

Operator's Manual

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Symphony™ Dispatch Console



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REV	DATE	REASON FOR REVISION
-	Jun/14	Initial release.
A	Nov/14	Updated for R2 release of software. Added Aux I/O, Base Station Modules, Paging and Encryption.
B	Jun/15	Updated for R3 release of software.
C	Jan/16	Updated for R4 release of software.
D	Jun/16	Updated for R5 release of software.
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L	Feb/22	Updated for R13 release of software. Added Advanced Backup Radio.

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1 SAFETY CONVENTIONS

The following conventions are used throughout this manual to alert the user to general safety precautions that must be observed during all phases of operation, service, and repair of this product. Failure to comply with these precautions or with specific warning elsewhere in this manual violates safety standards of design, manufacture, and intended use of the product. L3Harris assumes no liability for the customer's failure to comply with these standards.



The **WARNING** symbol calls attention to a procedure, practice, or the like, which, if not correctly performed or adhered to, could result in personal injury. Do not proceed beyond a **WARNING** symbol until the conditions identified are fully understood or met.



The **CAUTION** symbol calls attention to an operating procedure, practice, or the like, which, if not performed correctly or adhered to, could result in damage to the equipment or severely degrade the equipment performance.



The **NOTE** symbol calls attention to supplemental information, which may improve system performance or clarify a process or procedure.

2 INTRODUCTION

This manual describes the operation of the Symphony™ Dispatch Console. It includes instructions to be used by both the dispatch operator and the dispatch supervisor. Basic console operations as well as more advanced features are covered.

The manual is organized as follows:

- Section 2 Introduction: This section provides basic instructions for starting and stopping the application, using the keyboard and touchscreen, as well as accessing the online help.
- Section 3 Graphical User Interface: This section describes the two Symphony user interfaces. The Full Screen Display uses the entire primary monitor to provide a detailed view of, and quick access to, all communications assets. The Baton condenses the Full Screen Display into a compact toolbar that can run on a separate computer and work in conjunction with other applications such as Computer Aided Dispatch.
- Section 4 Console Audio System: This section describes the Symphony audio system including speakers, headsets, microphones, PTT controls, and audio routing.
- Sections 5 - 10 provide step by step instructions for using various Symphony features.
- Section 11 provides Customer Service contact information.

2.1 DESCRIPTION

The Symphony Dispatch Console provides high-performance computing and graphics capability to support 24-hour mission-critical dispatch operations. The completely solid-state design affords the high reliability needed for dependable communications in emergencies. The console uses digital audio technology to provide end-to-end secure communications that are free from unauthorized access. The Symphony Dispatch Console features a user-friendly interface, providing flexibility. In addition to the full screen console application, there is a compact user interface, or “Baton,” that is designed to minimize space consumed on CAD screens and is intended for frequently used functions. See Section 3 for more information on the user interface.

2.2 LICENSING

Many Symphony features are controlled by licensing. Therefore, a feature described in this manual may not be available on your console. Licensing can enable or disable an entire feature, or limit aspects of the feature (e.g., the number of speakers supported). The **About** dialog, available from the **Harris** menu, lists the features licensed for your console.

If a feature is not licensed, it is generally removed from the user interface to reduce confusion. However, some features are so integrated into the user interface, removing them is not feasible, so they are grayed out. The expected behavior for unlicensed features is noted with each feature herein.

2.3 START THE SOFTWARE

2.3.1 Full Screen Application




NOTE

The Symphony application can be configured by the administrator to launch automatically on PC power-up.

Consult with the system or dispatch administration personnel for power-up authorization. Verify that all cabling, power connections, and other system hardware are interconnected as required. Refer to the Installation and Configuration manual as necessary.

To launch the full screen application, choose **Symphony** from the Start screen. To open the start screen:

- Mouse: Click the **Start** button in the bottom-left corner of the screen (or point the mouse in the bottom-left corner of the screen until the **Start** button appears).
- Keyboard: Press the Windows® logo key .
- Touch screen: Tap the **Start** button in the bottom-left corner of the screen.

Symphony displays the splash screen shown in Figure 2-1. The screen indicates the boot progress via the check boxes for each boot step. It also provides valuable diagnostic information should the boot process fail.

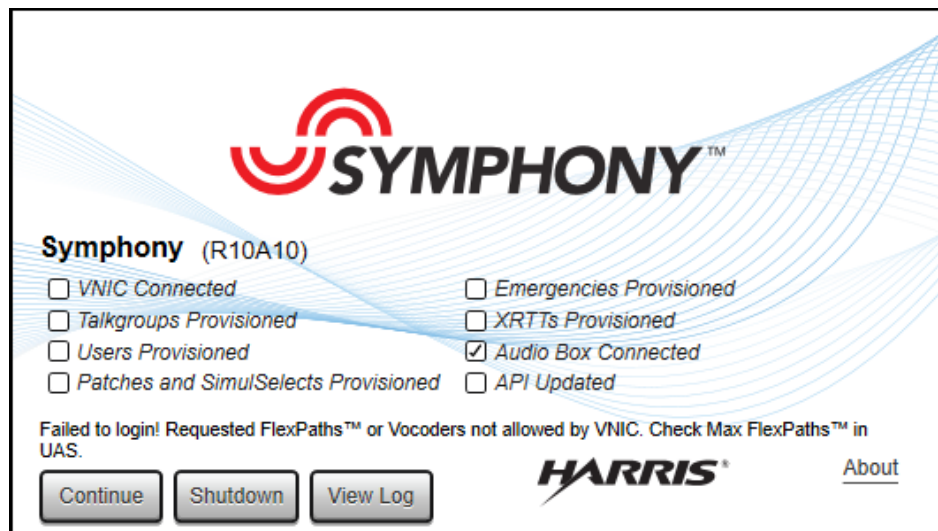


Figure 2-1: Symphony Full Screen Splash Screen

After startup, the console's Communication Modules are programmed based on the most recently saved operational session.




If the Symphony **Connection Manager** dialog (Figure 2-2) is displayed when opening the application, consult with your system administrator. This indicates there is an issue with console configuration.

Figure 2-2: Symphony Connection Manager

2.3.2 Baton

To launch the compact user interface, choose **Baton** from the Start screen. To open the Start screen:

- Mouse: Click the **Start** button in the bottom-left corner of the screen (or point the mouse in the bottom-left corner of the screen until the **Start** button appears).
- Keyboard: Press the Windows logo key .
- Touch screen: Tap the **Start** button in the bottom-left corner of the screen.



NOTE

If the Baton **Connection Manager** dialog (Figure 2-3) is displayed when opening the application, consult with your system administrator. This indicates there is an issue with the console configuration.

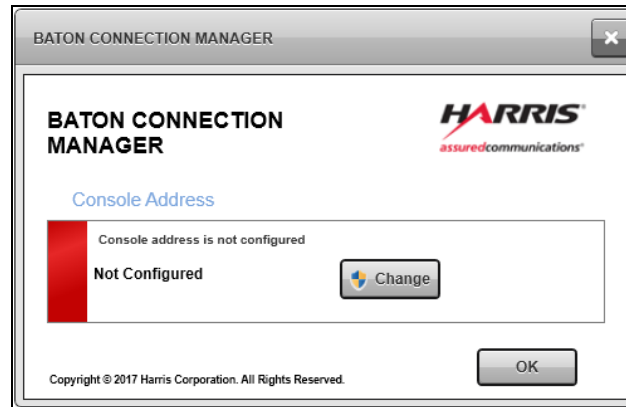
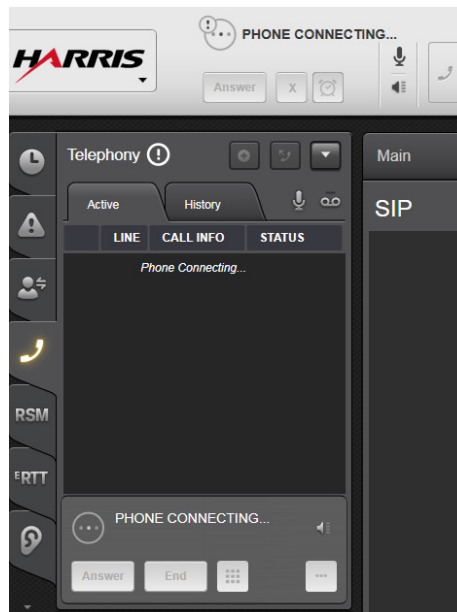


Figure 2-3: Baton Connection Manager

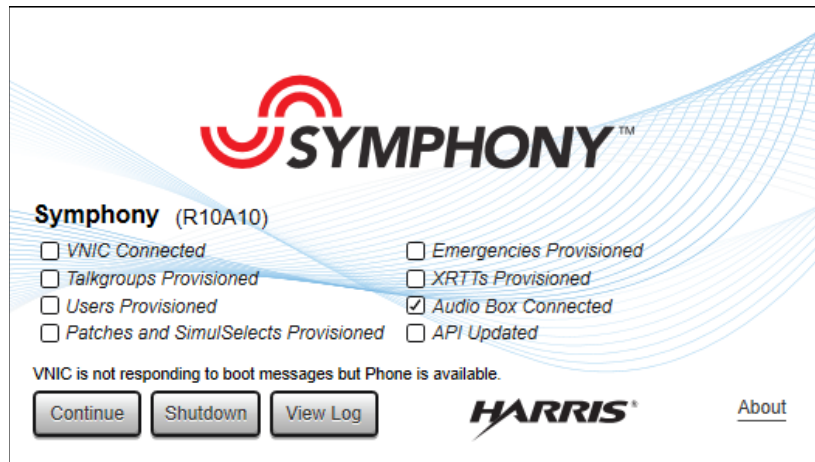
After startup, the Baton's Communication Modules are programmed based on the most recently saved operational session.

2.3.3 Error Conditions

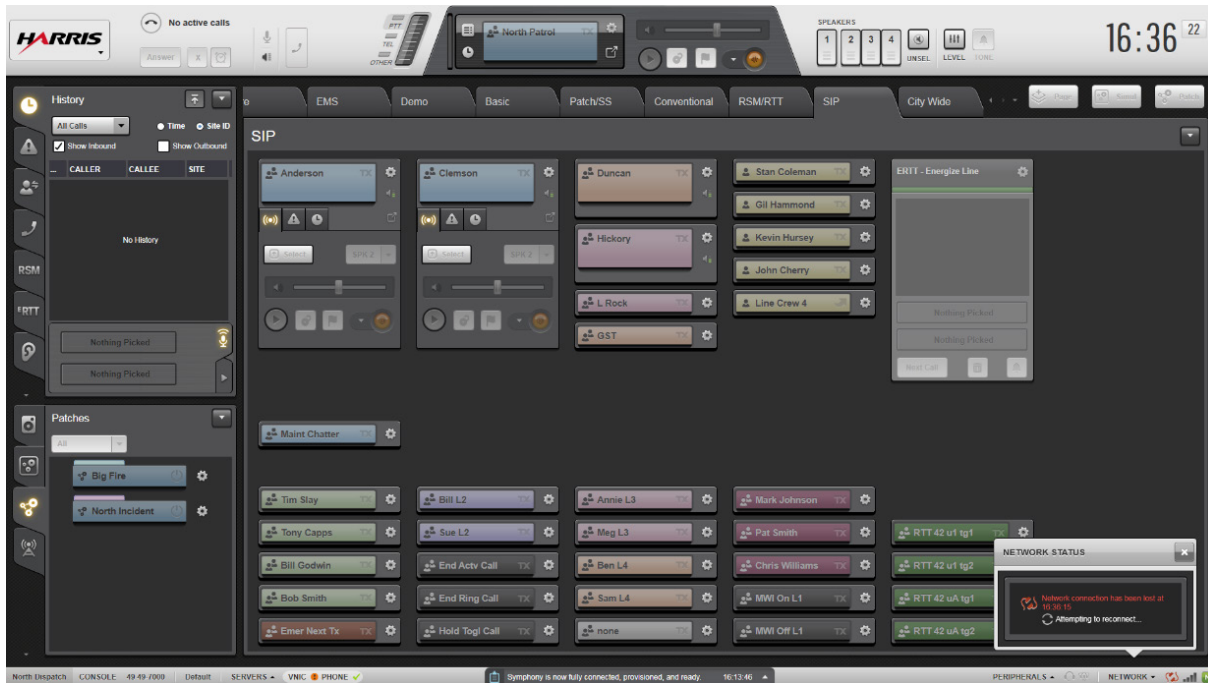
If Symphony cannot register with the PBX for SIP Telephony, the console continues to operate for radio communications, but telephony is not available. The degraded telephony state is indicated in multiple locations, as shown below. The text "Phone Connecting..." is shown in the Select Bar Telephony Control, the Active Call List, and the Call Control area. The ⓘ icon is shown in the Select Bar Telephony Control and Telephony Panel. The System History includes relevant error messages. The Baton provides similar indications on its equivalent controls. Contact your Systems Administrator to resolve the connectivity issue.



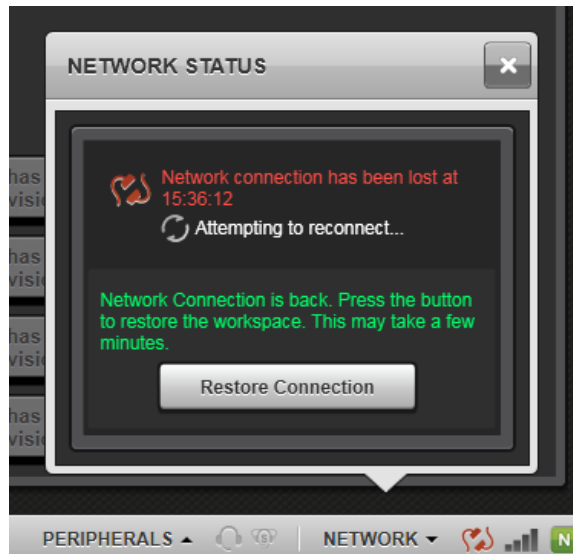
If the VIDA® core is not available when Symphony starts but the PBX is available, the console displays the following:



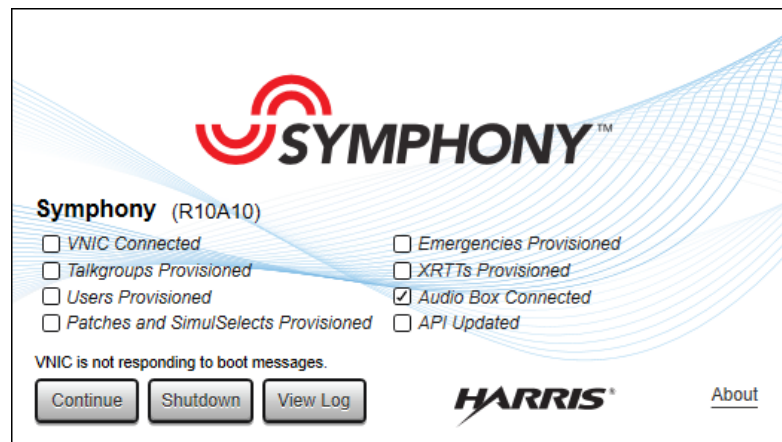
Press **Continue** to open Symphony in the SIP only mode. All radio communications modules are blanked out, but the telephony controls are available as shown below:



Once the VIDA core is available, the Network Status dialog requests permission to reconnect to the core as shown below. Reconnection to the core is best done when no phone calls are active as the GUI may be unresponsive while the initial provisioning takes place.



If both the VIDA core and PBX are unavailable, Symphony displays the following:



Press **Continue** to instruct Symphony to continue its attempts to connect to the core. When the VIDA core is available, Symphony completes the boot process as usual.

If connection to the VIDA core is lost once Symphony has booted, Symphony can be manually placed in Backup Radio (BUR) Mode. See the Installation and Configuration Manual for details. The headset and footswitch can then be used with a backup radio. During Backup Radio Mode, all of the telephony features are still accessible through the Telephony popup that is presented on the console. The Select Bar Telephony Control and Baton Telephony widget sections are also accessible during Backup Radio mode. Existing Telephony calls are maintained during the transition to Back-up Radio mode.



2.4 SHUTDOWN PROCEDURES

Shutdown of the console equipment may be necessary when a loss of AC power is anticipated and/or the equipment will otherwise remain idle for a long time. The console may be equipped with an optional Uninterruptible Power Supply (UPS), which can supply power to the console from storage batteries. If necessary, consult with the system or dispatch administration personnel when a decision must be made on whether or not to shut down.

When it is determined necessary to power-down the equipment, perform the shutdown in accordance with the following procedure:

1. If the full screen application or Baton is running, click the **Harris** logo and click **Exit**.
2. The Shutdown dialog is displayed as shown in Figure 2-4. This dialog notes any important states like emergencies that are still active. It also notes features like Patch that will be deactivated as part of the shutdown. Click **Yes** to proceed. Click **No** to return to the Symphony application.

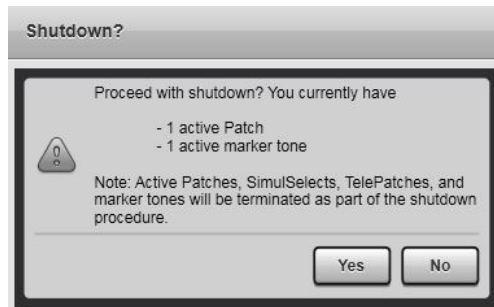


Figure 2-4: Shutdown Dialog

3. In Windows 10, use the mouse or touch screen to navigate to the bottom-left corner and click or tap the **Start** button. In Windows 8.1, use the mouse to point to the upper-right corner of the screen, move the mouse down and then click **Settings** on the Charms bar.

4. Tap or click **Power**, and then tap or click **Shut down**.
5. Turn off the monitor with its on/off power switch.
6. Turn off any other externally connected equipment, such as a Telephone Interface Unit.

2.5 USING THE MOUSE AND TOUCH SCREEN

Left-click using a mouse; touch an item with your finger using a touch screen; or a use combination of the two to perform a “press button” action on the display screen. To choose an item on a touch screen, touch the desired area of the screen with your finger.



NOTE

A touch is equivalent to a single left mouse button click.

Depending on your specific touch screen, you may also be able to touch the screen with any small blunt object, such as a pencil eraser.



CAUTION

Never touch the screen with any metal or sharp objects, as this can cause permanent damage to the screen.

2.6 ACCESSING THE ONLINE HELP

The Full Screen and Baton applications provide detailed online help. The online help is essentially a user guide for each application, providing descriptions, explanations, and detailed procedures. Access the online help for each application from the Windows Start menu, by pressing F1 while the application is in focus, or from the application’s Main Menu (see Section 3.1.1).

2.7 NOTES ON TELEPHONY

Symphony supports two kinds of telephony:

- **Call Director:** This feature connects a physical phone to the console as if the console were a headset for the phone. All telephony functions are controlled with the physical phone.
- **SIP VOIP Telephony:** This feature uses Voice over IP (VOIP) and Session Initiation Protocol (SIP) to integrate telephony features directly into the Symphony user interface. With this feature, Symphony replaces the physical phone and provides access to all telephony features including call transfer and conferencing.

Symphony currently supports only one telephony mode at a time. The user interface for Call Director and SIP telephony are similar, but SIP VOIP telephony provides many more features. Details of each interface are provided in the appropriate sections of this manual. Three terms are used throughout this document and in the user interface to refer to the telephony features:

- **CD or Call Director:** Refers to the Call Director feature.
- **PHONE or SIP Telephony:** Refers to the SIP VOIP telephony feature.
- **TELEPHONY:** Refers generically to telephony and includes Call Director and SIP VOIP telephony features.

3 GRAPHICAL USER INTERFACE

The Graphical User Interface (GUI) is what appears on the screen. Symphony supports two GUI interfaces: The Full Screen Display and the Baton. The former uses the entire primary monitor to provide a detailed view of, and quick access to, all communications assets. The latter condenses the Full Screen Display into a compact toolbar that can run on a separate computer and work with other applications such as Computer Aided Dispatch.



Figure 3-1: Full Screen Display



Figure 3-2: Baton

As a quick overview, the Full Screen Display has four main elements, described briefly below. Also described are how these elements are included into the Baton in order to minimize screen real estate.

- **Select Bar:** The Select Bar is the tool bar across the top of the Full Screen Display. This element contains the primary controls that must be available at all times. The Baton is basically a modified version of the Select Bar.
- **Sidebar Panels:** Features are organized into panels on the left side of the Full Screen Display. The same panels are hidden on the Baton until needed (a bank of panel buttons activates the popup panels).
- **Workspace:** The majority of the screen is dedicated to the Workspace which contains communications modules. The Workspace is divided into tabs to organize collections of modules. The Baton uses workspace widgets to show two modules from a workspace tab. A popup panel provides access to the other modules in the tab.

- **Status Bar:** The Status Bar is located on the bottom of the Full Screen Display and provides information on the status of Symphony connectivity to peripherals and VIDA servers. The Baton provides access to key status dialogs via buttons under the main **Harris** menu.

The following sections describe the basic arrangement and functions available on the Full Screen Display and Baton, as well as details on user input devices. Later sections provide step by step instructions for how to use each feature. The Full Screen Display and Baton share many common elements. This section starts with a description of common elements and ends with an overview of each GUI including how common elements are used and details on unique aspects of the interface.

3.1 COMMON CONTROL ELEMENTS

This section describes user interface elements that are primarily found on the Full Screen Display – Select Bar or on the main Baton. Complex elements found in multiple locations are covered as well. Figure 3-3 shows the elements shared between the Select Bar and Baton.

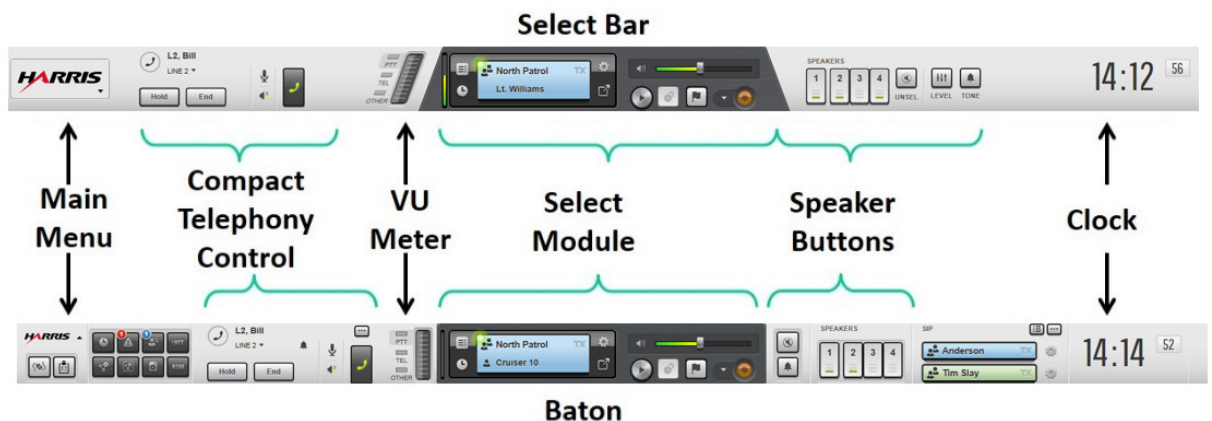


Figure 3-3: Common Controls on the Select Bar and Baton

3.1.1 Main Menu

On the Full Screen, click the **Harris** logo in the top left corner of the screen to open the application menu. On the Baton, click the **Harris** logo on the left side of the application to open the menu. Available options include:

- Open Directory:** Opens the Directory dialog. This menu offers a sub menu of the supported Directory tabs. The list of tabs depends on the licensed features on your machine. See Section 3.5.1 for more information.
- Open History:** Opens the Extended Call History for all entities. See Section 3.5.2 for more information.
- Open TextLink:** *Full Screen Application Only:* Opens the Radio TextLink client application in Symphony integrated mode. This menu option is available only if the Radio TextLink client is installed on the Symphony Console. See Section 9.4 for more information.
- Create Patch:** Opens the NEW/EDIT PATCH dialog. See Section 5.8 for more information.

- Create SimulSelect:** Opens the NEW/EDIT SIMULSELECT dialog. See Section 5.9 for more information.
- Switch to Next Monitor:** Moves the application to a different monitor.
- Configure:**
- Full Screen Application:* Allows the administrator to enter/exit Configuration Mode, switch the application to another monitor, configure GUI and audio settings and manage Speed Dial Lists. The console is non-operational when in Configuration Mode. Refer to the *Symphony Installation and Configuration Manual* (14221-3100-4100) for more information on configuration and the limitations of this mode. See Section 6.2.16 for more information on Managing Speed Dial Lists; **Symphony Configuration Utility → General → Telephony Format** option must be set to SIP and SIP must be licensed. The **Symphony Configuration Utility → Contacts → General → Enable Contact Modification** option controls Contacts and Speed Dial Lists modifications.
- Baton:* Allows the user to access the Manage Widgets dialog, manage Speed Dial Lists configure audio settings and change the volume level for audible tones including telephony ringing. The Manage Widgets dialog allows the administrator to add or delete extra controls on the Baton, such as the Speaker Buttons and the Workspace tabs. The number of Workspace tabs you can add is limited to the size/resolution of your monitor. See Section 2.3.2 for more information on the Baton. See Section 6.2.16 for more information on Managing Speed Dial Lists; **Symphony Configuration Utility → General → Telephony Format** option must be set to SIP and SIP is licensed. The **Symphony Configuration Utility → Contacts → General → Enable Contact Modification** option controls Contacts and Speed Dial Lists modifications.
- Setups:** *Full Screen Application Only:* Allows the user to switch setups. Allows the administrator to create and manage setups. See Section 5.13 for more on switching setups. Refer to the *Symphony Installation and Configuration Manual* (14221-3100-4100) for more information on creating and managing setups.
- Preferences:** Allows the user to Enable/Disable Global RTT Alarm or set phone preferences. The Global RTT Alarm menu option is available only if the **Symphony Configuration Utility → RTT/RSM → Settings → Allow Runtime Modification** option is enabled and RTT/RSM is licensed. This menu item contains an icon which shows the current status of the global RTT alarm (the icon is enabled if the global RTT Alarm is enabled); as well as text showing what action is available (e.g., if the global RTT Alarm is enabled, the text will read “Disable Global RTT Alarm and vice versa). The Phone Preferences menu option is available only if the **Symphony Configuration Utility → General → Telephony Format** option is set to SIP and SIP is licensed.

- Feature Panels:** *Baton Only:* Allows the user to access the Supervisory dialog and the System Status dialog. The Supervisory dialog is available only if the Discreet Listening and/or Radio Unit Monitor features are/is licensed or if a console is configured for the Console Enable Disable feature. These feature(s) also have to be enabled in the **Symphony Configuration Utility**. See Section 3.4.11. The System Status dialog shows the link status of VIDA elements. This is available only if the **Symphony Configuration Utility → System Status → Enable System Status Monitoring** option is enabled.
- About:** Opens the About dialog. The About dialog displays:
- Product Name
 - Product Version
 - Audio Box Build Number
 - Product Build Date
 - Licensed Features
 - Copyright Information
 - License Locking Code
- Help** Launches the Symphony and Baton online help.
- Exit:** Exits the Symphony application.

3.1.2 Call Director Control

The Call Director Control provides an indication of the CD on/off-hook status and serves as a transmit button when a headset is not used. See Section 6.1 for more details.

3.1.3 Compact Telephony Control

The Compact Telephony Control provides an always-available control that manages ringing and active telephony calls. The control is located on the Select Bar of the Full Screen Display and is referred to as the Select Bar Telephony Control. The control is located on the Baton and is referred to as the Telephony Widget. See Figure 3-4.

The Compact Telephony Control focuses on answering and managing calls. To place calls or access advanced telephony features, see the Telephony Panel in Section 3.4.5.

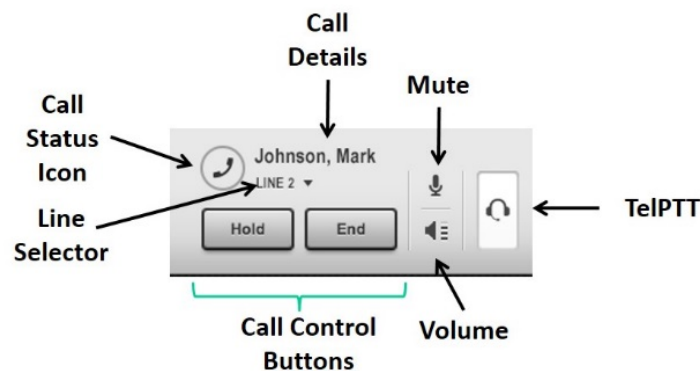
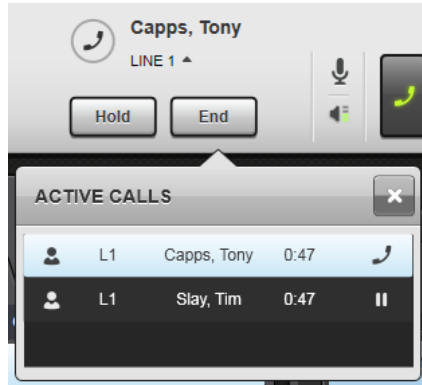


Figure 3-4: Compact Telephony Control

The Compact Telephony Control shown in Figure 3-4 has the following elements:

- Call Status Icon: This icon shows the current state of the call. See Table 3-7 for details.
- Call Details: This field includes information about the caller/callee if available. This can be an alias such as “Tim Slay” or a phone number.
- Line Selector: The line selector shows the currently selected line. Click on the Line Selector or the Call Details field to open the Active Call popup. This popup shows the active calls and allows the dispatcher to select a different call. This is a compact form of the Active Call List found in the Telephony Panel. See Section 3.4.5 for details.















