO does not remove the original caller on hold:

- a** Click **Hook Flash**.

Step outcome: All parties are joined into the conference call.

Transfer call

Step 6 If you would prefer to transfer the call rather than create a conference, follow the procedure above, then:

- a** Ensure the "position" entry is selected in the **Phone Connection** window.
- b** Click **Release**.

Result of this task: You are released from the call while your caller and the direct access party remain connected.

7.2.4 Manual multiple-party conference

If you want to conference a party which has no direct access button and that is not in your contacts, you can manually dial their phone number.

7.2.4.1 Smart pad

To facilitate the transition for existing Guardian PSAPs, the buttons from the previous version of Guardian still function although they launch the smart pad in the appropriate mode. For example, if you have the **DTMF Pad** button in your layout but the smart pad behavior has been enabled, you can continue to use the **DTMF Pad** button - it launches the smart pad in DTMF mode.

Consult with your administrator to determine if smart pad behavior has been enabled for your role.

Manual conference using smart pad

Step 1 Answer the incoming administrative call.

Step 2 Click **Smart Pad** or **DTMF Pad** if the window is not included in your layout.

Step outcome: **Smart Pad** is displayed in DTMF mode.

Example:

Hook flash button is enabled in DTMF mode

Smart Pad			
[Search Bar]			
FLSH	1	2 ABC	3 DEF
QXfer	4 GHI	5 JKL	6 MNO
CONF	7 PQRS	8 TUV	9 WXYZ
Call	*	0	#
SEND		EXIT	

Step 3 From the **Smart Pad** window, click **FLSH**.

Step 4 Wait until you hear a dial tone in your headset. Your caller's audio is usually disconnected.

- Step 5** Enter a prefix number if required.
More info: Verify with your administrator if a prefix such as "9" needs to be entered on your system. If no prefix is required, skip this step.
- Step 6** Dial the number of the third party required:
- a** If you make a mistake dialing the number or receive a busy signal, click **FLSH** to end the attempted conference and start again from Step 3.
Step outcome: A call is made to the third party - you do not see a new entry in the **Phone Connection** window. When the third party answers, they are connected with you only.
 - b** Click **FLSH** to conference in your original caller.
Step outcome: All three participants are conferenced into the call.

If CO automatically conferences original caller:

- c** Skip Step 6(b).

Transfer call

- Step 7** If you prefer to transfer the call rather than maintain a conference, follow the procedure above, then:
- a** Click **Release**.

Result of this task: You are released from the call while your caller and the third party remain connected. The phone call is cleared from your **Phone Connection** window.

Manual conference using smart pad with SIP REFER

- Step 1** Answer the incoming administrative call.
- Step 2** Click **Smart Pad** or **DTMF Pad** if the window is not included in your layout.
Step outcome: **Smart Pad** is displayed in DTMF mode.
- Step 3** From the **Smart Pad** window, click **FLSH**.
- Step 4** Wait until you hear a dial tone in your headset. Your caller's audio is usually disconnected.
- Step 5** Enter a prefix number if required.
More info: Verify with your administrator if a prefix such as "9" needs to be entered on your system. If no prefix is required, skip this step.

Step 6

Dial the number of the third party required:

- a** If you make a mistake dialing the number or receive a busy signal, select the third party entry in the **Phone Connection** window, click **Release**, and start again from Step 3.

Step outcome: A call is made to the third party - you see a new entry added to the call in the **Phone Connection** window. When the third party answers, all three participants are conferenced into the call.

If CO doesn't automatically conference original caller:

- b** Click **FLSH** to conference in your original caller.

Transfer call**Step 7**

If you prefer to transfer the call rather than maintain a conference, follow the procedure above, then:

- a** Ensure the "position" entry is selected in the **Phone Connection** window.
- b** Click **Release**.

Result of this task: You are released from the call while your caller and the third party remain connected. The phone call is cleared from your Phone Connection window.

7.2.4.2 DTMF pad

If **Smart Pad** is not enabled for your role, you can still make a manual conference using the **DTMF Pad**.

Manual conference using DTMF pad

Step 1

Answer the incoming administrative call.

Step 2

In the **Phone Connection** window, click on your caller's entry.

Step 3

Click **DTMF Pad**.

Step outcome: The **DTMF Pad** is displayed.

Example:

**Step 4**

From **DTMF Pad**, click **Hook flash**.

Step 5

Wait until you hear a dial tone in your headset. Your caller's audio is usually disconnected.

Step 6

Enter a prefix number if required.

Step outcome: Verify with your administrator if a prefix such as "9" needs to be entered on your system. If no prefix is required, skip this step.

Step 7

Dial the phone number of the third party required.

- a** If you make a mistake dialing the phone number or receive a busy signal, click **Hook flash** to end the attempted conference and start again from Step 4.

Step outcome: A call is made to the third party - you do not see a new entry in the **Phone Connection** window. When the third party answers, they are connected with you only.

- b** Click **Hook flash** to conference in your original caller.

Step outcome: All three participants are conferenced into the call.

If CO automatically conferences original caller:

- c** Skip Step 7(b).
- d** Click **Close** in the **DTMF Pad** window.

Transfer call

Step 8 If you prefer to transfer the call rather than maintain a conference, follow the procedure above until Step 7, then:

- a** Click **Release**.

Result of this task: You are released from the call while your caller and the third party remain connected. The phone call is cleared from your **Phone Connection** window.

Manual conference using DTMF Pad with SIP REFER

Step 1 Answer the incoming administrative call.

Step 2 In the **Phone Connection** window, click on your caller's entry.

Step 3 Click **DTMF Pad**.

Step outcome: **DTMF Pad** is displayed.

Step 4 From **DTMF Pad**, click **Hook flash**.

Step 5 Wait until you hear a dial tone in your headset. Your caller's audio is usually disconnected.

Step 6 Dial the phone number of the third party required.

- a** If you make a mistake dialing the phone number or receive a busy signal, select the third party entry in the **Phone Connection** window, click **Release**, and start again from Step 3.

Step outcome: A call is made to the third party - you see a new entry added to the call in the **Phone Connection** window. When the third party answers, all three participants are conferenced into the call.

If CO doesn't automatically conference original caller:

- b** Click **Hook flash** to conference in your original caller.
- c** Click **Close** in the **DTMF Pad** window.

Transfer call

Step 7 If you prefer to transfer the call rather than maintain a conference, follow the procedure above, then:

- a** Ensure the "position" entry is selected in the **Phone Connection** window.
- b** Click **Release**.

Result of this task: You are released from the call while your caller and the third party remain connected. The call is cleared from your **Phone Connection** window.

7.2.5 Automated answering system selection

The third party you conference into your administrative call may have an automated answering system requiring input from you such as a password or a menu selection.

Entering data via Smart Pad

- Step 1** When you hear the automated message asking for input, click **Smart Pad** or **DTMF Pad**, whichever is included in your layout.
Step outcome: The **Smart Pad** window is displayed in DTMF mode.
- Step 2** From **Smart Pad**, enter the numbers required.
Step outcome: The numbers entered are sent to the receiving party immediately.
- Step 3** Click **EXIT** when you do not need to enter any further data.
Step outcome: The **Smart Pad** window is closed when in floating mode.

Entering data via DTMF Pad

- Step 1** In the **Phone Connection** window, select the entry for the party requiring input.
- Step 2** When you hear the automated message asking for your input, click **DTMF Pad**.
Step outcome: The **DTMF Pad** window is displayed.
- Step 3** From **DTMF Pad**, enter the numbers required.
Step outcome: The numbers entered are sent to the receiving party immediately.
- Step 4** Click **Close** when you do not need to enter any further data.
Step outcome: The **DTMF Pad** window is closed.

7.2.6 Conference another telecommunicator

You may need to conference another telecommunicator from your PSAP to your active call. Your administrator may have configured position status buttons in your role layout. These buttons are similar to direct access buttons with the additional ability to notify you of the

status of the other users in your PSAP via their color. Section *Position status* on page 43 explains the various states that are reflected via the position status buttons.

Conference telecommunicator with status buttons

- Step 1** Answer the incoming administrative call.
- Step 2** Click the button representing the telecommunicator's position you want to join your call.

Result of this task: When the second telecommunicator answers the call, they are automatically conferenced to your call.

Transfer to telecommunicator with status buttons

- Step 1** Answer the incoming administrative call.
- Step 2** Click the button representing the telecommunicator's position.
- Step 3** As soon as you hear the ring or see an entry for the second telecommunicator in the **Phone Connection** window, click **Release**.

Result of this task: You are released from the call while your caller and the second telecommunicator remain connected.

Transfer with line appearance buttons

- More info:** The line appearance button changes color (default yellow) when the call has been answered and placed on hold.
- Step 1** Click the line appearance button.
- Step outcome:** The original telecommunicator is still on the call but in a hold state.
- Step 2** Select the original telecommunicator's entry in the **Phone Connection** window.
- Step 3** Click **Release**.

Result of this task: You remain on the call with the caller while the original telecommunicator is released from the call.

Conference telecommunicator using Smart Pad

Step 1 Answer the incoming administrative call.

Without conference button in layout

- a** Click **Smart Pad** if the window is not included in your layout.
- b** From **Smart Pad**, click **CONF**.

Step outcome: **Smart Pad** is placed in conference mode.

With conference button in layout

- c** Click **Conference**.

Step outcome: **Smart Pad** is displayed in conference mode.

Step 2 Type the phone number of the other telecommunicator's position.

Step 3 Click **SEND**.

Step 4 Click **EXIT**.

Step outcome: When the second telecommunicator answers the call, they are automatically conferenced to your call.

Transfer call

Step 5 Click **Release**.

Result of this task: You are released from the call while your caller and the second telecommunicator remain connected.

Conference telecommunicator using Dial Pad

Step 1 Answer the incoming administrative call.

Step 2 Click **Place Call**.

Step 3 In **Dial Pad**, type the phone number of the other telecommunicator's position.

Step 4 Click **Conference**.

Step 5 Click **Close**.

Step outcome: When the second telecommunicator answers the call, they are automatically conferenced to your call.

Transfer call

- Step 6** Ensure the “position” entry is selected in the **Phone Connection** window.
- a** Click **Release**.

Result of this task: You are released from the call while your caller and the second telecommunicator remain connected.

7.2.7 Joining two calls

If you have two or more calls answered at your position you can join calls together into a conference call. The calls to join do not need to be adjacent in the **Phone Connection** window; whichever call on hold you select is conferenced when you click **Join**. You can join more than two calls into a single conference call if needed.

At least one call must be active in order to join calls.

Join calls

- Step 1** Select the call to be conferenced into your active call from the **Phone Connection** window.
- Step 2** Click **Join**.

Result of this task: The selected call on hold is conferenced into your active call.

7.3 Releasing calls

Releasing calls on the Guardian system depends on which entry is selected in the **Phone Connection** window before you click **Release** or **Hook Flash**. The following procedures explain how to release the required party from an enhanced administrative call.

Release the entire call

Step 1 In the **Phone Connection** window, select the top line of the call to be released.

Step 2 Click **Release**.

Result of this task: All parties on the call are released and the call is cleared from your **Phone Connection** window.

Release yourself

Step 3 In the **Phone Connection** window, ensure the "position" entry is selected.

Step 4 Click **Release**.

Result of this task: You are released from the call and the call is cleared from your **Phone Connection** window but the other parties remain on the call.

Release party from multi-party call

Step 1 In the phone window, select the incoming call entry.

Step 2 Click **Hook Flash**.

Result of this task: You remain connected with the original incoming caller.

Release party from multi-party SIP REFER call

Step 1 In the **Phone Connection** window, select the SIP REFER party's entry.

Step 2 Click **Release**.

Result of this task: You remain on the call and only the selected party is released.

7.4 Placing call

You can make an outgoing call using one of the three procedures documented in this section. All new outgoing calls use your PSAP administrative phone lines.

Place call using direct access buttons

Step To call a direct access contact, single-click the direct access button.

Result of this task: A call is made to the direct access contact and the call is visible in your **Phone Connection** window.

Place call using contacts

Step 1 Click **Contacts**.
Step outcome: The **Contacts** window is displayed.

Step 2 Select the entry to call.

Step 3 Click **Call**.

Result of this task: A call is made to the entry selected and is visible in your **Phone Connection** window.

Place call using Smart Pad

Step 1 Click **Place Call** or **Smart Pad** (whichever is available in your layout).
Step outcome: The **Smart Pad** window is displayed in call mode.

Step 2 In **Smart Pad**, enter the phone number to call.

Step 3 Click **SEND**.

Result of this task: A call is made to the phone number entered and is visible in your **Phone Connection** window.

Place call using Dial Pad

Step 1 Click **Place Call**.
Step outcome: The **Dial Pad** window is displayed.

Step 2 In **Dial Pad**, enter the phone number to call.

Step 3 Click **Call**.

Result of this task: A call is made to the phone number entered and is visible in your **Phone Connection** window.

8 Text conversations

This chapter describes the procedures to answer, process, and make text calls.

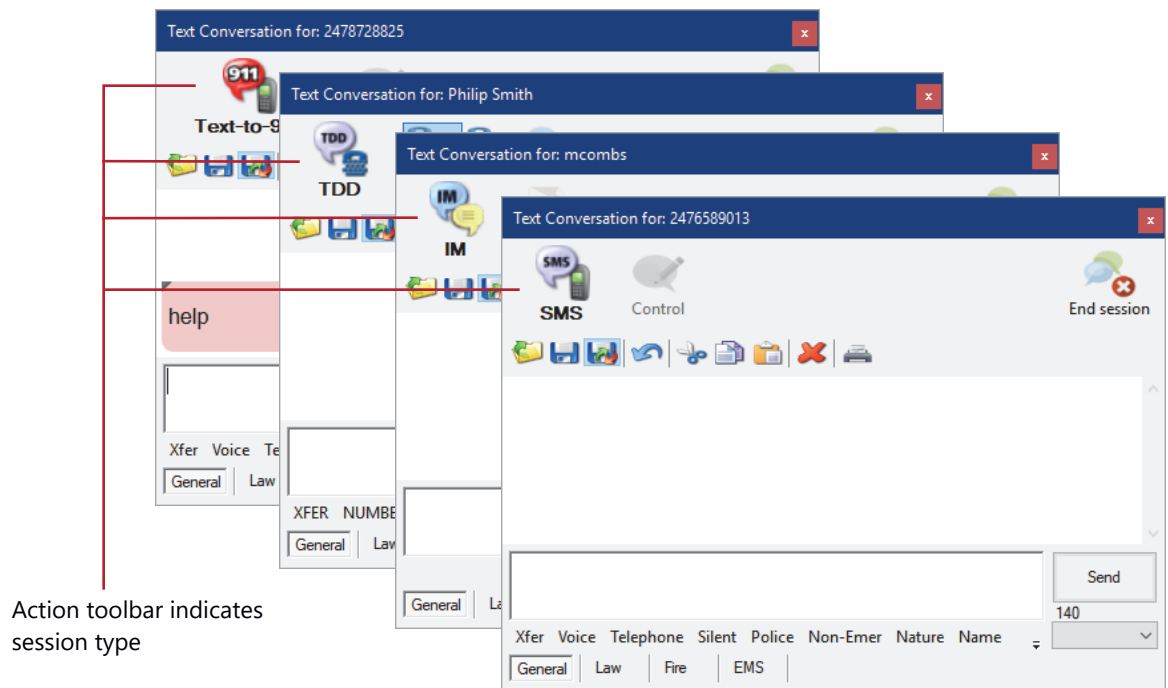
8.1 Text conversations

Guardian supports four forms of text conversations:

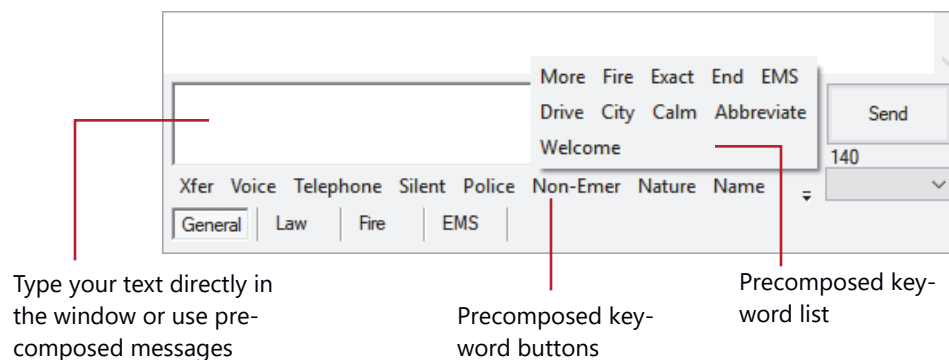
- 1 Text-to-9-1-1
- 2 TDD
- 3 Outgoing SMS messaging
- 4 Instant messaging

All text communication takes place within the **Text Conversation** window as shown in Figure 8.1. The **Text Conversation** window opens a session corresponding to the type of text conversation; i.e., Text-to-911, TDD, IM, or SMS.

Figure 8.1 Text Conversation window in all four modes



You can communicate with the caller or instant message with a telecommunicator by typing text directly or using precomposed messages as shown in Figure 8.2.

Figure 8.2 Precomposed messages in Text Conversation window


The precomposed messages are organized and displayed by category:

- General
- Police / Law
- Fire
- EMS

Access messages for category

Step 1 Click a category tab.

Step outcome: The precomposed message keywords are updated with messages related to the category.

Step 2 Click a message keyword or select a keyword from the list.

Step 3 Press **Send** or click **Enter** on your keyboard.

Precomposed messages may have been edited by your administrator and can be different for each type of text conversation; i.e., Text-to-911, TDD, SMS, and IM.

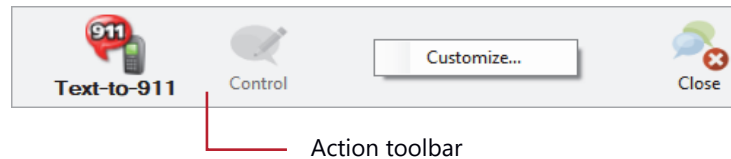
8.1.1 Customize font and button size

The font size displayed in the **Text Conversation** window can be customized. Changes made are in effect until you log out of Guardian.

Adjust font size

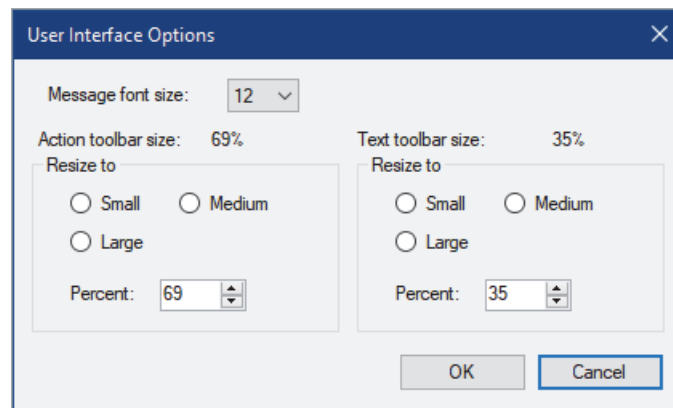
- Step 1** Right-click on the action toolbar of the **Text Conversation** window and select the **Customize** option.

Example:



Step outcome: The **User Interface Options** dialog box is displayed.

Example:



- Step 2** Set **Message font size** to the required font size.

- Step 3** Click **OK**.

Result of this task: The font is resized to the specified value when a new message arrives or is sent.

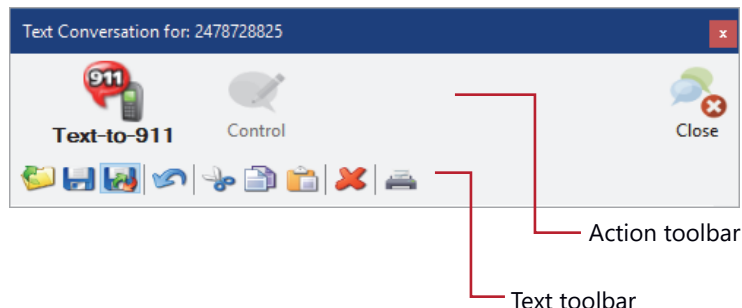
Adjust toolbar size

More info: You can customize the size of the buttons appearing on the action toolbar and the text toolbar.

Step 1

Right-click on the action toolbar of the **Text Conversation** window and select the **Customize** option.

Example:



Step outcome: The **User Interface Options** dialog box is displayed.

Step 2

Select **Small**, **Medium**, **Large**, or enter a custom percentage for the **Action** toolbar size.

Step 3

Select **Small**, **Medium**, **Large**, or enter a custom percentage for the **Text** toolbar size.

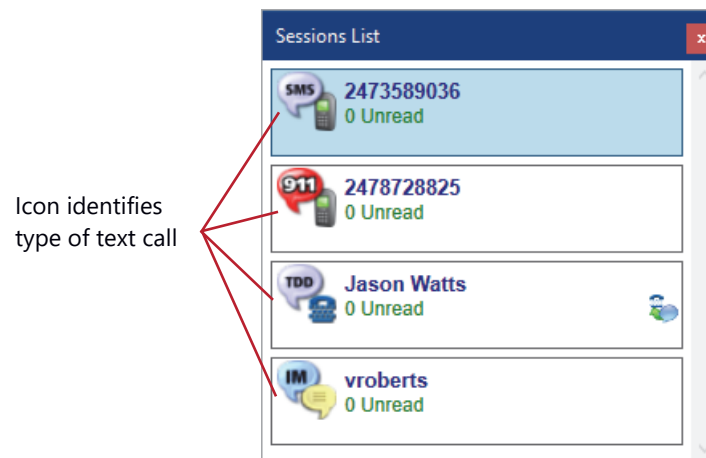
Step 4

Click **OK**.

Result of this task: The buttons on the action and text toolbar are resized.

8.1.2 Sessions list

The Guardian system allows you to answer a combination of text calls such as text-to-911, instant messaging, and/or TDD simultaneously. The **Sessions List** window helps you move between calls. The icon for each entry identifies the type of text call and displays the callers ANI or callerID (when available) as shown in Figure 8.3. Voice calls are not listed in the **Sessions List** window.

Figure 8.3 Sessions List window

If **Sessions List** is not available in your layout, you can launch it from the **View** menu.

Switch between text calls

Step 1 On the **View** menu, click **Sessions List**.

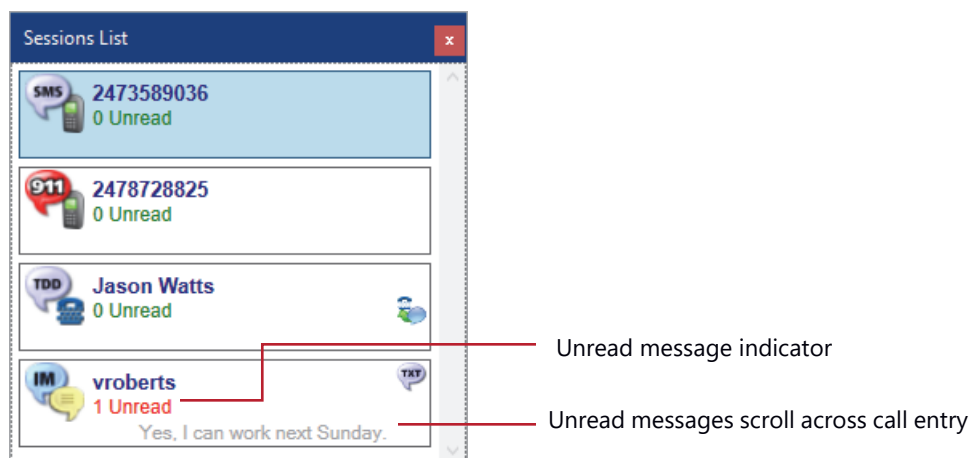
Step 2 Click an entry.

Result of this task: The selected call is brought into focus in the **Text Conversation** window.

Unread messages

Sessions List notifies you of new unread messages. The unread messages scroll across the **Sessions List** window as shown in Figure 8.4.

Figure 8.4 Sessions List with unread message

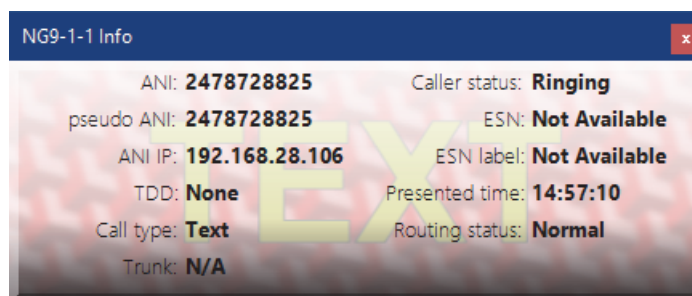


8.2 Text-to-911

The Guardian system can receive and process text-to-911 calls from wireless devices. The call is displayed in the **Phone Connection** window even though there is no audio established between the 9-1-1 text caller and you. When a text-to-911 call is detected, the **NG9-1-1 Info** window flashes "TEXT" instead of "9-1-1". The ALI is displayed for the call when available.

The **NG9-1-1 Info** window notifies you that the incoming call is a text call as shown in Figure 8.5. In addition, your administrator may have configured a different ring tone for text-to-911 calls to distinguish them from voice calls.

Figure 8.5 NG9-1-1 Info window for text-to-911 call



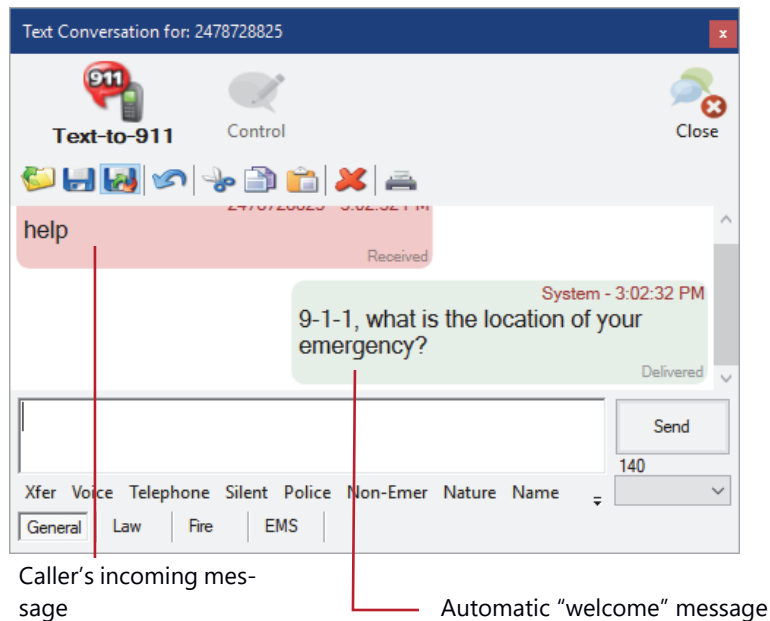
Answer text-to-911 call

Step

Answer the incoming text call as you would a voice call.

Step outcome: The **Text Conversation** window is presented with the caller's message. If configured by your administrator, an automatic "welcome" message is sent on your behalf.

Example:

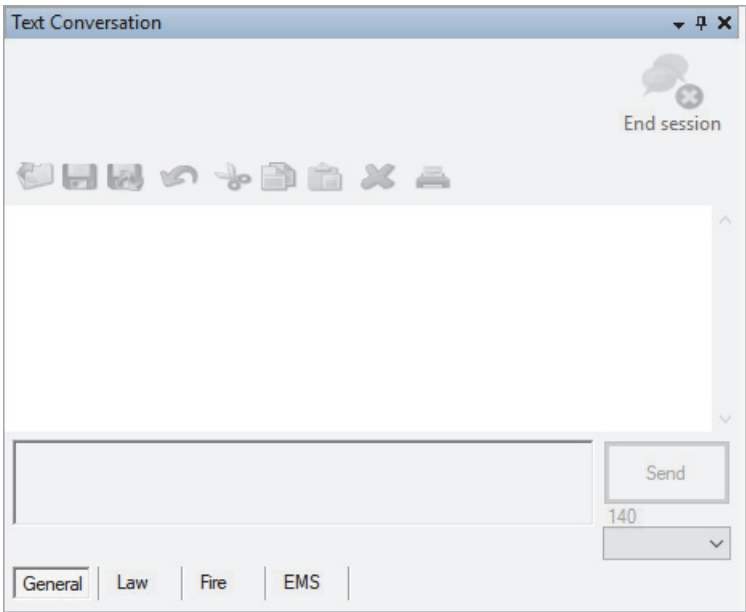


8.2.1 Mute, hold, and privacy

Text-to-911 calls only support placing the call/caller on hold as there is no voice connection with the caller to mute or place on privacy. The call is paused (placed on hold) automatically in the **Phone Connection** window when you answer a new voice or text call. If the **Text Conversation** window is not docked in your layout, the window is automatically closed when the text-to-911 call is paused.

If the window is docked in your layout and the call is paused, then the **Text Conversation** window for the call is disabled as shown in Figure 7.9.

Figure 8.6 Text Conversation window when text-to-911 call is placed on hold



Place text call on hold

- Step 1
- Ensure the text call is active.
- Step 2
- Click **Hold**.

Step outcome: The call is placed on hold and the caller’s status is paused in the **Phone Connection** window.

Example:

[Primary side] Telephone - 2475553001				
	Status	Name	Number	DNIS
☒	9-1-1 onhold	Hudson911		
☒	onhold	position	2475553001	
☒	paused	2478728825	2478728825	2475554001
TXT				

Remove text call from hold

- Step** You can remove a text call from hold by either of the following methods:
- a** Select the text call in **Sessions List**.
 - b** Ensure the text call is selected in the **Phone Connection** window and click **Hold**.

Result of this task: The text call is removed from hold and any active call is placed on hold automatically. Previous messages are displayed in the **Text Conversation** window.

8.2.2 Conference call on same Guardian system

You can conference another telecommunicator to a text-to-911 call. The telecommunicator does not need to be a member of your PSAP; they can be a user from another PSAP connected to the same Guardian system.

Two-way audio is established between the other telecommunicator and yourself. If the other telecommunicator's configuration supports text-to-911 calls, the **Text Conversation** window is automatically launched on their screen displaying the existing conversation.

You can conference another telecommunicator to your text call using a position status button, a direct access button for the telecommunicator's PSAP or ring group, or by manually dialing the telecommunicator's position.

Conference telecommunicator with position status or DA buttons

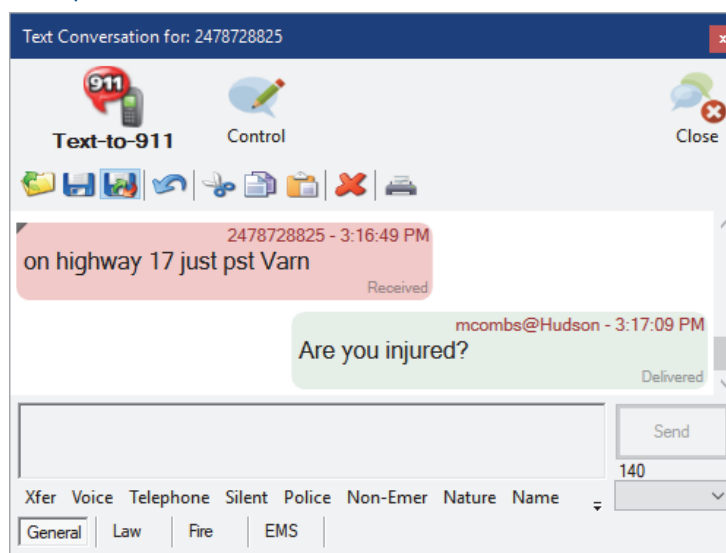
More info: If your layout includes position status buttons or ring group DA buttons, you can call the position directly or call the ring group the telecommunicator belongs to.

Step

Click the button representing the position/ring group you want to conference.

Result of this task: When the telecommunicator answers, they are conferenced into your call. If they have text-to-911 capability, the **Text Conversation** window is launched and displays any existing messages.

Example:



Conference telecommunicator manually

Without conference button in layout

- a Click **Smart Pad** if the window is not included in your layout.
- b From **Smart Pad**, click **CONF**.

Step outcome: **Smart Pad** is placed in conference mode.

With conference button in layout

- c Click **Conference**.

Step outcome: **Smart Pad** is displayed in conference mode.

Step 1

Type the phone number of the other telecommunicator's position.

Step 2 Click **SEND**.

Step outcome: The call appears as a text-to-911 call at the other telecommunicator's position and the routing status identifies the call as a transfer. When the other telecommunicator answers the call, they are automatically conferenced into the call and have two-way audio with you.

Caution Do not release yourself from the call unless the second telecommunicator has the ability to handle text-to-911 calls.

Take control of text call

More info: If you have been conferenced into a text-to-911 call, you see existing messages but you cannot send a message until you take control of the call.

Step From the **Text Conversation** window of a conferenced call, click **Control**.
Step outcome: Control of the call is granted to you. The text entry portion of the **Text Conversation** window is enabled. A notification of change in call control is displayed on both telecommunicators' screens but is not sent to the caller.

8.2.3 Conference call to different system

You can conference another telecommunicator from a different Guardian system to a text-to-911 call.

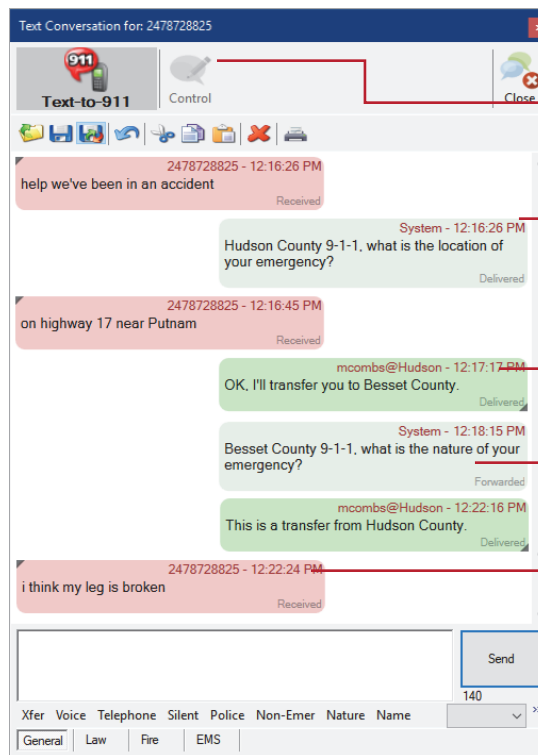
If the other system is text capable, a text call is established with the PSAP. When the call is answered, the **Text Conversation** window is automatically launched on their screen displaying the existing conversation and all three participants (caller, original telecommunicator, and second telecommunicator) are connected on the same text call. If an automatic "welcome" message is configured for the second PSAP, it is sent automatically - both the caller and original telecommunicator will see it.

If the other system is not text capable, a voice call is established between the original telecommunicator and the second telecommunicator as described in *Conference text-to-911 call* on page 199.

Conference telecommunicator on other system

- Step 1** Call the other PSAP using a direct access button, contacts, or by manually dialing.
- Step outcome:** When the second telecommunicator answers, they are conferenced into the text conversation.
- Step 2** Type your messages into the **Text Conversation** window.
- Step outcome:** Your messages are sent to all participants on the text call and their messages are visible on your screen.

Example:



Control button is disabled - you don't need to take control of the call to send your message

Sender is identified in each message

Messages from original T/C on first system

Messages from T/C on second system

Messages from Guardian are displayed on the right while messages from the caller are displayed on the left

Transfer call

- Step 3** Ensure the "position" entry is selected in the **Phone Connection** window.
- Click **Release**.
 - Click a disconnect reason button if available.

Result of this task: You are released from the call but the caller remains connected with the second telecommunicator.

The conversation, when displayed in the **Local Text Conversation** window, indicates the second telecommunicator by adding "[external]" in the *From* column as shown in Figure 8.7.

Figure 8.7 Local text conversation window displaying transferred call



File name	Date/Time	From	Action	Message
SMS_20180417_0846.2478728	2018-04-17 9:57:55 AM	2478728825	Received	yes, marker 82
SMS_20180417_0857.2478728	2018-04-17 9:58:27 AM	mcombs@Hudson	Delivered	OK, I'll transfer you to Besset County.
SMS_20180417_0908.2478728	2018-04-17 9:59:06 AM	mcombs@Hudson	Delivered	This is Hudson County with a car accident in your area.
SMS_20180417_0956.2478728	2018-04-17 9:59:20 AM	dchabert@Besset [external]	Forwarded	Are you injured?
SMS_20180417_1055.2478728	2018-04-17 9:59:27 AM	2478728825	Received	yes
SMS_20180417_1212.2478728	2018-04-17 9:59:40 AM	dchabert@Besset [external]	Forwarded	Where are you injured?
SMS_20180417_1216.2478728	2018-04-17 9:59:51 AM	2478728825	Received	I think my leg is broken.
SMS_20180420_1228.2478728	2018-04-17 10:02:57 AM	dchabert@Besset [external]	Forwarded	I've dispatched an ambulance to your location.

8.2.4 Conference text-to-911 call

You can conference a third party, voice only, into a text-to-911 call. Two-way audio between you and the third party is established. The audio between you and third party is recorded while the text messaging between you and the caller is saved as an attachment to the call.

Conference third party

Step 1 Answer the incoming text call.

Step 2 Conference a third party into a text-to-911 the same as with a regular 9-1-1 call using one of the following methods:

- Click the DA button for the third party.
- Select the third party from **Contacts** and click **Conference**.
- Use **Smart Pad** to manually dial the third party.

Step outcome: A call is made to the third party without placing the text call on hold.

Example:

The screenshot shows a software interface titled "[Primary side] Telephone - 2475553001". It contains a table with columns: Status, Name, Number, and DNIS. The table lists several entries: a "9-1-1 active" call with name "Hudson911", a "connected" call with name "position" and number "2475553001", a "9-1-1 connected" call with name "2478728825" and number "2478728825", and a "connected" call with name "DeWalt Trauma Center" and number "2477859622". There is also a "TXT" icon next to the "9-1-1 connected" entry.

Status	Name	Number	DNIS
9-1-1 active	Hudson911		
connected	position	2475553001	
9-1-1 connected	2478728825	2478728825	2475554001
connected	DeWalt Trauma Center	2477859622	

Result of this task: When the third party answers the call, two-way audio is established between you and the third party. The text messages are not available to third party since they connected via a voice call.

Info

Quick Transfer is disabled for text-to-911 calls since the third party may not support text-to-911 calls.

8.2.5 Disconnected text-to-911 call

There is a considerable potential for a text-to-911 call to be disconnected: caller's wireless device loses signal, becomes disabled, or loses power, etc. If the system detects that the caller is no longer available to receive messages, the call status in the **Phone Connection** window is updated to "9-1-1 disconnected" as shown in Figure 8.8. The call is not removed from the **Phone Connection** window; instead, the system provides you with the opportunity to attempt a voice callback or a text callback.

Figure 8.8 Phone Connection window when text-to-911 call is disconnected

[Primary side] Telephone - 2475553001			
Status	Name	Number	DNIS
9-1-1 active_Abn	Hudson911		
connected	position	2475553001	
9-1-1 disconnected	2478728825	2478728825	2475554001

The **Text Conversation** window is disabled if it is docked in your layout, otherwise it is closed automatically when the system detects that the caller is disconnected.

Info

A voice call is made to a disconnected text-to-911 call if the **Callback** button is used. A new SMS session and a text message is sent if the **Text Callback** button is used.

Call back text-to-911 call

Step While the call entry is still displayed in the **Phone Connection** window, click **Callback**.