- Mute: Mute the mic audio to the phone.
- Volume: Pop out the volume control.
- Call Control Buttons: These buttons provide the most used controls for a particular call situation. The buttons change depending on call state. For example, for a ringing call, the buttons are **Answer** and **Decline**; the buttons for an active call are **Hold** and **End**.
- TelPTT: The Telephony PTT is a multipurpose control that indicates when the call is active and/or in a TelePatch™ and provides a PTT control when telephony is in half duplex mode.
- More Button (Baton Only): The Baton includes a more button  to open the Telephony Panel Pop up for access to advanced telephony features.

Table 3-1 summarizes the call status icons.

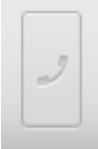
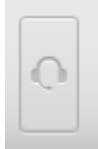
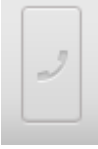
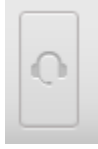




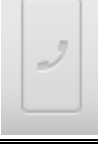



Table 3-1: Call State Table

ICON	DESCRIPTION
	No call is currently active
	Call is ringing. Applies to inbound and outbound calls.
	Call on the primary line is ringing. Applies to inbound and outbound calls.
	Call is active. Audio flows to and from the remote party.
	Call failed. See the System History for details on the call failure.

ICON	DESCRIPTION
	Call is on hold.
	Call has been placed on hold remotely (i.e., by the far side, not by Symphony).
	Calls are forwarded.
	Call is being transferred.
	Call is in a TelePatch.
	Call in TelePatch is on hold.

The Telephony PTT is a multipurpose control that indicates when the call is active and/or in a TelePatch and provides a PTT control when telephony is in half duplex mode. Table 3-2 summarizes the states of the Telephony PTT.

Table 3-2: Telephony PTT States

CALL STATE	HALF DUPLEX	FULL DUPLEX	HALF DUPLEX TELEPATCH*	FULL DUPLEX TELEPATCH
No Active Call			NA	N/A
Ringing			NA	N/A
Active				
Hold				

Click the volume control to open a multi-slider dialog as shown below:



The volume control includes:

- Default phone volume: The default volume used to begin every call.
- Call volume: Volume of the currently active call. This slider is not shown if there is no active call, or the call is a member of a conference.
- Radio volume: Volume of the radio entity in a TelePatch. Only shown when in a TelePatch.

The volume control shows the default volume if no calls are active and the volume of the active call if a call is active. The mute state is shown if any of the sliders are muted. The default sliders cannot be muted.

The mute control mutes the mic audio from the dispatcher as you would expect on a desk phone. The muted state is shown below:



3.1.4 Transmit Indications

Transmit indications provide valuable information about the types of transmissions that are in progress. Symphony supports multiple simultaneous transmissions; therefore, multiple indications are provided as shown in Figure 3-5 and Figure 3-6. Depending on licensing and configuration, your console may not have all of these indications.



The following describes the LEDs when active:

- PTT: Foreground PTT. Typically, the dispatcher mic, but can be foreground Pages.
- Tel: Mic Audio is being sent to the phone.
- Other: Background TX. The VU meter is inactive for background transmissions.



Figure 3-5: Full Screen Transmit Indications



Figure 3-6: Baton Transmit Indications



NOTE

The VU Meter and PTT LED can be used to troubleshoot cabling or configuration issues. If voice transmission is initiated and the PTT LED illuminates, but the VU Meter does not show activity, then the problem is likely with the mic, mic cable, or mic configuration. If the VU Meter shows activity, but the PTT LED does not illuminate, then the problem is likely with the PTT switch.

3.1.5 Select Module

Figure 3-7 is an example of the Select Module as displayed in the Select Bar.

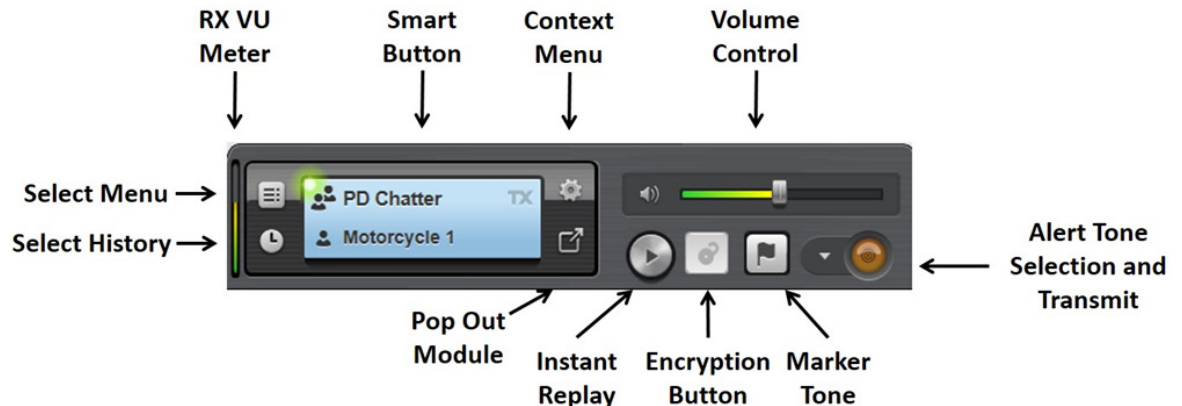



Figure 3-7: Select Module

3.1.5.1 Receive VU Meter

The Receive VU Meter provides a visual indication of the select receive audio level. Since the select module can support multiple audio streams simultaneously, the Receive VU Meter shows the sum of all select audio including received calls and loser audio on the select group or SimulSelected groups. The meter does not include unselect audio directed to speaker 1, such as Priority Groups, Discrete Listening, Radio Unit Monitor, local tones, or call playback.

The meter moves up and changes color as the audio level increases. Red indicates that the audio level is high and may be distorted. The meter can be configured to show the audio level before or after Automatic Level Control. Check with your administrator to determine how your console is configured. The meter does not move if the select module is muted.

3.1.5.2 Select Menu

Click  to open the Select Menu. The Select Menu provides a convenient way to change the select group to the Default Select entity, Preferred Select entity, or a recent entity that has occupied the Select Module.

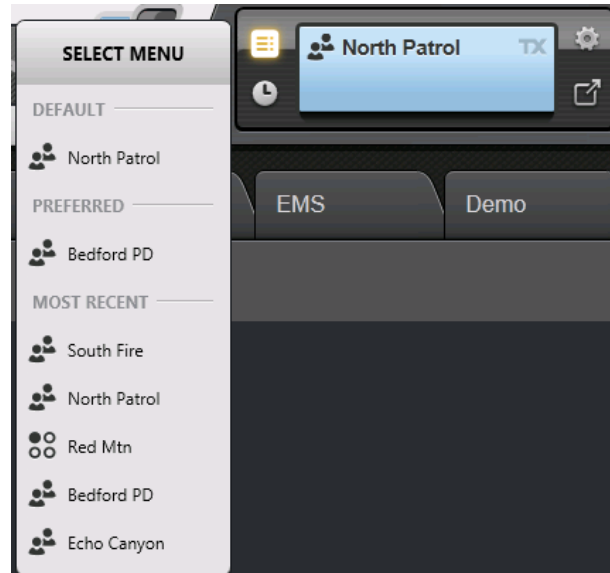


Figure 3-8: Select Menu

- The Default Select Entity (DSE) is set in the Directory (see Section 3.5.1). Only one DSE is allowed. DSE can be set by the administrator and the administrator may not allow the console user to change it.

When a Default Select Entity (DSE) is configured:

- The Select module will contain the Default Select Entity (DSE) on startup. If the DSE is part of a remote Patch, then the Patch is selected.
- DSE is stored per setup. The DSE for a setup is selected when the dispatcher changes to that setup.
- When a Patch is deactivated, Symphony normally selects the previously selected entity. If the previously selected entity cannot be selected, then Symphony will attempt to select the entity before that. If the list of previously selected entities is exhausted, the DSE is selected.

When a Default Select Entity (DSE) is not configured:

- The Select module contains the most recently selected entity group or unit on startup. The console will not activate a SimulSelect automatically on startup. If the most recent selected entity is a group that is now in a remote Patch, the Patch is selected.
- If a Patch is torn down while selected, the console will select the most recently selected group or unit. If the group is in a Patch, then that Patch is selected.

See Section 5.2.3 for more information on the Default Select Entity.


- Preferred Select entities are set in the Directory. The number of Preferred Select entities allowed is configurable by the administrator.

- The last few entities that have been selected are available from the Select menu. The number of recent Select entities visible is configured by the administrator.



If you cannot designate an entity “As Default Select” in the Directory dialog, the feature has been disabled by the administrator.

3.1.5.3 Select History

Click  to open the Select History (Figure 3-9 and Figure 3-10). The Select History provides the user with a list of the most recent calls which have occurred on the entity that currently occupies the Select Module. It also displays calls that have occurred on the previously selected entities when they were selected with one exception; Advanced Backup Radio Select History is separate from the Select History of all other entities. The number of calls displayed is configured by the administrator. The user can navigate between the call back controls and playback controls within the Select History dialog. See Section 5.6 for more information on Audio Playback.

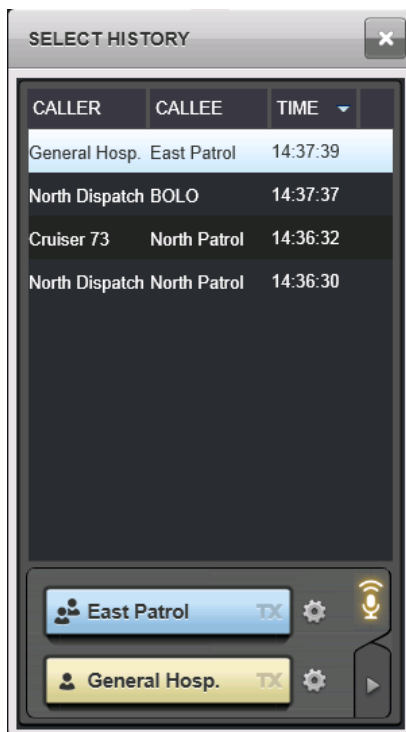


Figure 3-9: Select History – Callback Controls

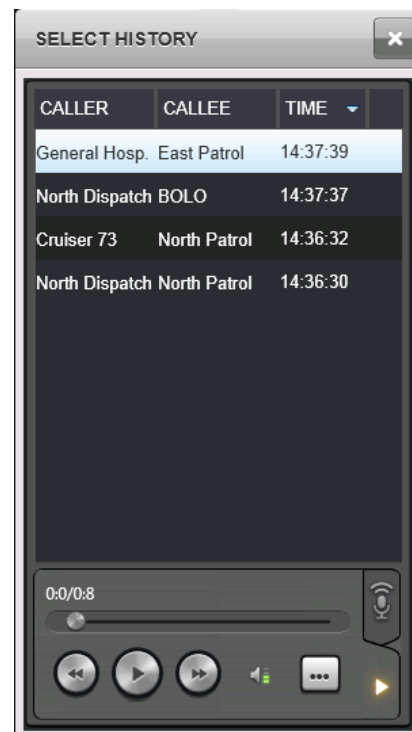


Figure 3-10: Select History – Playback Controls

3.1.5.4 Instant Replay

The Instant Replay button replays the last incoming call on the selected entity.

3.1.5.5 Alert Controls

The Alert controls allow the user to choose and transmit an alert tone to the entity. See Section 5.11.6 for more information on alert tones.

3.1.6 Speaker Buttons

The number of speaker buttons displayed is determined by how many speakers are physically connected to the console [maximum of four (4) can be displayed on GUI]. The buttons have two elements. The top part shows the speaker number. The bottom part of the button shows the activity on the speaker. When there are no calls, the activity bars remain gray. If there is one active call, the bottom activity bar lights up green. If there are two active calls, the bottom two activity bars light up green. If there are three or more active calls, all three activity bars light up green. Click a speaker button to open a limited call history of all entities assigned to the speaker (see Figure 3-11 and Figure 3-12). From here, the console user can call back the highlighted entity or playback the call for the highlighted entity on the Playback tab. In the Full Screen Application, the Speaker buttons are available in the Select bar. In the Baton, the administrator may configure them to be available on the application.

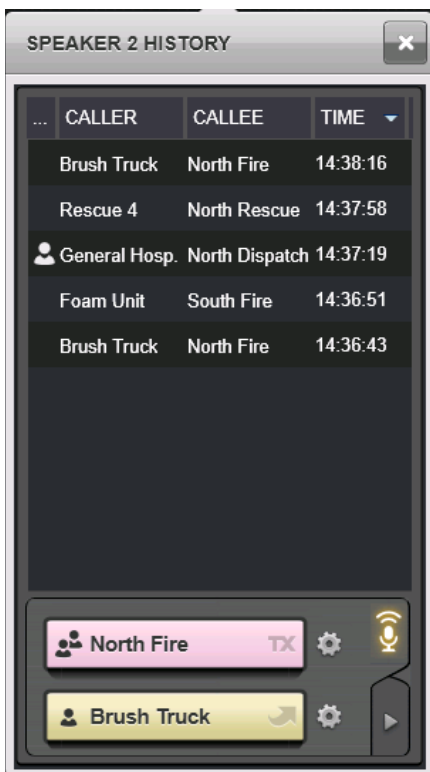


Figure 3-11: Speaker History – Callback Tab

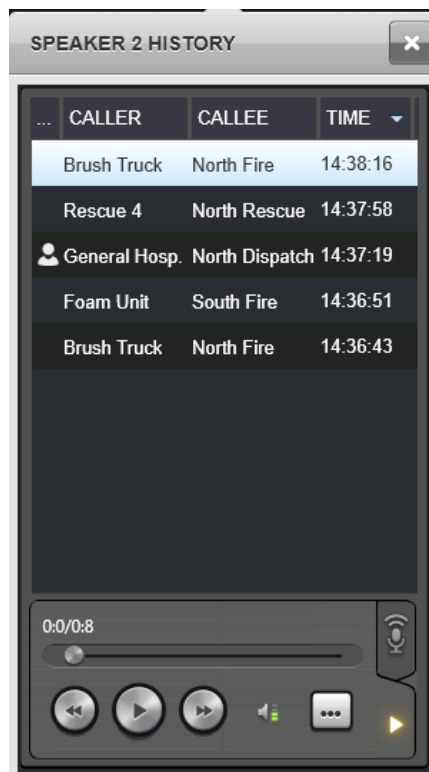


Figure 3-12: Speaker History – Playback Tab

In addition to the Speaker Buttons, this area of the Select Bar also has a button to mute all unselect entities and a button to mute audible tones.



Mutes everything except select audio and the Advanced Backup Radio Module.

Clicking this button starts a countdown timer ²⁹ UNSEL. If you do not manually unmute the modules by clicking this button again, the mute times out based on an administrator-defined time. Click the button again to manually unmute.



Mutes audible tones. Only applies to tones sounding at the time the button is clicked. You will hear all new incoming tones (e.g., emergency). Click the button again to silence them.



Level: Sets the volume level for audible tones including telephony ringing. This control is not available on the main Baton. It must be accessed from the Baton main menu.

3.1.7 Clock Panel

The clock panel displays the current time. The time is displayed in hours: minutes: seconds. The time format, whether 12 or 24-hour (default), is configured by the administrator.

3.1.8 Dial Control

The dial control allows you to place a phone call to the party highlighted in the associated dialog or panel. This control is used in multiple locations including the SIP Telephony Sidebar Panel and the Extended History dialog. If only one line is supported, the control is just the **Dial** button. If multiple lines are supported, the control includes the **Dial** button and the line selector. Pressing the line selector opens the line selector menu which lists the available lines, indicates the current and primary line, and shows the current number of calls on each line. Use this menu to select the desired outbound line. Once the line is selected, press the **Dial** button to place the call.

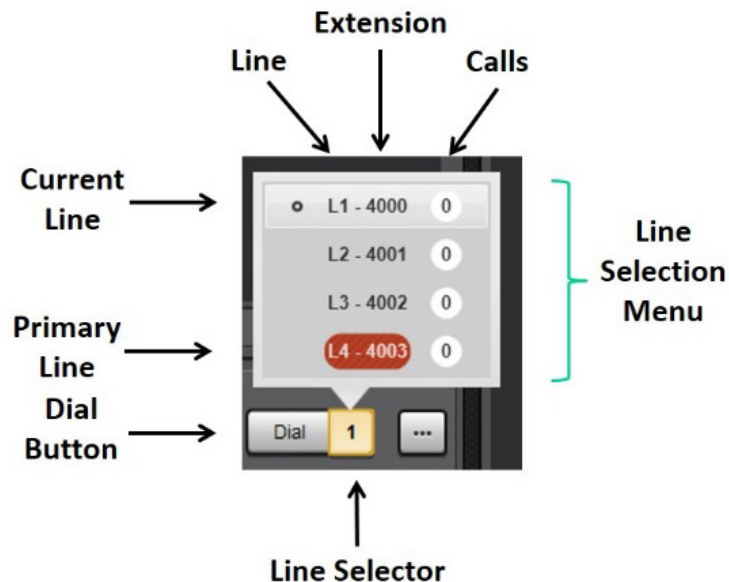


Figure 3-13: Line Selector

3.2 COMMON STATUS ELEMENTS

This section defines the common status elements that are available on the Full Screen Status Bar and on the Baton. Figure 3-14 shows the placement of these elements on the Baton and Status Bar.

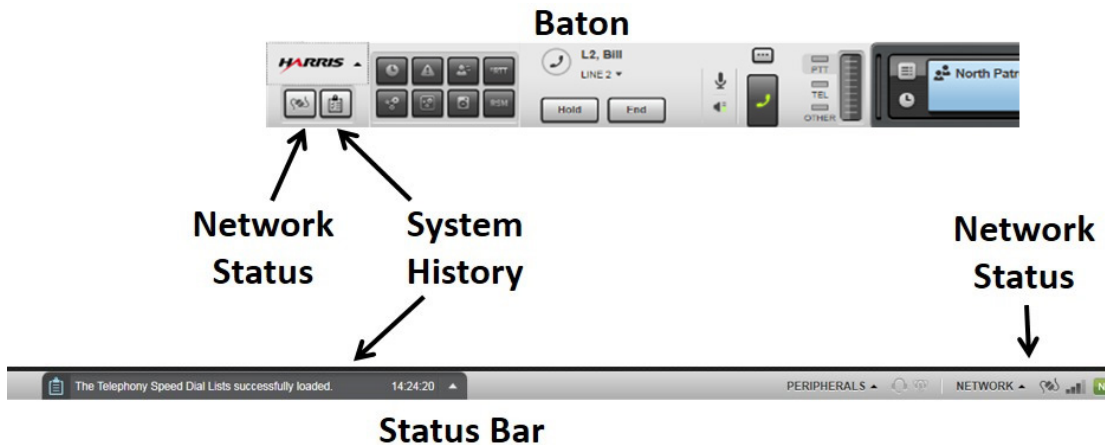




Figure 3-14: Common Status Elements

3.2.1 System History

The System History dialog displays information about system and console tasks. It lists informational messages such as “VNIC is now connected” and also errors and failures that come up during the console’s normal operation. On the Baton, click the  button to access the System History in a dialog. On the Full Screen display, the System History appears in the center of the Status Bar; click  to expand (Figure 3-15). In the unexpanded form, the Status Bar only indicates the most recent event in system history. When expanded, it shows the full list of events in the System History. Scroll through the list to see older history entries.

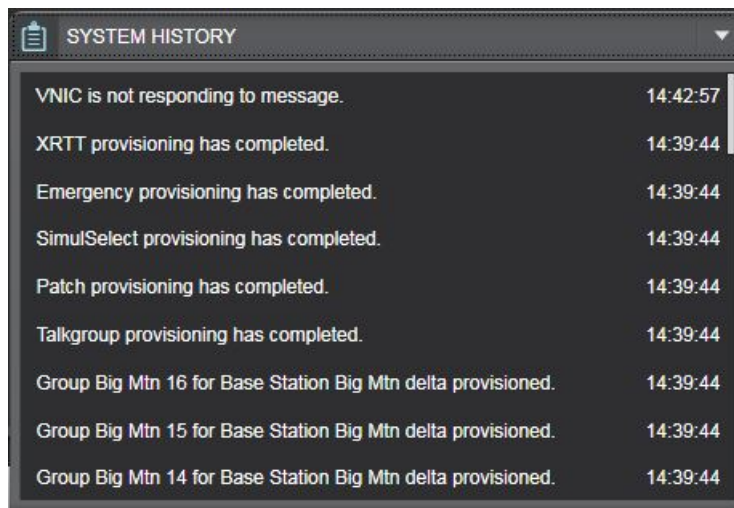





Figure 3-15: System History - Expanded

3.2.2 Network Status

The Network Status Dialog shows information about the current network connection status, FlexPath usage, and connection quality in detail. On the Baton, click  to access the Network Status dialog. On the Full Screen display, click  next to **NETWORK** to expand the Network Status dialog. Next to , there are three indicators that also show the current network status, FlexPath usage, and connection quality. When the network connection is good, the Network Status dialog indicates that it is connected (Figure 3-16).

Contact the administrator if the Network Connection is lost (Figure 3-17), if the FlexPaths have been exceeded (Table 3-3), or the connection quality icon is orange or red (Table 3-4).

During the network connection/reconnection process, visual indications may lag actual radio system audio. Once the “Symphony is now fully connected, provisioned, and ready” message is in the system messages area, the console has reached a steady state connection to the radio system, and the screen indications have caught up to the radio system audio.

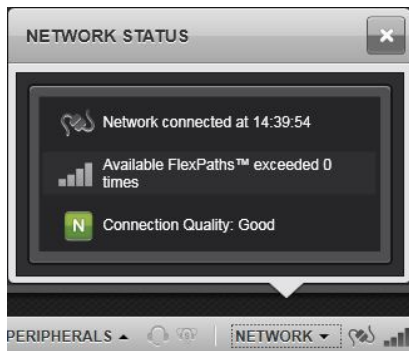


Figure 3-16: Network Status – Expanded

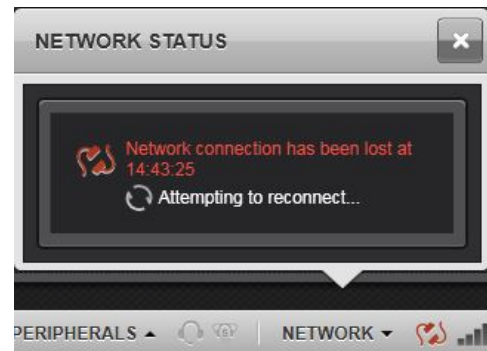


Figure 3-17: Network Status Connection Lost

Table 3-3: Available FlexPaths










ICON	DESCRIPTION
	100% of FlexPaths are available.
	75% of FlexPaths are available.
	50% of FlexPaths are available.
	25% of FlexPaths are available.
	0% of FlexPaths are available.
	FlexPaths exceeded; calls will be missed due to lack of FlexPaths.

Table 3-4: Connection Quality

ICON	DESCRIPTION
	Green: Good Connection
	Orange: Warning
	Red: Serious network problem – contact your administrator

3.3 COMMON COMMUNICATION MODULES

A Communication Module is the fundamental component for communicating through the console. Each Communication Module represents a single entity, whether a talkgroup, a radio unit, a conventional channel, another console, a Patch, a SimulSelect, a page, a base station, or an aux I/O. When an entity is programmed into the Workspace, all audio related to that entity, if applicable, is routed to the console. Modules provide incoming call monitoring and outgoing console-originated call transmissions. The Full Screen display supports three module sizes: Normal, Small, and Mini (see Figure 3-18, Figure 3-19, and Figure 3-20). The Baton supports Mini modules only; however, you can pop-out these modules from the context menu to make them normal sized, moveable modules.

This section provides an overview of the module elements shared by multiple module types. Feature-specific modules and module elements (e.g., tabs and Smart Buttons) are discussed in detail in the associated feature section.

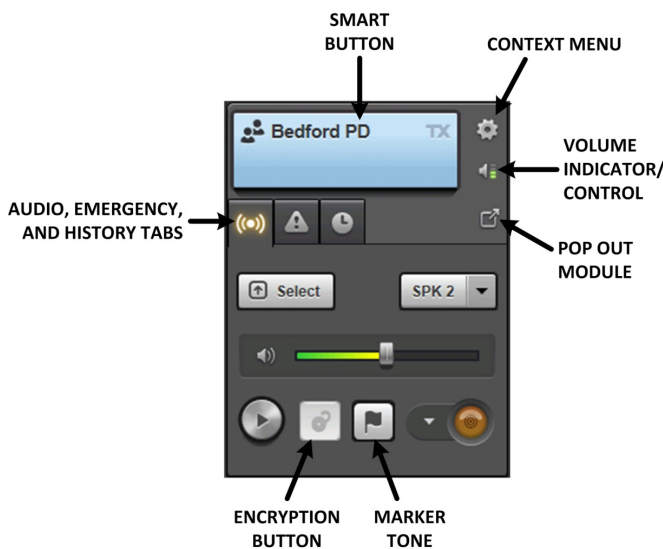


Figure 3-18: Normal Module

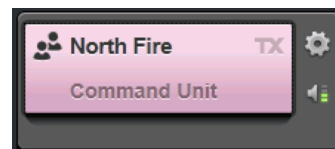


Figure 3-19: Small Module

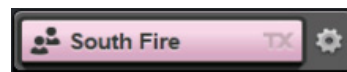


Figure 3-20: Mini Module

3.3.1 Smart Button

Mini Communication Modules have a one-line Smart Button (Figure 3-21). Normal and small Communication Modules have a two-line Smart Button (Figure 3-22). This section provides an overview of Smart Button behavior on modules that communicate directly with an entity, such as a group or user. However, Smart Button behavior may vary for specific kinds of modules such as Paging or Aux I/O modules. Subsequent sections detail module-specific Smart Button behavior that deviates from this section.



Figure 3-21: One Line Smart Button

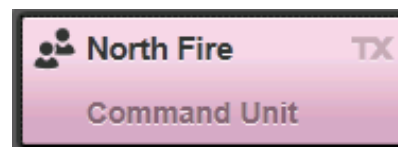


Figure 3-22: Two Line Smart Button

Each type of Smart Button provides slightly different information and controls to the user. Generally, the Smart Button provides the user a way to:

- Transmit to the entity programmed to the module. Click the button to transmit (PTT).
- Determine the type of entity programmed to the module.
- Determine who is transmitting to the console.
- Determine the current action of the module (i.e., transmit, ringing, initiate call, etc.).
- Determine activity of the incoming call.



Smart Button behavior for modules such as Paging and Aux I/O deviates from the generalities described in this section. Details for these modules are covered in later sections.

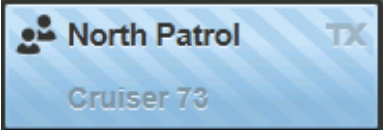
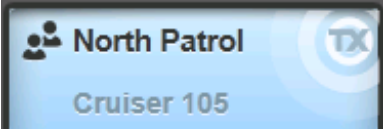
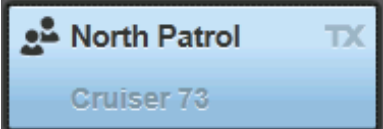
When activity is occurring on the second line of the one-line Smart Button, the button alternates between the first line (entity type icon, entity name, and Action Area) and the second line (Activity Indicator, Information, and State Info). Each is shown periodically until the activity on the second line completes, at which point the first line is shown. When an emergency is active on the module, the entity type icon alternates with the emergency indicator.

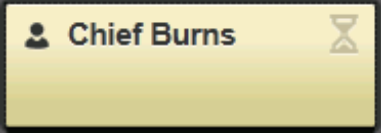


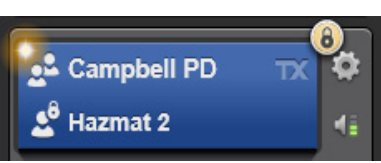
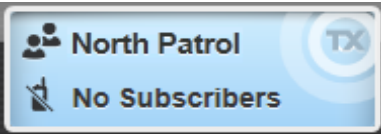
If the Caller ID Display Time is greater than 0, the second line of the One Line Smart button continues to be displayed grayed out for the configured time period (0 to 15 seconds) after the call completes. This time is configured by the administrator. The two-line Smart Button continues to display the last caller after the call ends, but the last caller is grayed out.

The text fields of the Smart Button where the alias or ID is displayed support up to 14 characters. The configuration of the console determines if the Alias or ID is presented on the Smart Button. If the ID is presented, the format for units will be ww-rrrr-aaaa-uuuu, where w = WACN, r = region, a = agency, and u = Unit ID or Group ID. If the ID is greater than 14 characters, it is truncated with three dots (i.e., "...") at the end. The full ID is displayed when the mouse pointer is hovered over the ID. Patches and SimulSelects still display their alias.