Result of this task: A voice call is made to the caller and a new entry is added to the call in the **Phone Connection** window. When the caller answers you are connected to them via a voice call.

Text back text-to-911 call

Step 1 While the call entry is still displayed in the **Phone Connection** window, click **Text Callback**.

Step outcome: A new Text Conversation window is launched and a SMS session to the caller's callback number is started. The new session is added to the Session List window.

Step 2 Enter your message and click Send.

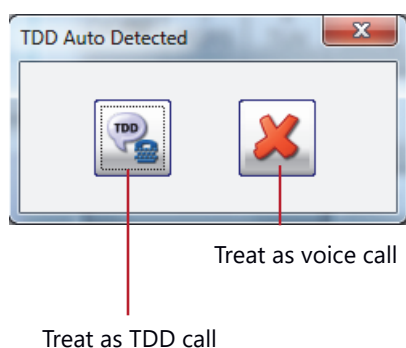
Result of this task: The message is sent to the caller's callback number.

8.3 Hearing impaired calls

The Guardian system supports auto-detection of hearing impaired calls. Since detection is based on the incoming tones, detection occurs after the call is answered. Automatic detection supports two forms of notification:

- 1 A pop-up dialog box is displayed as shown in Figure 8.9

Figure 8.9 TDD Auto Detected window



- 2 A change in color of the **TDD** button is displayed as shown in Figure 8.10

Figure 8.10 TDD auto detect indicator



Your administrator may have chosen either of these indicators for your PSAP or they may have decided not to enable automatic detection. If automatic detection is not enabled, you

can still communicate with hearing impaired calls although you have to launch TDD mode manually.

Answer incoming TDD call

Step 1 Answer the incoming call.

With pop-up indicator

More info: If the system detects a hearing impaired call, the **TDD Auto Detected** dialog box is presented on your screen.

a Click **Treat as TDD**.

With color indicator

More info: If the system detects a hearing impaired call, the **TDD** button in your layout changes color.

b Click **TDD**.

With no indicator

More info: If the system does not detect a TDD call or your system is not configured for automatic detection, you can still communicate with a hearing impaired caller.

c Click the **TDD** button in your layout.

Step outcome: The Text Conversation window opens in TDD mode and a “welcome” message is sent automatically.

Step 2 Type the message or select a precomposed message keyword.

Step 3 Click **Send** or press **Enter** on your keyboard.

8.3.0.1 Voice carry over mode

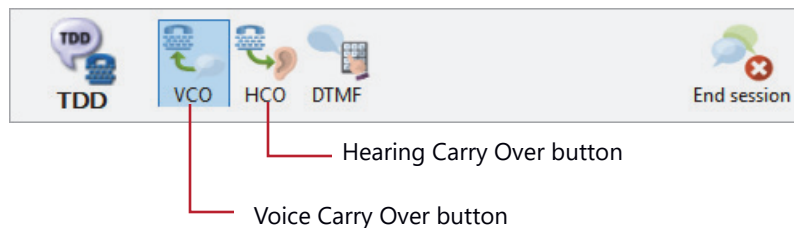
Tip: By default, the system puts your position in VCO mode when you access TDD manually so you can hear the caller and/or any background audio.

Voice carry over (VCO) mode allows you to hear any audio on the call such as background noise. It is also useful for caller's who have difficulty with their hearing only. In this instance, the caller sends a voice message and you respond using a text message.

8.3.0.2 Hearing carry over mode

If your caller notifies you that they can hear you, just not speak to you, you can place the system in hearing carry over (HCO) mode. In this instance, you would read your caller's message and verbally reply to your caller.

Figure 8.11 VCO and HCO buttons on the action toolbar



Voice carry over mode

More info: You only need to enable VCO mode when auto detection is enabled. VCO mode is enabled automatically when TDD is launched manually.

- Step 1** Answer the incoming call.
- Step 2** Click **Treat as TDD** or the colored **TDD** button.
Step outcome: The **Text Conversation** window is displayed.
- Step 3** From the **Text Conversation** window, click **VCO**.
- Step 4** Listen to your caller and reply by typing your response.

Hearing carry over mode

- Step 1** Answer the incoming call.
- Step 2** Click **TDD**.
Step outcome: The **Text Conversation** window is displayed.
- Step 3** From the **Text Conversation** window, click **HCO**.
Step outcome: The bottom portion of the **Text Conversation** window is disabled.
- Step 4** Reply to the caller verbally.

8.3.1 Conference hearing impaired calls

Hearing impaired calls can be conferenced, and transferred, the same as regular voice calls. It is good practice to stay on the call until the conference is connected to announce the nature of the call before releasing yourself from the call to complete the transfer.

8.3.1.1 Conference T/C to hearing impaired call

You can conference another telecommunicator to your hearing impaired call as you would with a regular call. If auto detection is enabled, the second telecommunicator is displayed the pop-up notification or the **TDD** button changes color from which they can launch the **Text Conversation** window once they have answered the call. Any messages that have already passed between you and the caller are not viewable by the second telecommunicator. However, the second telecommunicator has two-way audio with you and hears any incoming audio from the TDD caller.

Both you and the second telecommunicator can communicate with the incoming caller via the **Text Conversation** window but you must take turns sending messages to the caller. Text messages from you and the second telecommunicator are sent to the caller but be aware that the text messages in the **Text Conversation** window do not identify which telecommunicator wrote the message.

Transfer hearing impaired call

- Step 1** Answer the incoming call.
- Step 2** Click **TDD**.
Step outcome: The **Text Conversation** window is displayed.
- Step 3** Conference the second telecommunicator.
- Step 4** Wait for the second telecommunicator to answer the call.
More info: The second telecommunicator has two-audio with you.
- Step 5** Notify the second telecommunicator to launch the **Text Conversation** window if auto-detection is not enabled.
- Step 6** Ensure the "position" entry is selected in the **Phone Connection** window.
 - Click **Release**.
 - Click a disconnect reason button if available.

Result of this task: You are released from the call but the caller remains connected with the second telecommunicator.

8.3.2 DTMF tone detection

In the event that you receive a call and hear no voice audio from the caller, after challenging the call with TDD, you may want to attempt tone detection. DTMF tone detection provides you with the ability to ask the caller to respond to you by pressing a key on their phone. The system will then interpret the key pressed and display it on the screen for you.

VCO mode is enabled automatically while in DTMF detection mode so you can hear any background audio.

Enable DTMF tone detection

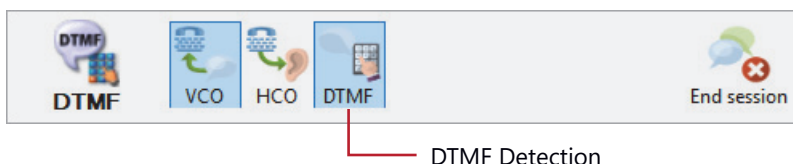
Step 1 Answer the incoming call.

Step 2 Click **TDD** in your layout.

Step outcome: The **Text Conversation** window opens in TDD mode. The “welcome” message is sent automatically.

Step 3 Click **DTMF Detection** on the action toolbar.

Example:

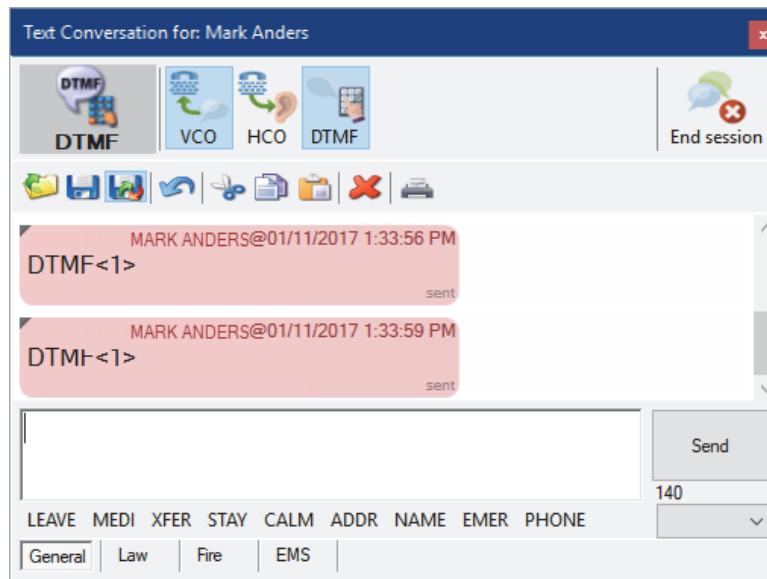


Step 4

Ask the caller to respond by pressing a key on their phone.

Step outcome: When the caller presses a number on their phone it is interpreted and displayed in the **Text Conversation** window.

Example:

**Step 5**

When you are not expecting anymore tones or ending the call, click **End Session**.

Step outcome: The **Text Conversation** window is closed.

- a Click **Release**.
- b Click a disconnect reason button if available.

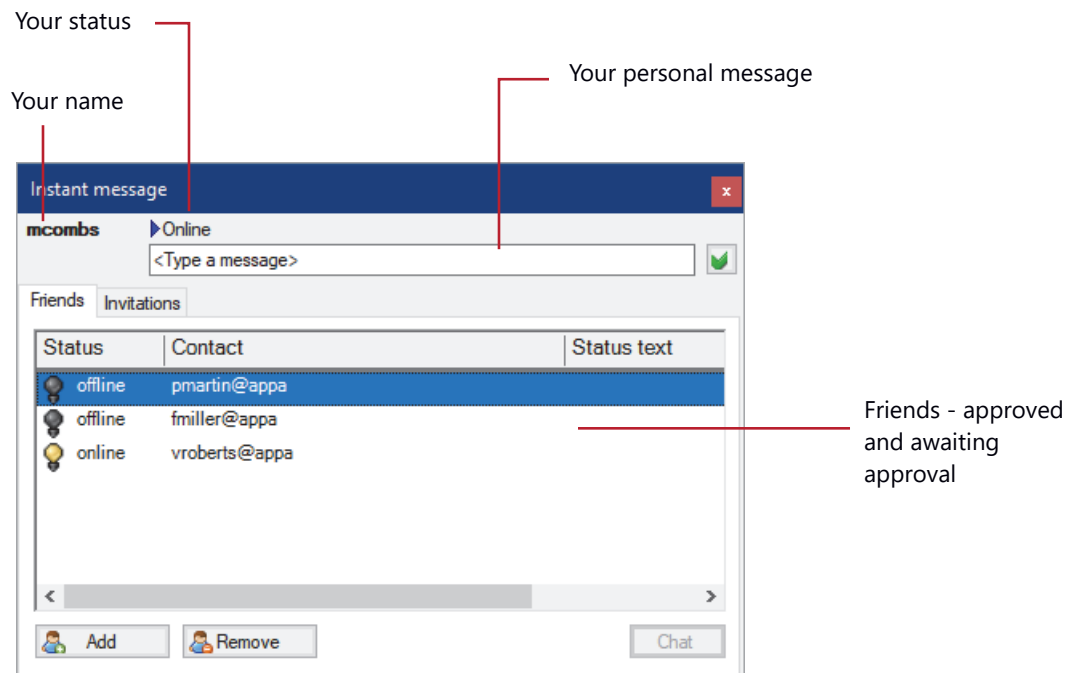
8.4 Instant messaging

The Instant Messaging (IM) feature can interface between other IM users on the system and with users of external IM applications, if configured. Instant messages are saved to the local position and are viewable by all logged in users from the **Local Text Conversations** window.

Menu: View > Instant Message

If the **Instant Message** window is not visible in your layout, launch it from the View menu as a floating window, as shown in Figure 7.28.

Figure 8.12 Instant Message window



Add IM friend

More info: Before sending instant messages to other users, you must send the user an invitation and await approval.

Step 1

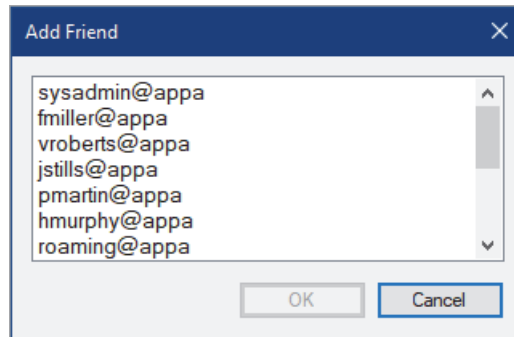
On the **View** menu, click **Instant Message**.

Step outcome: The **Instant Message** window is displayed.

Step 2 Click the **Add** button from the **Friends** tab.

Step outcome: The **Add Friend** dialog box is displayed listing all user accounts available for the system.

Example:



Step 3 Click the email address of the friend to add.

Step 4 Click the **OK** button.

Result of this task: An invitation to be added as a friend to your IM list is sent to the email address. You must be approved by the user before you can start sending messages.

Approve/refuse invitation

More info: When another user adds you to their friends list, you receive a request to be added as a friend. You can choose to accept or refuse the invitation.

Step 1 On the **View** menu, click **Instant Message**.

Step outcome: The **Instant Message** window is displayed.

Step 2 Click the **Invitations** tab.

Step 3 Select a request and click the **Approve** or **Refuse** button.

Remove IM friend

Step 1 On the **View** menu, click **Instant Message**.

Step outcome: The **Instant Message** window is displayed.

Step 2 Select the friend from the **Friends** tab.

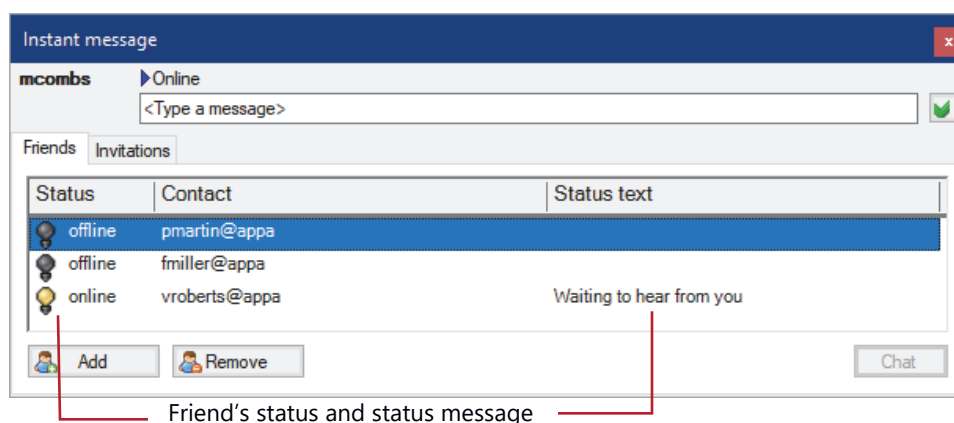
Step 3 Click the **Remove** button.

Result of this task: The friend is removed from your friends list. Your status is set to disabled on the friend's instant message window.

8.4.1 Status

You can update your status to allow friends to see if you are available, busy, away, etc. Your status is displayed to your friends in their instant message window as shown in Figure 8.13.

Figure 8.13 Instant Message window displaying friend's status



Modify status

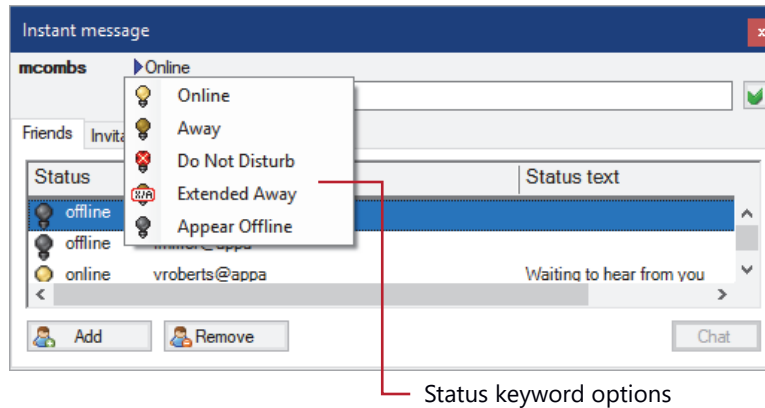
Step 1 On the **View** menu, click **Instant Message**.

Step outcome: The **Instant Message** window is displayed.

Step 2 Click on the status keyword in the **Instant Message** window.

Step outcome: A list displaying other statuses is displayed.

Example:



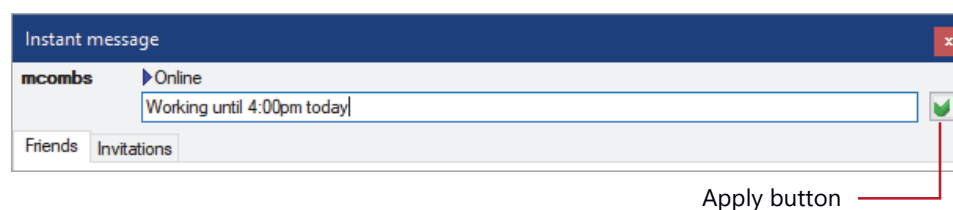
Step 3 Select a status from the list.

Result of this task: Your status is immediately updated in your friends' instant message window.

8.4.2 Personal message

You can add a custom message that is displayed to your friends in their instant message window as shown in Figure 8.14.

Figure 8.14 Personal message in IM window



Add personal message

- Step 1** On the **View** menu, click **Instant Message**.
Step outcome: The **Instant Message** window is displayed.
- Step 2** Type your personal message.
- Step 3** Click the **Apply** button.

Result of this task: Your message is immediately updated in your friends' instant message window.

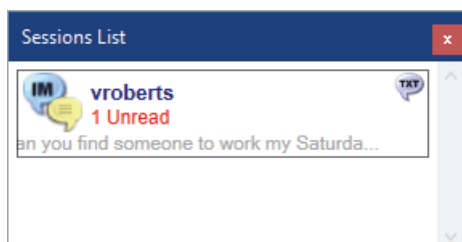
Chat using instant messaging

- Step 1** On the **View** menu, click **Instant Message**.
Step outcome: The **Instant Message** window is displayed.
- Step 2** Select a friend from the **Friends** tab.
- Step 3** Click the **Chat** button.
Step outcome: The **Text Conversation** window is displayed.
- Step 4** Type your message in the lower portion of the window.

Step 5 Click the **Send** button.

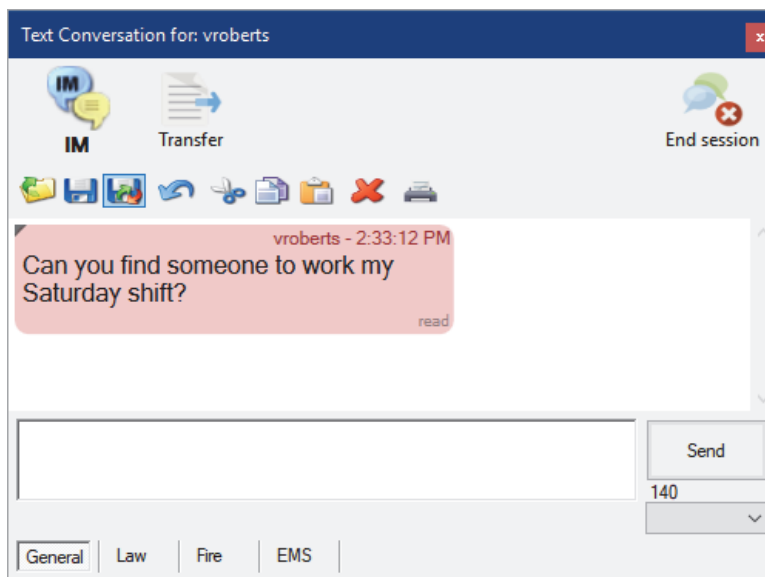
Step outcome: The **Sessions List** window on your friend's position is updated to indicate they have an incoming message from you.

Example:



More info: When your friend clicks on the indicator in the Sessions List window, their **Text Conversation** window is displayed.

Example:

**Step 6** Click the **End session** button when you want to end the conversation.

8.4.3 Transfer files

You can transfer files to instant messaging friends. The maximum size of a transferred file is 2MB.