The Smart Button's color is configurable by the administrator. Table 3-5 describes the Smart Button's visual appearance associated with each state of the Communication Module:

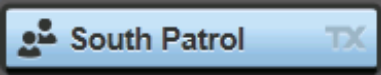


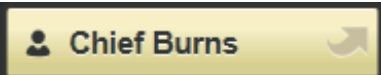

Table 3-5: Smart Button Indicators

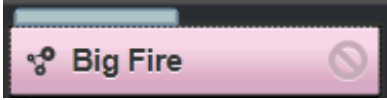
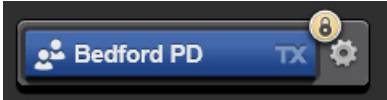

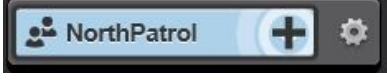
INDICATOR	DESCRIPTION
	<p>Muted: A striped-layer appears behind the content within the Smart Button indicating the muted state. The stripes disappear when unmuted.</p>
	<p>Transmitting: In this state, radiating ripples appear on the top right corner of the Smart Button.</p>
	<p>Idle: When transmission stops, the ripples disappear, sending the Smart Button to the Idle state. The activity indicator and the information area also update accordingly.</p>

INDICATOR	DESCRIPTION
	<p>In Progress: displayed until the outbound call is completely setup. This typically applies to a confirmed call waiting on a ready reply from the VIDA system.</p>
	<p>Unable to TX: displayed when the call placed on the module is busy, queued, or denied.</p>
	<p>Incoming Call: A green dot appears in the top left corner of the Smart button when there is an incoming call. On the two-line Smart Button, the caller's ID is bold and black, and the icon indicating the type of caller blinks next to the ID. On the one-line Smart button, the text alternates between the caller ID and callee ID.</p>
	<p>Incoming Encrypted Call: An orange dot appears in the top left corner of the Smart button when there is an incoming encrypted call. On the two-line Smart Button, the caller's ID is bold. The icon indicating the type of caller includes a lock symbol and blinks next to the caller's ID. On the one-line Smart button, the text alternates between the caller ID and callee ID.</p>
	<p>Outgoing No Subscribers Call: The "No Subscribers" text and icon are displayed in the information area of the Smart Button while the border flashes white. A modified grant tone is played when there are no units listening.</p>


The icon in the top-right corner of the Smart button indicates what action will occur when the button is pressed. Table 3-6 describes what action will occur on PTT when certain icons are displayed.

Table 3-6: Icons

ICON	DESCRIPTION
	<p>Transmit to the entity.</p>
	<p>Activate SimulSelect.</p>
	<p>Activate Patch.</p>
	<p>Ring radio (acknowledged Individual Calls).</p>
	<p>Cannot transmit; the module is receive-only. No action occurs.</p>

ICON	DESCRIPTION
	Indicates Patch or SimulSelect activation failed.
	Transmit Encrypted: A badge containing a locked padlock indicates the module is enabled for encrypted transmission. Note: the badge may or may not appear based on configuration. See Section 3.3.5.7 for details.
	Transmit Unencrypted: A badge containing an unlocked padlock indicates the module is not enabled for encrypted transmission. Note: the badge may or may not appear based on configuration. See Section 3.3.5.7 for details.
	TelePatch Call: A plus sign (+) appears in the top right corner of the Smart button when this entity is TelePatched to the local Call Director. The module is receive-only.

3.3.2 Context Menu

The Context Menu control  is available on all Communication Modules. This control provides quick access to menu options common to most modules such as Select, entity specific options such as Deactivate Patch, as well as informational controls. The exact options in the menu depend on the entity type, the state of the entity, and licensed features.

Options available on most module types:

- Pop Out Module: “Pops out” the Communication Module, making it a normal-sized, moveable resizable module. This provides the console user access to all module functions.
- Open History: Opens the module’s extended call history, filtered to the entity that is programmed to the module (see Section 3.5.2).
- Open Directory: Opens the Directory Dialog, filtered to the entity programmed to that module (see Section 3.5.1).
- Select: Selects the module.
- Activate/Deactivate Marker Tone: Activates or deactivates Marker Tone if the Marker Tone feature is licensed. See Section 7.3 for more details.
- Set Encryption On/Off: Set the transmit encryption state of the module to on or off if the Encryption feature is licensed, keys are properly configured, and the entity is not in an active I-Call. See Section 5.10 for more details.

Options available on specific module types:

- I-Call (Accept, Reject I-Call, End I-Call)
- Patch (Activate Patch, Delete Patch, Deactivate Patch, Edit Patch, Save Patch)
- SimulSelect (Delete SimulSelect, Edit SimulSelect, Save SimulSelect)
- Base Station (Channel, Refresh)
- Paging (Send without SimulSelect)

3.3.3 Volume Indicator/Control

This control indicates the relative volume level of the Communication Module. Click to open a vertical slider volume control. This control also allows the dispatcher to mute the entity.



Figure 3-23: Vertical Slider Volume Control

Close the vertical slider by clicking anywhere else on the Symphony application while it is opened. All other controls on the console, including Smart Buttons are blocked while the vertical slider is visible.

3.3.4 Pop-Out Module

The Pop-Out Module control provides a way to open a dialog containing a normal-size, moveable, resizable Communication Module. This provides the console user access to all module functions. The module is vertically resizable (not horizontally resizable).

3.3.5 Audio Tab (Normal Module)

The Audio tab (Figure 3-24) contains controls used for audio functionality.

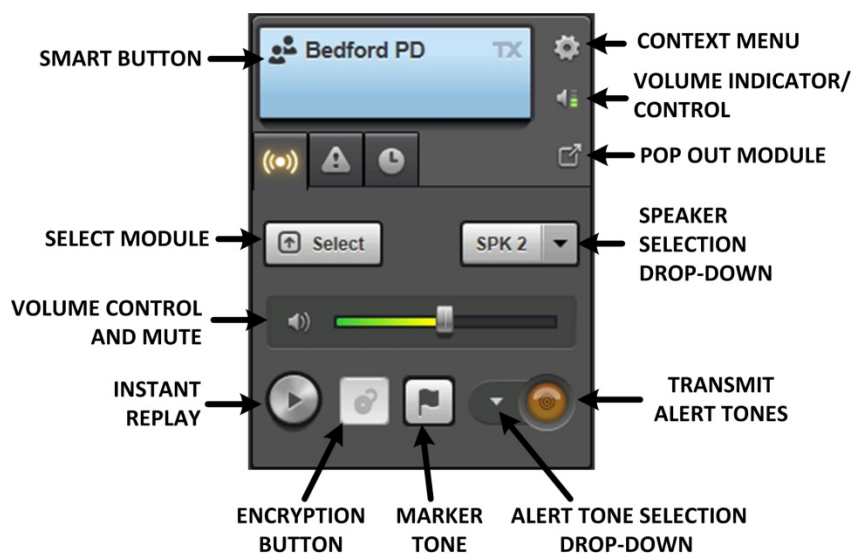


Figure 3-24: Audio Tab

3.3.5.1 Select a Module

The **Select** button selects the module and moves it to the Select Module (see Section 3.1.5). See Section 5.2.2 for more information on Select Modules.

3.3.5.2 Volume Control and Mute




The Audio tab of a normal module has a horizontal slider volume control (Figure 3-25). Click and drag the slider bar to increase or decrease volume for the module. Click the speaker icon  on the slider volume control to mute the module.  and a diagonal striped pattern on the Smart Button indicates the module is muted. Click the  icon to unmute.



Figure 3-25: Volume Control

You can also control the volume and mute the module by clicking the volume indicator as indicated in Section 3.3.3.

3.3.5.3 Speaker Drop-Down

The speaker drop-down allows the user to designate a speaker for incoming audio and associated sidetones if this entity is not selected. If this entity is the selected entity, the speaker drop-down does not change the speaker for the incoming audio since the selected speaker will play that audio. However, when this entity gets unselected, the speaker choice will be respected. See Section 4.1 for more information.

3.3.5.4 Instant Replay

The instant replay button allows the user to replay the last incoming call on the entity.

3.3.5.5 Alert Controls

The alert controls allow the user to choose the type of alert tone from a drop-down and transmit the tone to the module entity.

3.3.5.6 Marker Tone Button

The Marker Tone Button enables and disables Marker Tones as described in Section 7.3. This button is only available if the Priority Marker Tone feature is licensed.

3.3.5.7 Encryption Button

The Encryption Button controls the transmit encryption state of the module. The possible states of the button are shown in Figure 3-26.

- **Unsupported:** The button is grayed out if encryption is not licensed or if the module does not have an encryption key and therefore cannot support encryption.
- **Unencrypted:** The button shows an unlocked padlock to indicate the module is enabled for unencrypted transmission.
- **Encrypted:** The button glows orange and shows a locked padlock to indicate the module is enabled for encrypted transmission.

Press the Encryption Button to toggle between encrypted and unencrypted transmission if encryption is supported.



Figure 3-26: Encryption Button State

3.3.6 Emergency Tab (Normal Module)

The Emergency tab allows the console user to declare, acknowledge, and clear an emergency on the module. For talkgroup and Patch modules, this tab provides a list of all units who have declared an emergency on that entity. For user modules, this tab provides information about the alert such as the location it originated from, the group (if any) associated with the alert and when the alert was declared. When an emergency has not been declared by any of the members, the only control that appears on the tab is the **Declare** button. When an emergency has been declared by a member, the area around the Smart Button turns red and there are controls to acknowledge/clear the emergency declaration.

The emergency tab on the Unit Module and Group Module function differently because they respond to Group Emergencies and Unit Alerts, respectively. See Section 5.11 for more information on emergency operation.

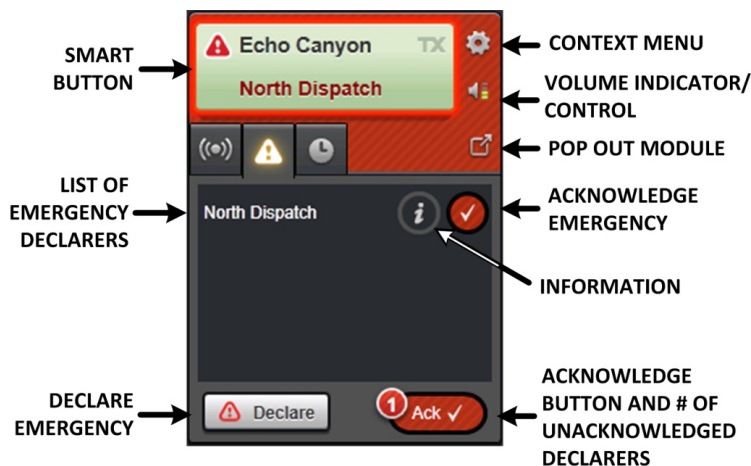


Figure 3-27: Group Module in Emergency

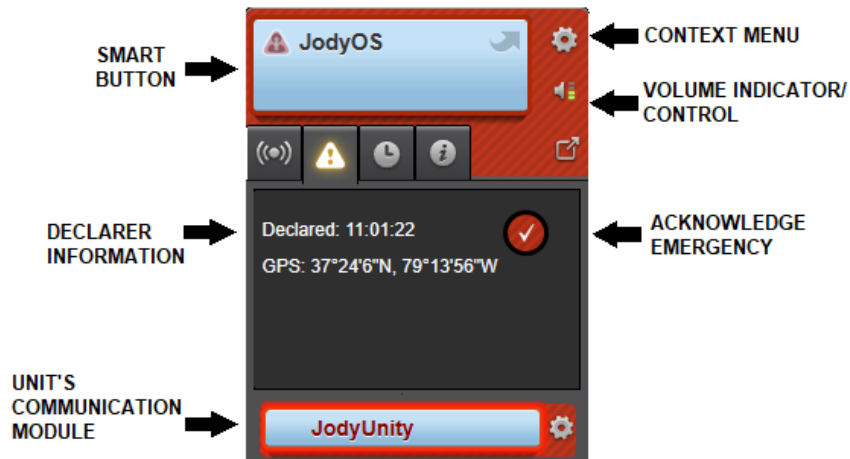


Figure 3-28: Unit Module in Emergency

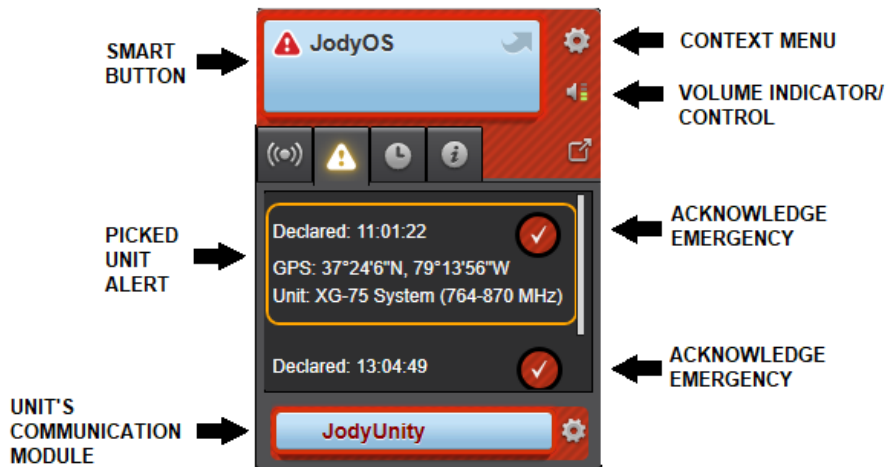


Figure 3-29: Unit Module with Multiple Emergencies

3.3.6.1 Declare (Only Group and Patch Modules)

Press and hold the **Declare** button for one second to declare an emergency on the entity.

3.3.6.2 Information




The Information button allows the user to query information about the entity that declared the emergency. The information includes:

- ID/alias of the declaring user.
- The declaration stamp.
- ID/alias of the acknowledging user.
- The acknowledged timestamp if acknowledged.
- GPS Location of originator if available – (OpenSky® Unit Module only).
- The radio model that declared the unit alert (OpenSky and VIDA ID only).

3.3.6.3 Emergency Declarers (Group and Patch Modules)

This list displays the individuals who have declared an emergency on the entity of the module. The list items include a button to acknowledge the emergency and to query details of the emergency (i.e., show the Information Dialog).

3.3.6.4 Acknowledge Emergency (Group and Patch Modules)

The  and  buttons allow the user to acknowledge an emergency. These buttons are displayed until the emergency is acknowledged. After the emergency is acknowledged,  changes to .

3.3.6.5 Clear Emergency (Group and Patch Modules)

The **Clear** button allows the user to clear the emergency. If the console is configured to allow clearing an emergency without an acknowledgement beforehand, or if all emergencies associated with the module entity have been acknowledged, the **Clear** button appears enabled. This button is not displayed unless there is an active emergency declared.

3.3.6.6 Ack/Clear Unit Alert Button (Unit Module Only)

The **Ack/Clear** button allows the user to acknowledge and clear a unit alert. The button is displayed as **Ack** unless the system has been configured to clear without acknowledge. After the unit alert is acknowledged, the **Clear** button appears. When the user presses the **Clear** button, the user alert information, **Clear** button, and group module are removed. There will be one **Ack/Clear** button for each unit alert declared by a user. This can happen when the user is logged into multiple radios and is supported on OpenSky and VIDA ID enabled systems.

3.3.6.7 Declarer Information (Unit Module Only)

The Declarer Information displays the time, GPS position, and radio model information for each declared unit alert. Not all of this information is available in every case. GPS location is only available for OpenSky units. The radio model information is available for OpenSky and VIDA ID enabled units.

OpenSky and VIDA ID users may be logged into multiple radios at the same time. When unit alerts are declared on multiple radios by the same user, the declarer information area becomes a list. One of the unit alerts is picked (orange border). The Unit's Communication Module changes to the picked unit alerts.

3.3.7 History Tab (Normal Module)

The History tab contains specific call history data for the entity type programmed to the module. The maximum number of entries in the list is configurable (default = 100).

- Group – the group history contains the call history and emergency declare history for this group.
- Individual – the individual history contains the call history for the unit and the local console, and the emergency declare history for the unit.
- Patch – contains the call history and emergency declare history for all the groups associated with the Patch while the Patch is active.
- SimulSelect – contains the call history for all the groups and units associated with the SimulSelect while the SimulSelect is active.

- Base Station – contains the call history for channels.
- Advanced Backup Radio – contains the call history and emergency declare history that has occurred on the ABR's selected talkgroup.

See Section 5.5 for more information on Call History.

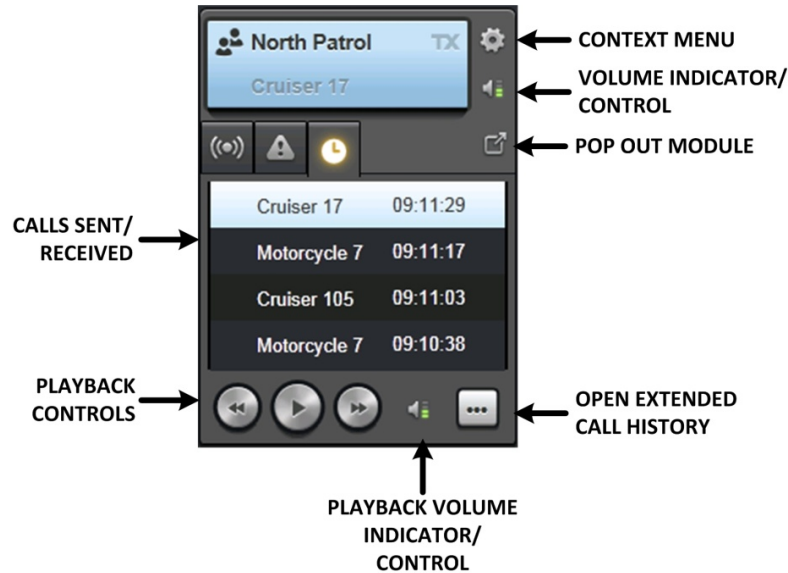




Figure 3-30: History Tab

3.3.7.1 Calls Sent/Received

The Calls Sent/Received list can be configured by the administrator to only display calls received on this module, or to display transmitted and received calls. This list includes the alias/ID of the caller and the timestamp of the call.

3.3.7.2 Open Extended Call History

Click  or click  and choose **Open History** to view the extended call history. The entity associated with this Communication Module is used as the history filter and context. See Section 3.5.2 for more information.

3.3.7.3 Playback Controls/Volume

Highlight one or more calls from the list to activate the playback controls. See Section 5.6 for more information on using the Playback Controls.



Adjusting the playback volume on one module changes it for ALL modules.

3.3.8 Information Tab (Individual Module)

The Information tab shown in Figure 3-31 provides information about the individual including the user alias and ID, as well as controls to communicate with the individual by means other than an I-Call.

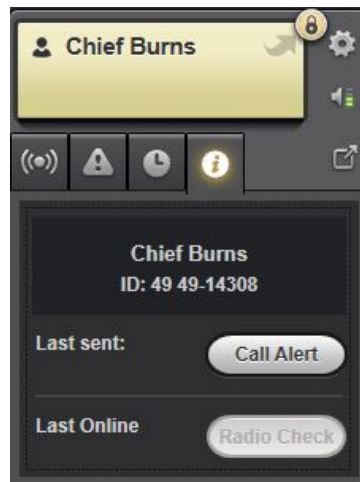


Figure 3-31: Information Tab

3.3.8.1 Call Alert

The Call Alert feature sends a message to a user's radio to alert the user that dispatch is trying to reach him/her. The radio rings or sounds a distinctive tone to get the user's attention, though other actions may be available through radio programming.

Press the **Call Alert** button to initiate the message. If the radio is currently registered on the system, the text to the left of the button indicates the time the message was sent. If the radio is not registered on the system, the text will indicate the message failed to reach the user.

3.3.8.2 Radio Check

The Radio Check feature sends a message to a user's radio to verify the radio is connected to the VIDA system and is reachable.

Press the **Radio Check** button to initiate the message. If the radio is currently registered on the system and can be reached, the text to the left of the button indicates the time the message was sent. If the radio is not registered on the system or cannot be reached, the text indicates the message failed to reach the user.

3.3.9 Members Tab (Patch/SimulSelect Modules)

The Members Tab is only available on Patch and SimulSelect Modules. This tab displays all entities included in the Patch/SimulSelect. Click the **i** button to view detailed information about the Patch/SimulSelect and click **Edit** to modify the Patch/SimulSelect.

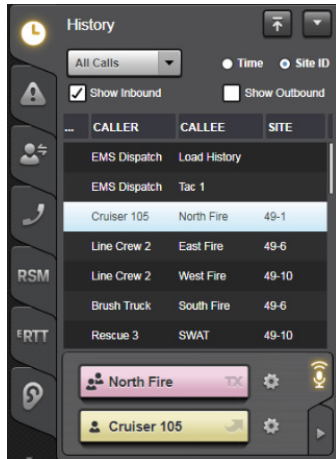


Figure 3-32: Patch Module Members Tab

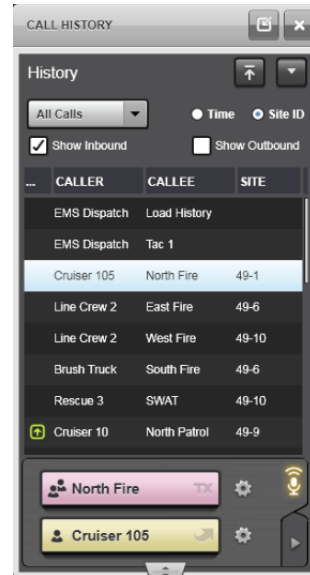
3.4 COMMON PANELS

Panels provide a summary of relevant activity, alert the dispatcher to any activity requiring action, and provide the data and controls necessary to take action. Panels are organized by feature, providing an overview of important activity that may be distributed across multiple modules or workspace tabs. In the Full Screen Display, the panels are typically located on the left side of the screen and are referred to as Sidebar Panels. On the Baton, panels are hidden to save space. The dispatcher presses a control to pop out the panel when needed and to hide it when not in use. Baton panels are referred to as Popup Panels.

Figure 3-33 shows the history panel for the Full Screen Display and Baton. The core elements are the same, they just have different wrappers and are accessed by tabs on the Full Screen, and buttons or menu on the Baton. This manual only describes one version of a panel, typically the side panel, but both versions of the panel work the same way unless specified otherwise.




**Full Screen Display
Sidebar Panel**



**Baton
Popup Panel**

Figure 3-33: Sidebar Panel vs. Popup Panel

3.4.1 History Sidebar Panel

The History Sidebar Panel shows calls sent or received, with the most recent calls displayed at the top of the list. Click  to access the History Sidebar Panel from the Baton.

Specify how to filter calls displayed from the filter drop-down: All Calls, Select Only, or Unselect Only. Check the appropriate box to specify Inbound or Outbound calls.

Choose an entry from the list and click the mini-module's Smart Button to call the highlighted entity. The Display Time/Site ID option allows you to display the time of calls in the history or the Site ID.

See Section 5.5 for more information on Call History.

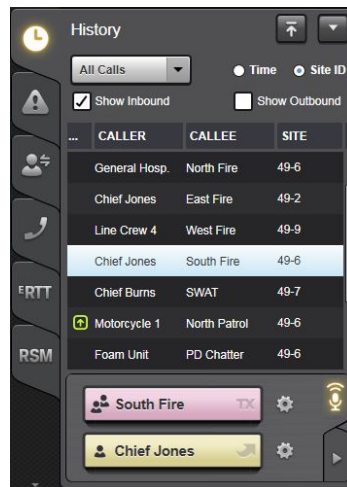


Figure 3-34: Call History Sidebar Panel

3.4.1.1 Menu Drop-Down

The options available from the drop-down menu are:

- Open Directory – Opens the Directory dialog. See Section 3.5.1 for more information.
- Open History – Opens the History dialog. See Section 3.5.2 for more information.

3.4.1.2 Filter Drop-Down

Allows the user to filter the Call History list using the following criteria:

- All Calls
- Select Only
- Unselect Only

3.4.1.3 Inbound/Outbound Checkboxes

Show Inbound – When checked, shows calls from everyone else.

Show Outbound – When checked, shows calls from this console.

3.4.1.4 Display Time or Site ID

These options determine whether the call time or Site ID is displayed in the fourth column.

3.4.1.5 Call History Entries

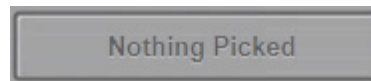
The Call Entries list is a subset of all incoming calls received by this console. The number of calls displayed is configurable (default = 100).

3.4.1.6 Context Menu

The context menu provides access to additional functionality not provided by the mini Communication Module.

3.4.1.7 Callback Tab

The Callback tab provides the console user with a mini Communication Module to call back the selected entity from the call history list. When no entry is selected from the call history list, the mini module appears as shown below:



Generally, call back modules show the current state of the module, not the state of the module at the time of the highlighted call. For example, if the highlighted call was an encrypted group call, but the group is currently unencrypted, the group call back module shows the current state of the group as unencrypted. If the module state differs from the received call and you want to reply with the same module state, pop-out the module from the context menu and change the module to the desired state before replying.

3.4.1.8 Playback Tab

Highlight one or more calls from the list to activate the playback controls. See Section 5.6 for more information on using the Playback Controls.

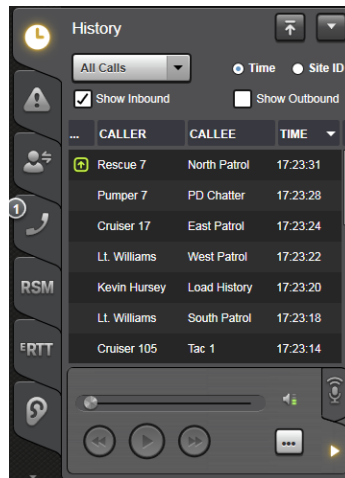



Figure 3-35: History Sidebar Playback Tab

3.4.2 Emergency Sidebar Panel

The Emergency Sidebar Panel displays all active emergencies and user alerts from all monitored entities. If the “Route Unprogrammed Emergencies” option is enabled, then all emergencies in the system are displayed in this panel. Click  to access the Emergency Sidebar Panel from the Baton. A “badge” on the tab or Baton button indicates the number of active emergencies (see Figure 3-36).

From this tab, you can acknowledge and clear the emergency, or you can transmit on the emergency group, no matter which Workspace tab is currently selected.

Refer to Section 5.11 for more information on emergency operation.

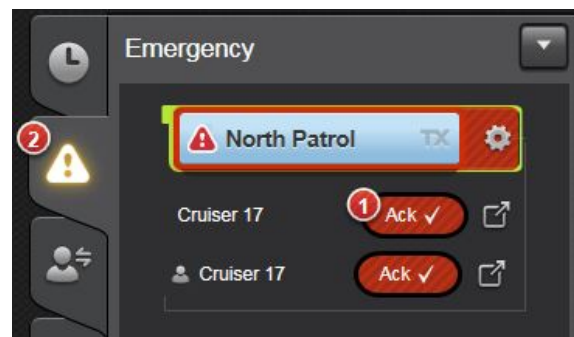


Figure 3-36: Emergency Sidebar Panel

3.4.2.1 Menu Drop-Down

The options available from the drop-down menu are:

- Open Directory – Opens the Directory dialog. See Section 3.5.1 for more information.
- Open History – Opens the History dialog. See Section 3.5.2 for more information.

3.4.2.2 Context Menu

The context menu provides access to additional functionality not provided by the mini Communication Module.

3.4.2.3 Emergency Badge

When there are unacknowledged group emergencies or unit alerts in the Emergency Panel or on a Workspace tab, a numeric badge appears on the panel and Workspace tab. The number indicates the total number of active (i.e., not cleared) group emergencies and unit alerts in the panel or on the Workspace tab. As emergencies and alerts are acknowledged, the number is reduced. The badge is displayed until the emergency is cleared.

3.4.2.4 Declarer Information


The name/alias of the entity declaring the emergency or unit alert is displayed here.



3.4.2.5 Action Button

The action and text displayed before pressing the button depends on the emergency type and the emergency state of the entity.

- Group Emergencies: When **Ack** with a checkmark and a number indicator is displayed, pressing this button acknowledges all unacknowledged declarers. If all of the declarers have been acknowledged or the system has been configured to allow bypassing acknowledge, the **Clear** button appears.
- Unit Alerts. When **Ack** with a checkmark is displayed, pressing this button acknowledges the unit alert and the **Clear** button appears.

3.4.3 I-Call Sidebar Panel

The I-Call Sidebar Panel displays a list of individual calls that are in progress, as well as a log of previous calls. Click  on the Baton to access the I-Call Sidebar Panel. A bell badge on the I-Call Sidebar tab or Baton button indicates an incoming acknowledged call. A number badge on the I-Call Sidebar tab or Baton button indicates the number of unique individuals with unanswered calls. That is, multiple missed calls from the same individual are only counted once on the badge.

Click  to direct-dial a contact or click  to choose a pre-defined contact from a list. The remainder of this section provides a brief overview of the I-Call Sidebar Panel organization and controls. See Section 5.4 for details on communicating with individual calls.

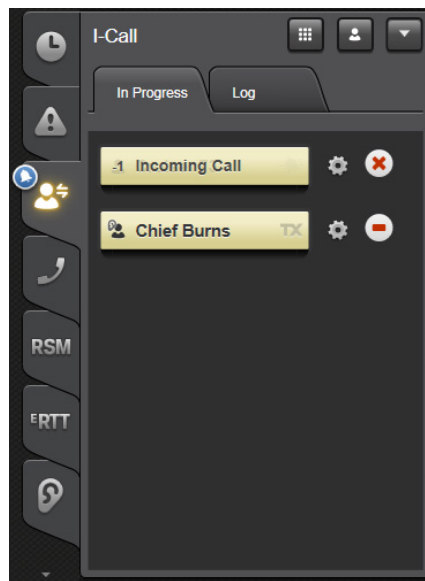


Figure 3-37: I-Call Sidebar – In Progress Tab

3.4.3.1 Menu Drop-Down

The options available from the drop-down menu are:

- Open Directory – Opens the Directory dialog. See Section 3.5.1 for more information.
- Open History – Opens the History dialog. See Section 3.5.2 for more information.

3.4.3.2 Keyboard Control

The Keyboard control, shown in Figure 3-38, allows the console user to start an individual call by entering the callee unit ID. After entering the unit ID, the **Start/Load** button changes to **Start** or **Load** for acknowledged or unacknowledged I-Calls respectively. Set the encryption state of the call by clicking the encryption button. See Section 5.10 for more details on encryption. Click the

Start/Load button to close the keyboard and start the I-Call. See Section 5.4 for details on initiating acknowledged and unacknowledged I-Calls.



Figure 3-38: I-Call Keyboard Control

The User ID follows the P25 format ssss-uuuuuuuu where ssss is the P25 System (which is equivalent to the VIDA Region) and uuuuuuuu is the P25 unit (which is equivalent to the VIDA agency concatenated with the VIDA 4-digit unit [aaaa-uuuu]). The P25 Wide Area Communications Network (WACN) identifier is not used since I-Calls to foreign systems are not currently supported.

If the unit has the same P25 system (VIDA Region) as the console, only enter the unit. Leading 0s are not required. When entering a VIDA agency + unit, leading 0s are not required for the agency, but are required for the unit. For example, agency 14 and unit 102 would be entered as 140102. If the agency is 0, it can be omitted. The previous example with agency 0 could be entered as 102.

If the unit has a different P25 system (VIDA Region) than the console, enter the system, hyphen, unit. In the example above, if the system was 17, the unit would be entered as 17-140102.

If the user ID is known, the alias is shown below the ID. This is a good way to verify the correct ID has been entered.

3.4.3.3 Contact List

The Contact List, shown in Figure 3-39, allows the console user to initiate an Individual Call from a list of contacts. This list supports searching and filtering. To place a call, highlight a user in the list. The Start/Load button changes to **Start** or **Load** for acknowledged or unacknowledged calls respectively. Set the encryption state of the call by clicking the encryption button. See Section 5.10 for more details on encryption. Click the **Start/Load** button to close the Contact List and start the I-Call. See Section 5.4 for details on initiating acknowledged and unacknowledged I-Calls.

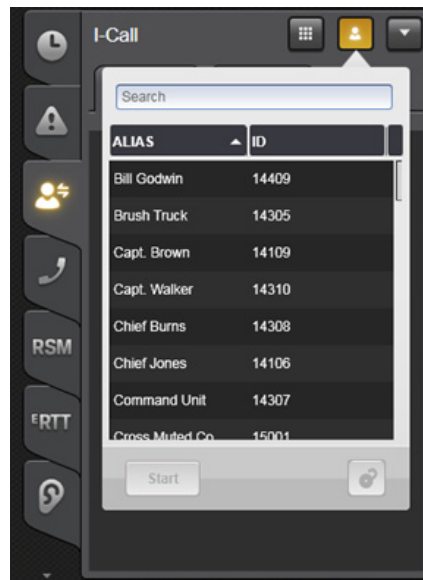



Figure 3-39: I-Call Contact List

3.4.3.4 In Progress Tab

The In Progress tab contains a mini module for each individual from which an I-Call has been received. The I-Call mini module style is slightly different than workspace mini modules, but it contains the same Smart Button and Context Menu controls. Press the appropriate Smart Button to respond to an active call or initiate a new call to a previous caller. The mini modules remain in the In Progress Tab until the Delete Icon  is pressed.

3.4.3.5 Log Tab

The Log tab, shown in Figure 3-40, provides a list of individual calls that have been initiated or received since the application last started. Each entry includes the call direction, unit alias, start time and an info icon which reveals detailed call information when clicked. The length of time that calls are maintained in the log is configurable with a default of 7 days.

The initial missed I-Call from each caller increments the Missed Call Counter badge on the I-Call panel tab. Multiple missed I-Calls from the same caller only increment the counter once. The latest missed call from each individual is denoted with a blue border. Consecutive entries with the same direction and unit are consolidated into one entry with the number of duplicate entries in parentheses following the individual's alias. For example, if Chief Burns called two times in a row, the entry would be Chief Burns (2). Click the info icon to see details about each consolidated call.

The Missed Call Counter decrements when a missed caller is called back or when the missed individual is dismissed from the log. To dismiss an individual, highlight the latest call from the individual in the log (with blue border) and press the blue Dismiss Notification Button. Dismissing the call also stops the Missed I-Call Alarm.

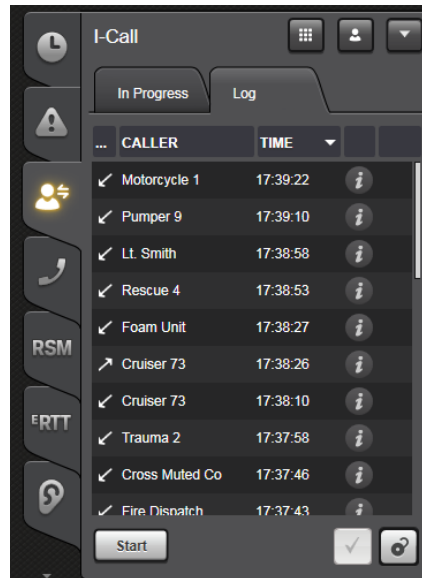


Figure 3-40: I-Call Log Tab

To start a call from the log, highlight a user from the list. The Start/Load button changes to **Start** or **Load** for acknowledged or unacknowledged calls respectively. Set the encryption state of the call by clicking the encryption button. See Section 5.10 for more details on encryption. Click the **Start/Load** button to start the I-Call. See Section 5.4 for details on initiating acknowledged and unacknowledged I-Calls.

3.4.4 Call Director Sidebar Panel

The Telephony Sidebar Panel allows the dispatcher to view and control active telephony (Call Director) calls. The panel consists of two tabs, the History Tab which displays records of each telephony call placed or received and the Active Tab which shows information and controls for the current active telephony call.

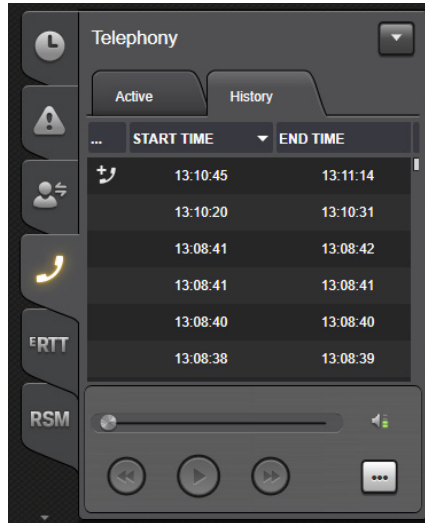


Figure 3-41: Telephony Sidebar Panel

Active Tab:

When there is an active Call Director call, the Active tab will show an entry identifying the call information. The call information includes three columns of information, LINE, CALL INFO, and STATUS. LINE identifies the line number that the call is on (L0 for Call Director calls) and CALL INFO identifies the type of call, e.g., Call Director. STATUS will show the amount of time that the call has been active and also if the call is currently TelePatched to the radio system.

History Tab:

The History tab in the Telephony Sidebar Panel displays all Call Director calls that have been made or answered from the Symphony dispatch console and allows the user to play back any of these calls. Highlight one or more calls from the list to activate the playback controls. See Section 5.6 for more information on using the Playback Controls.

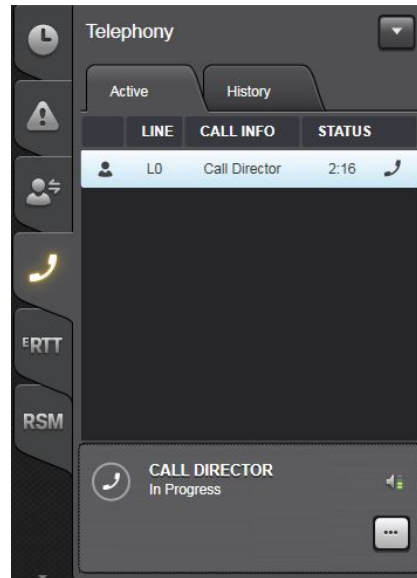



Figure 3-42: Active tab

When it is required that an active call be TelePatched to another entity (a Talkgroup, radio Patch or Base Station) the More button  can be used to open the Call Director TelePatch panel. Reference the Call Director operation section for more detail.

3.4.4.1 Call Director TelePatch Operations

Symphony provides the ability to connect the local call director to one or more “radio” entities, thus allowing an incoming caller on the phone system to communicate directly with radio end users in the field. This feature is known as Call Director TelePatch. The types of entities which may be patched to the phone in this manner are Talkgroups, Base Station channels, and active radio Patches. The entity being patched to the phone may be either selected or unselected. During TelePatch operations, normal dispatch operations can continue as normal. For example, while TelePatched to an unselect entity, if a module for a different entity is selected, transmits may be performed via the Select Module Smart Button or the footswitch or the desk mic’s PTT switch while communications with the TelePatch is accomplished using the CD PTT button in the Select Bar.

3.4.5 SIP Telephony Sidebar Panel

The Telephony Sidebar Panel, shown in Figure 3-43, provides the required controls and indications to place, receive, and manage SIP telephony calls. The panel comprises a set of global controls and several feature tabs. The **Active** tab shows a list of telephony calls that are currently in progress, and the **History** tab displays records of each telephony call placed or received.

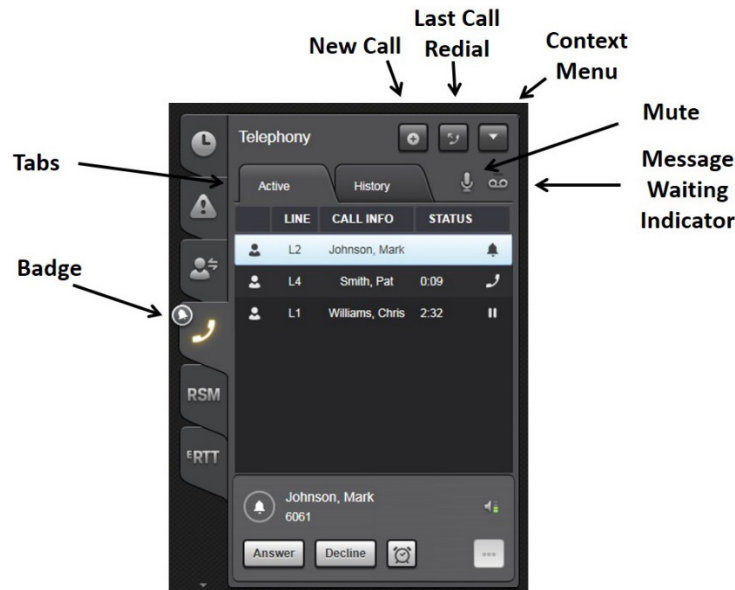
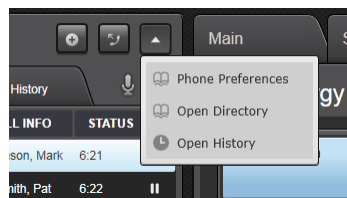


Figure 3-43: Telephony Sidebar Panel Global Controls

The Telephony global controls are shown in Figure 3-43 and include the following:

- **New Call Button:** This button opens or closes the New Call popup which is used to initiate calls. It includes contact lists and a dial pad.
- **Last Number Redial:** Pressing this button dials the last number dialed by the dispatcher.
- **Mute:** This control mutes the mic for the entire telephony interface (i.e., not per line). This is intended to be a temporary mute for a quick private conversation. Place the call on hold to mute a line for an extended period of time.
- **Context Menu:** This menu includes the following menu items:



- **Phone Preferences:** Opens a telephony preferences dialog to set telephony preferences like Call Forwarding. This dialog also provides detailed information about each line.
 - **Directory:** Opens the Contacts tab in the Directory Dialog.
 - **History Button:** Opens the Phone Calls tab in the Extended History Dialog.
- **Message Waiting Indicator.** This control comprises an LED and the voicemail icon . The LED indicates at least one message is waiting for at least one line when the LED is active.

The icon indicates at least one line has been configured for voicemail when active (i.e., not greyed out). If only one line is configured and voicemail is configured for it, clicking on the icon calls voicemail. If multiple lines are configured, then clicking on this icon provides a menu that shows the voicemail state of each line. For each line, the line number is greyed out if the line is not configured for voicemail and the LED is active if the line has at least one message waiting. Click on a line in the menu to call to the appropriate voicemail.



- Badge: Shows a bell when a new call is ringing and the number of active calls otherwise. If there are no active calls, the badge is not present.
- Tabs:
 - Active Tab: Lists the current active calls.
 - History Tab: Lists call history including dialed and received calls.

Details of each tab and the New Call Panel are covered in subsequent sections.

3.4.5.1 Telephony Sidebar Panel Active Tab

The Active Tab provides controls and indications to manage active calls. It has two major sub components as shown in Figure 3-44: The Active Call List (ACL) and Call Control Area (CCA).

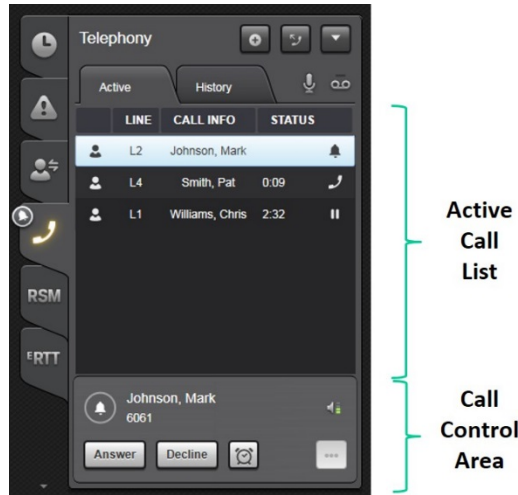


Figure 3-44: Telephony Panel - Active Tab





The Active Call List (ACL) shows all the active calls and their current state. The highlighted call in the ACL is loaded in the Call Control Area providing all controls necessary to manage the call.

Active calls include unanswered ringing calls and calls that have been answered including those on hold. Each active call has an entry in the active call list with the following information from left to right:

- Icons: The left most column provides an icon to identify the call type. These include:
 - - Single Party calls.
 - - Conferenced calls.
- Line: This column indicates the line associated with the call (e.g., L1 for line 1). The primary line is indicated by a red circle in the background.
- Call Info: This column contains the phone number or alias of the caller/callee.
- Status: This column includes a call state icon from Table 3-7 and call timing information. The timer information indicates the call duration when the call is active and the hold time when the call is on hold. The time format is mm:ss for times under an hour and hh:mm for times an hour or greater in length (hh = hour, mm = minutes, and ss = seconds).

Table 3-7: ACL - Call State Table

ICON	DESCRIPTION
	Call is ringing. Applies to inbound and outbound calls.
	Call is active. Audio flows between the dispatcher and remote party. That is, you can speak to and hear the remote party.
	Call failed. See the System History for details on the call failure.
	Call is on hold.

ICON	DESCRIPTION
	Call has been placed on hold remotely (i.e., by the far side, not Symphony).
	Call is being transferred. Depending on your system, the transfer may occur so quickly that this icon does not appear.
	Call is in a TelePatch.
	Call in TelePatch is on hold.

The highlighted call in the ACL is loaded in the Call Control Area providing the controls necessary to handle the call. The CCA is similar to the Compact Telephony Control but provides more call control features.

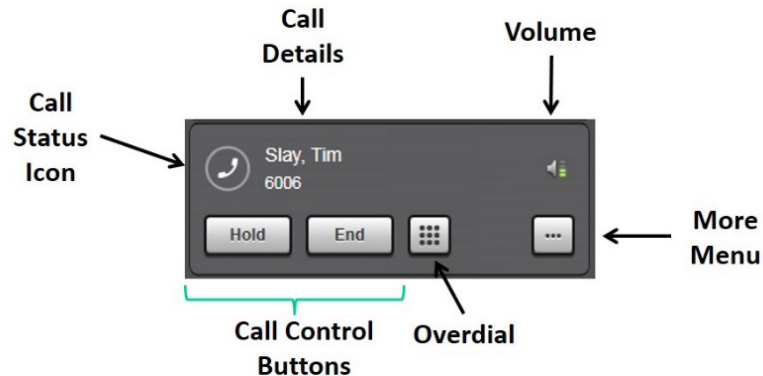

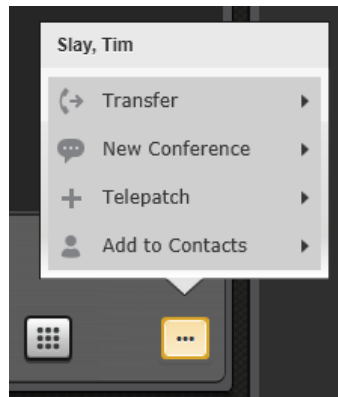


Figure 3-45: Call Control Area

The Call Control Area provides the following:

- **Call Status Icon:** This icon shows the current state of the call. The icons are the same as those used in the compact telephony control. See Table 3-1 for details.
- **Call Details:** This field provides information about the caller/callee. The first line shows the alias if available or Unknown Party if an alias is not available. The second line shows the phone number.
- **Volume:** Click this icon to pop out the volume control for the call. This is just like the volume control used on communication modules.
- **Call Control Buttons:** These buttons provide the most used controls for a particular call situation. The buttons change depending on call state. For example, for a ringing call, the buttons are **Answer** and **Decline**; for an active call, the buttons are **Hold** and **End**.

- More Button  provides access to the Advanced Features Menu such as Transfer, conference, and TelePatch via a popup menu. The menu items will depend on the state of the call in the CCA.



- Overdial Popup: During active calls, the Overdial Popup is available to access DTMF keys for overdialing. Numbers exceeding the width of the display scroll out of view.



3.4.5.2 Telephony Sidebar Panel History Tab

Calls to and from the Console appear in the History tab. This tab is used to play back call recordings and call back parties from previous calls. Figure 3-46 shows the tab's primary components.

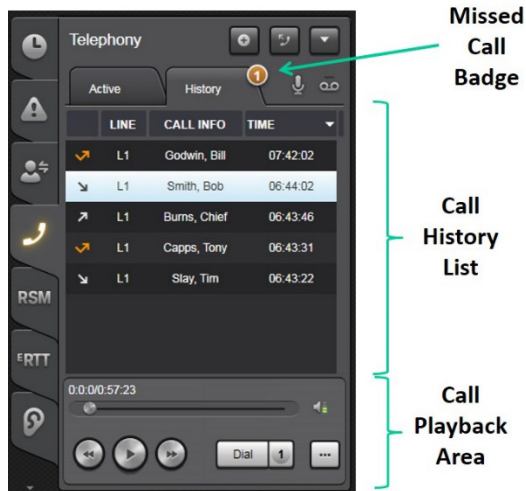


Figure 3-46: Telephony Panel – Call History

The Missed Call badge shows the number of missed calls. This badge increments each time a call is missed. It is cleared anytime the History is opened (i.e., changes from the Active Tab to the History Tab).

The call history list contains an entry for each call placed or received. Each entry includes the following columns:

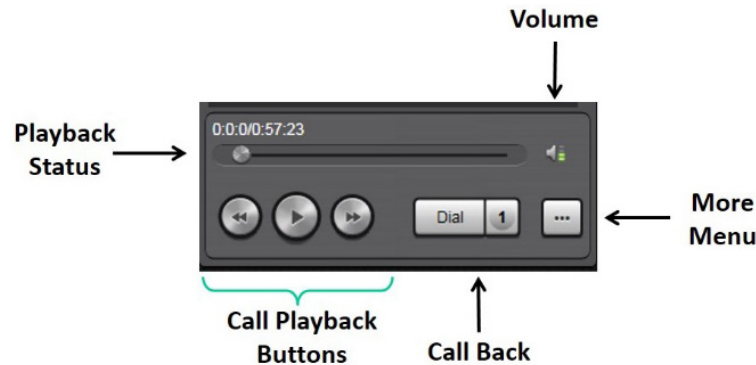
- Icons: The left most column includes a call disposition icon from Table 3-8.

Table 3-8: History Tab - Call Disposition Table


ICON	DESCRIPTION
↘	Inbound calls to the console.
↗	Outbound calls from the console.
↘ (with red slash)	Outbound call failed. See the System History for information on the call failure.
↘ (with orange arrow)	Missed Call.
🕒	Call Back Reminder. This icon marks calls tagged for later call back.

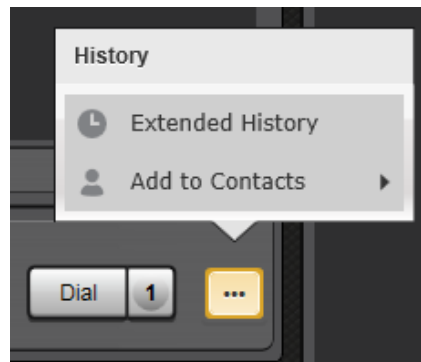
- Line: This column indicates the line associated with the call (e.g., L1 for line 1). The primary line is indicated by a red circle in the background.
- Call Info. This column contains the phone number or alias of the caller/callee.
- Time: This column contains the time the call was initiated.

Highlighting a call in the Call History List loads the call in to the Call Playback Area which provides controls to playback the recorded call or place a call to the call party.



The Call Playback Area provides the following:

- **Call Playback Status:** This indication provides the playback progress via text and a graphical progress bar. The text format is <position in call>:<call length> where both fields have the format <hour.minute.second>. So, 0.0.1; 0.1.21 indicates the playback is one second into a call that is one minute and twenty-one seconds long.
- **Volume:** Click on the volume icon to pop out the volume control for call playback. This volume setting applies to all played back phone calls.
- **Call Playback Buttons:** These buttons stop and start the playback and jump to the next or previous call.
- **More Button:** The More Button  provides menu access to advanced features such as Add to Contacts. The menu also includes an option to open the Extended History Dialog.



- **Call Back Control:** Dial control to initiate a call back to the highlighted party in the Call History list. See Section 3.1.8 for details.

3.4.5.3 New Call Popup

The new call popup provides access to call targets by providing a dial pad and multiple contact views. The popup can be initiated with the **New Call** button or from certain menu items in the Advanced Feature Menu using the **More** button. The menu bar at the top provides buttons to select the specific view. Starting on the left, the views are Dial Pad, Contacts, Favorites, and Speed Dial.

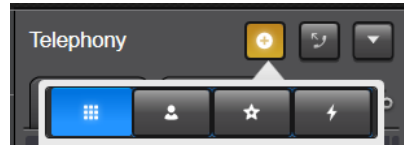


Figure 3-47: New Call Menu Bar

3.4.5.4 Dial Pad View

The Dial Pad View allows the dispatcher to dial a number with the mouse, touchscreen, or the keyboard. Enter the number and press **Dial** to place the call. This view is only used to initiate new calls; it is not used for overdiel.



Figure 3-48: Dial Pad

3.4.5.5 Contact Views

The Contacts Views allow the dispatcher to select contacts from a filtered list of all contacts. The views include Contacts which includes all contacts, Favorites which limit the contacts to a frequently used subset, and Speed Dial Lists which organize contacts into categories. The typical contact view is shown in Figure 3-49.



Figure 3-49: Typical Contact View

The typical contact view includes the following:

- **Search field:** This field filters the contacts based on the search criteria. The search is character-by-character like all other Symphony search fields.
- **Contact List:** List of contacts matching the search criteria (if any). Contacts are listed in alphabetical order. Each entry has two columns:
 - **Name:** This column shows the contact name. If a contact has multiple numbers, only the first number listed shows the contact name. The remaining numbers for the contact have blank name fields. This allows you to easily associate all the numbers for a contact.
 - **Number:** This column shows an icon to identify the number type (work, home, mobile, or other) and the associated phone number. The icon comprises the first letter of the number type (i.e., w for work) in a grey circle. The circle is white if the contact was created locally by the user. If the number is too long for the column, it will be truncated with “...” at the end. Hover the mouse over the number for a tool tip containing the complete number.
- **Call control buttons that depend on the call type.** Common buttons include:
 - **Dial Control:** Closes the New Call Panel and initiates a call to the highlighted contact in the contact list. See Section 3.1.8 for details.
 - **Close Button:** Closes the New Call Panel.

The Speed Dial View organizes contacts into categorized lists for easy access. The view shows a list of categories with the count of the contacts in each category. Highlight a list and press Show to reveal the list of contacts in that category.

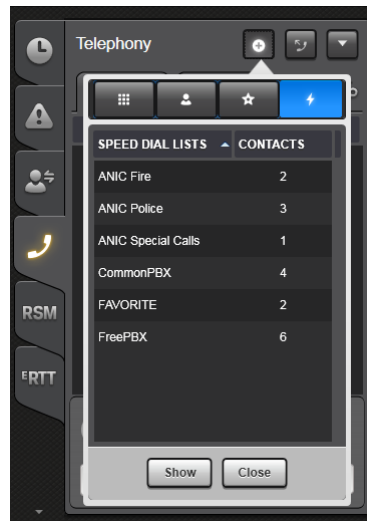



Figure 3-50: Speed Dial List

3.4.6 Speaker Sidebar Panel

The Speaker Sidebar Panel consists of a list of tabs – each representing a specific speaker on the console. The number of tabs is determined by how many speakers are connected to the console. In each speaker tab, mini Communication Modules are provided representing each group assigned to that speaker. Click  on the Baton to access the Speaker Sidebar Panel.

Patches, SimulSelects, and Individuals are not displayed in the speaker panel because each of these has their own specific Sidebar Panel that provides interaction for the user.

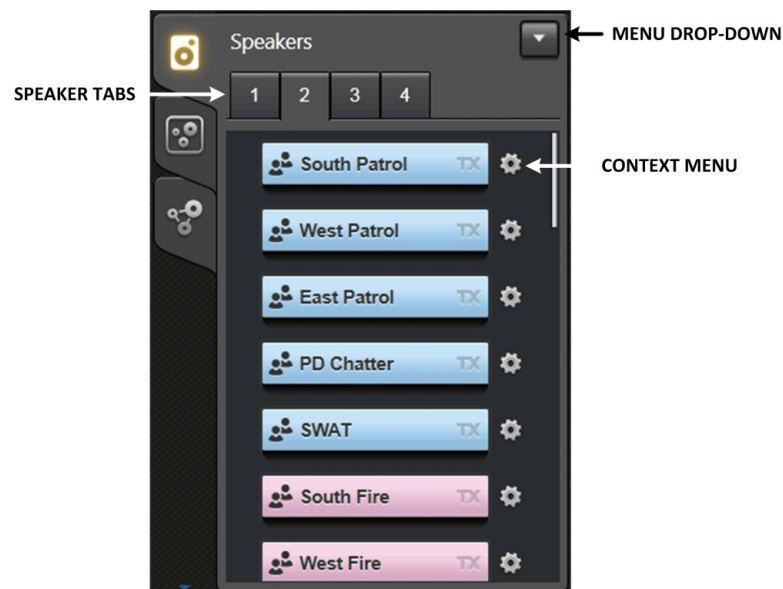


Figure 3-51: Speaker Sidebar Panel

3.4.6.1 Menu Drop-Down

The options available from the drop-down menu are:

- Open Directory – Opens the Directory dialog. See Section 3.5.1 for more information.
- Open History – Opens the History dialog. See Section 3.5.2 for more information.


3.4.6.2 Context Menu

The context menu provides access to additional functionality not provided by the mini Communication Module.

3.4.6.3 Speaker Tabs

Each tab contains all entities assigned to that speaker.

3.4.7 Patch Sidebar Panel

The Patch Sidebar Panel displays all patches and allows the user to create new patches. Click  on the Baton to access the Patch Sidebar Panel. See Section 5.8 for more information on patches.