Transfer file

Step 1 On the **View** menu, click **Instant Message**.

Step outcome: The **Instant Message** window is displayed.

Step 2 Select a friend from the **Friends** tab.

Step 3 Click the **Chat** button.

Step outcome: The **Text Conversation** window is displayed.

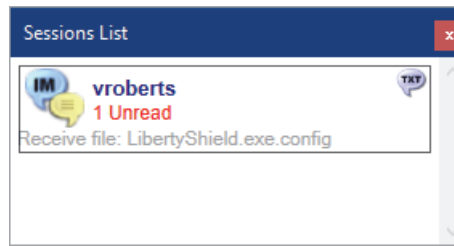
Step 4 Click the **Transfer** button on the action toolbar.

Step 5 Navigate to the location of the file to transfer.

Step 6 Click the **Transfer** button.

Step outcome: The **Sessions List** window on your friend's position is updated to indicate they have a file transfer from you.

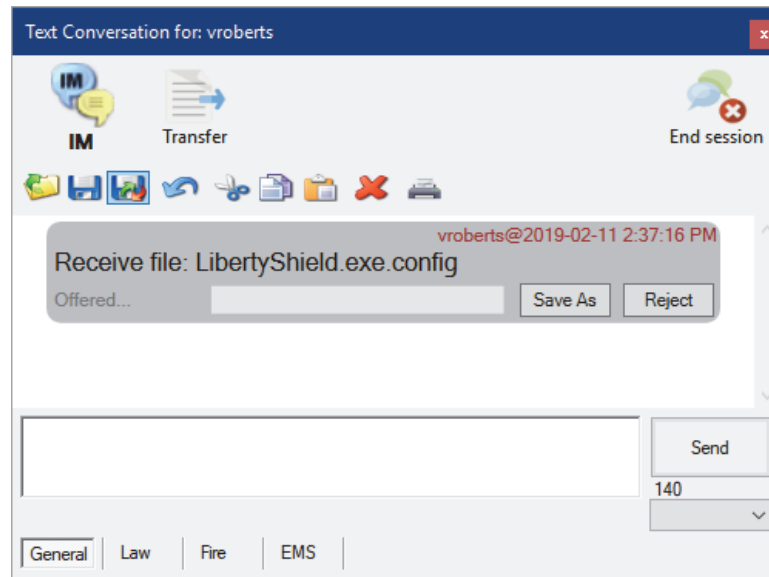
Example:



Accept transferred file

More info: If you receive a file transfer, click on the indicator in the **Sessions List** window, your **Text Conversation** window is displayed.

Example:



Step 7

Click the **Save As** button to save the transferred file to your position.

Step 8

Click the **End session** button when done.

9 User tools

This chapter describes the tools and features available in Guardian that allow you to adjust and save volume settings, join and leave ring groups, change your password, and create personal contacts and greetings.

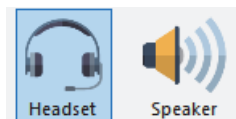
Depending on the privileges assigned to your role not all of these features may be available and should be disregarded.

9.1 Audio source

Depending on your system's configuration, you may have the ability to redirect your caller's audio to your position's external speakers. This feature can be a benefit if you experience difficulty discerning your caller's enunciation and require another individual to listen to the caller, or maybe an individual near you can act as a translator for your caller.

It is easy to identify if the audio can be redirected by verifying if the **Speaker** button is included in your layout.

Figure 9.1 Headset/speaker audio source buttons



Switch audio output

Step 1 Ensure the active call is selected in the **Phone Connection** window.

Step 2 Click **Speaker**.

Step outcome: The audio is redirected to the position's external speakers. Ensure the volume dial on the speakers is high enough to hear the audio.

Return audio to headset

Step 3 Click **Headset**.

Result of this task: The audio is returned to the headset/handset.

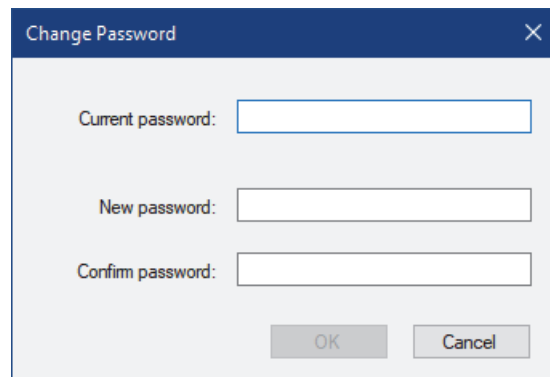
9.2 Change password

You can change your existing password at any time.

Menu: Tools > Change Password

When you launch the change password tool, the **Guardian - Change Password** window is displayed as shown in Figure 9.2.

Figure 9.2 Guardian - Change Password window



The screenshot shows a standard Windows-style dialog box titled "Change Password". It features three text input fields stacked vertically, each preceded by a label: "Current password:", "New password:", and "Confirm password:". At the bottom of the dialog, there are two buttons: "OK" and "Cancel". The dialog box has a dark blue title bar with a close button (X) on the right.

9.2.1 Password rules

- Password must be a minimum of 6 characters in length.
- The password is case sensitive so be aware of using lower- and uppercase.
- Any combination of alphanumeric characters, including special characters, is allowed.
- There is no time limit applied to the password.
- There is no minimum complexity requirement for the password.

Change your password

- Step 1** On the **Tools** menu, click **Change Password**.
- Step 2** Enter your current password.
- Step 3** Enter your new password twice.
- Step 4** Click **OK**.

Info It is good practice to log out and log back in using your new password to ensure that it has been applied successfully after changing it.

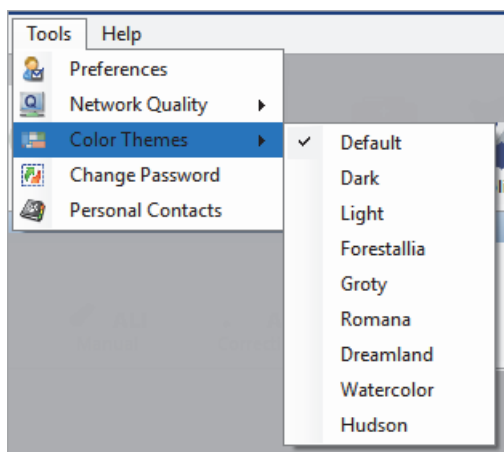
9.3 Color themes

Guardian supports more than one color theme for your interface. If the feature is enabled, your administrator may assign a customized theme to your interface that is applied automatically when you log in. They may also provide you a selection of themes from which you can choose a theme for your current session. A **Default** theme is available at the top of the list which applies colors specified within Microsoft Windows.

If your role is assigned the privilege, you can view available themes from the **Tools** menu.

Menu: **Tools** > **Color Themes**

Figure 9.3 Color themes are selectable from the Tools menu



Change color theme

Step 1 On the **Tools** menu, click **Color Themes**.

Step 2 Select a theme.

Result of this task: The theme is applied immediately and remains in effect until you log out. You can change the theme again while you are logged into a Guardian session.

Restore default theme

- Step 1** On the **Tools** menu, click **Color Themes**.
- Step 2** Select **Default**.

Result of this task: The colors designated within Microsoft Windows are applied immediately.

9.4 PAC II Settings

The PAC II unit allows you to connect an auxiliary audio device to your headset and allows you to configure the Push-To-Signal (PTS) button on your handset/headset for either Push-To-Talk (PTT) or Push-To-Mute (PTM) mode. Your role must be assigned the privilege to access this feature.

9.4.1 Auxiliary audio selection

If your position is equipped with a PAC II unit, your administrator may offer you the ability to listen to external audio sources while you are on-hook (not on a call.) The PAC II unit supports input from up to three external devices. For example, the audio from a TV monitor in your center may be connected, the audio from a radio source, and a cable to which you can connect your own music player.

Menu: **Tools > PAC II Settings > Auxiliary**

The Auxiliary feature allows you to select which source is currently sending audio to your headset.

Audio received from an attached auxiliary device plays in your headset when you are not on an active call or connected to a radio console. The audio from the auxiliary device is suspended during an active call and resumes when the call is ended.

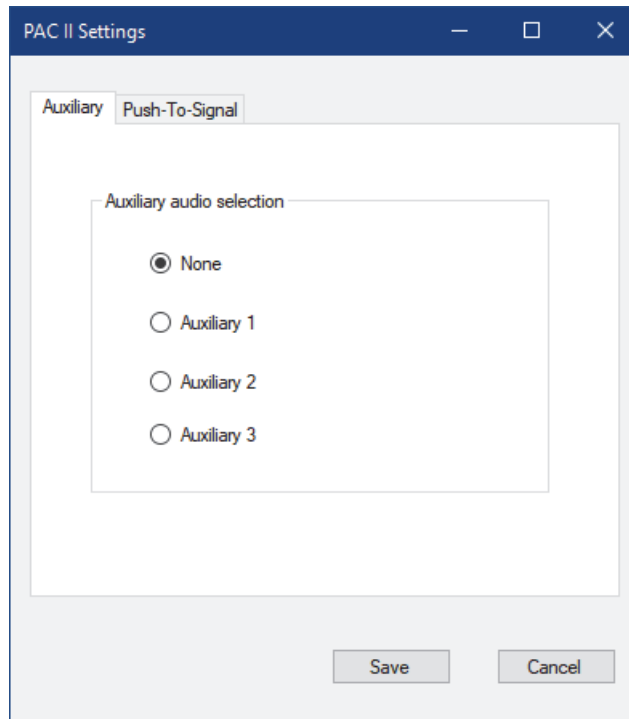
Up to three devices can be physically connected to the PAC II unit but your headset can only receive audio from one source at a time.

Select audio source

Step 1 On the **Tools** menu, click **PAC II Settings**.

Step outcome: The **PAC II Settings** window is displayed.

Example:



Step 2 Ensure the **Auxiliary** tab is selected.

Step 3 Select the audio source to connect.

Step 4 Click **Save**.

Result of this task: Audio from the selected source plays in your headset.

9.4.2 Push-to-signal

In call taker mode, the PTS button on a handset/headset can be configured to signal PTT or PTM mode. You can configure each jack box independently for either mode.

Menu: **Tools > PAC II Settings > Push-To-Signal**

9.4.2.1 PTT mode

When a jack box is configured for PTT, the handset/headset is muted by default. The action of pressing the PTS button while on a call, activates the handset/headset microphone. The microphone remains active while the PTS button is depressed/enabled or until the active call is released or placed on hold. If the PTS button is activated before a call is answered, the PTS button status is ignored.

The PTT status is indicated on the status bar of the position.

9.4.2.2 PTM mode

When a jack box is configured for PTM, the action of pressing the PTS button while active on a call (microphone is live) mutes the microphone of the handset/headset locally in the PAC II. The mute condition persists until the PTS button is released. If the PTS button is activated before a call is answered, the PTS button status is ignored.

The PTM status is indicated on the status bar of the position.

Selecting PTS mode

More info: If your account is granted the privilege, the PTS mode can be modified from within Guardian. The setting is applied to the position and you need log out and back into Guardian to activate the change.

Step 1

On the **Tools** menu, click **PAC II Settings**.

Step 2

Select the **Push-To-Signal** tab.

Step outcome: The **PAC II Settings** window is displayed.

Example:

The screenshot shows the 'PAC II Settings' window with the 'Push-To-Signal' tab selected. The window has two tabs: 'Auxiliary' and 'Push-To-Signal'. Under 'Push-To-Signal', there are two sections: 'Jack box 1' and 'Jack box 2'. Each section contains a 'PTS mode' dropdown menu, both currently set to 'Ignore'. At the bottom of the window, there is a 'Note: These changes will only take effect on the next login.' and two buttons: 'Save' and 'Cancel'.

Step 3

In the **PTS mode** list, click the mode for jack box 1.

More info: To disable the PTS button, select **Ignore**.

Step 4

In the **PTS mode** list, click the mode for jack box 2.

More info: To disable the PTS button, select **Ignore**.

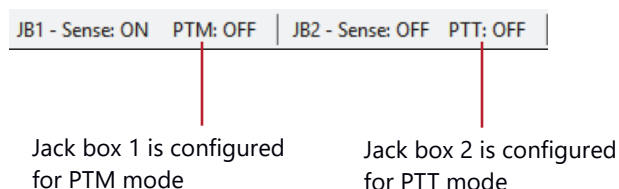
Step 5

Click **Save**.

Result of this task: The changes are saved to the position's configuration file. You need to log out and log back in for the changes to take effect.

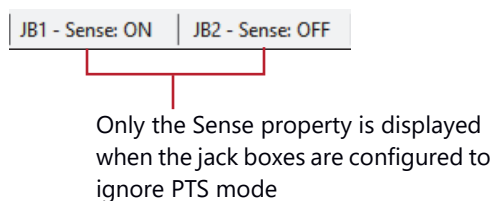
The status bar in Guardian displays the mode of each jack box as shown in Figure 9.4.

Figure 9.4 Status bar for jack boxes configured with different PTS modes



If a jack box is configured to ignore the PTS mode, only the sense property for the jack box id displayed as shown in Figure 9.5.

Figure 9.5 Status bar when jack boxes are configured to ignore PTS mode



9.5 Personal contacts

The Guardian system provides you with the ability to add contacts which are connected to your user account. This ensures that these contacts are available whenever you log in to Guardian regardless of the position you are logged in at. It also ensures that these contacts are only available to your user account - no one else on the system can view your personal contacts.

Menu: Tools > Personal Contacts

Once you create personal contacts, they are combined with the public contacts, which are managed by your administrator, in the **Contacts** window providing you with one merged location for all contacts.

9.5.1 Alternate phone number

An alternate phone number can be configured for the contact. The alternate phone number is dialed automatically if a failure occurs when the primary phone number is dialed or if a busy notification is received from the primary phone number.

Info

Alternate phone numbers are dialed when a busy notification is received for SIP, SS7, and ISDN circuits only and may not apply to the phone lines at your PSAP. Verify with your administrator.

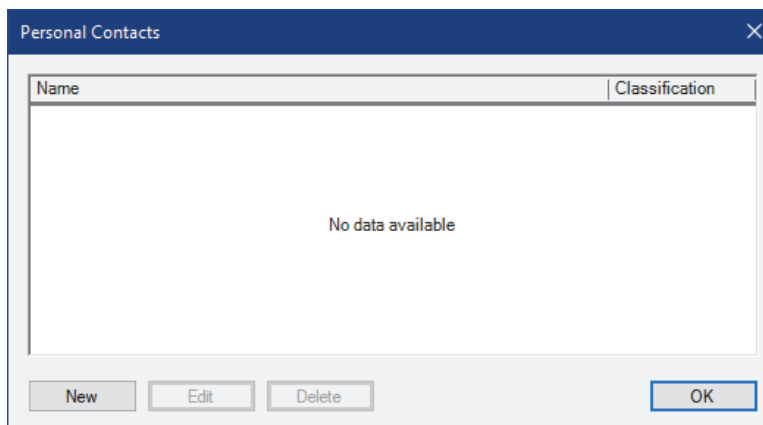
Create personal contact

Step 1

On the **Tools** menu, click **Personal Contacts**.

Step outcome: The **Personal Contacts** window is displayed.

Example:



Step 2

On the **Personal Contacts** window, click **New**.

Step outcome: The **Add Contact** dialog box is displayed.

Example:

The screenshot shows the 'Add Contact' dialog box. It has a title bar with 'Add Contact' and a close button. The dialog is divided into two main sections: 'General' and 'Custom fields'. The 'General' section contains fields for 'Name:' and 'Classification:', both with red exclamation mark icons indicating they are required. Below these is a checkbox for 'Conference'. A section titled 'Phone number/URI details' contains a 'Phone number/URI:' field with a red exclamation mark icon. At the bottom, there are two sub-tabs: 'Incoming' (selected) and 'Alternate'. Under 'Incoming', there is a checked checkbox for 'Reuse incoming line' and four text input fields labeled 'Dial pattern 1:', 'Dial pattern 2:', 'Dial pattern 3:', and 'Dial pattern 4:'. At the bottom right are 'OK' and 'Cancel' buttons.

Step 3

Enter a **Name** for the contact.

More info: This is how the contact appears in the **Contacts** window.

Step 4

Enter a **Classification** for the contact.

More info: This is the classification the contact is filtered under in the **Contacts** window.

Step 5

Enter the **Phone number** of the contact.

More info: Verify with your administrator whether phone numbers need to be entered as 7- or 10-digit numbers, if "1" is required for long distance, or if any other phone codes need to be provided. You can use dashes to make it easier to read the number.

Step 6

Clear the **Reuse Incoming Line** check box.

More info: Leave checked only if dial patterns are supplied.

- Step 7** Ignore the dial pattern fields unless advised differently by your administrator.
- Step 8** Click the **Alternate** tab.
- a** Enter an alternate phone number if applicable.
 - b** Select **Disable alternate number on busy** if you do not want to dial the alternate number automatically when a busy notification is received.
- Step 9** Click **OK**.
- Step 10** Click **OK** on the **Personal Contacts** window.

Edit personal contact

- Step 1** On the **Tools** menu, click **Personal Contacts**.
Step outcome: The **Personal Contacts** window is displayed.
- Step 2** On the **Personal Contacts** window, select the entry to edit, and click Edit.
Step outcome: The **Edit Contact** dialog box is displayed.
- Step 3** Modify the data in the fields as required.
- Step 4** Click **OK**.
- Step 5** Click **OK** on the **Personal Contacts** window.

Delete personal contact

- Step 1** On the **Tools** menu, click **Personal Contacts**.
Step outcome: The **Personal Contacts** window is displayed.
- Step 2** On the **Personal Contacts** window, select the entry to remove, and click **Delete**.

Result of this task: The contact is removed from the **Personal Contacts** window and will no longer be listed in the **Contacts** window.

9.6 Personal greetings

The Personal Greetings feature eliminates the repetitive nature of answering calls and stating the same greeting. Instead, if your PSAP permits, you can record your standard greeting and configure it to play automatically when you answer an incoming call. You can create separate greetings for emergency and administrative calls. You can create multiple greetings for both emergency and administrative calls by specifying the call type by DNIS as explained in *Assign greeting to call type* on page 229.

When a call is answered, you and your caller hear the recorded greeting play, and you resume normal call handling tasks after the recorded greeting plays out.

Menu: Tools > Greeting Recorder

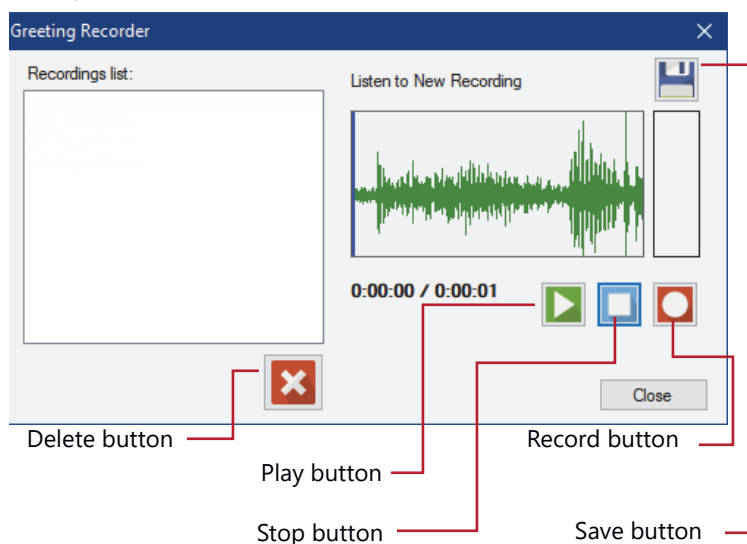
Creating personal greetings is a two-part process: you record the greeting, and then you assign the greeting to a type of call.

Record personal greeting

Step 1 On the **Tools** menu, click **Greeting Recorder**.

Step outcome: The **Greeting Recorder** window is displayed.