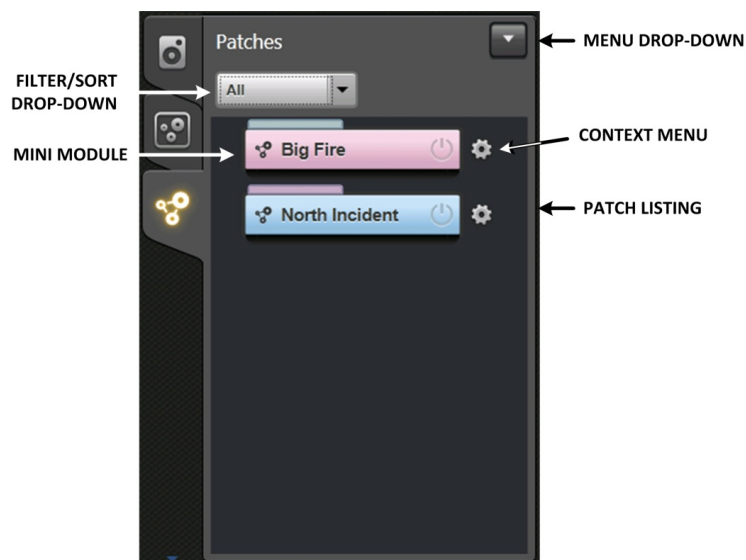


Figure 3-52: Patch Sidebar

3.4.7.1 Menu Drop-Down

The options available from the drop-down menu are:

- Open Directory – Opens the Directory dialog. See Section 3.5.1 for more information.
- Open History – Opens the History dialog. See Section 3.5.2 for more information.
- Create Patch – Opens the NEW/EDIT PATCH dialog (see Section 3.4.7.5).

3.4.7.2 Context Menu

The context menu provides access to additional functionality not provided by the mini Communication Module.

3.4.7.3 Filter Sort Drop-Down

The Filter/Sort drop-down allows the user to filter and sort patches according to the following criteria:

- All
- Local Active
- Local Inactive
- Remote Active

3.4.7.4 Patch Listing

All active and inactive local patches, as well as remote patches are displayed in the list. Each mini Communication Module shows a different Patch name. The Patch listing provides a separator displaying the locally defined patches on the top of the separator, and remote patches below the separator. The separator is not displayed when either remote or local patches are filtered out of the list.

3.4.7.5 New/Edit Patch Dialog


The New/Edit Patch dialog allows the user to create predefined patches to be activated at a later time or activated as soon as the Patch is defined.



Figure 3-53: New/Edit Patch Dialog

- Patch Name – Assign a name for the Patch (Big Fire in Figure 3-53).
- Entities Filter Drop-Down – Determines which entities will be included in the Available Entities List. Options include All Entities, Groups, and Base Stations.
- Search – Allows the console user to search for available entities.
- Clear List – Clears all entities from the Patch Member list.
- > – Adds the selected entity(ies) to the Patch Member list.
- < – Removes the selected entity(ies) from the Patch Member list.
- Override Key – When checked, overrides the default Patch key with key chosen from the drop-down box.
- Activate on Save – The newly defined Patch is saved and activated when **Save & Close** is clicked. This only applies to the last defined Patch prior to closing the dialog.
- Select on Save – The newly defined Patch is saved, activated and selected when **Save & Close** is clicked. This only applies to the last defined Patch prior to closing the dialog.
- Save & Close – This button saves the new Patch and closes the dialog.
- Cancel – Cancels the operation and closes the dialog.

3.4.8 SimulSelect Sidebar Panel

The SimulSelect Sidebar Panel displays all SimulSelects and allows the user to create new SimulSelects. Click  on the Baton to open the SimulSelect Sidebar Panel. See Section 5.9 for more information on SimulSelect operation.

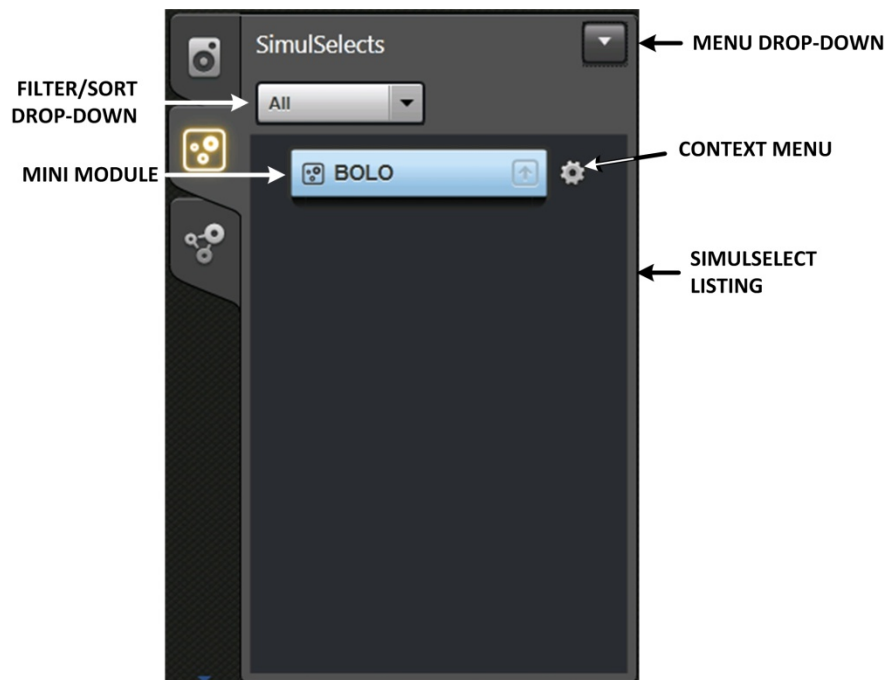


Figure 3-54: SimulSelect Sidebar

3.4.8.1 Menu Drop-Down

The options available from the drop-down menu are:

- Open Directory – Opens the Directory dialog. See Section 3.5.1 for more information.
- Open History – Opens the History dialog. See Section 3.5.2 for more information.
- New SimulSelect – Opens the NEW/EDIT SIMULSELECT dialog.

3.4.8.2 Context Menu

The context menu provides access to additional functionality not provided by the mini Communication Module.

3.4.8.3 Filter Sort Drop-Down

The Filter/Sort drop-down allows the user to filter and sort patches according to the following criteria:

- All
- Local Active
- Local Inactive
- Remote Active

3.4.8.4 SimulSelect Listing

All active and inactive local SimulSelects, as well as remote SimulSelects are displayed in the list. Each mini Communication Module shows a different SimulSelect name. The SimulSelect listing provides a separator displaying the locally defined SimulSelects on the top of the separator, and remote SimulSelects below the separator. The separator is not displayed when either remote or local SimulSelects are filtered out of the list.

3.4.8.5 New/Edit SimulSelect Dialog

The New/Edit SimulSelect dialog allows the user to create predefined SimulSelect to be activated at a later time or activated as soon as the SimulSelect is defined.

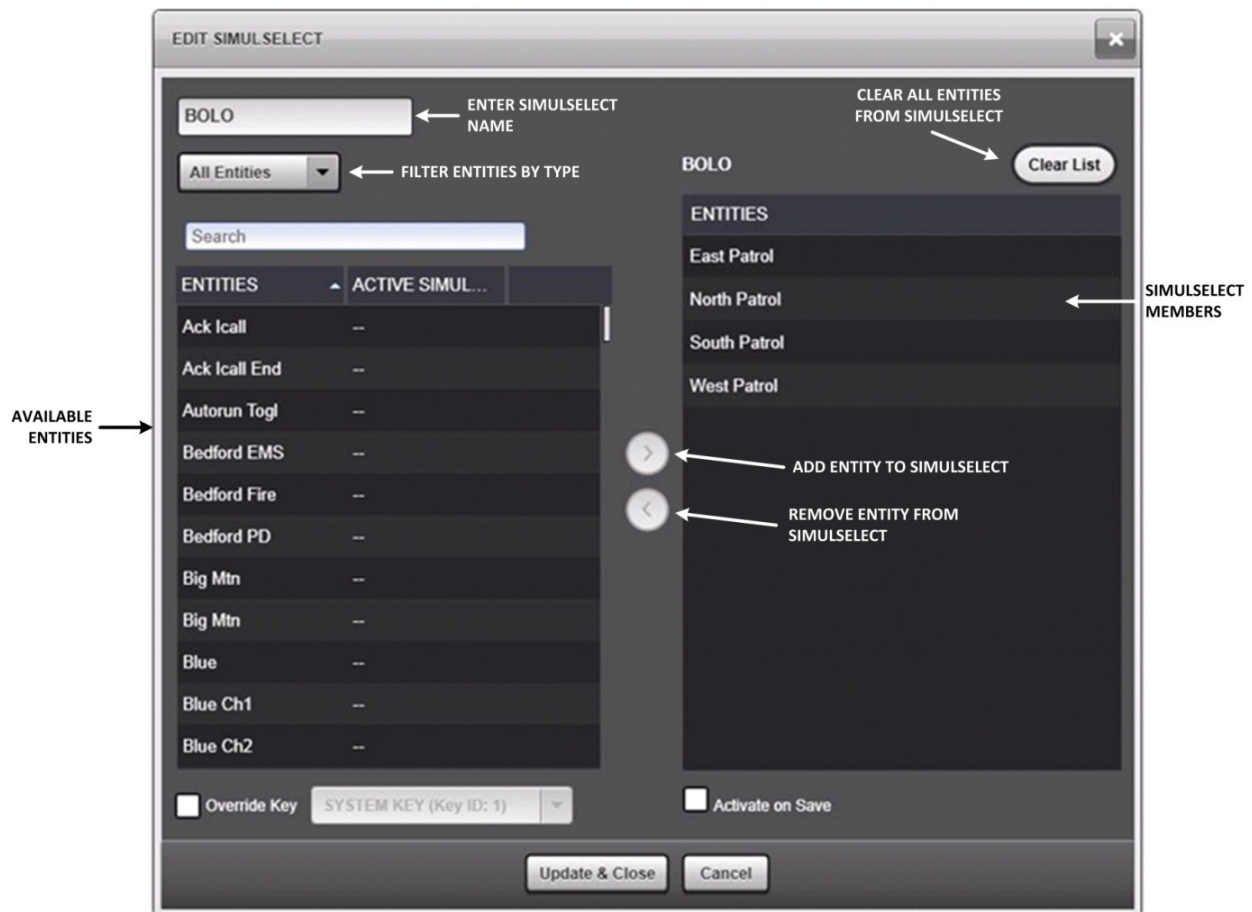


Figure 3-55: New/Edit SimulSelect Dialog

- SimulSelect Name – Assign a name for the SimulSelect (BOLO in Figure 3-55).
- Entities Filter Drop-Down – Determines which entities will be included in the Available Entities List. Options include All Entities, Groups, Users, and Base Stations.
- Search – Allows the console user to search for available entities.
- Clear List – Clears all entities from the SimulSelect Member list.
- > – Adds the selected entity(ies) to the SimulSelect Member list.
- < – Removes the selected entity(ies) from the SimulSelect Member list.
- Override Key – When checked, overrides the default SimulSelect key with key chosen from the drop-down box.
- Activate on Save – The newly defined SimulSelect is saved and activated when **Save & Close** is clicked. This only applies to the last defined SimulSelect prior to closing the dialog.
- Save & Close – This button saves the new SimulSelect and closes the dialog.
- Cancel – Cancels the operation and closes the dialog.

3.4.9 RSM Sidebar Panel

The RSM Sidebar Panel shows all radio units that have sent a Radio Status Message with Unit, Time, or Status. Radios can only be in one state at a time and a radio will appear in RSM Sidebar Panel only when they send an RSM message during a console session. Radios stay in the RSM Sidebar Panel list until the console is rebooted. When a radio sends a new status, it appears at the top of the list with its new status. Its previous entry in the list is removed. The most recent radio message is displayed at the top of the list. Click **RSM** to access the RSM Sidebar Panel from the Baton.

Use the drop-down to specify a status to filter by.

Choose an entry from the list and click the mini-module's Smart Button to call the highlighted entity.

See Section 9.2.1 for more information on Radio Status Message Modules.

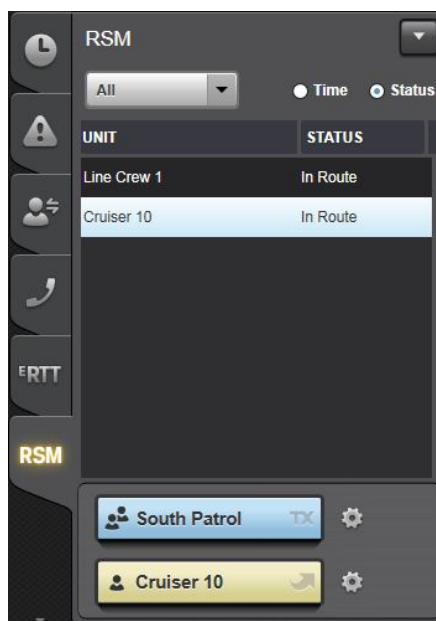


Figure 3-56: RSM Sidebar

3.4.9.1 Menu Drop-Down

The options available from the drop-down menu are:

- Open Directory – Opens the Directory dialog. See Section 3.5.1 for more information.
- Open History – Opens the History dialog. See Section 3.5.2 for more information.

3.4.9.2 Filter Drop-Down

Allows the user to filter the Radio Status Message list using the following criteria: All or a list of all configured Status codes in the console.

3.4.9.3 Display Time or Status

These options determine whether the message time or status is displayed in the second column.

3.4.9.4 Radio List

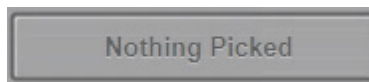
The RSM side bar panel shows all radios that have sent a status message with Unit, Time or Status columns. The most recent radio message is displayed at the top of the list. The number of radio messages displayed is configurable (default = 100).

3.4.9.5 Context Menu

The context menu provides access to additional functionality not provided by the mini Communication Module.

3.4.9.6 Callback Module

The Callback module provides the console user with a mini Communication Module to call back the selected entity from the Radio Status Message list. When no entry is selected from the Radio Status Message list, the mini module appears as shown below:



3.4.10 ERTT Sidebar Panel

The ERTT Sidebar Panel organizes and manages all received RTT messages. This helps the dispatcher to quickly view how many units have pending requests for all RTT codes. The Sidebar Panel also allows you to quickly respond to an RTT message through a group or individual call.

Dispatchers can share a pool of RTT messages, further enhancing the efficiency of dispatch operations. Actions from one dispatcher are reflected on the consoles of other dispatchers if the Remote option is enabled.

Click **RTT** to access the ERTT Sidebar Panel from the Baton.

Use the drop-down to filter RTT messages by Request. Choose an entry from the list and click the group mini-module's Smart Button to call the talkgroup associated with the entry or the unit mini-module's Smart Button to call the unit. See Section 9.2.1 for more information on ERTT Modules.

The ERTT Sidebar Panel, shown below, combines all the ERTT entries into a single panel. The ERTT panel in the Baton is identical. The panel has the following elements from top to bottom:

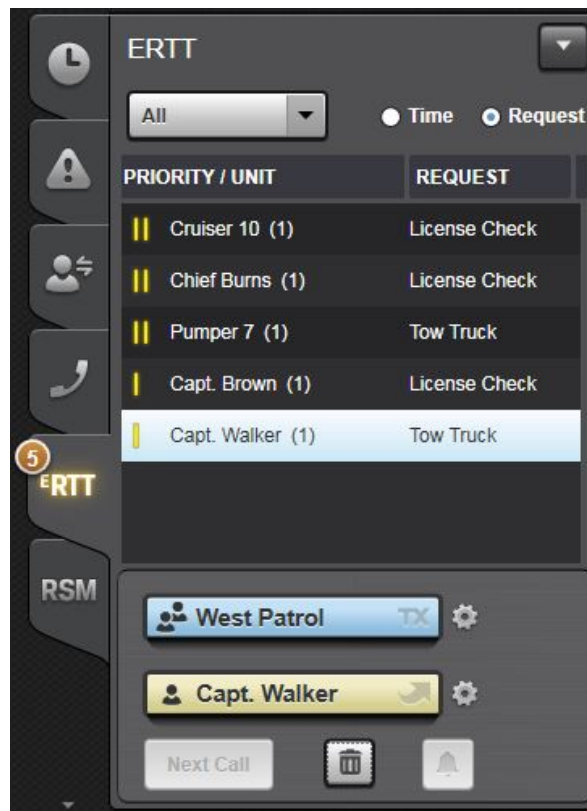


Figure 3-57: ERTT Sidebar

3.4.10.1 Menu Drop-Down

The options available from the drop-down menu are:

- Open Directory – Opens the Directory dialog. See Section 3.5.1 for more information.
- Open History – Opens the History dialog. See Section 3.5.2 for more information.

3.4.10.2 Filter Drop-Down

This control lists each of the ERTT codes. However, if all requests need to be displayed no matter the RTT code, then choose the option “All” from the drop-down. When “All” is chosen, RTTs are ordered by the same rules as an ERTT module, but without regard for the RTT code. That is, all High priority requests show up at the top followed by Normal priority requests. Within the High priority requests or the Normal priority requests, the entries are ordered by oldest first.

3.4.10.3 Display Time/Request Radio Buttons

These controls allow the user to set the column on the right to Time or the Request code. By default, the Request code is displayed.

3.4.10.4 RTT Message List

This lists RTT messages that are unanswered or answered but not yet deleted. The listed RTT messages and their order are controlled by the Filter selection and by system configuration.

The entry listing includes:

- **Priority Icon:** Indicates the priority of the entry (High or Normal). This uses the same icons as the High Priority Indicator.
- **Alias** for the unit that generated the RTT message. The alias can be configured to show the unit alias or RTT associated group alias. The default is unit alias. If the UI is configured to show IDs instead of aliases, then the alias is replaced by the appropriate ID. This setting is controlled by the console administrator.
- **Request count:** The number of times the unanswered request has been sent. The count follows the alias in parentheses.
- **Time/Request associated with the list entry.** If the Time option is selected, then the time of the RTT is listed. If the Request option is selected, then the RTT type is listed.
- Answered calls are indicated with gray text and gray priority icon.

3.4.10.5 ERTT Control Buttons

When the messages are filtered by specific RTT message, then the controls work identical to the RTT Module. See Section 9.3.1.1 for more information. This section describes how the controls work when the “All” filter option is selected.

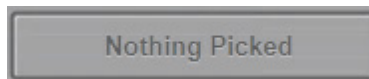
- **Next Call Button:** The Next Call button is grayed out.
- **Delete Button:** The Delete button removes the highlighted entry from the RTT Message List. It also removes the same RTT message from the RTT Module. If the RTT message is deleted from the RTT Module, then the same entry in the RTT Sidebar Panel is deleted. Deleting an unanswered RTT message silences the alarm associated with the deleted RTT.
- **Tone Off Button:** The Tone Off button terminates RTT Ringing tone(s) for all unanswered RTTs. In this mode, it works like a global RTT Tone off. The button has two states: disabled and active. In the disabled state, the button cannot be pressed. In the active state, the button is raised and can be pressed. This indicates that there are unanswered RTTs that have not been silenced. Pressing the button stops all alarms for all RTTs and the button switches to the grayed-out state until a new alarm is generated.

3.4.10.6 Context Menu

The context menu provides the ability to open the Directory and History dialogs.

3.4.10.7 Mini Modules

Mini modules return an I-Call or group call to the issuing radio of the highlighted entry. If the mini module is not valid for any reason, it will not be populated with a Smart button. When there are no messages because they have all been deleted or none have been generated then the mini module appears as shown below:



3.4.11 Supervisory Sidebar Panel

The Supervisory Sidebar Panel supports three supervisory features: Discreet Listening, Radio Unit Monitoring (RUM) and Console Enable/Disable. The Discreet Listening feature enables the dispatcher to monitor active VIDA I-Calls between two remote units. The Radio Unit Monitoring feature allows the dispatcher to monitor a remote unit by initiating a VIDA I-Call or group call from the remote unit to the dispatcher's Symphony console. Both features are receive-only. The Console Enable/Disable feature allows a supervisory console to enable/disable a console under its control. The 'Console' tab will only be visible on supervisory consoles when the supervisory console has a valid console configured for this feature.



Discreet Listening and Radio Unit Monitor features must be separately licensed and enabled before they can be used.

The Supervisory Sidebar Panel contains three tabs:

- The 'I-Call' tab supports the Discreet Listening feature.
- The 'Radio' tab supports the Radio Unit Monitoring (RUM) feature.
- The 'Console' tab supports the Console Enable/Disable feature.



Figure 3-58: Supervisory Sidebar

All three tabs share the controls in the top right corner of the panel. These controls provide the following functions from left to right:

- Keypad Control: Provides a keypad to enter the ID of the target unit for Discrete Listening or RUM.
- Contact List: Provides a list of provisioned contacts from which to select the target unit for Discrete Listening or RUM.
- Menu: Provides quick access to the History and Directory dialogs.

3.4.11.1 Shared Controls

This section describes the Supervisory Sidebar Panel shared controls in detail.

3.4.11.2 Menu Drop-Down

The options available from the drop-down menu are:

- Open Directory – Opens the Directory dialog. See Section 3.5.1 for more information.
- Open History – Opens the History dialog. See Section 3.5.2 for more information.

3.4.11.3 Keyboard Control

The Keyboard control, shown in Figure 3-59, allows the console user to program a unit by entering its Unit ID. After entering a valid Unit ID, the **Add/Continue** button is enabled. If in the I-Call tab of the Supervisory panel, click the **Add** button to program the unit onto the I-Call tab. In the Radio tab of the Supervisory panel, click the **Continue** button to program the unit onto the Radio tab. Click the **Cancel** button to close the keyboard without programming the unit. See Section 9.5 for details on using the Discreet Listening feature, and Section 9.6 for more information about the Radio Unit Monitor feature. The Keyboard control is not available for Console Enable/Disable.



Figure 3-59: Supervisory Keyboard Control

The format used for unit IDs is the same as that used in the I-Call Sidebar Panel keyboard control. See Section 3.4.3.2 for details.

3.4.11.4 Contact List

The Contact List, shown in Figure 3-60, allows the console user to program a unit from a list of contacts. This list supports searching and filtering. To program a unit, highlight a user in the list. The **Add/Continue** button is enabled. In the I-Call tab of the Supervisory Sidebar Panel, click the **Add** button to program the unit onto the I-Call tab. In the Radio tab of the Supervisory panel, click the **Continue** button to program the unit onto the Radio tab. Click the **Cancel** button to close the keyboard without programming the unit. See Section 9.5 for details on using the Discreet Listening feature, and Section 9.6 for more information about the Radio Unit Monitor feature.

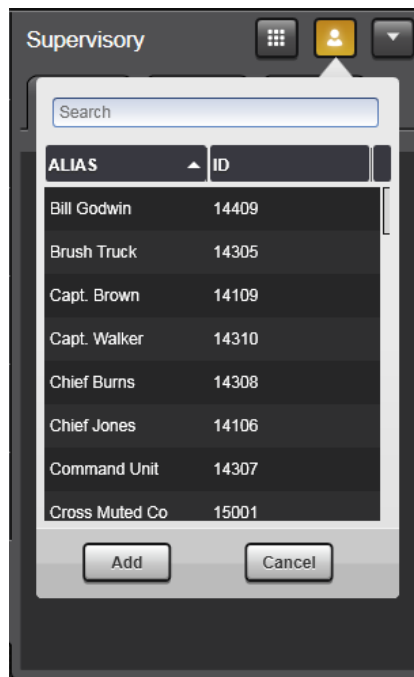



Figure 3-60: Supervisory Contact List

3.4.11.5 I-Call Tab (Discreet Listening)

The I-Call tab displays all units that have been programmed for Discreet Listening and allows the dispatcher to add new or remove existing units. See Section 9.5 for more information on the Discreet Listening feature.



Figure 3-61: I-Call (Discreet Listen) Tab

The I-Call tab displays a unit module and a Delete Icon  for each unit that has been programmed for Discreet Listening. When one of these units is involved in an active VIDA I-Call, the associated module displays pertinent call indications including:

- Active Call Indicator – The LED on the top left of the smart button turns green when the Target unit is involved in an unencrypted VIDA I-Call and amber when the call is encrypted.
- Other Endpoint – The Alias/ID of the other unit that is involved with the target unit in the active VIDA I-Call is displayed on the bottom of the smart button.
- Call Direction Indicator – The arrow points away from the caller and towards the callee in the active VIDA I-Call.
- Supervisory Icon – A blinking Supervisory Icon is displayed on the left side of the bottom line to clearly indicate the call is a Discrete Listening call.

The unit modules remain on the tab until the Delete Icon is pressed. At this point, the module is removed from the I-Call tab and the Discrete Listening is terminated for the associated target unit.

3.4.11.6 Radio Tab (Radio Unit Monitor)

The Radio tab allows the dispatcher to initiate Radio Unit Monitor (RUM) on a unit. See Section 9.6 for more information on the Radio Unit Monitor feature.

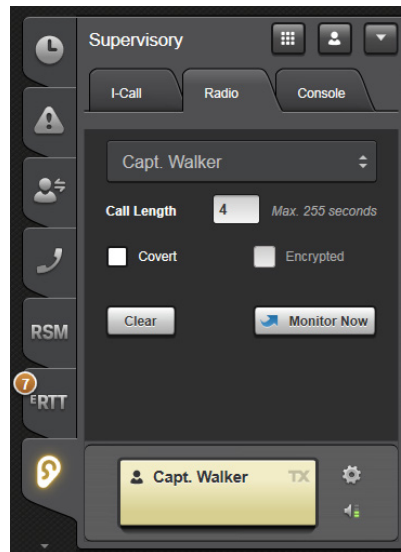




Figure 3-62: Radio Unit Monitor

The Radio tab has the following controls and indications once a unit has been programmed into the tab:

- Alias/ID of the unit that has been programmed for Radio Unit Monitor.
- Call length text field: The Radio tab displays the default Call Length, as configured in the Symphony Configuration tool by the administrator. This parameter is available for editing only if allowed by the administrator.
- Covert checkbox: When checked, the radio will not indicate to the radio user that it is transmitting the RUM call. This checkbox is checked or unchecked by default, as configured by the administrator. This parameter is available for editing only if allowed by the administrator.
- Encrypted checkbox: When checked, the radio transmits the RUM call encrypted. This is checked or unchecked by default, as configured by the administrator. This parameter is available for editing only if allowed by the administrator.
- Clear button: This button clears the Radio tab. To initiate another Radio Unit Monitor, click the Keyboard control  or Contacts  buttons.
- Monitor Now button: This button initiates the Monitor request. Clicking this button prompts the radio to start a Radio Unit Monitor call. If this call is configured to be an I-Call, then the Radio Unit Monitor call is initiated by the radio with the console as a VIDA I-Call. If this call is configured to be a group call, then the Radio Unit Monitor call is initiated by the radio on the configured talkgroup as a group call.
- RUM Target Module: This group or unit module shows the progress of the RUM call.


3.4.11.7 Console Tab (Console Enable/Disable)

The Console tab allows the dispatcher to disable and re-enable other Symphony and C3 Maestro^{IP} consoles. See Section 9.7 for more information on the Console Enable/Disable feature.



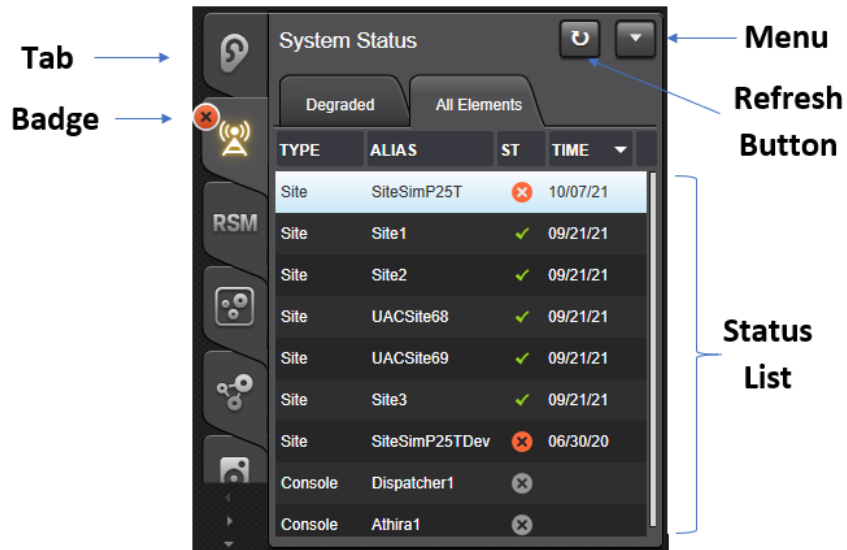
Figure 3-63: Console Enable/Disable

The Console tab has the following controls and indications once a unit has been programmed into the tab:

- Alias/ID of the console that has been selected for enabling or disabling.
- Enable/Disable radio buttons: Allows dispatcher to select the desired action – enabling the selected console or disabling the selected console.
- Clear button: This button clears the Console tab. To initiate another Console Enable/Disable action, click the Contacts  button.
- Send button: This button initiates a command to enable or disable the remote console. Clicking this button prompts the console to issue a command for enabling/disabling the console. Text will appear below the button to notify the dispatcher of the command's success or failure.
- Console Enable/Disable Target Module: This unit module shows the selected console.

3.4.12 System Status Sidebar Panel

The System Status Panel shows the link status of VIDA elements. If a link is down, the element may still be operational, but Symphony cannot communicate with it. An element in VIDA can represent a single site, such as a P25 site, a single simulcast system, another Symphony Dispatch Console, or an aggregation of devices such as conventional base stations. In the latter case, VIDA gateways typically aggregate multiple base stations into a single site. If the link to the gateway is lost, the System Status Panel shows a single site down, but multiple base stations modules may indicate the failure state. Consult with your system administrator to understand which communications assets are covered by each site. The System Status feature must be licensed and configuration in the Symphony Configuration Utility is required for this Sidebar Panel to be visible.



The System Status Sidebar Panel contains the following items:

- **Menu:** The menu provides options to open the directory dialog (Base Station tab if the Base Station feature is licensed; otherwise, it opens to the Groups tab) and the history dialog (radio tab).
- **Tabs:** The **All Elements** tab shows all monitored VIDA elements and their current status. The Degraded tab only shows VIDA elements that are in a degraded state. If all monitored elements are operational, the Degraded tab shows “No Degraded Elements.” Both tabs have the Status List described below.
- **Status list:** The Status List contains an entry for each system element reported by the tab. Each entry provides the following information:
 - **Type:** The type of VIDA element. Base station gateways are labelled Station; all other elements are labelled Site.
 - **Alias:** The site alias if available; otherwise, the site ID is shown.
 - **Status:** The current status of the site. A check mark indicates the link is up, ✘ indicates the site link is down, and ✘ indicates the console link is down or that the specified console is not supported by the system status panel.
- **Time:** Time or date that the current event state became active. Time is used if the event changed on the current date.
- **Badge:** The badge appears when at least one element is down. The badge blinks for 10 seconds after each state change.
- **Refresh Button:** This updates console entries with the latest link status.

The System Status Panel shows the last known status of individual sites and consoles. If Symphony loses connectivity to the VIDA core, the System Status Panel continues to show the state before the connection to the core was lost. Once connectivity is restored, the System Status Panel updates with the latest link status.

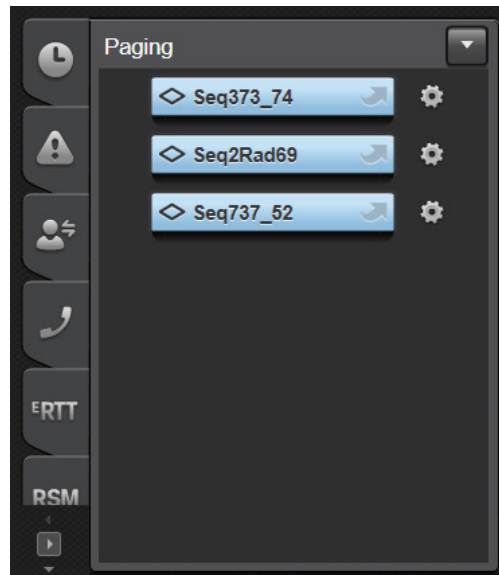
Symphony only reports the status of sites that are administratively configured and reported by the system. There are cases where the VIDA system does not know the status of a site and therefore does not report it to Symphony. If an expected site is not included in the System Status panel, consult with your system administrator.

The System Status Panel is available on the Baton via the main menu: **Main Menu → Feature Panels → System Status**.

3.4.13 Advanced Paging Sidebar Panel

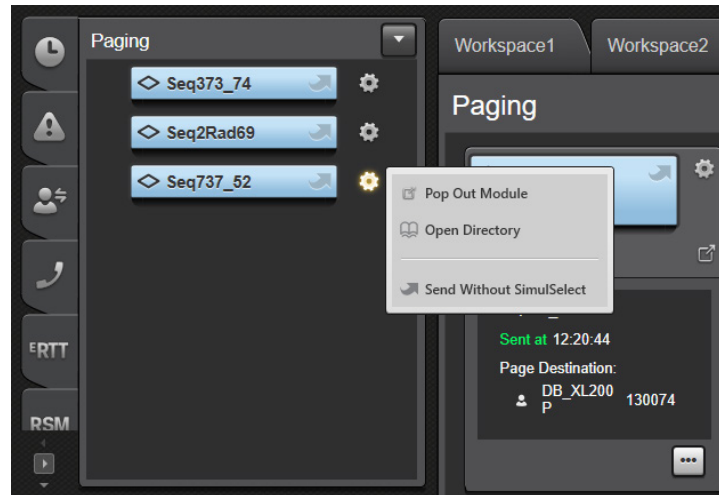
The Advanced Paging Sidebar Panel shows all the sequences that were attempted during the last Stack Page or Composite Page, displaying a miniature Smart Button for each in the order they were attempted. It is a convenient way to retry any sequences that were not sent successfully. The System History Panel will identify any failed sequences and explain why they failed, so any issues can be resolved before retrying. This Sidebar Panel allows you to retry one of the sequences individually or to retry multiple sequences with one action, as explained below.

The Advanced Paging feature must be licensed and enabled via the Symphony Configuration Utility Paging Tab.



The Advanced Paging Panel includes the following elements:

- **Mini Smart Button:** identifies each sequence in the recently attempted Stack Page or Ad-hoc Stack Page. These buttons are displayed top to bottom in the order they were attempted.
- **Gear Icon:** beside each Mini Smart Button to pop out a context menu of options that can be used in conjunction with each page sequence on the side bar panel.



The following options can be accessed from this context menu:

- **Pop Out Module:** pop out a full-size paging module for the corresponding sequence.
- **Open Directory:** display a directory listing of all pre-defined sequences and stacks.
- **Send Without SimulSelect:** send the corresponding page but do not SimulSelect the targets of all the sequences listed on the side panel.

3.5 COMMON DIALOGS

3.5.1 Directory Dialog

The Directory dialog provides access to all provisioned and configured entities including Groups, Users, Base Stations, Aux I/O Points, and Paging Sequences. Click the tab corresponding to the type of entity to display. If a provisioned entity is added or removed while the console is in operation, the Directory updates to reflect the changes and a message is added to the System History to note the change. The Search field allows you to search for entities by alias, ID number, or any text-based field.

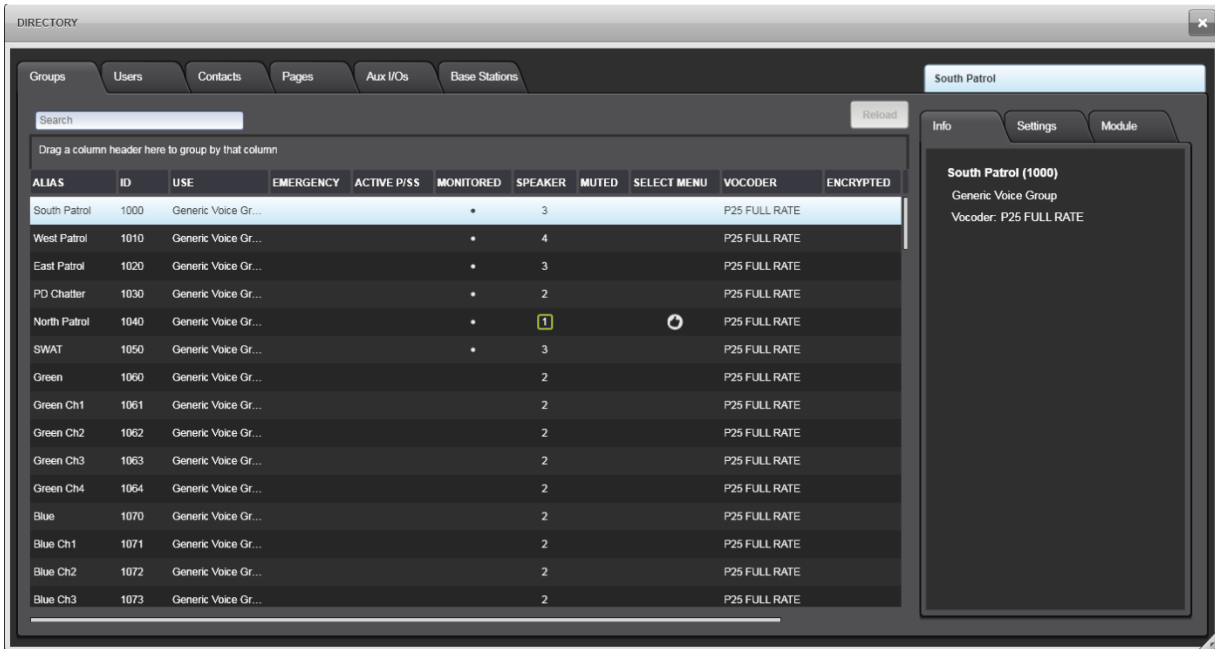



Figure 3-64: Directory Dialog

This section focuses on the Group, User, and Contact Tabs. Other tabs, such as Aux I/O, will be discussed with their associated feature. Table 3-9 and Table 3-10 define each column in the Group, User, and Contacts tabs. Click a column header to sort entities by that column or click  and specify the filter criteria. Click and drag a column header as shown in Figure 3-65 to group entities by that column.

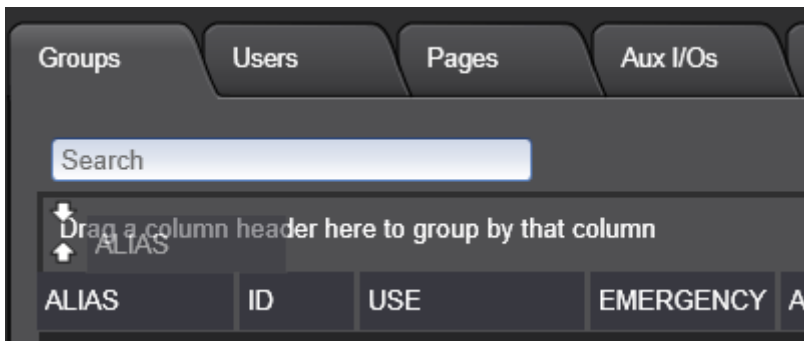


Figure 3-65: Group Columns

Table 3-9: Directory Columns – Group Tab








COLUMN	DESCRIPTION
ALIAS	The alias for the group.
ID	The ID number of the group.
USE	The group type: <ul style="list-style-type: none"> • Generic Voice Group • Conventional Station Voice Group • All-Call Voice Group • IP Console Intercom Voice Group • Announcement Voice Group
EMERGENCY	If the group is in emergency,  is displayed in this column.
ACTIVE P/SS	Indicates if the group is part of an active Patch or SimulSelect.
MONITORED	Console receives calls on this group. Entities that are programmed on a workspace are set to be monitored by default.
SPEAKER	Shows the currently assigned speaker.  indicates selected.
MUTED	 Indicates the current mute state for the group.
SELECT MENU	Indicates that the group is included in the Select Menu.  indicates the group is the Default Select Entity.  indicates the group is a Preferred Select Entity.
VOCODER	The vocoder last used on the group: <ul style="list-style-type: none"> • P25 HALF RATE • OPENSKY® AMBE+ 2™ • AMR • AMR WB • EDACS® IMBE™ (Not supported) • OPENSKY AMBE® (Not supported) • P25 FULL RATE • ADPCM
ENCRYPTED	If the group supports encryption, this column will contain a padlock icon. The lock is locked  if the group is encrypted and unlocked  if the group is unencrypted. This column is empty otherwise.
ACTIVE MARKER TONE	Indicates if the group has an Active Marker Tone.

Table 3-10: Directory Columns – User Tab



COLUMN	DESCRIPTION
ALIAS	The alias for the user.
ID	The ID number of the user.
UNIT ALERT	Indicates if the user has an active Unit Alert.
ACTIVE SIMUL	Indicates if the User is included in an active SimulSelect.
SPEAKER	Shows the currently assigned speaker.
MUTED	Indicates the current mute state for the user.
SELECT MENU	Indicates if the user is in the select menu.
VOCODER	The vocoder last used on the user.
ENCRYPTED	Indicates the current console transmit encryption state. The lock is locked  if the state is encrypted and unlocked  if the state is unencrypted. The column is empty if encryption is not supported or configured.

Table 3-11: Directory Columns – Contacts Tab

COLUMN	DESCRIPTION
LAST	The Last name of the contact.
FIRST	The First name of the contact.
WORK	The Work telephone number of the contact.
MOBILE	The cellphone number of the contact.
HOME	The Home telephone number of the contact.
OTHER	An extra field for additional telephone number of the contact.
AFFILIATION	The affiliation of the contact. For example, the contact's company or department.
LOCAL	Indication if the contact is a Local Contact or Admin Contact.
HOTKEY	The associated hotkey for the contact.

The Directory Dialog offers a search box to quickly filter items of interest. The search box applies to all text-based columns. Enter the search string in the search box. Searching and filtering take place as you type. The dialog continuously updates the list of matching entries with matching characters highlighted in yellow. Delete the search characters to return to the full directory list.

The dialog includes a **Reload** button to reload the data in the tab. This is used when you jump to the directory from the context menu of a module. The Directory opens the appropriate tab for the module (i.e., Group tab for a group module) and only shows the data for the module in question. Press the **Reload** button to show all data for the tab.

3.5.1.1 Info Tab

The Info tab provides a text-based summary of key parameters from the Directory Columns for the selected entity. The Info Tab for Contacts includes controls to place a call to a contact and edit local contacts. See Section 6.2.15 for more details.

3.5.1.2 Settings Tab

The Directory dialog supports choosing multiple entities for changes on the **Settings** tab. After making changes on the **Settings** tab, click **APPLY**. The following options are available from the Settings tab:

CONSOLE	Monitor group with console – When this option is checked, the console user can monitor and interact with this group even if it is not programmed to a Communication Module on the Workspace. The setting will persist until the Symphony is restarted or the current setup is changed to a different one, when defaults are restored. Defaults are to monitor groups that are programmed on a workspace in the current setup and to monitor unprogrammed groups when this check box is filled.
UNSELECT SPEAKER	Specify the speaker assigned to the highlighted group(s).
SELECT MENU	Specify the Select List for assignment (Not in Select, In Preferred Select, or As Default Select). The maximum number of entities allowed in the Preferred Select list is configured by the administrator. When the maximum is reached, the last one in the list is removed (i.e., oldest entry is removed). If “As Default Select” is chosen, the selection box is grayed out. You must set another entity “As Default Select” to re-enable the selection box for this entity.
RX STATE	<i>(Group tab)</i> When this option is checked, you can ONLY receive on this group (cannot transmit or declare/acknowledge/clear emergencies). If this option is checked for any group included in a local Patch before the Patch is active, the Patch is receive-only. If the Patch is a remote Patch, you can still transmit on the Patch.
VOLUME	Adjusts volume for all highlighted entities.

3.5.1.3 Module Tab

The Module tab displays a normal-sized Communication Module for the selected entity.

3.5.2 Extended History Dialog

The Extended History dialog provides a detailed history of calls with audio. Calls are filtered depending on how the history is accessed. For example, if the History dialog is accessed from a module’s menu, only the call history for that module is displayed. If the History dialog is accessed from the main menu or one of the Sidebar Panels, the full call history is displayed. Call history is further divided by call type: Radio Call, Phone Call, or ABR Call. Each type of call appears on its own tab in the Extended History dialog.

See Section 5.5 for more information on Call History.

3.5.2.1 Radio Call History

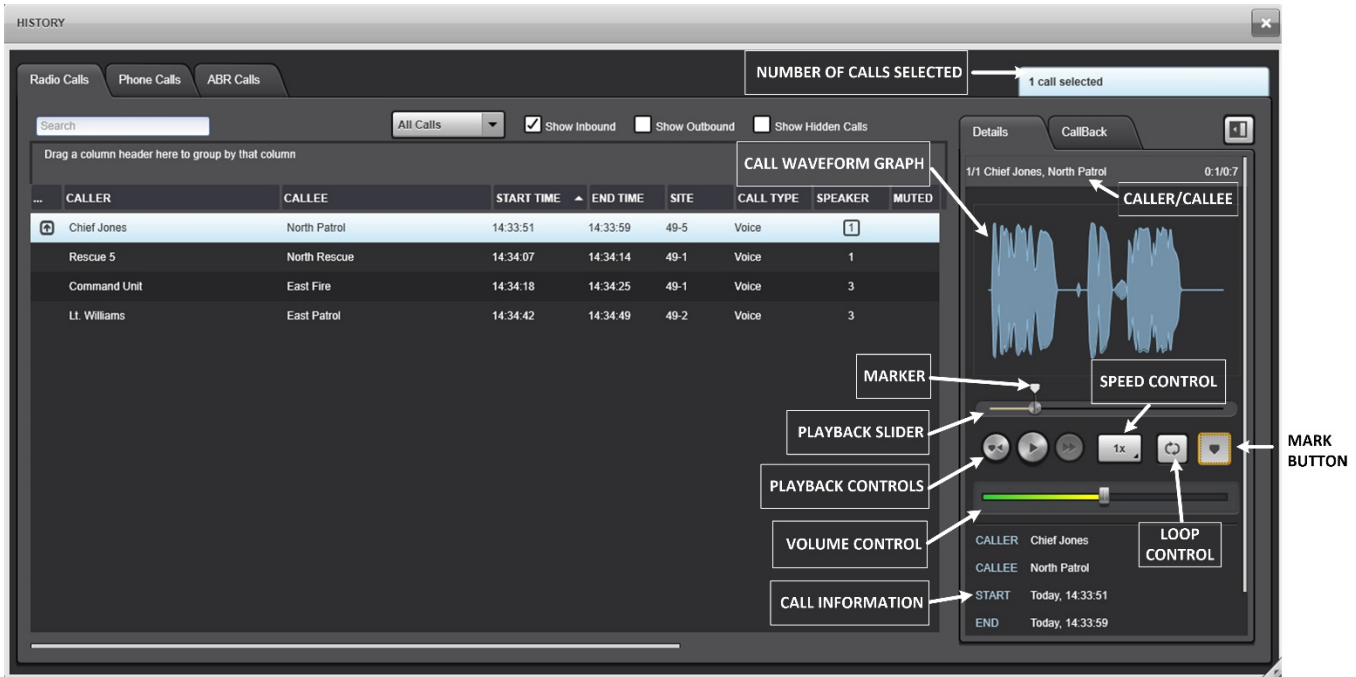


Figure 3-66: Extended History Dialog - Radio Calls

Table 3-12 lists and describes the columns displayed in the History dialog for radio calls. Click a column header to sort entities by that column or click and specify filter criteria. Click and drag a column header as shown in Figure 3-67 to group entities by that column.

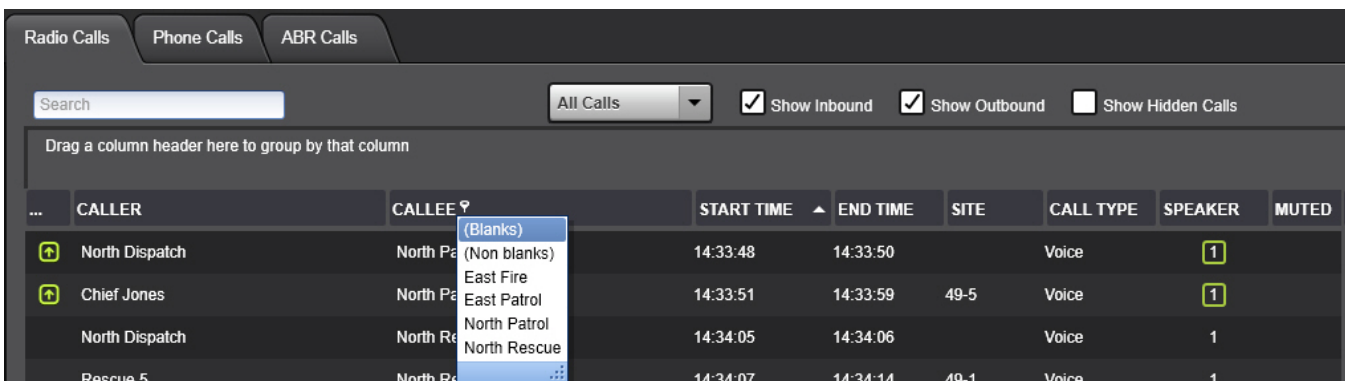




Figure 3-67: Group by Entity Columns

Table 3-12: History Dialog Columns (Radio Calls)

COLUMN	DESCRIPTION
...	Icons to highlight aspects of calls like selected, muted, in emergency, etc.
CALLER	The caller for the call.
CALLEE	The callee for the call.
START TIME	When the call started (HH:MM:SS).
END TIME	When the call ended (HH:MM:SS).
SITE	The site ID used for the call.

COLUMN	DESCRIPTION
CALL TYPE	Indicates the type of call including Voice, Marker Tone, Alert Tone, and page. Outbound calls are properly tagged since the console knows which call it placed. Inbound calls may not provide enough information to identify the specific call type, in which case the default Voice type will be used.
SPEAKER	The speaker assigned to the entity at the time of the call.
MUTED	Indicates whether or not the entity was muted at the time of the call.
SELECT	Indicates if the call was in the Select module when the call occurred.
LOSING AUDIO	Indicates that at least part of this call could not be transmitted to the radios. Typically, a call of higher priority pre-empted this call.
ENCRYPTED	If the entity supports encryption, this column will contain a padlock icon. The lock is locked  if the entity is encrypted and unlocked  if the entity is unencrypted. This column is empty otherwise. If the locked icon is red, the call could not be decrypted because the entity did not have an encryption key specified.
VOCODER	Indicates the Vocoder used on this call.


The Extended History Dialog offers several ways to quickly find items of interest:

- **Search Box:** The search box applies to the Caller and Callee columns. Enter the name or number of the caller or callee. Searching and filtering take place as you type, and matching characters are highlighted in yellow. Delete the search characters to return to the full call history list.
- **Show Inbound:** Checking this box shows inbound calls. Unchecking the box hides inbound calls.
- **Show Outbound:** Checking this box shows outbound calls. Unchecking the box hides outbound calls.
- **Show Hidden Calls:** Certain call types can be hidden by configuration. This typically applies to background calls that could clutter up the call history such as Marker Tones or Alert Tones. Hidden calls do not appear on any other call history container irrespective of Show Inbound/Outbound settings. However, the extended Call History provides this check box to show hidden calls when checked.

3.5.2.1.1 Details Tab

The Details tab includes contextual information about the highlighted call, as well as playback controls. The Details Tab includes:

Playback Controls

Highlight one or more calls from the list to activate the playback controls. See Section 5.6 for more information on using the Playback Controls. Special use of the back button  is covered in this section.

Call Waveform Graph

The Call Waveform Graph displays the highlighted call's audio levels graphically.

Call Information

The call information displayed includes the caller, callee, start/end time, and site ID.

Call Times

The Start and End Times display the date using MM/DD/YYYY HH:MM:SS format. When the date is the current date, "Today" is displayed instead of MM/DD/YYYY. The time format could also use (AM/PM) if configured for non-military time.

Playback Slider

The playback slider tracks the playback progress and allows you to move to different points in the call to focus on specific portions of the playback. The slider represents the entire length of the call. So, moving the slider to the middle of the slider control moves the playback point to the middle of the call. Click and drag the slider or click on the new location to move the slider. If the slider is moved while the call is playing back, the playback stops during the movement and restarts automatically at the new location.

Playback Speed Control

The speed control changes the speed of playback while maintaining the original pitch. Click the speed control to reveal a list of supported speeds. Values less than 1 slow down the playback (e.g., .5 is half speed) while values greater than 1 speed up the playback. The speed cannot be changed during playback. Changing the speed only applies to the current call being played back. Highlighting a new call for playback or closing the History Dialog resets the playback speed to normal speed (i.e., 1x).

Mark Button

The mark button sets the starting point for playback and locks it until the Mark Button is pressed again to clear the mark. The marked position is indicated by the mark icon above the slider. This allows the user to select a point of interest in the call and quickly return to it using the back button.

To set the mark:

- Find the point of interest on the slider by clicking on the point or by pausing the playback at the desired point.
- Press the mark button.
- The mark button will glow, and the mark icon will appear above the slider.
- At any time, press the back button to move to the mark. If the call is playing back when the back button is pressed, the call will continue playing at the mark point.
- To repeat a section of the call, set the mark and speed control, then press play. Click the back button to restart at the mark and repeat as needed.
- Press the glowing mark button to clear the mark.

Marker Icon

The marker icon shows the current marked position set by the mark button.

Loop Control

The Loop Control allows the console user to put playback calls in the loop mode. Press the loop control button to activate loop mode. The button glows to indicate the loop mode is active. Press the loop control button again to deactivate loop mode.

When the loop control is active and multiple calls are selected, the calls play one after the other. When all calls have been played, the first call begins playing again, continuing until you click the Loop Control to stop. If only one call is selected, that call plays over and over until you click the Loop Control to stop.

To use loop mode:

1. Highlight one or more calls in the call list.
2. Press the loop control button to activate loop control mode. The button glows to indicate the change in state.
3. Press the Play button to start playing the highlighted call(s).
4. To pause playing back calls, press the pause button. This will not disable the loop mode.
5. To end loop mode, press the loop control button or highlight another call in the call list.

The loop control can only be changed when the **Play** button is not pressed (i.e., no call is playing back). Changing the playback list disables the loop mode and stops the current playback.

If the mark is set, the loopback will play to the end of the highlighted call and loopback to the marked position.

3.5.2.1.2 CallBack Tab

The CallBack tab displays a normal-sized Communication Module for the highlighted entity. The caller is displayed on the bottom as shown in Figure 3-68.

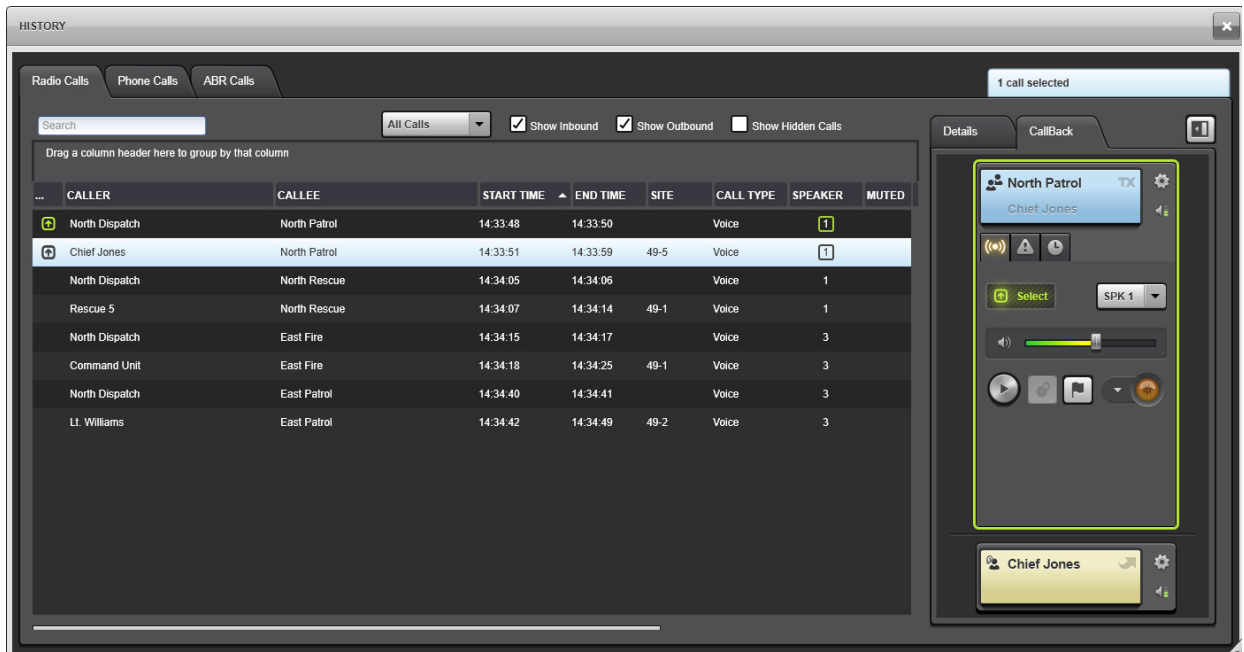


Figure 3-68: CallBack Tab

3.5.2.2 Phone Call History

The Extended Phone Call History displays either Call Director calls or SIP Telephony Calls depending on the current telephony configuration. The phone call history is similar to the radio history, so only the differences are addressed in this section.