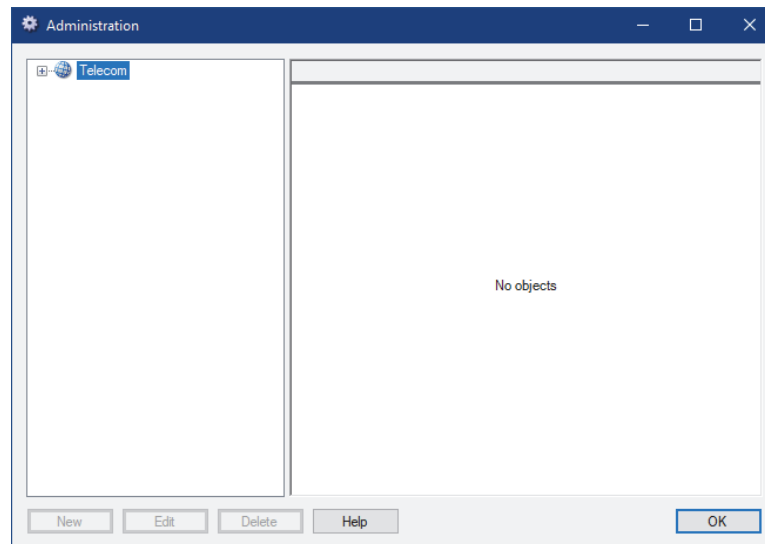
Example:



- Step 2** Click **Record** and speak your greeting into your headset/handset.
- Step 3** Click **Stop** when you are finished.
- Step 4** To listen to your recording, press **Play**.
- Step 5** To save your recording, press **Save**.
a Provide a name for the recording.
b Click **Save**.
- Step 6** To delete the recording, click **Delete**.
- Step 7** Click **Close** when done.

Assign greeting to call type

- Step 1** On the **Tools** menu, click **Administration**.
Step outcome: The **Administration** window is displayed.
Example:



Step 2 In the **Telecom** group, click **Automatic Greetings**, click **Personal Greetings**, and then click **New**.

Step outcome: The **Add Personal Greeting** window is displayed.

Example:

Add Personal Greeting

Changes do not take effect until the next log in of the position. No sound is loaded

Name:

Type: None Emergency AdminEmergency Admin DNIS

Greeting: ...

Record greeting... 00:00:00 / 00:00:00 OK Cancel

Step 3 Enter a **Name** for the greeting.

Step 4 Select the **Type** of call this greeting is assigned.

- If **DNIS** is selected, specify the number (DNIS) used to route this call to a position. Confirm the correct values with your administrator.

Step 5 Click **Browse**.

Step 6 Select on the WAV file to use as the greeting.

Step 7 Click **Open**.

Step 8 Click **OK**.

Note: Your audio files are not deleted from the position that was used to make the recording. You have to manually delete the file from the position.

More info: You can repeat steps 1 through 7 to add another recorded greeting for a different call type or DNIS.

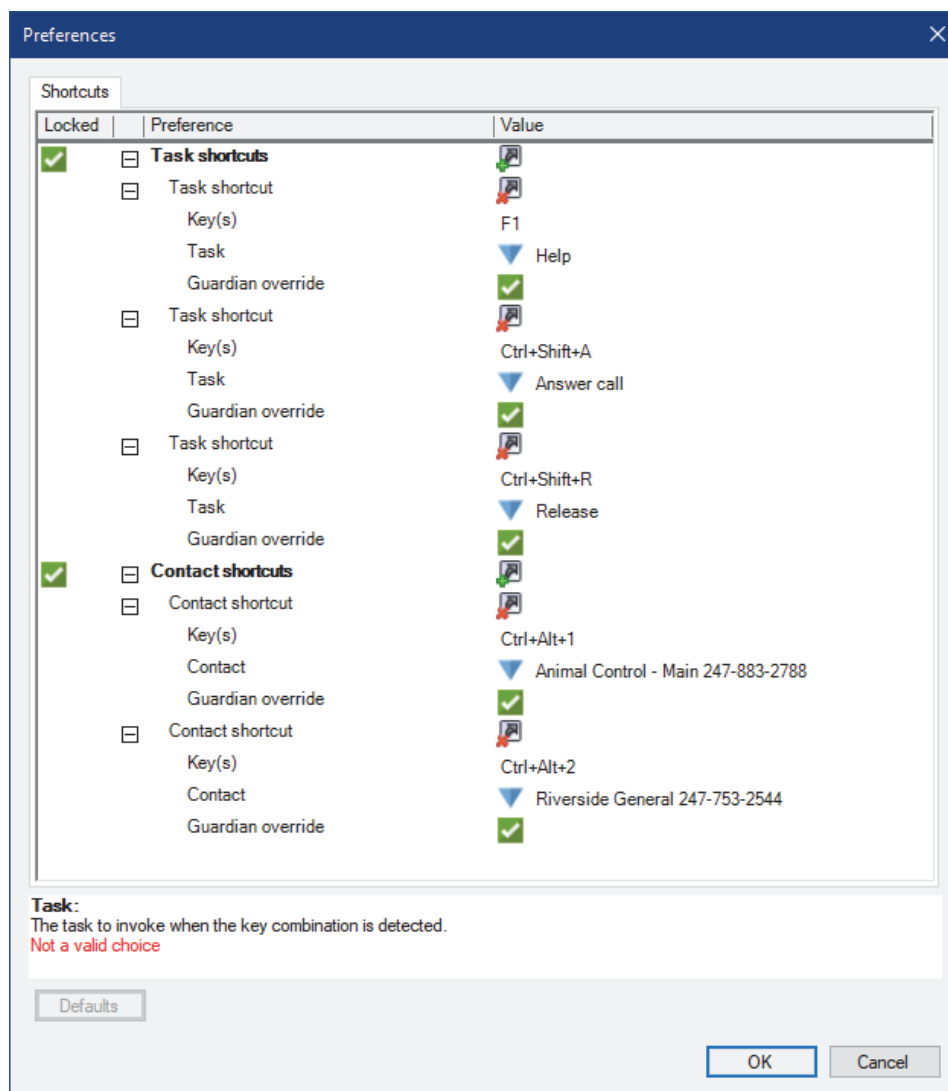
9.7 Preferences

If your system has been configured with keyboard shortcuts for tasks and/or contacts, you can view the shortcut assignments from the **Preferences** window. Your role may also be assigned the ability to create additional shortcuts or modify existing shortcuts. To create new shortcuts, refer to the *Guardian Administration Guide*.

Menu: **Tools > Preferences**

When you launch preferences from the **Tools** menu, the **Preferences** window is displayed as shown in Figure 9.6.

Figure 9.6 Preferences window



If the locked option is checked, your role does not have the privilege to modify the shortcuts.

The shortcuts can be created for application actions such as answer, release, mute, or for contact shortcuts so that you can place a call to a contact using a keyboard shortcut.

9.8 Share IRR

The Share IRR feature allows you to share an Instant Recall Recorder (IRR) recording with one or more other users within your tenant. From the **Share IRR** window, you can share phone call recordings as well as radio recordings if your position is equipped with radio integration.

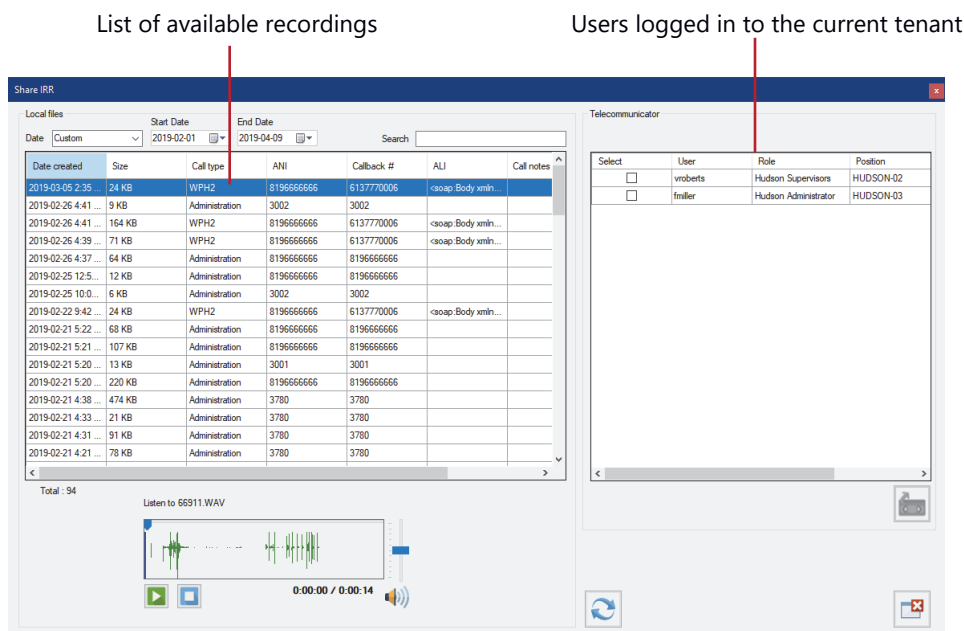
Your administrator may have granted your role the ability to send recordings only, receive recordings only, or both.

9.8.1 Send recordings

You can send any recording available on your position to any other user(s) currently logged in to Guardian within your tenant. You can access the **Share IRR** window from the **View** menu.

Menu: View > Share IRR

Figure 9.7 Share IRR window



The following button commands are available from the **Share IRR** window.

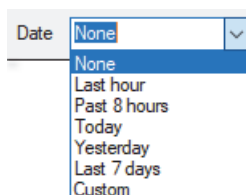
	Select an entry and click Play to hear the recording play to your default listening device; i.e., headset/handset.
	Click Pause to stop playing the selected recording while maintaining your position in the playback so you can resume listening from the same point in the recording.
	Click Stop to stop playing the selected recording and reset playback to the beginning of the recording.
	Click Refresh to update the list of recordings and the list of users logged in to Guardian within your tenant group.
	Click Share to send the recording to the selected user(s).
	Click Close to close the Share IRR window.

9.8.1.1 Search recordings

From the **Share IRR** window, you can search for a recording from a specific date, search for a recording based on a callback number, or search for a recording using any term within the **ALI**, **Call notes**, or **TDD conversation** columns if available.

The **Date** list, shown in Figure 9.8, allows you to search for recordings:

- From the last hour
- From the past 8 hours
- From today
- From yesterday
- From the last 7 days
- From a specific date/date range by selecting a start and end date

Figure 9.8 Date search options

You can execute a custom search by entering the criteria in the **Search** box. The list of recordings are filtered with each character you type in the box. The system searches the following fields for the criteria entered:

- Call type
- ANI
- Callback #
- ALI display
- Call notes
- TDD conversations

To ensure you share the correct recording, you can listen to a recording prior to sharing it. If you are listening to a recording and an incoming call arrives, the ring tone for the call plays in your headset/handset simultaneously with the recording playback. When you answer the call, the recording stops playing automatically. You can also choose to listen to a recording while you are on an active call - the audio from the recording is heard only on your headset/handset, even if you have another telecommunicator conferenced to your call.

You can view the raw ALI for the call, any call notes entered during the call, or any TDD conversation recorded for the call, as shown in Figure 9.9, when you click on the column entry.

Figure 9.9 View raw ALI from listing of recordings

Date created	ANI	Callback #	ALI display	Call Notes	TDD Conversation
2019-02-01 2...	2475116002	2478293652	<soap:Body ...		
2019-02-01 2...	2475116001	2478649612	<soap:Body ...		
2019-02-01 2...	2475116001	2478362544	<soap:Body ...		
2019-02-01 1...		!			
2019-02-01 1...	2475116003	2478659021	<soap:Body ...		

X


```

<soap:Body
  xmlns:soap="http://schemas.xmlsoap.org/soap/envelope/">
  <VerifyCustomerInfoResponse
    xmlns="VIXXI_WEBSERVICE/PSAPLookup">
    <VerifyCustomerInfoResult><CusPhone>2478649612</CusPhone>
    <CusPhoneExist>true</CusPhoneExist><CusName>
    Bowl</CusName><Street_Val /><City_Val /><State_Val /><Zip_Val
    /><PSAPPhone>2479111003</PSAPPhone><PSAPName>
    HUDSONVILLE</PSAPName><PSAPID>2475116001</PSAPID>
    <ESN>109<ESN /><LATA_ID>LAT-3</LATA_ID><RateCenter>
    RC3</RateCenter><WireCenter>WC3</WireCenter><Street_Name
  
```

View the content of the ALI, call notes, or TDD conversation when you click on the field

9.8.1.2 Recipient list

The **Telecommunicator** panel in the **Share IRR** window lists all the users who are currently logged in to Guardian within your tenant group. The users are listed in the order they logged in. For each user, you can view their current logged in role, the position they are logged into, and their status.

Recordings can only be sent to users with the required privilege. Users who do not have the privilege to receive recordings cannot be selected in the **Telecommunicator** panel. The list of users displayed in the panel is current as of the time you launch **Share IRR** - you can update the list of users and recordings via the **Refresh** button.

You can send a recording to multiple users by selecting more than one user in the **Telecommunicator** panel.

Launch Share IRR

- On the **View** menu, click **Share IRR**.
- Click the **Share IRR** icon in the layout.

Step outcome: The **Share IRR** window is displayed and lists all the recordings available on the current position and all the users currently logged in to Guardian.

Search recordings

- Step 1** Launch **Share IRR**.
Step outcome: The **Share IRR** window is displayed.

Search by specific date/date range

- Step 2** In the **Date** box, click **Custom**.
Step 3 In the **Start Date** list, click the beginning date to search.
Step 4 In the **End Date** list, click the last date to search.

Step outcome: All IRR recordings on the current position between the specified dates, inclusively, are listed.

Example:

Local files							
Date	Custom	Start Date	2019-02-01	End Date	2019-02-04	Search	
Date created	Size	Call type	ANI	Callback #	ALI	Call notes	TDD conversation
2019-02-01 2:36 PM	7 KB	WPH2	2475116002	2478293652	<soap:Body ...		
2019-02-01 2:35 PM	8 KB	WPH2	2475116001	2478649612	<soap:Body ...		
2019-02-01 2:01 PM	56 KB	WPH2	2475116001	2478362544	<soap:Body ...		
2019-02-01 12:25 ...	29 KB	02/01/19 12...		!			
2019-02-01 12:25 ...	12 KB	WPH2	2475116003	2478659021	<soap:Body ...		

Search by phone number

- Step 5** In the **Search** box, type the phone number.
Step outcome: All IRR recordings that contain the phone number in the **ANI**, **Callback #**, **ALI**, **Call notes**, or **TDD conversation** fields are listed.

Example:

Local Files							
Date	None	Start date	2019-02-01	End Date	2019-02-01	Search 247864	
Date created	Size	Call type	ANI	Callback #	ALI display	Call Notes	TDD Conversation
2019-02-01 2:35 PM	8 KB	WPH2	2475116001	2478649612	<soap:Body ...		

Search by keyword/text

Step 6 In the **Search** box, type the keyword/text.

Step outcome: All IRR recordings that contain the keyword/text in the **ANI**, **Callback #**, **ALI**, **Call notes**, or **TDD conversation** fields are listed.

Example:

Local files							
Date	None	Start Date	2019-02-01	End Date	2019-03-05	Search	laur
Date created	Size	Call type	ANI	Callback #	ALI	Call notes	TDD conversation
2019-02-19 9:41 AM	597 KB	RESD	2478641244	2478641244	<soap.Body ...		2/19/2019 9:37:...

View all recordings

Step 7 In the **Date** box, click **None**.

Step outcome: All available recordings on the current position are listed.

More info: You can also combine a date and phone number/keyword/text search.

Share recording

Step 1 Launch **Share IRR**.

Step outcome: The **Share IRR** window is displayed.

Step 2 In the **Local files** panel, find and select the recording to share.

Step 3 In the **Telecommunicator** panel, select the recipient(s).

- If needed, click **Refresh** to update the **Telecommunicators** list.

More info: The **Select** option for users who do not have the privilege to receive recordings is disabled.

Example:

Select option is disabled as user does not have privilege to receive recordings

Select	User	Role	Position	Status
☐	pmartin	Hudso...	HUDSON-03	Idle
☐	vroberts	Hudso...	HUDSON-02	Idle

Step 4 Click **Share**.

Result of this task: The recording is sent to the selected recipient(s). The results of the transfer are displayed at the bottom of the window.

Example:



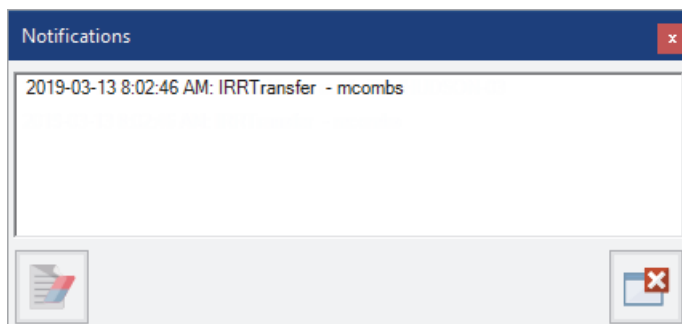
9.8.2 Receive recordings

When a recording is shared by another user, the **Notifications** button, if available in your layout, changes color to indicate that you have a new notification as shown in Figure 9.10. When you click on the button, the **Notifications** window is launched and displays an entry for the shared recording.

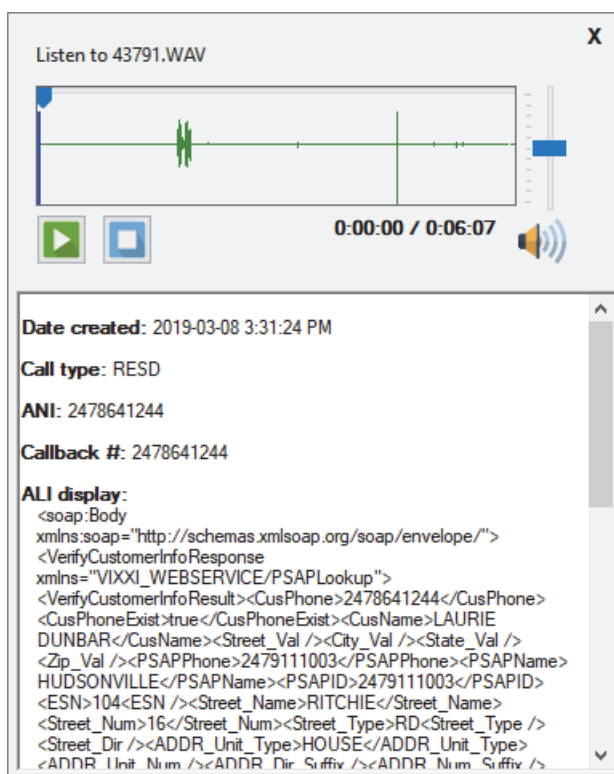
Figure 9.10 Notifications button indicates new message via color



If your layout does not include the **Notifications** button but includes the **Notifications** window, an entry for the shared recording is added to the window when a recording is received as shown in Figure 9.11.

Figure 9.11 New entry for shared recording is displayed in Notifications window

When you click on the message in the **Notifications** window, you are presented with the ability to listen to the recording on your default device and view the ALI and any call notes or TDD conversation text as shown in Figure 9.12.

Figure 9.12 Click on the IRR transfer message to view recording information

If you are listening to a recording and an incoming call arrives, the ring tone for the call plays in your headset/handset simultaneously with the recording playback. When you answer the call, the recording stops playing automatically. The recording dialog box also closes automatically.

You can also choose to listen to a recording while you are on an active call - the audio from the recording is heard only on your headset/handset, even if you have another telecommunicator conferenced to your call.