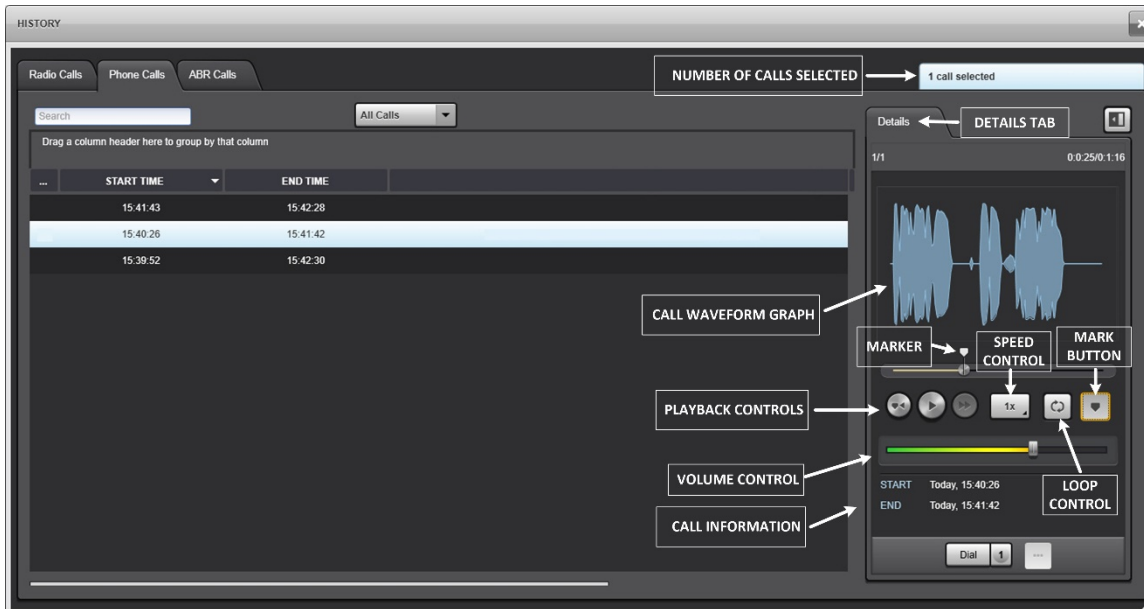


Figure 3-69: Extended Call History - Call Director Phone Calls

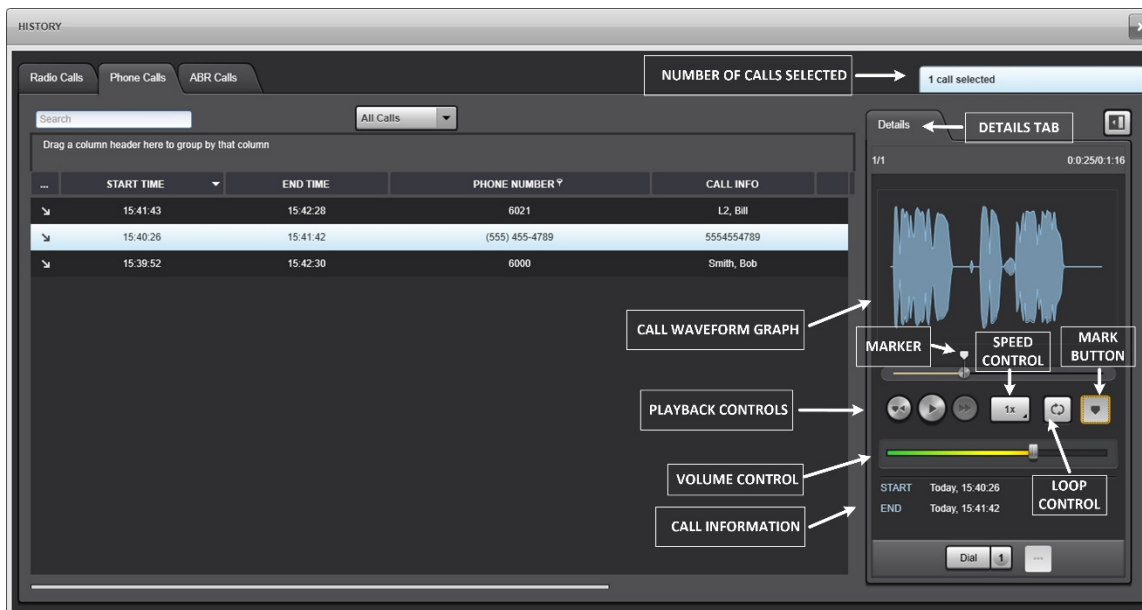



Figure 3-70: Extended Call History – SIP Phone Calls

Table 3-13 lists and describes the columns displayed on the Phone History dialog. Click a column header to sort entities by that column or click  and specify filter criteria. Click and drag a column header as shown in Figure 3-71 to group entities by that column.

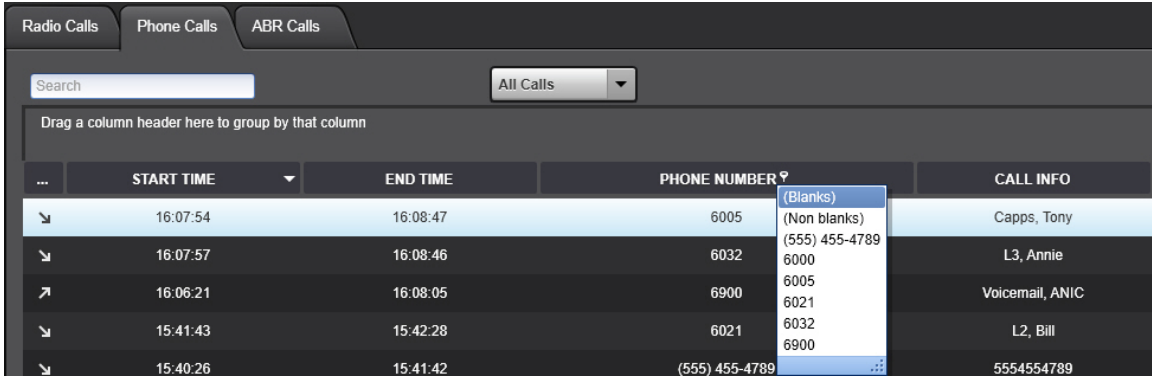


Figure 3-71: Group by Phone Call History Column

Table 3-13: History Dialog Columns (Call Director Phone Calls)

COLUMN	DESCRIPTION
...	Icons to highlight aspects of phone calls, e.g., call director, patch, etc.
START TIME	When the call started (HH:MM:SS).
END TIME	When the call ended (HH:MM:SS).
CALL TYPE	Indicates the type of call, e.g., including Call Director.

Table 3-14: History Dialog Columns (SIP Phone Calls)

COLUMN	DESCRIPTION
...	Icons to highlight aspects of phone calls, e.g., inbound, outbound, etc. See Section 3.4.5.2 for list of icons.
START TIME	When the call started (HH:MM:SS).
END TIME	When the call ended (HH:MM:SS).
DURATION	Indicates the duration of the call.
PHONE NUMBER	Indicates the phone number of the other party of the telephone call.
CALL INFO	Indicates the name, if configured in the directory. Displays the Phone number if the contact is not currently present in the directory.
LINE	Indicates the line associated with the call (e.g., L1 for line 1). The primary line is indicated by a red circle in the background.
TELEPATCH	Indicates the entity that was TelePatched during this call.

The Phone Call tab with SIP Telephony includes a drop-down filter button (next to the search box) which filters calls by type such as inbound, outbound, or missed calls.

3.5.2.2.1 Details Tab

The Details tab includes contextual information about the highlighted call, as well as playback controls. The Details Tab includes:

Playback Controls

Highlight one or more calls from the list to activate the playback controls. See Section 5.6 for more information on using the Playback Controls.

Playback Slider

The playback slider for phone calls works just like radio calls except that the waveform only shows one minute of the call at a time.

Since phone calls are typically much longer than the standard radio call, recording of phone calls is managed differently. Rather than recording each phone call as one continuous call, phone calls are recorded in one-minute segments and stored as a series of files on disk. When the dispatcher chooses to playback a phone call, the Symphony console manages loading each minute segment as it is needed to playback the call contiguously. The slider still represents the entire call. When changing the location of the slider, the waveform updates to show the minute associated with the current slider position.

Call Waveform Graph

The Call Waveform Graph displays the highlighted call's audio levels graphically.

Call Information

The call information displayed includes the start/end time and call sub-type.

The Start and End Times display the date using MM/DD/YYYY HH:MM:SS format. When the date is the current date, "Today" is displayed instead of MM/DD/YYYY. The time format includes (AM/PM) if configured for non-military time.

Playback Speed Control

The speed control changes the speed of playback while maintaining the original pitch. Click the speed control to reveal a list of supported speeds. Values less than 1 slow down the playback (e.g., .5 is half speed) while values greater than 1 speed up the playback. The speed cannot be changed during playback. Changing the speed only applies to the current call being played back. Highlighting a new call for playback or closing the History Dialog resets the playback speed to normal speed (i.e., 1x).

Mark Button

The mark button sets the starting point for playback and locks it until the Mark Button is pressed again to clear the mark. The marked position is indicated by the mark icon above the slider. This allows the user to select a point of interest in the call and quickly return to it using the back button.

To set the mark:

- Find the point of interest on the slider by clicking on the point or by pausing the playback at the desired point.
- Press the mark button.
- The mark button will glow, and the mark icon will appear above the slider.
- At any time, press the back button to move to the mark. If the call is playing back when the back button is pressed, the call will continue playing at the mark point.
- To repeat a section of the call, set the mark and speed control, then press play. Click the back button to restart at the mark and repeat as needed.

Press the glowing mark button to clear the mark.

Loop Control

The Loop Control allows the console user to put playback calls in the loop mode. Press the loop control button to activate loop mode. The button glows to indicate the loop mode is active. Press the loop control button again to deactivate loop mode.

When the loop control is active and multiple calls are selected, the calls play one after the other. When all calls have been played, the first call begins playing again, continuing until you click the Loop Control to stop. If only one call is selected, that call plays over and over until you click the Loop Control to stop.

To use loop mode:

1. Highlight one or more calls in the call list.
2. Press the loop control button to activate loop control mode. The button glows to indicate the change in state.
3. Press the Play button to start playing the highlighted call(s).
4. To pause playing back calls, press the pause button. This will not disable the loop mode.
5. To end loop mode, press the loop control button or highlight another call in the call list.

The loop control can only be changed when the Play button is not pressed (i.e., no call is playing back). Changing the playback list disables the loop mode and stops the current playback.

If the mark is set, the loopback will play to the end of the highlighted call and loopback to the marked position.

Dial Control (SIP Telephony only)

The Dial Control allows you to call back the party highlighted in the call list.

More Menu (SIP Telephony only)

The More Menu reveals advanced features, such as adding the highlighted party to the local contacts list.

3.5.2.3 ABR Call History

The ABR call history is similar to the radio call history.

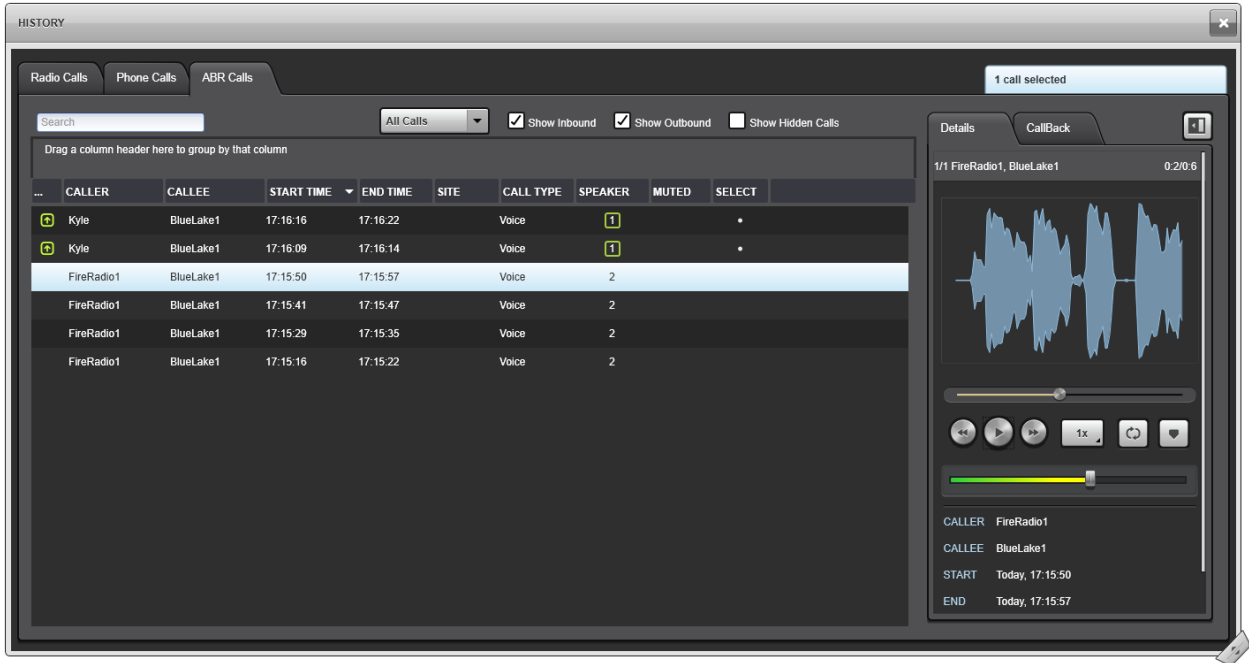


Figure 3-72: Extended History Dialog - ABR Calls

Table 3-15 lists and describes the columns displayed on the ABR History dialog. Click a column header to sort entities by that column or click and specify filter criteria. Click and drag a column header as shown in Figure 3-72 to group entities by that column.

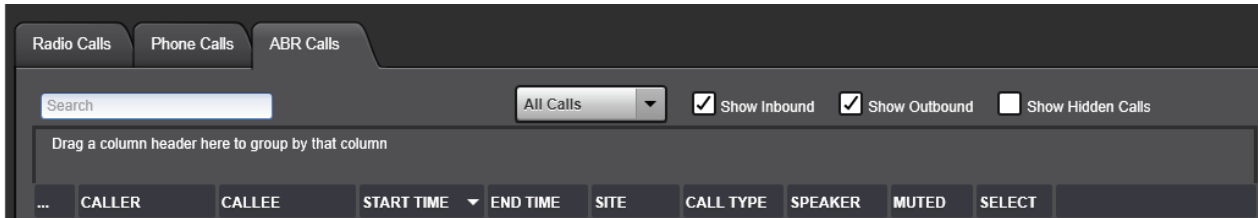


Figure 3-73: Group by ABR Call History Column

Table 3-15: History Dialog Columns (ABR Calls)

COLUMN	DESCRIPTION
...	Icons to highlight aspects of calls like selected, muted, in emergency, etc.
CALLER	The caller for the call.
CALLEE	The callee for the call.
START TIME	When the call started (HH:MM:SS).
END TIME	When the call ended (HH:MM:SS).
SITE	The site ID used for the call.

COLUMN	DESCRIPTION
CALL TYPE	Indicates the type of call including Voice, Marker Tone, Alert Tone, and page. Outbound calls are properly tagged since the console knows which call it placed. Inbound calls may not provide enough information to identify the specific call type, in which case the default Voice type will be used.
SPEAKER	The speaker assigned to the entity at the time of the call.
MUTED	Indicates whether or not the entity was muted at the time of the call.
SELECT	Indicates if the call was in the Select module when the call occurred.

The ABR Calls tab includes a drop-down filter button (next to the search box) which filters calls by type such as inbound, outbound, or missed calls.

3.6 QUICK ACTION KEYS

Refer to Table 3-16 for a list of quick action key combinations.



NOTE

Only one quick action key combination should be pressed at a time. Multiple concurrent quick action keys (ex: attempting to increase the selected module's volume via quick action key while simultaneously transmitting on a selected module via quick action key) are not supported.



NOTE

To avoid unexpected results such as screen rotation when using Quick Action keys, disable the Intel Graphics Driver hotkey commands by clicking on the Intel HD Graphics System Tray icon → **Graphic Options** → **Hot Keys** → **Disable**.



NOTE

Press the Num Lock key on the console's keyboard to enable the Num Lock function. This will ensure that Quick Action keys and keys on a Programmable Keypad function properly when the corresponding Quick Action Keys include the (Number Pad) as part of their KEY COMBINATION.

Table 3-16: Quick Action Keys

KEY COMBINATION	ACTION
CTRL + T	Transmits on the Selected Module.
CTRL + "+" (Number Pad)	Increases Selected Module Volume.
CTRL + "-" (Number Pad)	Decreases Selected Module Volume.
CTRL + M	Mutes Selected Module.
CTRL + U	Mutes all Unselected Modules.
CTRL + O	Mutes audible alert tone.
CTRL + R	Replays last Select call.
CTRL + L	Transmits alert tone on Selected Module.
CTRL + ,	Changes the Selected Module to the previous Selected entity.
CTRL + 1 (Number Pad)	Changes the preferred alert tone type to Alert on Selected Module.
CTRL + 2 (Number Pad)	Changes the preferred alert tone type to Warble on Selected Module.
CTRL + 3 (Number Pad)	Changes the preferred alert tone type to Pulse on Selected Module.

KEY COMBINATION	ACTION
CTRL + 4 (Number Pad)	Changes the preferred alert tone type to Custom1 on Selected Module.
CTRL + 5 (Number Pad)	Changes the preferred alert tone type to Custom2 on Selected Module.
CTRL + ALT + 1 (Number Pad)	Changes the preferred alert tone type to Alert and transmits the Alert tone on Selected Module.
CTRL + ALT + 2 (Number Pad)	Changes the preferred alert tone type to Warble and transmits the Warble tone on Selected Module.
CTRL + ALT + 3 (Number Pad)	Changes the preferred alert tone type to Pulse and transmits the Pulse tone on Selected Module.
CTRL + ALT + 4 (Number Pad)	Changes the preferred alert tone type to Custom1 and transmits the Custom1 tone on Selected Module.
CTRL + ALT + 5 (Number Pad)	Changes the preferred alert tone type to Custom2 and transmits the Custom2 tone on Selected Module.
CTRL + P	Enables/Disables marker tone on Selected Module.
(ALT + Z) + H	Opens the History Sidebar Panel.
(ALT + Z) + P	Opens the Patch Sidebar Panel.
(ALT + Z) + S	Opens the SimulSelect Sidebar Panel.
(ALT + Z) + E	Opens the Emergency Sidebar Panel.
(ALT + Z) + I	Opens the I-Call Sidebar Panel.
(ALT + Z) + A	Opens the Speaker Sidebar Panel.
(ALT + Z) + M	Opens the main menu.
(ALT + Z) + T	Opens the Telephony Sidebar Panel.
(ALT + Z) + R	Opens the RTT Sidebar Panel.
(ALT + Z) + U	Opens the RSM Sidebar Panel.
(ALT + Z) + B	Opens the Advanced Paging Sidebar Panel.
(ALT + Z) + V	Opens the Supervisory Sidebar Panel.
(ALT + Z) + Y	Opens the System Status Sidebar Panel.
(ALT + Z) + C	Opens the Cross Mute Sidebar Panel.
SHIFT + ALT + G	Opens the group directory tab.
SHIFT + ALT + U	Opens the user directory tab.
SHIFT + ALT + C	Opens the contacts directory tab.
SHIFT + ALT + H	Opens the extended history dialog.
SHIFT + ALT + B	Opens the Base Station Control directory tab.
SHIFT + ALT + A	Opens the Aux I/O directory tab.
SHIFT + ALT + P	Opens the Paging directory tab.
SHIFT + ALT + 1 (Number Pad)	Activate/Deactivate Patch1 (Custom Keypad Key P1).
SHIFT + ALT + 2 (Number Pad)	Activate/Deactivate Patch2 (Custom Keypad Key P2).
SHIFT + ALT + 3 (Number Pad)	Activate/Deactivate Patch3 (Custom Keypad Key P3).
SHIFT + ALT + 4 (Number Pad)	Activate/Deactivate Patch4 (Custom Keypad Key P4).
SHIFT + ALT + 5 (Number Pad)	Activate/Deactivate Patch5.
SHIFT + ALT + 6 (Number Pad)	Activate/Deactivate Patch6.

KEY COMBINATION	ACTION
SHIFT + ALT + 7 (Number Pad)	Activate/Deactivate Patch7.
SHIFT + ALT + 8 (Number Pad)	Activate/Deactivate Patch8.
SHIFT + CTRL + 1 (Number Pad)	Activate/Deactivate SimulSelect1 (Custom Keypad Key S1).
SHIFT + CTRL + 2 (Number Pad)	Activate/Deactivate SimulSelect2 (Custom Keypad Key S2).
SHIFT + CTRL + 3 (Number Pad)	Activate/Deactivate SimulSelect3 (Custom Keypad Key S3).
SHIFT + CTRL + 4 (Number Pad)	Activate/Deactivate SimulSelect4 (Custom Keypad Key S4).
SHIFT + CTRL + 5 (Number Pad)	Activate/Deactivate SimulSelect5.
SHIFT + CTRL + 6 (Number Pad)	Activate/Deactivate SimulSelect6.
SHIFT + CTRL + 7 (Number Pad)	Activate/Deactivate SimulSelect7.
SHIFT + CTRL + 8 (Number Pad)	Activate/Deactivate SimulSelect8.
CTRL + TAB	Goes to the next Sidebar Panel, if one of the Sidebar Panels has focus.
CTRL + SHIFT + TAB	Goes to the previous Sidebar Panel, if one of the Sidebar Panels has focus.
ALT + X	Opens the Radio TextLink client application (if installed) in Symphony integrated mode or brings the already running Radio TextLink client application to the foreground.
F1	Opens the online help.
SIP TELEPHONY ONLY	
ALT + A	Answer Ringing Call: Answers the oldest ringing call and auto places the current call on hold.
ALT + H	Hold: Places the current active call on hold. Pressing again resumes.
ALT + E	End call: Hangs up the current active call.
ALT + T	Transmit to phone in half duplex mode. When the headset is not plugged in, press this hotkey while speaking. Release when finished talking.
ALT + M	Mute/Unmute the mic audio to the telephone.
ALT + B	Decline the oldest ringing call and set the Callback reminder.
ALT + L	Redial last number.
ALT + 0-9 (Number Pad)	Speed dial (1-10); Number Pad 0 = 10
ALT + F	Transfer: <ul style="list-style-type: none"> • If the current call is not tagged as the target for a transfer, opens the transfer menu in the Telephony Panel for the current call and the dial pad. This is the equivalent of pressing the More button, then selecting Transfer, and selecting To Number to access the dial pad. Note the Baton opens the Telephony popup as well as the dial pad. • If the current call is tagged as the target for a transfer, complete the transfer. This is the equivalent of pressing the More button and selecting Complete Transfer. See Section 6.2.8 for more details.

KEY COMBINATION	ACTION
ALT + C	N-way conference: <ul style="list-style-type: none"> If no call is active, no action is taken. If the current call is not tagged as the target for a conference, opens the conference menu for the current call and the dial pad. This is the equivalent of pressing the More button, then selecting New Conference, and selecting To Number to access the dial pad. Note the Baton opens the Telephony popup as well as the dial pad. If the current call is tagged as the target for a conference, completes the conference join. This is the equivalent of pressing the More button and selecting Complete Conference. See Section 6.2.10 for more details.
ALT + N	New call popup with Dial Pad.
ALT + O	Opens overdial for the current active call, closes the overdial for the current active call, or takes the call off hold.
0-9,*,#,NumPad 0-9	Send DTMF tones while overdial is open and dial numbers when the dial pad is open.

3.7 FULL SCREEN DISPLAY

The Full Screen Display, shown in Figure 3-74, is organized like most modern user interfaces. Important elements that are used frequently are located in a toolbar at the top known as the Select Bar. Feature-based panels are located on the Sidebar to the left while the primary Workspace consumes the majority of the screen, providing access to communications modules. The application status is summarized at the bottom in the Status Bar.



Figure 3-74: Sample Full Screen Display

The following briefly describes the primary elements of the full screen display:

- Select Bar – The Select Bar hosts the Select Module, the Select Bar Telephony Control, the menu button, the clock, and the audio widgets. It is prominent and persistent, ensuring quick and easy access to important controls.

- **Sidebar Panels** – Sidebar Panels are organized by feature such as I-Call or Emergency. They provide a summary of relevant activity, alert the dispatcher of any activity requiring action, and provide the data and controls necessary to take action. Sidebar Panels provide overviews and summaries of important activity that the dispatcher would otherwise have to search for separately on the various module pages. See Section 3.4 for more information on Sidebar Panels.
- **Workspace** – The Workspace organizes communications modules into tabs containing related modules, e.g., Police and Fire. It also provides controls to initiate functions, like Patch and SimulSelect, which require picking multiple modules. The Workspace is discussed in more detail later in this section.
- **Status Bar** – The Status Bar displays information about the mode (backup radio versus console mode), user, setup, servers, logging, accessories, and network connection. The Status Bar is discussed in more detail later in this section.

3.7.1 Workspace

The Workspace organizes communications modules into tabs as show in Figure 3-75. Each tab contains a unique collection of communications modules though a communications module can be in more than one tab.

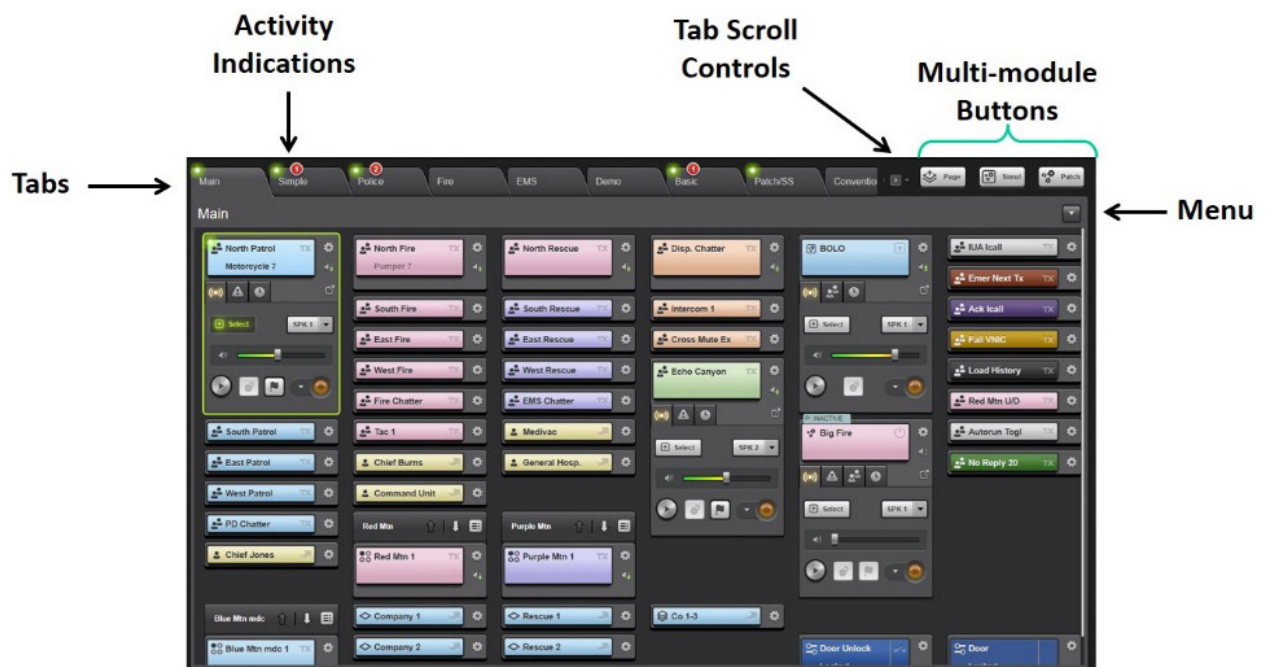
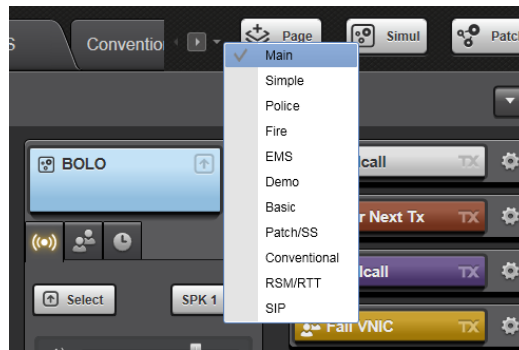


Figure 3-75: Workspace Elements

The Workspace elements include:

- **Tabs:** Each tab contains a unique collection of communications modules. Click on the desired tab to show the modules in that tab.
- **Activity Indications:** Each tab has activity indications that show when calls are in progress or that the tab contains an entity in the emergency state. This allows the dispatcher to quickly and easily identify activity occurring on all tabs.

- Tab Scroll Controls: When there are more tabs than can be shown at once, a set of navigation arrows appear to scroll through the available tabs. There is also a drop-down menu to directly select any tab.



- Multi-module Buttons: These buttons initiate features that require picking multiple modules such as Patch or SimulSelect. These buttons are detailed later in this manual.
- Menu: The menu provides quick access to the Extended History and Directory dialogs.

3.7.2 Web Browser Workspace

If the Symphony workstation is licensed and configured for the Web Browser feature, the dispatcher can use one Web Browser Workspace and its one to four pre-configured subtabs to access selected VIDA System components such as the Unified Administration System (UAS) and Regional Network Manager (RNM).