Listen to received recording

- Step 1** Launch the **Notifications** window if not docked in your layout:
- Click the **Notifications** button.
 - On the **View** menu, click **Notifications**.
- Step 2** In the **Notifications** window, click the "IRR transfer" entry.
Step outcome: The recording dialog box is displayed.
- Step 3** Click **Play**.
Step outcome: The recording plays to your default device.
- Step 4** If included, scroll down in the dialog box to view the ALI, call notes, or TDD conversation text.
- Step 5** Close the recording dialog box:
- Click the **Close** button in the dialog box.
 - Click anywhere outside the dialog box.

9.9 Volume

You can control the volume of your headset, microphone and speakers independently of each other. The volume control affects the wave setting of your playback device.

Menu: View > Volume > Volume

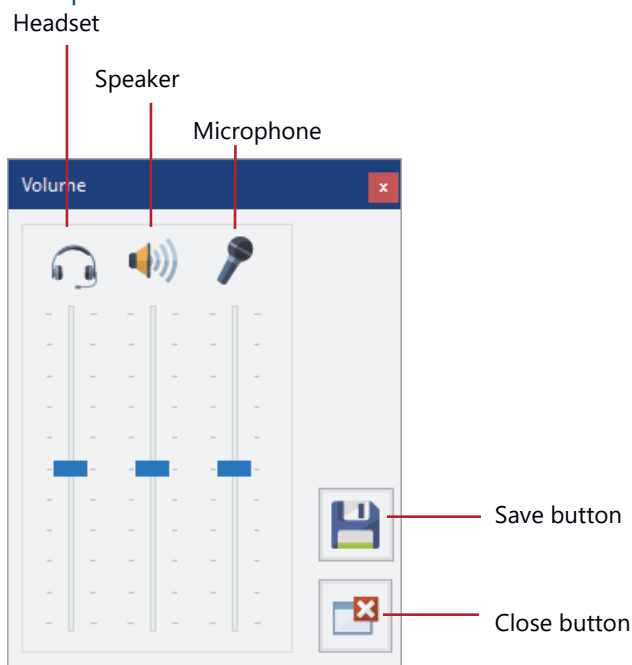
You may have more sliders visible in the **Volume** window due to the privileges assigned to your role. This section describes the sliders that are visible to all users regardless of privilege assignment.

Adjust volume

Step 1 Drag the sliders for each device to set the volume.

More info: The sliders control, from left to right, your headset volume, your external speaker volume, and your microphone volume.

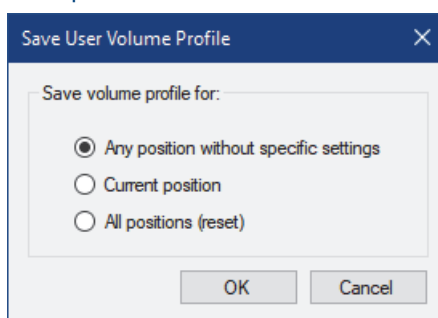
Example:



Step 2 Click **Save**.

Step outcome: The **Save User Volume Profile** dialog box is displayed.

Example:



Step 3 Select for which position to save the settings.

More info: Refer to Table 8.1 for the save options available in the **Save User Volume Profile** dialog box.

Step 4 Click **OK**.

Step 5 Click **Close** to close the **Volume** window.

Note: While the **Volume** window is open, changes applied to certain external volume controls may not be reflected.

Table 9.1 Save user volume profile settings

Setting	Description
Any position without specific settings	Saves this profile for your user account for any position within your tenant group that currently does not have a profile. If you have previously saved a profile for a position, it is not overwritten with this profile.
Current position	Saves this profile for your user account for the current Guardian position only.
All positions (reset)	Saves this profile for your user account for all positions within your tenant group and overwrites any existing profiles saved for your user account.

9.9.1 Volume meter

The Volume Meter feature gages the strength of audio to and from your position.

Menu: **View > Volume > Volume Meter**

You can use this tool to measure:

- Audio levels you send to the caller through the microphone.
- Audio levels you receive from a caller through the headset or speaker.

Using the **Volume Meter** and the **Volume** windows in conjunction, you can set input and output volume levels to an optimal range.

Adjust microphone volume using volume meter

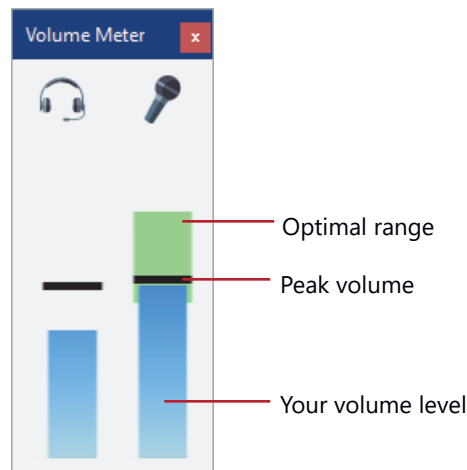
Step 1 Launch **Volume Meter**.

Step outcome: The **Volume Meter** window is displayed.

Step 2 Turn off any distracting sound sources to ensure you capture just your own voice in the microphone.

Step 3 Speak into the microphone of your headset/handset as if you are answering a call.

Example:



Step 4 Continue to speak into the microphone. Perform one of the following:

- If your levels are within the optimal range, close the window.
- If your levels are outside the optimal range, go to the next step.

Step 5 Open the **Volume** window.

More info: Refer to Figure 8.1.

Step 6 Monitor the **Volume Meter** while speaking into the microphone.

Step 7 From the **Volume** window, drag the slider for the microphone volume up or down until the levels are within the optimal range.

Step 8 Click **Save**.

Step outcome: The **Save User Volume Profile** dialog box is displayed.

Step 9 Select for which device/user account to save the settings.

More info: Refer to Table 8.1 for the save options available in the **Save User Volume Profile** dialog box.

Step 10 Click **OK**.

Step 11 Click **Close** to close the Volume window.

Adjust headset/speaker volume using volume meter

- Step 1** Ensure you have an active call.
- Step 2** Launch **Volume Meter**.
Step outcome: The **Volume Meter** window is displayed.
- Step 3** Listen to the caller using a headset or speaker.
More info: The meter indicates the speaker/headset volume level (the blue zone). The black band indicates the peak volume.
- Step 4** Continue to listen. Perform one of the following:
- If you are able to hear the caller, close the window.
 - If you have trouble hearing the caller, go to the next step.
- Step 5** Open the **Volume** window.
More info: Refer to Figure 8.1.
- Step 6** While listening to the caller; view the volume meter.
- Step 7** From the **Volume** window, drag the slider for the headset volume up or down until you are satisfied with the volume.
- Step 8** Click **Save**.
Step outcome: The **Save User Volume Profile** dialog box is displayed.
- Step 9** Select for which device/user account to save the settings.
More info: Refer to Table 8.1 for the save options available in the **Save User Volume Profile** dialog box.
- Step 10** Click **OK**.
- Step 11** Click **Close** to close the Volume window.

10 Guardian alarm notification

This chapter describes the various alarm notifications and alarm tools available with the Guardian answering position that notify you when predefined conditions occur. Alarms may not necessarily dictate any immediate action and may only be informational. However, it is best to consult with your administrator if any, and what, actions are required when alarms are displayed on your position.

10.1 Server switchover

A redundant Guardian system is equipped with a primary and secondary server. Your position connects to the primary server under normal operating conditions. The system automatically switches your position to the secondary server if a service-affecting failure is detected with the primary server.

A combination of the status bar and the ticker message bar notifies you of the status of the servers and identifies which server your position is currently connected to.

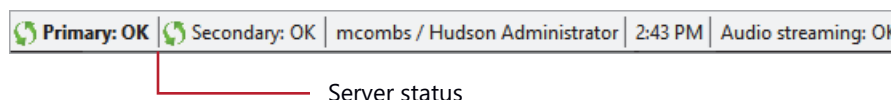
To identify if your Guardian system has two servers, you can view the status bar on your position.

Note: Your system administrator may also have configured the system to play an audible alarm; explained in *Audible alarms* on page 253.

10.1.1 Both servers operational

If your system has two servers and they are both operational, "Primary" and "Secondary" are listed on the status bar and their status is set to "OK" as shown in Figure 10.1. The server displayed in bold indicates the server to which your position is currently connected.

Figure 10.1 Server status



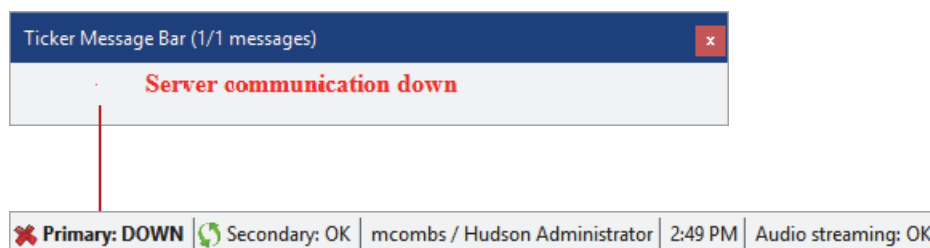
Redundant system with both servers available and operational; "Primary" in bold notifies you that your position is connected to the primary server

10.1.2 Communication failure on primary server

If a communication error occurs with the primary server but call handling capability is not effected, the status bar changes the status of the primary server and a ticker message notifies you of the problem as shown in Figure 10.2. In this scenario your position is still connected to the primary server, as indicated by the primary server status displayed in bold, even though system communications are handled by the secondary server.

Caution Notify your administrator immediately if this scenario occurs.

Figure 10.2 Ticker message when server status changes

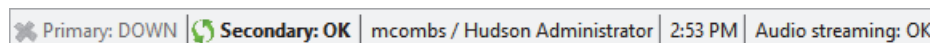


The ticker message bar scrolls a message stating "Server communication down" and the status bar changes to show you that the primary server is "Down." Primary remains bold because call handling is still being handled by the primary side.

10.1.3 Primary server failure

If the primary server experiences an error resulting in a failure of call handling capability, the system automatically switches your position to the secondary server. The status bar indicates the switch by displaying the secondary server in bold and changing the status of the primary server as shown in Figure 10.3.

Figure 10.3 Connected to secondary server



Redundant system when the primary server experiences a problem with call handling switches your position to the secondary server.

If you were on an active call at the time of the switchover, the call is reconnected to your position after the position has switched to the secondary server. If the system cannot reconnect the call to your position, the call is presented as abandoned and you can attempt a call back.

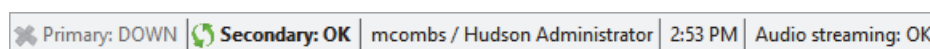
If your system is configured with audible alarms, a short alarm tone plays when the switchover occurs.

10.1.4 Deliberate switchover

There may be instances when your administrator advises you that your position is being purposefully forced to switch to the secondary server. The system automatically switches your position to the secondary server when it detects you are not on an active call.

The status bar notifies you that your position is connected to the secondary server while the primary server did not experience a failure as shown in Figure 10.4.

Figure 10.4 Connected to secondary server



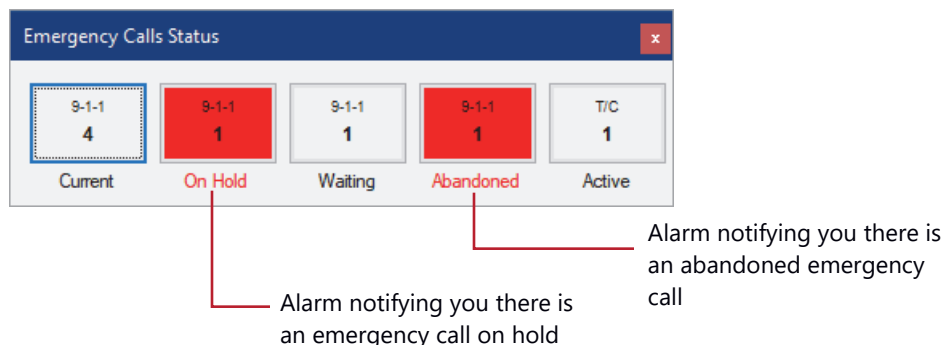
The primary server is dimmed to notify you that you are connected to the secondary server even if the primary server is operational.

10.2 Tenant status alarms

Your administrator may have alarms configured when calls are in a certain state for a specified amount of time or when a certain number of calls are present. These alarms are visible on the two tenant status bars: **Emergency Calls Status** and **Admin Calls Status**.

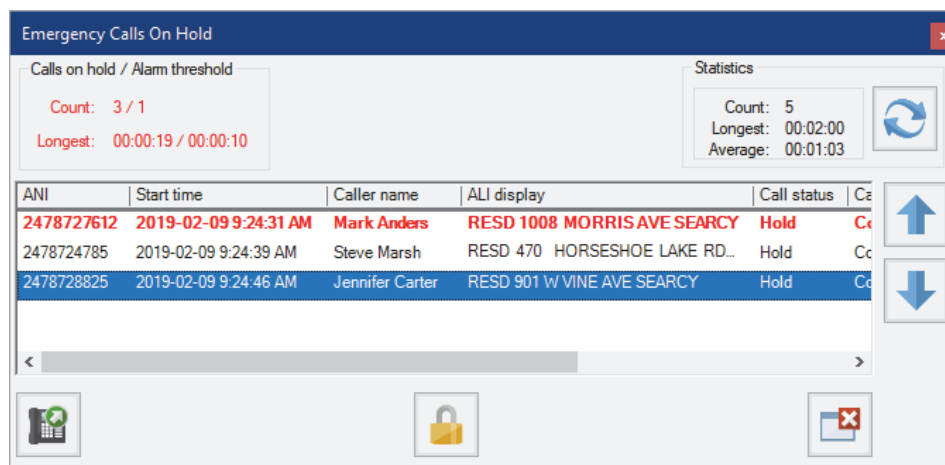
The button representing the state that triggered the alarm changes color (default red), notifying you that an alarm is triggered as shown in Figure 10.5. Your administrator may also have configured a message to scroll across the ticker message bar when the alarm is triggered.

Figure 10.5 Emergency Calls Status bar



When you click on the button displaying an alarm state, the appropriate tenant status window is displayed. By default, the call that triggered the alarm is displayed in red (shown in Figure 10.6), although your administrator may have customized the color properties for your PSAP. Additionally, if more than one call has triggered the same alarm condition, the oldest call can be configured to display in bold.

Figure 10.6 Emergency Calls On Hold window



After receiving the alarm, handle the call as you would normally. The alarm is cleared when the call has been addressed.

10.3 Ticker message alarms

Your administrator may have configured the ticker message bar to display alarm messages when your position experiences problem with audio quality or connecting to a server.

If enabled, the following system messages may scroll across the ticker bar:

Note: Always notify your administrator whenever any of these messages are displayed in your ticker message bar.

1 Switch communication down

- This message is primarily seen on simplex system; i.e., non-redundant systems. The message indicates that there is a call handling and processing failure with the server which is service affecting. Notify your administrator/support staff immediately.
- You may see this message for a brief moment on a redundant system before the system automatically switches you to the secondary server.

2 Server communication down

- If you see this message, it indicates that there is a communication problem with the server. If you are on an active call, your current call is not affected. However, no new incoming calls are sent to any position experiencing this failure. Notify your administrator/ support staff.

3 Voice communication down

- This message indicates that the position attempted to register with a proxy and failed. Notify your administrator/ support staff.

4 Audio quality alarm

- If you see this message in the ticker message bar, it indicates that your position has audio but that the audio quality is less than optimal; for example, the audio may be choppy or sound robotized. Notify your administrator/support staff for assistance in rectifying the audio quality issue.

10.4 Notifications

The **Notifications** window displays a message when certain events occur within the current tenant group. A message is added when a position has been forced to busy out or log out because a telecommunicator did not answer an ACD ring group call within the specified RNA (ring no answer) time. The window also displays a message when a transfer-to-IVR action fails due to the IVR server being unavailable. Messages notifying you that the notification feature has stopped communicating with the server are also displayed.

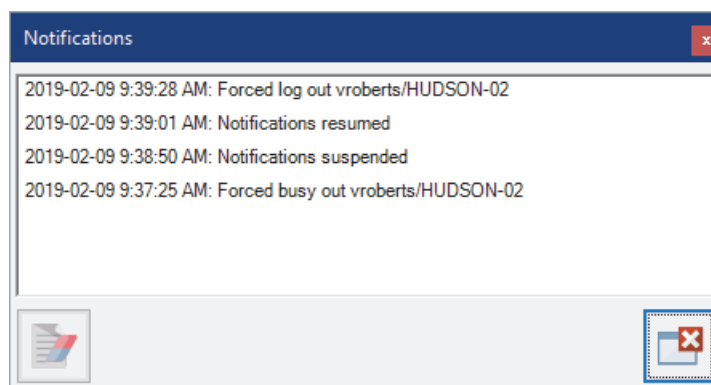
You role must be assigned one of the notification privileges in order to access the **Notifications** window. Your administrator may have chosen to display any combination of the available notifications for your role.

If **Notifications** is not visible in your layout, launch it from the **View** menu as a floating window.

Menu: **View > Notifications**

Each notification includes the date and time of the event and a brief message describing the event. Notifications related to forced busy out and forced log out events also include the user account and position affected as shown in Figure 10.7.

Figure 10.7 Notifications window



If a **Notifications** button is included in your layout, the button changes color (default green) when new notifications are present as shown in Figure 9.8.

Figure 10.8 Notifications button indicating new notifications (default color)



The original color of the **Notifications** button is restored after the **Notifications** window is closed; notification messages do not need to be erased, they are removed when you log out.

Your administrator may have configured the system to play an audible alarm when a new notification is received. The audible alarm may be configured to play in your headset or

your position's external speakers as specified by your administrator. Your administrator may have chosen to play an alarm for all notification messages or only select events.

View notifications

More info: When the **Notifications** window is docked in your layout, new notifications are automatically displayed in the window. When the **Notifications** button is included in your layout, the color of the button changes to indicate new notifications are available.

Step Click **Notifications** in the layout.

Result of this task: The **Notifications** window is launched and displays new notifications.

Clear notifications

Step 1 Click **Notifications** if the window is not included in your layout.

Step 2 Click **Clear List**.

Result of this task: The list of current notifications is erased from the **Notifications** window.

10.5 Audible alarms

Your administrator may have configured audible alarms to play when certain conditions occur with your position. This alarm is a short tone that may play in your headset/handset or on the position's external speakers depending on configuration settings.

If your role is configured for automatic call answer, the zip tone played when a call is answered is typically different from the tone playing to notify you of an alarm condition.

The following conditions can trigger an alarm notification:

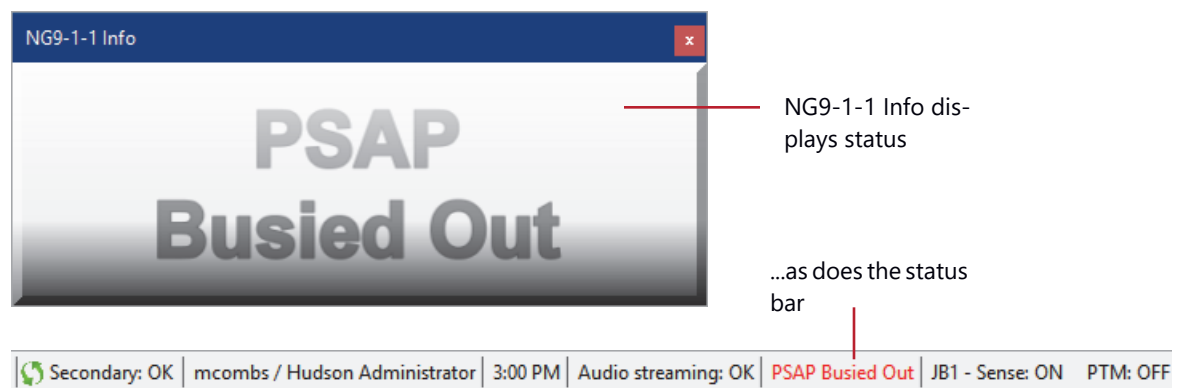
- Your position switches from one server to the other.
- Your position is disconnected from the primary or secondary server.
- Your position experiences audio issues such as choppiness or distortion.
- Your position experiences network problems.
- A position has been forced busied out.

- A position has been forced logged out.
- A system IVR is not available.
- You attempt to make busy when you are the last member of a ring group.

10.6 PSAP make busy warning

The system supports placing your entire PSAP in busied out mode; for example, if you need to evacuate the premises. When your administrator places the PSAP in busied out mode, the **NG9-1-1 Info** window and status bar display the status as shown in Figure 10.9.

Figure 10.9 NG9-1-1 Info window and status bar indicate PSAP busied out status

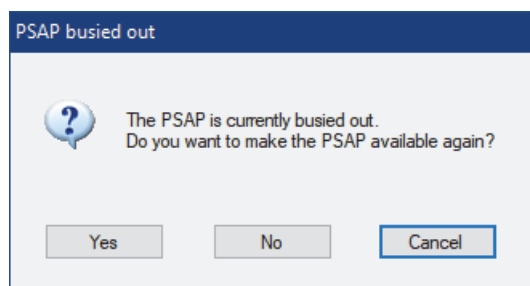


Note: While the PSAP is in busied out mode, no calls are routed to any answering positions belonging to the PSAP.

10.6.1 Log in when busied out

If your role does not have the ability to deactivate PSAP make busy mode, you may log in to a Guardian position but the PSAP remains busied out.

If your role can deactivate PSAP make busy mode, you are presented with a warning requesting confirmation to continue when you log in as shown in Figure 10.10.

Figure 10.10 PSAP busied out warning

Remove PSAP from busied out state

More info: After entering your credentials, the PSAP busied out warning is presented.

Step Click **Yes**.

Result of this task: You are logged in to Guardian and all members of the PSAP are removed from the busied out state.

Log in and remain in PSAP busied out mode

More info: After entering your credentials, the PSAP busied out warning is presented.

Step Click **No**.

Result of this task: You are logged in to Guardian but you and all other members of the PSAP remain in the busied out state. The **NG9-1-1 Info** window displays the state.

Cancel log in attempt

More info: After entering your credentials, the PSAP busied out warning is presented.

Step Click **Cancel**.

10.7 ALI status

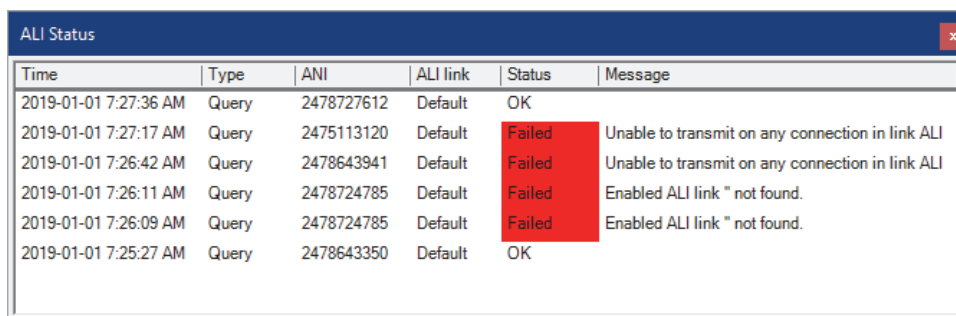
The **ALI Status** window displays historical status information relating to the ALI queries executed from your position. The information provided in **ALI Status** is relative to your current position and contains information since you last logged in. The statistics are cleared once you log out of Guardian.

If a query failed, the message column provides you with information related to the type of failure experienced.

If **ALI Status** is not visible in your layout, launch it from the **View** menu bar as a floating window.

Menu: View > ALI Status

Figure 10.11 ALI Status window



The screenshot shows a window titled "ALI Status" with a close button in the top right corner. Inside the window is a table with the following data:

Time	Type	ANI	ALI link	Status	Message
2019-01-01 7:27:36 AM	Query	2478727612	Default	OK	
2019-01-01 7:27:17 AM	Query	2475113120	Default	Failed	Unable to transmit on any connection in link ALI
2019-01-01 7:26:42 AM	Query	2478643941	Default	Failed	Unable to transmit on any connection in link ALI
2019-01-01 7:26:11 AM	Query	2478724785	Default	Failed	Enabled ALI link " not found.
2019-01-01 7:26:09 AM	Query	2478724785	Default	Failed	Enabled ALI link " not found.
2019-01-01 7:25:27 AM	Query	2478643350	Default	OK	

A Acronyms

Acronyms

The following table lists the common acronyms found in this document.

Table A.1

Acronym	Expanded form
ACD	Automatic Call Distribution
ACK	ACKnowledgement
AGC	Audio Gain Control
ALI	Automatic Location Identification
ANI	Automatic Number Identification
ASCII	American Standard Code for Information Interchange
CAD	Computer Aided Dispatch
CDR	Call Detail Record
CO	Central Office
COS	Class Of Service
DA	Direct Access
DID	Direct Inward Dialing
DNIS	Dialed Number Identification Service
DTMF	Dual Tone Multi Frequency
E9-1-1	Enhanced 9-1-1

Table A.1

Acronym	Expanded form
ECRF	Emergency Call Routing Function
EMS	Emergency Medical Service
ESD	Electrostatic Discharge
ESInet	Emergency Service IP network
ESN	Emergency Service Number
ESP	Emergency Services Platform
ESRN	Emergency Services Routing Number
ESRP	Emergency Services Routing Proxy
ESZ	Emergency Service Zone
FQDN	Fully Qualified Domain Name
FTC	Fixed Transfer Code
GCCI	Guardian Call Control Interface
HELD	HTTP-Enabled Location Delivery Protocol
IEC	International Electrotechnical Commission
IP	Internet Protocol
LAN	Local Area Network
MSRP	Message Session Relay Protocol
NAK	Not Acknowledged
NENA	National Emergency Number Association
NG9-1-1	Next Generation 9-1-1
PAC	Position Audio Control
pANI	pseudo ANI
PBX	Private Branch eXchange
PIDF-LO	Presence Information Data Format - Location Objects
PSAP	Public Safety Answering Point
PTM	Push To Mute
PTS	Push To Signal

Table A.1

Acronym	Expanded form
PTT	Push To Talk
PTZ	Pan Tilt Zoom
RNA	Ring No Answer
RN	Resource Number
QoS	Quality of Service
SIP	Session Initiation Protocol
SMS	Short Message Service
SNMP	Simple Network Management Protocol
SO	Sheriff Office
SOAP	Simple Object Access Protocol
SR	Selective Router
STA	Selective Transfer Agency
T/C	Telecommunicator
TDD	Telecommunication Device for the Deaf
TDM	Time Division Multiplexing
URI	Uniform Resource Identifier
URN	Uniform Resource Number
URL	Uniform Resource Locator
VoIP	Voice over Internet Protocol
XSLT	Xml Stylesheet Language Transformation

TDD abbreviations

B TDD abbreviations

The following are universally used TDD terms, followed on the next page by abbreviations commonly used in text communication:

Table B.1

Abbreviation	Meaning
GA	Go Ahead (your turn to type)
Q or QQ	Question mark
SK	Stop Keying (your party is ready to hang up)
SKSK	Hanging up or hung up
TDD	Telecommunication device for the deaf
TTY	Teletypewriter
XXX	Erasing an error

Table B.2

Abbreviation	Meaning
ABT	about
ACCT	account
AMBL, AMBU	ambulance
AMT	amount
ANS MACH	answering machine
ANS	answer
APPT	appointment
APR	April
APT	apartment
ASAP	as soon as possible
ASST	assistant
ATTN	attention
BAL	balance
BBQ	barbecue
BCUZ	because
BDAY	birthday
BDRM	bedroom
BECUZ	because
BIZ	business
BKGD	background
BLDG	building
BLVD	boulevard
BSMT	basement
BYE	goodbye
C	Celsius
CD	could
CHCK	check

Table B.2

Abbreviation	Meaning
CI	cochlear implant
CIR	circle (address)
CLD	could
CLL	call
CLR	clear
CMTY	community
CO	company
CONT	continue
CONT	continuing
COS, COZ	because
CT	court
CTR	center
CUD	could
CUL	see you later
CUST	customer
CUZ	because
DEC	December
DEPT	department
DOB	date of birth
DR	doctor, drive (address)
E	east
EDUC	education
EMPL	employment
ETC	etcetera
EXT	extension
F	Fahrenheit
FEB	February

Table B.2

Abbreviation	Meaning
FIGS	figures
FONE	telephone
FRI	Friday
FRIG	refrigerator
FT	fort (proper noun)
FWD	forward
FYI	for your information
GOVT	government
HAHA	laughing
HH	hard of hearing
HLD	hold
HLTH	health
HOH	hard of hearing
HOSP	hospital
HR	hour
HVE	have
HWY	highway
ID	identification
ILY	I love you
IM	instant message
IMPT	important
INC	incomplete
INC	incorporated
INFO	information
INS	insurance
INT, INTERP	interpreter
INTL	international

Table B.2

Abbreviation	Meaning
IRD	Inland Revenue Department
JAN, JANU	January
JR	junior
JUL	July
JUN	June
KITC	kitchen
LIVRM	living room
LN	lane (address)
LTR	letter
LTRS	letters
LV	leave
LVG	leaving
MACH	machine
MAR	March
MD	medical doctor
MGR	manager
MIN	minute
MINS	minutes
MISC	miscellaneous
MON, MOND	Monday
MORN	morning
MR	Mister
MRS	Missus
MSG, MSGE	message
MTG	meeting
MYOB	mind your own business
N	north

Table B.2

Abbreviation	Meaning
NATL	national
NBR	number
NE	northeast
NITE	night
NO	number
NOV	November
NOYB	non of your business
NU, NUMR	number
NW	northwest
NXT	next
OCT	October
OFC, OFFC	office
OIC	oh, I see
OK	okay
OPR	operator
OXOXOX	hugs and kisses
PBLM	problem
PC	personal computer
PD	paid
PHNE	phone
PIN	personal identification number
PL	place (address)
PLM	problem
PLS, PLSE	please
PPL	people
PRBLM	problem
PRIV	private

Table B.2

Abbreviation	Meaning
PRO	professional
PROBLY	probably
PROF	professional
PSBLE	possible
PT	physical therapy
PYMT	payment
Q	question mark
R	are
RD	road
RECD	received
RECHD	reached
REFR	referral
REG	regular
REP	representative
RM	room
RPT	repeat
RT	route
RWD	rewind
S	south
SAT	Saturday
SD	should
SE	southeast
SERV	service
SEPT	September
SHD, SHUD	should
SR	senior
SRVC	service

Table B.2

Abbreviation	Meaning
ST	Saint (proper noun) or street
STRM	storm
SUN	Sunday
SVC	service
SW	southwest
TBSP	tablespoon
TECH	technical (or) technician
TEL, TELE	telephone
TERP	interpreter
TERR	terrace (address)
THK	thank
THKS	thanks
THO	though
THRU	through
THU, THUR	Thursday
TKS	thanks
TK	thank
TNKS	thanks
TMR, TMRW, TMW	tomorrow
TSP	teaspoon
TTYL	talk to you later
TUE, TUES	Tuesday
TV	television
TY	thank you
UR	you are (or) your
U R	you are
U	you

Table B.2

Abbreviation	Meaning
URS	yours
VOC REHAB	vocational rehab
W	west
WD	would
WED, WEDN	Wednesday
WHT	what
WINZ	work and income
WK	week
WKEND, WKND	weekend
WLCOME	welcome
WLD, WUD	would
XMAS	Christmas
YR	year

Icon glossary

PHONE CONNECTION WINDOW STATUS ICONS

The following icons are displayed at the beginning of each entry in the **Phone Connection** window to indicate the status of the call participant.

Table C.1

Icon	Description
	The audio for the call is directed to the headset/handset. Only active calls display an audio device icon in the status column.
	The audio for the call is directed to the external speakers. Only active calls display an audio device icon in the status column.
	The call is in monitor mode. There is audio but no microphone connectivity.
	A ringing emergency call that has not been answered.
	A ringing administrative call that has not been answered.

Table C.1

Icon	Description
	A ringing text-to-911 that has not been answered.
	An active call entry. There is audio and microphone connectivity.
	An active SIP REFER call entry. There is audio and microphone connectivity.
	The call is monitored by the current position.
	The call party is muted. There is audio but no microphone connectivity.
	The call party is on privacy. There is no audio but there is microphone connectivity.
	The call party is on hold. There is no audio or microphone connectivity. When you place a call entry on hold, the caller on hold hears a beep tone every 10 seconds until the participant is removed from hold.
	The call party is placed on exclusive hold. There is no audio or microphone connectivity for all participants on the call. When you place the call on exclusive hold, the participants on the call hear a beep tone every 10 seconds until the call is removed from exclusive hold. Only the original telecommunicator to place the call on exclusive hold can remove the call from exclusive hold.
	A text-to-911 call on hold.
	The call is in a queue.

Table C.1

Icon	Description
	Call is connecting to a position.
	The call is being dialed.
	A busy signal is being received for the call leg.
	The call party is disconnected.
	Call has been unexpectedly disconnected, the system is reconnecting the call on an alternate route or to the alternate server.
	The status of the call party is undetermined.
	Status is unavailable or unable to connect.

D Release history

This appendix outlines the features introduced with each product release of Guardian.

Release 19.0.1

PSAP MAKE BUSY ENHANCEMENT

The Tenant Make Busy feature has been renamed to the PSAP Make Busy feature to better match industry naming conventions. The feature allows a user with the correct privilege to deny all calls to a given tenant so that calls are rerouted by Guardian to an alternate route or rejected with a specific SIP cause (default cause 486) so that the incoming ESRP can alternate route the calls.

Release 19.0

BUSY OUT REASONS

An user can associate a reason for enabling make busy mode on their answering position.

LAST TELECOMMUNICATOR BUSY OUT WARNING

A warning message is presented to the telecommunicator when they attempt to busy out notifying them they are the last active member of a ring group.

LAYOUT COLOR THEMES

This feature allows a user with the correct privilege to define colors that are applied to the interface as a selectable theme.

CALL NOTES ENHANCEMENT

The call notes feature has been enhanced to support editing and adding to existing notes from the call history windows which are sent to MIS or a third-party system using i3 logs.

Release 18.1

ABILITY TO SWITCH PUSH-TO-SIGNAL MODE

When a PAC II is configured for call taker mode, the push-to-signal button on a handset/headset can be configured to signal push-to-talk mode or push-to-mute mode. Each jack box can be configured independently.

Release 18.0

TRANSFER TEXT-TO-911 BETWEEN SYSTEMS

The system has been enhanced to support the transfer of text-to-911 calls between different Guardian systems using MSRP. When both systems are equipped with support for SIP MSRP, a text-to-911 call is transferred as a text call. If receiving system does not support SIP MSRP, a voice conference is initiated.

Release 17.3

TENANT DASHBOARD

Real-time statistics can be displayed in a configurable dashboard with data displayed per named subtenant groups. The dashboard consists of three statistical sections: Grade of Service, Call Activity, and Position Call Status.

Release 17.2

GUARDIAN APPLICATION ALERT

The Guardian icon on the answering position's status bar can be configured to light up when a new call arrives at the position to notify the user if their attention is in another application.

TEXT CALL BACK

The system provides the ability to send an outgoing SMS message when either a voice or text-to-911 emergency call is abandoned.

Release 17.1

PROCEDURAL MESSAGES

The procedural messages feature allows predefined messages to be displayed to the telecommunicator during the processing of a call. The messages displayed are determined by the action the telecommunicator performs via the various command buttons in the application.

ADDITIONAL CALLER INFORMATION

This feature introduces the ability to query an external system for additional location data when an emergency call is received by the Guardian system. The external server can be queried on the ANI, callback number, or other data element received with the call or from the original location data.

Release 17.0

DISCONNECT BUTTONS WITH REASON

This feature provides the ability to define unlimited reasons for releasing a call or a participant of a call. Custom action buttons are provided to telecommunicators to log the reason a call was released or the reason a participant was released from the active call. The disconnect reason is logged in the End Media event for statistical purposes.

CALL HISTORY ENHANCEMENT

Disconnect reasons and transfer reasons are available within the call history windows for historical calls.

AUTOMATIC FORCED BUSY OUT AND FORCED LOG OUT

This feature provides the ability for the Guardian system to automatically busy out a position or even log out the user from the position when a call presented to the position is not answered within the specified ring no answer (RNA) time. A notification can be configured to be sent automatically to a specified role or role group when the system forces a position to busied out status or logs out the current user from the position.

TELECOMMUNICATOR CALL HISTORY

This feature provides the ability for a telecommunicator to view emergency call history for all the calls they participated in since they last logged in to Guardian from the Telecommunicator Call History window.

NOTIFICATIONS

This feature provides the ability to send predefined notifications to telecommunicators when their role has been assigned the privilege.

Release 16.1

TOOLBAR EDITOR ENHANCEMENT

Individual tenant status buttons can be placed in the layout. The buttons function the same as the buttons on the Tenant Status Bar - they launch the associated tenant status window. The label for the buttons displays the number of calls in the corresponding state - current, on hold, ringing, and so on. Buttons for both emergency and administrative calls can be added to the layout, and the images used for the buttons can be customized.

Release 16.0

ALI CORRECTION ENHANCEMENT

The ALI email feature allows corrections made to the ALI by a telecommunicator to be sent to a preconfigured email address as a HTML attachment. The feature can be configured to

send the corrections to multiple email addresses. Each tenant group can have different email addresses configured.

CONTACTS ENHANCEMENT - ALTERNATE NUMBER

Contacts now support the configuration of a secondary phone number that is dialed automatically in the event the call to the primary phone number returns a busy notification or another condition occurs resulting in the destination not being reached.

TENANT STATUS ENHANCEMENT - ELAPSED TIME SINCE STATUS CHANGE

A new column in the Active Telecommunicator tenant status window displays the elapsed time since the user's status has changed. Users can view how long a user has been idle, how long a user has been on a call, or how long a user has been busied out.

SUPPLEMENTAL ALI NOTE ENHANCEMENT - EXPORT, PRINT, AND CUSTOM FIELDS

Up to four additional categories are configurable for supplemental ALI notes per tenant group. Users can print the contents of a note from the Supplemental ALI Note window while on an active call or a user with manage privileges can print a note from the management window. Supplemental ALI note entries can be exported to an external CSV file from the Supplemental ALI Note Manager window.

GUI IMPROVEMENTS - PHASE I

Numerous improvements to the appearance of the user interface are available including:

- Customizable font type and size for all textual windows
- Customizable icon images for command buttons
- Flat-look effect for layout
- Customization of NG9-1-1 Info window
- Support for icon images up to 256x256 pixels

Additional customization of direct access toolbars including:

- Display label only
- Scalable buttons up to 600%
- Customizable label fonts
- Customizable button width and height
- Customizations are saved to layout per user/role

Release 15.1

INTERACTIVE VOICE RESPONSE TO SILENT CALLS

Telecommunicators can send silent calls to an interactive voice response queue where an announcement asks the caller to press a specific digit(s). When the system detects the correct digit(s) was pressed, the call will be sent to a designated ring group for processing. If an incorrect digit is pressed, or no digit is pressed, in the configured time period, the call will be released.

Release 15.0

TRAINEE MONITORING

Ability for a supervising user to automatically monitor all calls handled by another user such as a trainee.

EXCLUSIVE HOLD

A user can place a call on exclusive hold which will prohibit the call from being retrieved by any user other than the original user to place the call on hold.

CALLER HISTORY

A user will have the ability to retrieve the complete call history for the current caller, based on the callback number. Administrators can configure the call history retention period by number of days and/or number of records.

Release 14.1

DISPLAY CONCISE ALI IN TENANT STATUS WINDOWS

A subset of ALI fields can be displayed in the tenant status windows as it currently is displayed in the Call History window.

USER-CONTROLLED PAC II RELAY CONTACT

The system provides the user the ability to switch between relay 1 and 2 at the position. The primary purpose of the feature is to support a mode of radio integration where a

headset/handset could be connected to secondary jack box and provide a jack sense to the radio console.

Release 14.0

NG911 TEXT-TO-911 CALLS

This feature consists of receiving text-to-911 (SMS messages) over MSRP as a PSAP or multi-tenant terminating ESRP. The operational procedures of receiving and handling emergency SMS will closely resemble handling a TDD call including the current call distribution capability of the platform. Outgoing SMS messages are supported. Transfer of text calls is supported when the transfer agency resides on the same Guardian controller.

TDD/TTY AUTO-DETECT INDICATOR

In addition to the current pop-up indicating a TDD call, the system will provide an alternate option to light up the existing TDD button, change color to green, to indicate that the system has detected a TDD call.

SUPPLEMENTAL ALI

The system will provide a comment field that allows the telecommunicator, with the proper permissions, the ability to add, modify, or delete comments about the caller - based on the call back number. The comments will be stored in a server-based database so that existing comments will be displayed each time the same telephone number initiates an emergency call.

SORT AND FILTER CALL HISTORY BY CALLBACK NUMBER

The Call History window will support the ability to sort entries by callback number.

Release 13.0

ALI CORRECTION

When a user prints out ALI corrections they have made, both the original ALI and the corrected version will be included in the printout.

SAVE AUDIO/VOLUME LEVELS PER USER

Allow to save audio settings configured with volume control to a user account so that a user logging into a position with the same PAC II equipment will have their saved audio settings applied automatically. If the user logs into a position without PAC II equipment, the audio settings in the position's configuration file will be applied.

MERGE DIAL PAD AND DTMF PAD

A new "smart pad" will be made available as an alternative for the dial pad and the DTMF pad. Users will no longer have to launch separate pads when making calls, conferencing, or transferring.

CALL HISTORY "MY CALLS" FILTER

This feature will allow telecommunicators to view only their calls from the Call History window. A "My Calls" button will be added to the Call History window which will display only the calls where the Telecommunicator column match the currently logged in user account name.

Release 12.2.2

TDD VOICE CARRY OVER WITH TEXT

When a telecommunicator challenges a silent emergency call via TDD, the system will automatically place the telecommunicator in Voice Carry Over mode so that the telecommunicator can hear the caller and any available background audio.

Release 12.2

VIRTUAL POSITION

The Virtual Position feature provides an option where the resource number is bound to the user. A user configured as a roaming user can log in to any roaming-enabled position at any PSAP in the system and maintain the layout and call handling capabilities configured in their own PSAP. When a user is set as a roaming user, they are still members of ring groups and have the same line appearance settings as when they are logged into a position in their PSAP.

WEBSITE LINKS

A telecommunicator may require access to a website, for example an SMS (Short Message Service) web page. An administrator can provide access to specific web sites, as well as limit the navigation tools available to a telecommunicator.

ENHANCED CONTACT SEARCH FUNCTIONALITY

To locate a contact in a large list, you can begin typing the first characters of the name or phone number; the list is refined as each letter is entered and the match is highlighted. The list can also be filtered by classification, or scrolled.

TRUNK-BASED RECORDING - CONTINUOUS RECORDING

When this feature is enabled, the recorder continues to record, even when the call is placed on hold.

Release 12.1

RING GROUP ASSIGNMENT

The Ring Group Assignment feature allows positions to view and log in to and log out of ring groups to which they are assigned, provided they have the appropriate privileges enabled.

RING GROUPS DASHBOARD

The Ring Group Dashboard feature displays real-time statistics on call traffic and handling by providing a summary of the activity of all tenant groups, ring groups and users.

RING TONES AND ZIP TONES

The Ringer Modes Manager feature allows an administrator to set the following ringer modes for each call type and call state:

- three distinct ring tones to identify admin, emergency or abandoned calls
- two distinct ring tones, to identify the call state for admin, emergency and abandoned calls
- one auto answer zip tone, to identify auto answer for admin, emergency and abandoned calls

ACD ENHANCEMENT

Ring group members can belong to any combination of the following ring group categories:

- Physical (position or phone device)
- User
- Role
- Role Group

ROUND ROBIN

Round robin call distribution routes calls to telecommunicators in the order that they logged into the ring group. Calls are rotated through the list to ensure each member receives a comparable number of calls. Instead of starting at the top of the call list for each new call, the system keeps track of which telecommunicator received the last call, and a new call is routed to the next telecommunicator.

WRAP UP

The Wrap up feature allows an administrator to set a specified length of time to wrap up the last call before the telecommunicator becomes available to receive the next ACD call.

BUSY OUT OPTIONS

The Busy out feature allows an administrator to set whether direct dial, line appearance, and ring group calls are displayed in the Phone Connection window when the position is in a busied out, or wrap up state. A position can busy out when a call is displayed in the phone window, provided they have the appropriate privilege enabled.

ABANDONED CALL REPLACEMENT

The Abandoned Call Replacement feature allows an abandoned call to be replaced by a new call when the ANI is the same. When a second call with the same ANI arrives in the system, the first abandoned call is removed and the second call is considered active.

SOUND MANAGER

The Sound Manager feature allows an administrator to manage the sound resources that alert a telecommunicator when a call has been automatically answered or when an emergency, administrative or abandoned call is presented to a position.

Release 12.0

PREFERENCES

The preferences feature contains options that an operator can set to customize the functionality of their position for the role in which they are logged in. For example, an operator

can create a keyboard shortcut to call a contact with the press of a single key, or create a keyboard shortcut to open the administration window.

SHORTCUT KEY ENHANCEMENT

Shortcut keys are now configured within a preferences template. To view keys that are mapped to a shortcut, you must view the preference template.

Guardian shortcut key functionality can now override other applications with the same mapped shortcut key. This is an option that must be enabled.

INTELLIGENT CONTACT

The Intelligent Contact feature allows a telecommunicator to press a single direct access button to transfer and conference calls to different phone numbers or * codes based on the incoming route.

AUTOMATIC GREETINGS

The Automatic Greetings feature eliminates the repetitive nature of answering each call with the same greeting. A telecommunicator can record a variety of greetings to automatically play for a caller.

DTMF DETECTION

The DTMF Detection feature allows a caller to communicate with a telecommunicator when the caller is unable to speak. In DTMF mode, a caller can respond to a telecommunicator's questions by pressing numbers on their telephone key pad.

Release 11.0.0.1

STATUS BAR ENHANCEMENT

When a PAC II device is installed on the position, the status of jack box 1 and 2 is displayed.

ENHANCED VOLUME CONTROL

When a PAC II is configured, and the Volume Calibration privilege is enabled, a user can adjust and calibrate the volume for all input and output audio devices connected to the PAC II

JACK BOX MUTE BUTTON

When a PAC II is configured for the position, the Jack box mute (JB1 Mute and JB2 Mute) buttons allow the telecommunicator to mute one or both Jack boxes from the application window.

Release 11.0

ALI PRESENTATION ENHANCEMENT

An administrator can specify the format of the ALI and PIDF-LO information in the Caller Information window. The choices are: None, cpeResult_old.htm, cpeResult.htm, cpeResult_pidflo.html, and cpeResult_pidflo_raw.html.

CONTACT ENHANCEMENT

Allows users to import and export contacts. Users can filter based on a role to which contacts are imported and exported.

PHONE CONNECTION WINDOW ENHANCEMENT

When a telecommunicator receives a call from a ring group, the name of the ring group is displayed in the Phone Connection window.

TICKER MESSAGE ENHANCEMENT

A Now! button runs the ticker message immediately.

ENHANCED LOG IN

When a system is deployed in redundant mode, telecommunicators have the option to choose which server to log into. When more than one server is configured, an Options button, Redundant check box and server options are displayed.

FORCE SIMPLEX OPTION

During a system upgrade or maintenance cycle, the telecommunicator can choose the Force Simplex option to change the server on which they are logged. The telecommunicator can then be connected to the active server.

STATUS BAR ENHANCEMENT

The status bar at the bottom of the window indicates to which server the position is connected.

RINGER MODES MANAGER

This feature allows the user to delete ringer modes.

VIEW CALL HISTORY ENHANCEMENTS

You can easily navigate a large spreadsheet by hiding, unhiding and re-ordering columns.

CALL HISTORY ENHANCEMENTS

- When a telecommunicator calls back a caller using the Call Back button. The Called Back column is updated in the Call History window to indicate the caller has been called back.

- When a telecommunicator displays the ALI from a past call, the telecommunicator can print the ALI information from the call.
- An administrator can set the maximum number of calls stored.

TENANT STATUS WINDOW ENHANCEMENT

The tenant status window provides live monitoring of resources and are typically used by PSAP supervisors. The status windows are available from the View menu for emergency and now for admin calls.

LAYOUT MANAGER

This feature allows the telecommunicator to delete layout configurations that are saved to the database. Depending on assigned privileges you can delete your layout or layout configurations saved for other roles.

Release 10.2

PHONE CONNECTION WINDOW ENHANCEMENT

When an incoming call is routed through a ring group, the name of the ring group is displayed in the Phone Connection window. When a telecommunicator from the ring group transfers the call to a new ring group, the name of the new ring group is displayed in the Phone Connection window.

ALLOW BUSIED OUT POSITIONS TO VIEW QUEUED CALLS

When a telecommunicator is busied out, the telecommunicator can view calls in the queue and answer calls from the queue.

QUEUED CALL ICON

A new icon in the Phone Connection window indicates that a call is in the queue.

RECONNECTED ICON

An icon indicates when a call has been unexpectedly disconnected, and the system is reconnecting the call on an alternate route, or an internode route.

Release 10.1

GUARDIAN CALL CONTROL INTERFACE

The Guardian Call Control Interface is a status bar located on the operator position that indicates the connection status of the Guardian with the CAD device.

OPERATOR STATUS INDICATOR USING SPEED DIAL BUTTONS

Speed Dial buttons can be configured with an operator position phone number and can be configured to indicate the current state of the operator positions in the user's tenant group by means of the button color.

HELD AND LOST PROTOCOL SUPPORT

Guardian supports new query protocols such as HELD and LoST to determine location and routing information for the call and supports both manual and automated TN/ESN table updates for new ANIs.

LOCATION TYPE SUPPORT

Guardian supports the following 2 location types:

- Geodetic
- Civic

SELECTIVE TRANSFER AGENCY ENHANCEMENT

Guardian supports URI and URN addresses and displays them with Selective Transfer Agency information when they are available.

CONTACTS ENHANCEMENT

Guardian supports URI addresses for contacts in lieu of telephone numbers.

NETWORK QUALITY

The Network Quality feature allows the user to:

- view audio quality for any device on the system that requires streaming, for example headsets, speakers, and radio speakers
- view network quality, for example view any delays on the network
- set thresholds that audio quality cannot exceed without raising an alarm

CALLER INFORMATION WINDOW ENHANCED TO DISPLAY PIDF-LO

The Caller Information window displays one or more forms containing caller information retrieved from the ALI or PIDF-LO databases as each incoming 9-1-1 call is received or answered.

Depending on the configuration for your role, not all the following buttons may be displayed.

- ALI - choosing this option displays call information, street address, geographic information, transfer information and call taker notes.
- Location - choosing this option displays the caller name, country, city, county, neighborhood, landmarks, street address, floor, unit number, etc.
- Notes - choosing this option displays a notes dialog box where an operator can enter additional information.
- Raw - choosing this option displays PIDF-LO and ALI raw data.

TENANT STATUS ENHANCEMENT

The Tenant Status bar and associated status windows for Emergency and Admin calls display live PSAP status information for use by supervisors.

Release 10.0

ABANDONED CALL STATUS WINDOW

With the press of a button, all abandoned 9-1-1 calls are displayed. The following information is presented:

- ANI of the call
- Call start/arrival time
- Call Status
- Operator Position
- Operator Name

From the window the operator can call back the abandoned call.

CONTACTS, IMPORTING AND EXPORTING

Contacts can be imported from other systems such as Outlook, into the Guardian Contacts list. You can also export Guardian contacts for backup purposes or to transfer them to another server.

TDD AUTOMATIC DETECTION

The system detects baudot tones automatically and provides visual indication to the call taker.

TDD ENHANCEMENTS

The TDD window allows an operator to communicate by typing a conversation or by using pre-composed messages. The VCO and HCO features allow an operator to communicate with a caller using a combination of voice and text messages.

TICKER MESSAGE

Various informational messages can be configured and displayed in a ticker (one line scrolling) window. The window will also display in red alarms indicating a malfunction such as a loss of server connectivity or loss of audio stream.

SINGLE LOG IN

Allows the user to log in the system without entering a user name or password.

The system will use Windows log in credentials

CALL TAKER NOTES

Allows a Call Takers to take notes while on a call in order to record an event or status of significance.

SHORTCUT KEYS

Shortcut keys allow the operator to perform a task by pressing a single key. Shortcuts can be created, modified or deleted for each role.

OPERATOR STATUS INDICATOR

Speed Dial buttons can be configured to indicate the current state of the operator positions in the user's tenant group by means of the button color. The color indicates in following for each position: not logged in, call ringing, busy with a call, busied out and position ready to take a call.

QUICK TRANSFER

The Quick Transfer button transfers a call and automatically releases the Call Taker from the call when the called party answers.

TDD ENHANCEMENTS

The TDD window allows an operator to communicate by typing a conversation or by using pre-composed messages. The VCO and HCO features allow an operator to communicate with a caller using a combination of voice and text messages.

Index

Index

A

- abandoned call replacement 80
- abandoned emergency calls 80
 - answer 81, 101
 - answer and callback 82
 - answer and text back 82, 101
 - queued 101
- active telecommunicators 82
 - commands 83
 - elapsed time 86
 - force log out 83
- additional caller information 25
- admin calls on hold
 - answer call 77
- AdminEmergency 19
- administrative calls 20
 - answer 149, 167
 - release 163, 183
 - SIP REFER 168
- ALI corrections 135
 - email 137
 - print 137
- ALI status 256
- ALI tools
 - close 14
 - correction 14
 - email 14
 - manual 14
 - print 14
 - refresh 14
 - RMT print 14

- alternate server 8
- audible alarms 253
- audio quality alarm 251
- audio source 245
 - change 217
- automated answering system 127, 159, 180
- automatic greeting 99
 - adjust volume 100
 - disable 100
 - stop 100
- automatically answered calls 97
- auxiliary audio 220

B

- busy out warning 36

C

- call alert 16
- call history 55
 - ALI corrections 57
 - apply filter 60
 - call back 58
 - commands 56
 - delete entry 59
 - my calls 60
 - prevent updates 59
 - print ALI 58
 - text back 59
 - view ALI 57

- view conversation 58
- call monitoring 74
 - barge 76
- call notes 141
 - add 142
 - edit 145
- caller history
 - view 147
- caller information 21
 - arrange tabs 21
 - display after call release 22
- caller status 20
- color themes 245
 - change 219
 - restore 220
- communication failure 247
- contacts 51
 - alternate phone number 124
 - commands 52
 - conference 124, 154, 173
 - conference SIP REFER 174
 - personal 224
 - quick transfer 125, 154
- current admin calls 72
 - answer call 73
 - commands 73
 - monitor call 73
- current emergency calls 72
 - answer call 73
 - commands 73
 - monitor call 73

D

- dial pad 47
 - commands 48
 - manual conference 126, 158
 - manual quick transfer 127, 159
- direct access 37, 153
 - alternate phone number 121
 - behavior 38
 - conference mode 121, 151, 170
 - conference SIP REFER 171
 - quick transfer 173
 - quick transfer mode 123

- transfer only 39
- disconnect reasons 33
- disconnected calls 103
 - callback 103
 - text back 103
- DTMF pad 49
 - conference 119
 - conference SIP REFER 120
 - manual conference 177
 - manual conference SIP REFER 179
- DTMF tone detection
 - enable 206

E

- emergency calls 19
 - answer 97
 - answer out of sequence 98
 - manual ALI 134
 - refresh ALI 133
 - release 131
- emergency calls on hold
 - answer call 77
- exclusive hold 32

F

- fixed transfer codes 111
 - conference 112
 - NG9-1-1 conference 113
 - NG9-1-1 release 113
 - NG9-1-1 transfer 114
 - release 112
 - SIP REFER 114
 - transfer 112

H

- hearing carry over 204
- hearing impaired 202
 - conference 205
 - transfer 205
- hold 30, 31

I

- instant messaging 207
 - add friend 208
 - add personal message 212
 - chat 212
 - invitation 209
 - modify status 210
 - remove friend 209
 - transfer file 214
- IRR 92
 - play 94

J

- join calls 131, 162, 183

L

- line appearances 40
 - answer 42
 - color 41
 - retrieve from hold 42
- local text conversation 199
- local text conversations 90
- log in 6
 - alternate server 9
 - roaming account 7
- log out 11

M

- make busy 34
- make busy reasons 36
- manual xfer 49
- menu bar 13
- mute 29, 30

N

- next generation 9-1-1 5
- NG9-1-1 info 18
- notifications 251
 - clear 253
 - view 253

P

- password
 - change 218
 - rules 218
- personal contacts
 - create 225
 - delete 227
 - edit 227
- personal greetings 228
 - assign 229
 - record 228
- phone connection 15, 26
 - automatic expansion 28
- PIDF-LO 23
- place call 164, 184
- position status 43
- precomposed keyword 188
- preferences 245
 - shortcuts 232
- primary server failure 248
- privacy 30, 31
- procedural messages 88
- PSAP make busy
 - disable 255
 - log in 255
 - warning 254
- push-to-signal 221
 - PTM mode 222
 - select mode 223

Q

- quick transfer mode 153

R

- redial list 66
 - commands 68
- ring group assignments 53
 - log in 54
 - log out 54
- roaming account 7
- routing status 20

S

- selective transfer agency 104
 - conference 105
 - NG9-1-1 conference 107
 - NG9-1-1 release 108
 - NG9-1-1 STA info 109
 - NG9-1-1 transfer 108
 - release 105
 - SIP REFER 109
 - STA info 106
 - transfer 106
- server communication down 251
- server switchover 247
- sessions list 190
- share IRR 233
 - receive recordings 239
 - recipient list 236
 - search recordings 237
 - send recordings 233
 - share recording 238
- smart pad 45
 - commands 46
 - conference 117
 - conference SIP REFER 118
 - DTMF awareness 47
 - manual conference 125, 155, 175
 - manual conference SIP REFER 176
 - manual quick transfer 127, 157
- status bar 14
- supplemental ALI notes 138
 - add 140
 - modify 139
 - print 141
 - view 139
- switch communication down 251

T

- taskbar alert indicator 16
- TDD
 - answer 203
- TDD auto detect 202
- telecommunicator call history 62
 - ALI corrections 64
 - call back 65

- commands 63
 - print ALI 65
 - text back 66
 - view ALI 64
 - view conversation 65
- telecommunicator monitoring 84
 - barge 84
 - escape 85
- tenant dashboard 86
- tenant status alarms 249
- tenant status bar 70
- tenant status buttons 72
- text conversation 188
 - action toolbar 189
 - text toolbar 190
- text-to-911 19, 192
 - answer 193
 - call back 201
 - conference 200
 - conference different system 198
 - conference same system 196
 - control 197
 - hold 194
 - remove hold 195
 - text back 201
- ticker 89
- ticker message alarms 250

V

- voice carry over 203
- voice communication down 251
- volume
 - adjust 242
 - save settings 243
- volume meter
 - adjust 244

W

- waiting admin calls
 - answer call 79
- waiting emergency calls 78
 - answer call 79
- website links 91

wrap up 98
end 99





A Subsidiary of
Comtech Telecommunications Corp.

Release 19.1
C2214114-30 SR1

Call Toll Free (North America): 1 (888) SOLACOM (765-2266)
Call Direct (Worldwide): 1 (819) 205-8100

Visit: www.solacom.com

Copyright © 2019

Information presented within this document is subject to change without notice. Solacom and the Solacom logo are trademarks of Solacom, a subsidiary of Comtech Telecommunications Corp. All other trademarks are the property of their respective owners.