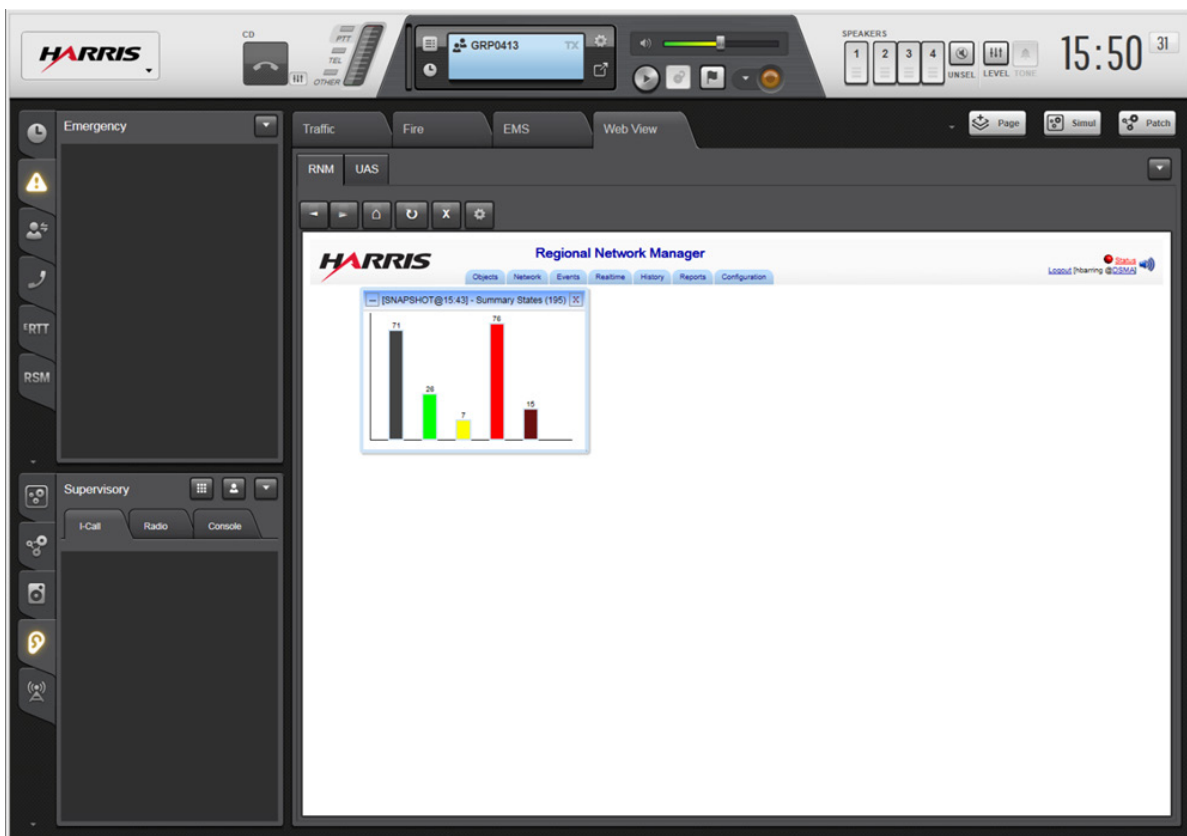


Figure 3-76: Web Browser Workspace

The top of the Web Browser Workspace includes one to four subtabs, each assigned to its own URL. Click on each subtab to access its assigned URL and display it.

3.7.2.1 Status Bar

The Status Bar displays information about the mode (e.g., Backup Radio versus Console Mode), user, setup, servers, logging, accessories, and network connection. The Status Bar consists of six regions each supporting a different control. Some of the controls open an information dialog to display more detailed information. See Section 3.2 for details on the System History and Network Status.

3.7.2.2 Console Identification

The Console Identification control on the Status Bar is divided into two areas. The first area (to the left) displays the dispatcher's Alias. Hover over this area using the mouse to see your numeric User ID. The second area displays the text "CONSOLE" with the console's PSAP ID displayed to the right. The console's ID is comprised of the Wide Area Communications Network (WACN), P25 System (also known as the VIDA region) region, and console ID respectively.



3.7.2.3 Setup

The Setup control on the Status Bar shows the name of the current setup. If your administrator has configured multiple setups, switch between them using the Main Menu. This area on the Status bar will change to the name of the new setup each time you switch setups.



Figure 3-78: Setup Name Example

3.7.2.4 Server Status

The Server Status section displays a SERVERS button which opens a dialog showing a list of servers and their current status. To the right of the SERVERS button, there is a summary area that displays information for the important servers. Click ▲ to expand the Servers dialog (Figure 3-79). The Servers information dialog displays the name and state for all servers in use by the console. The states for the server include: Connected, Disconnected, Poll Failing, Not Licensed, and Not Configured.

- PHONE – The IP Private Branch Exchange (PBX) that provides telephony communication utilizing SIP and RTP. The state Partially Connected indicates that at least one, but not all lines successfully registered with the PBX.
- VNIC – The server that allows you to talk to the radio system (and have the radios talk back).
- Audio Box – The other part of the Symphony Dispatch Console that plays audio and interfaces with accessories.
- Aux I/O Server – The server that communicates with all of the I/O points monitored by the Symphony Dispatch Console. Specifically, the server is the Regional Network Manager which communicates with one or more Network Sentries which manage the physical I/O points.

- Key Management Facility – The server that provides Symphony with encryption information including encryption keys and bindings between talkgroups and keys.

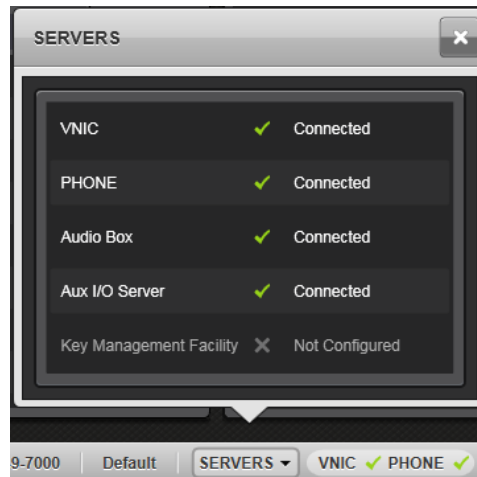



Figure 3-79: Server Status

3.7.2.5 Peripherals

The Peripherals dialog displays the connection status of all peripherals, such as supervisor and dispatcher headsets. Click  to expand the PERIPHERALS dialog (Figure 3-80).

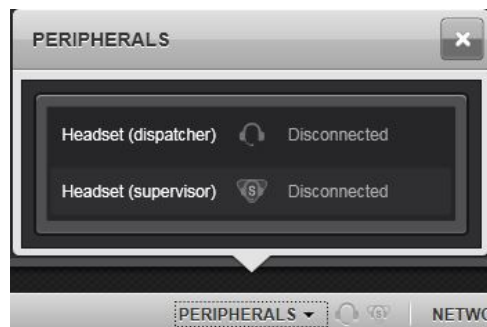


Figure 3-80: Peripherals - Expanded

3.8 BATON

The Baton is designed to minimize screen space and improve efficiency when used in conjunction with a Computer Aided Dispatch application. Most of the Full Screen functionality is provided with the Baton. Figure 3-81 is an example of the Baton display. The Baton can be operated on the Symphony Console or on a remote Computer Aided Dispatch computer. In the latter case, the Symphony Dispatch Platform is still required to provide connectivity to the VIDA system and audio accessories, while the Baton runs on the CAD computer to provide an integrated user experience which shares the same screen, keyboard, and mouse between Symphony and the CAD program.



Figure 3-81: Sample Baton Display – Unique Elements

Many of the Baton elements are similar to those on the Select Bar and have been covered earlier. The items unique to the Baton are shown in Figure 3-81 and are detailed in this section. The Baton is configurable, so your Baton may have different widgets than shown in this manual.

To move the Baton window, press **Alt + Space** to pop out the window menu, and then press **M** to move.




NOTE

Move the Baton using the cursor keys only. Once the baton is correctly located, “lock” the location by pressing **Enter**. The move process has to be started using the cursor keys. Moving the mouse after hitting a cursor key makes the placement difficult to control.

The Baton hides panels to save space. The panels are available as popups that are activated by the buttons shown below. These buttons are just like the tabs used on the Full Screen Display Sidebar. The buttons use the same icons and badges as the equivalent Sidebar Panel. Press the button to pop out or close the associated panel. This bank of buttons may not include all available panels. Some panels have dedicated controls in a widget such as the Compact Telephony Control. Rarely used panels are available through the main menu to conserve space.



The Workspace Widget provides access to a workspace tab defined in the Full Screen Display. The widget contains two modules and controls access to other workspace modules. Click  to change the displayed Workspace tab as shown in Figure 3-82. The Workspace list corresponds to the Workspaces defined for the Full Screen display.

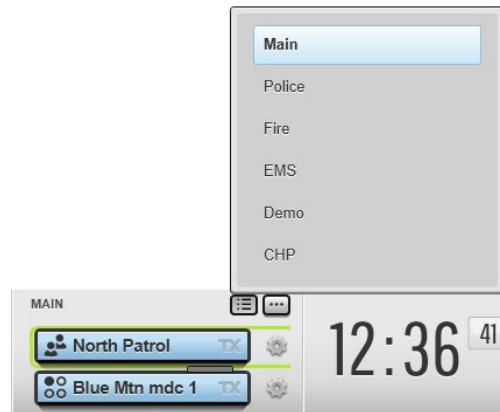



Figure 3-82: Changing the Workspace Tab

Click  to bring up the Workspace panel to access all modules on the Workspace and change the displayed modules on the Baton (top or bottom) as shown in Figure 3-83. The Workspace panel includes all modules on the associated Full Screen Display Workspace tab. Use the context menu beside each module to pop-out normal size modules with more controls. Click the box to the left of the module to change the modules position on the Baton. A small menu appears allowing you to select the top or bottom position.

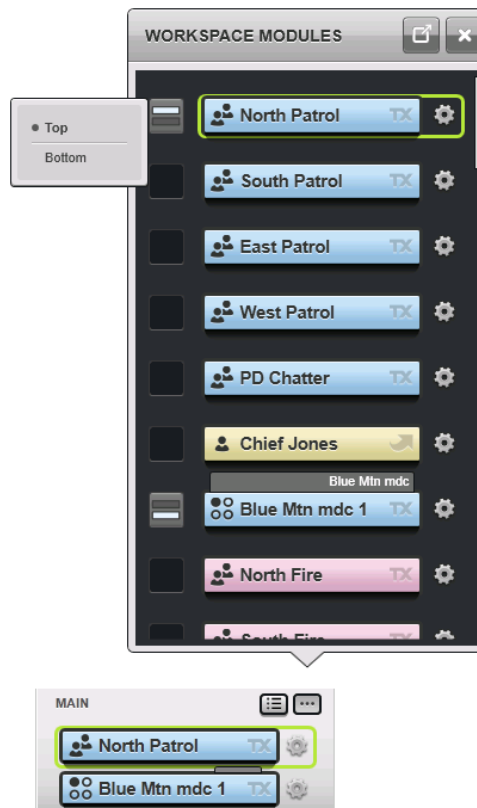


Figure 3-83: Workspace Panel

4 CONSOLE AUDIO SYSTEM

At the console, audio received from a radio unit or another console is heard in either a speaker or a headset. During a console transmission, dispatcher voice audio is applied to a microphone (“mic”) connected to the console. The specifics on console receive and transmit audio functions are described in this chapter.

The console can be configured with a telephony style handset and hook switch instead of a headset. This handset can be operated in one of two ways depending on how Symphony is configured by the administrator.

4.1 SPEAKER AND HEADSET

The console can be equipped with up to eight speakers, one of which is designated as the “select” speaker. This is “speaker 1” in the console system. All other speakers connected to the console are designated “unselect” speakers (7 maximum). If present, these speakers are numbered 2 through 8. In most console systems, only one unselect speaker is utilized—speaker 2. All speakers used with the console have a rotary volume control located on the speaker’s front panel. This volume control is in addition to the volume control that is associated with individual Communication Modules.

Headset types include one designated as the “operator” headset and one designated as the “supervisor” headset. Each headset has speaker elements, normally referred to as “earphones,” a volume control, and a microphone for use during console transmissions. In some cases, separate headset jacks are provided at the console for simultaneous connection of both an operator headset and a supervisor headset.

As mentioned above, each speaker and headset has a mechanical volume control that provides local volume adjustment. For unselected modules that share a common unselect speaker, changing the speaker’s volume setting with the mechanical control changes the audio level of all the summed unselect audio. This is in contrast to module volume control where volume settings can be controlled on an individual module basis.

4.2 TELEPHONY-STYLE HANDSET

The optional, telephony-style handset has two hardware components. The handset itself contains a speaker, microphone, and a PTT button. The associated hook switch provides a cradle to place the handset in when not in use. The hook switch signals the console when the handset is picked up by the dispatcher. The handset becomes enabled while picked up and disabled when resting on the hook switch.

The Symphony administrator can configure how audio is routed to and from the handset. There are two possible modes of operation. In the first mode, the handset replaces the headset hardware. All audio routing and PTT signals behave as if the handset were a headset. Lifting the handset off the hook switch is equivalent to inserting the headset. In the second mode, the handset works primarily for phone audio, and radio audio is always routed to the speakers even if the handset is off hook. Refer to Section 4.12.1.3 for more information on SIP Telephony with a Handset. Note that the second mode applies only to SIP Telephony.

4.3 ENABLING AND DISABLING HEADSETS

To enable a headset, it must be plugged into the headset jack. The Symphony Dispatch Console will automatically detect that the headset is plugged in and begin routing audio appropriately. Symphony will detect a headset as an operator or supervisor headset based on the jack that it is plugged into.

4.4 MICROPHONE BASICS

Microphone types that can be used with the console include a headset mic, a desk mic, and a gooseneck mic. When the supervisor footswitch or headset PTT is used while an operator is PTTing from another source, the current call remains active, but the Supervisor Headset Mic audio replaces the Desk Mic or Operator headset Mic Audio.



NOTE

The Gooseneck microphone is designed for close-talking operation. For best performance, speak as close to the microphone as comfort permits. A distance of 1 to 5 cm (0.4 to 2 inches) provides the greatest discriminations between voice and background noise.



NOTE

Although the desktop microphone gain is adjustable, the microphone is intended to be used relatively close to the talker, about 12 inches. Setting the gain for operation at longer distances makes the microphone more susceptible to picking up background room noise.

Table 4-1 describes the Microphone and call Destination when PTT from a single source.

Table 4-1: Microphone and Call Destination

PTT SOURCE	MICROPHONE USED	TRANSMIT TO
NO HEADSETS INSERTED		
Desk Mic PTT	Desk Mic	Select Entity
Operator Footswitch	Desk Mic	Select Entity
Supervisor Footswitch	Desk Mic	Select Entity
Mouse/Touchscreen (Unselect Module)	Desk Mic	Clicked Module
Mouse/Touchscreen/Keyboard (Select Module)	Desk Mic	Select Entity
ONLY OPERATOR HEADSET INSERTED		
Desk Mic PTT	Desk Mic	Select Entity
Operator Footswitch	Operator Headset Mic	Select Entity
Supervisor Footswitch	Desk Mic	Select Entity
Mouse/Touchscreen (Unselect Module)	Operator Headset Mic	Clicked Module
Mouse/Touchscreen/Keyboard (Select Module)	Operator Headset Mic	Select Entity
Operator Headset PTT	Operator Headset Mic	Select Entity
ONLY SUPERVISOR HEADSET INSERTED		
Desk Mic PTT	Desk Mic	Select Entity
Operator Footswitch	Desk Mic	Select Entity
Supervisor Footswitch	Supervisor Headset Mic	Select Entity

PTT SOURCE	MICROPHONE USED	TRANSMIT TO
Mouse/Touchscreen (Unselect Module)	Supervisor Headset Mic	Clicked Module
Mouse/Touchscreen/Keyboard (Select Module)	Supervisor Headset Mic	Select Entity
Supervisor Headset PTT	Supervisor Headset Mic	Select Entity
OPERATOR AND SUPERVISOR HEADSETS INSERTED		
Desk Mic PTT	Desk Mic	Select Entity
Operator Footswitch	Operator Headset Mic	Select Entity
Supervisor Footswitch	Supervisor Headset Mic	Select Entity
Mouse/Touchscreen (Unselect Module)	Operator Headset Mic	Clicked Module
Mouse/Touchscreen/Keyboard (Select Module)	Operator Headset Mic	Select Entity
Operator Headset PTT	Operator Headset Mic	Select Entity
Supervisor Headset PTT	Supervisor Headset Mic	Select Entity

4.5 VOLUME INDICATIONS

The VU (Volume Unit) Meter (Figure 4-1) indicates outgoing audio levels during a console foreground transmission to the radio system and to SIP telephone calls. Foreground audio includes mic audio, alert tones and foreground pages. The VU meter does not display indications for mic audio transmitted to Call Director. This panel is located in the top of the Full Screen display and to the left of the Select Module on the Baton. See Figure 3-74 for the VU location.



Figure 4-1: VU Meter - Full Screen Display



Figure 4-2: VU Meter - Baton

When transmitting a call from the console to a selected module, the unselected modules (instant transmit), or a SIP telephone call, the VU meter panel indicates the audio level that is applied to the open microphone. The LEDs on the side of the VU meter indicate the transmission type. PTT LED indicates a foreground call to the radio system. The Tel LED indicates an SIP Telephone call, and the Other LED indicates background transmissions such as PMT and background pages.

When transmitting, it is best to speak directly into the open microphone in a normal voice level. Shouting in a microphone is never necessary. The best voice audio clarity at the receiving units can be achieved by maintaining, on average, a mid-range VU meter indication while speaking into the microphone. However, occasional maximum and minimum indications during a PTT sequence are normal and should be expected. A minimum indication almost always briefly appears between spoken words. To achieve an optimum audio level, adjust voice level and/or mouth-to-microphone distance as necessary during transmissions.

Excessive background noise during a PTT sequence may cause indications to appear on the VU meter panel when the open microphone picks up this audio. This should be avoided whenever possible to prevent background noise from being transmitted.

4.6 PRIORITY GROUPS

Groups/Units can be assigned to speaker 1 (select speaker) as their unselect speaker. This makes them a “priority” group. Audio from these groups and units is sent to the select device; the select speaker or headset earpieces.

- These groups are NOT selected.
- They are muted when mute all unselect is pressed.
- They are not transmitted to when a select transmit action is performed.

4.7 RECEIVE AUDIO ROUTING

Audio received from the “selected” module and priority groups is heard in the “select” speaker. However, if a headset is connected to the console, the select speaker may be muted (muting is optional, dependent on setting “Set Select Speaker Always On,” set by the administrator), in which case select audio is heard only in the connected headset.¹

Audio received from “unselected” modules is heard in an “unselect” speaker, typically speaker 2. Receive audio that is simultaneously received from two or more unselected modules shares an unselect speaker and the audio is summed (mixed) together in the speaker. The console can be equipped with up to seven unselect speakers, identified as speakers 2 through 8. Each programmed Communication Module has an assigned unselect speaker to which the module’s incoming audio is routed when the module is unselected. The factory default unselect speaker is speaker 2. If allowed by the administrator, each programmed module’s unselect speaker can be changed from the default unselect speaker to any available unselect speaker. To change a module’s current unselect speaker, select from the speaker selection drop-down (see Figure 3-24). The “Can Change Entity Unselect Speaker” configuration setting controls this behavior.

When an emergency exists on one or more programmed Communication Modules, audio received from units calling on the groups is routed per standard methods. If the emergency group’s module is selected, audio from units calling on the group is heard in the select speaker. If an emergency group’s module is unselected, audio from units calling on the group is heard in the module’s unselect speaker. Alarm audio is routed in the same way.

4.8 AUDIO FOR INTERCOM CALLS

Audio for Intercom calls is routed using the standard select and unselect audio routing rules. Intercom transmit operations also follow the standard select and instant TX rules.


4.9 AUDIO FOR PATCHES

Audio routing for calls received on an active Patch occurs similarly to non-Patch operations. If a Patch is selected, incoming call audio on the Patch is routed to the select speaker or the headset. If a Patch is unselected, incoming call audio on the Patch is routed to the unselect speaker.²

When a Patch activates, all of its modules’ volume settings equalize to the highest setting among the patched modules. To adjust a Patch’s volume setting, choose a module in the Patch or the Patch module and perform volume up or down actions as required. The volume indicators on all of the patched modules change simultaneously.

¹ During local Call Director operations, select audio is directed to the select speaker while a headset is in use.


² If the console is equipped with more than one unselect speaker and modules in the patch have different unselect speaker settings, receive audio on the patch is routed to the unselect speaker as determined by the module with the lowest unselect speaker number.

Muting one module in an active Patch mutes all modules in the Patch. During a Patch mute condition,  is displayed next to the volume indicators in all of the patched modules. To mute a Patch, choose any module in the Patch and toggle the module mute function on (see Section 5.2.7). Repeat this operation to unmute a muted Patch. If a muted Patch is deactivated or cleared, all modules in the Patch return to their pre-patched conditions. Independent module volume adjustments can then be performed.

Muting all unselected modules via the consoles “mute all” function mutes all modules in an unselected Patch. If a Patch is selected during a mute all condition, none of the modules in the Patch are muted—the Patch does not mute.

4.10 AUDIO FOR SIMULSELECTS

All modules included in a SimulSelect are selected when the SimulSelect is activated. Unlike a Patch, the volume settings of the SimulSelected modules are not equalized. The volume setting of each module can be individually controlled. Audio summation occurs in the select speaker/headset when two or more calls are concurrently received on modules included in a SimulSelect.

Modules within a SimulSelect can be individually muted. To mute a module, choose the desired module and toggle the module mute function on. Repeat this operation to unmute a module. While a module is muted,  replaces the volume indication in the module. The “mute all” function does not affect a SimulSelect since all modules in the SimulSelect are selected.

4.11 AUDIO FOR CALL DIRECTOR

Call Director supports local operations where the dispatcher is connected to the local phone caller/callee and TelePatch operations where the dispatcher and phone caller/callee are connected to a radio entity such as a talkgroup or radio patch. Audio routing for these operational modes depends on whether or not the headset is plugged in. For TelePatches, the audio routing also depends on whether the radio entity is selected or not.

4.11.1 Local Operations

If a headset is used during local CD operations, audio is “full-duplex,” much like a normal telephone conversation. In other words, both parties always hear each other speak during the conversation and no console PTT action is necessary when the dispatcher speaks to the CD telephone callee/caller. The dispatcher speaks into the headset’s microphone and the CD telephone callee/caller is heard in the headset’s earphones. Refer to Table 4-2.

If a headset is *not* used during local CD operations, audio is “half-duplex from the dispatcher’s point-of-view. In other words, a console PTT action is necessary when the dispatcher speaks to the CD telephone callee/caller. Also, the CD telephone callee/caller’s voice mutes (at the select speaker) during the console PTT. The dispatcher speaks into the open microphone and the CD telephone callee/caller is heard in a console’s select speaker. Refer to Table 4-2.

Table 4-2: Audio Routing During Local CD Operations

	DESK MIC & SPEAKER OPERATIONS	HEADSET OPERATIONS
Headset/Handset Hears:		Telephone Audio
Select Speaker Output:	Telephone Audio & Select Audio	Select Audio
Unselect Speaker Output:	Unselect Audio	Unselect Audio

	DESK MIC & SPEAKER OPERATIONS	HEADSET OPERATIONS
Telephone Hears:	Dispatcher (half-duplex)	Dispatcher (full-duplex) excluding dispatcher PTTs to the radio system
Dispatcher Talks To Phone With:	CD Control Button (PTT necessary)	"hands-free" (no PTT necessary)
Dispatcher Talks To Radios With:	Unselected ("Instant") TX, Selected TX or Microphone PTT	Unselected ("Instant") TX, Selected TX or Footswitch PTT

4.11.2 TelePatch operations

Audio routing for Call Director TelePatch varies slightly from the routing which occurs with local CD. In addition, audio routing is handled differently when the entity to which the phone is patched is an unselect entity as opposed to the select entity. The CD TelePatch interaction is Full Duplex only when a headset is attached and the Phone is TelePatched to the select entity. In all other scenarios the interaction is Half-Duplex requiring the dispatcher to key the CD PTT button in order to communicate on the CD TelePatch. The following tables detail how the audio (both outbound and inbound) is routed during CD TelePatch operations

4.11.2.1 **CD TelePatch with a Headset**

4.11.2.1.1 TelePatch to Select Entity

In this scenario, the outbound audio for the CD TelePatch is sampled from the headset mic and is sampled as full-duplex. The mic is always active, no PTT is required to communicate with the TelePatch. Should the dispatcher wish to communicate with an entity outside the CD TelePatch, he/she may key on the Smart Button of the entity's module. This will temporarily interrupt transmission to the CD TelePatch and allow the outbound audio to be directed to the unpatched entity. As soon as the dispatcher un-keys, the headset mic reverts to full-duplex communication with the TelePatch. Incoming audio from both radios and the phone are routed to the headset earpiece.

Table 4-3: Audio Routing During CD TelePatch to Select Entity

	DESK MIC AND SPEAKER OPERATIONS	HEADSET OPERATIONS
Headset Hears:	N/A	TelePatch Audio (phone and radio)
Select Speaker Output:	TelePatch Audio (phone and radio)	N/A
Unselect Speaker Output:	Unselect Audio	Unselect Audio
TelePatch (Phone and Radio) Hears:	Dispatcher (half-duplex)	Dispatcher (full-duplex) excluding dispatcher PTTs to the radio system
Dispatcher Talks To TelePatch (phone and radio) With:	CD Control Button (PTT necessary)	"hands-free" (no PTT necessary)
Dispatcher Talks To Radios With:	Unselected ("Instant") TX, Selected TX or Microphone PTT	Unselected TX using module Smart Button



When CD TelePatched to the select entity with a headset attached, the foot switch and the headset PTT button are not operational.

4.11.2.1.2 TelePatch to an Unselect Entity

In a CD TelePatch to an unselect entity with the headset attached, the outgoing audio is also sampled from the headset mic. However, the mic operates as half-duplex, requiring the dispatcher to key the CD TX button in the select bar in order to transmit to the CD TelePatch (both phone and radios). This scenario causes the CD TelePatch's inbound audio to be routed to the unselect speaker assigned to the entity which is patched to the phone and the dispatcher will hear both phone and radio traffic in that speaker.

Table 4-4: Audio Routing During CD TelePatch to UnSelect Entity

	DESK MIC AND SPEAKER OPERATIONS	HEADSET OPERATIONS
Headset Hears:	N/A	Select audio
Select Speaker Output:	TelePatch Audio (phone and radio)	N/A
Unselect Speaker Output:	TelePatch Audio (phone and radio) and other Unselect Audio	TelePatch Audio (phone and radio) and other Unselect Audio
TelePatch (Phone and Radio) Hears:	Dispatcher (half-duplex) from desk mic	Dispatcher (half-duplex) from headset mic
Dispatcher Talks To TelePatch (phone and radio) With:	CD Control Button (PTT necessary)	CD Control Button (PTT necessary)
Dispatcher Talks To Radios With:	Unselected ("Instant") TX, Selected TX or Microphone PTT	Unselected ("Instant") TX, Selected TX or Microphone PTT

4.11.2.2 CD TelePatch Without a Headset

When the Symphony console sets up a CD TelePatch with no headset attached, outbound audio is always sampled from the main mic (typically the desk mic or gooseneck mic) in half-duplex. Transmit occurs to both phone and radios in the TelePatch when the dispatcher keys the CD TX button in the select bar. Inbound audio follows typical routing patterns for radio calls, i.e., if the phone has been patched to the select entity, inbound audio (both phone and radio) will be heard in the select speaker. If, however, the phone has been patched to an unselect entity, all inbound audio on the TelePatch will be heard on the unselect speaker assigned to the entity to which the phone is patched.

4.11.3 Phone Sidetone, Volume, and Mute

Phones connected to Symphony via Call Director produce sidetone (low-level audio diverted from the phone's microphone to its earpiece to provide feedback to the phone operator) which Symphony has no ability to remove or filter out. As a result, some of the audio controls during a CD TelePatch behave differently from what a dispatcher might expect. The effect of this sidetone is most noticeable when trying to adjust the volume and/or mute one or more of the components of the CD TelePatch. The effect is the same whether CD is patched to the select or an unselect entity and whether using headset or speakers. This behavior is detailed in the following table:

Table 4-5: CD TelePatch Volume/Mute Controls

OPERATION	EFFECT
Mute TelePatched Entity	Audio from phone and radio (via phone sidetone) heard in speaker
Mute Phone	Phone audio no longer heard in the speaker; radio audio still present
Mute TelePatched Entity and Phone	Neither Phone nor Radio audio heard in speaker

4.12 AUDIO FOR SIP TELEPHONY

SIP Telephony supports multiple calls simultaneously, but the dispatcher can only interact with one call at a time. This is like a desk phone where you have to place a call on hold to answer another call. Symphony allows you to easily switch between calls, but only one call is routed to the dispatcher while other calls are on hold.

SIP Telephony supports local operations where the dispatcher is connected to a phone call and TelePatch operations where a phone call is connected to a radio entity such as a talkgroup, or radio patch. Audio routing depends on the operational mode and whether the headset is plugged in or not. An optional telephony style handset is also available as a replacement for the headset hardware. This will be installed and configured by the dispatch administrator. TelePatch has additional dependencies covered later in this section.

Placing a call on hold means different things for different call types:

- **Single Party Call:** Hold stops the media flow between the console and the call party just like hold on any desk phone.
- **Conference:** Placing a conference on hold stops the media flow between the console and the conference. However, the conference continues for all other members of the conference.
- **Conference Member:** Placing an individual conference member on hold stops the media flow between conference member and the conference. However, the conference continues for all other members of the conference including the dispatcher.
- **TelePatch:** Placing a TelePatch on hold stops the media flow between the console and the TelePatch. However, the radio entity and phone call in the TelePatch continue to communicate with each other.

4.12.1 SIP Telephony Local Operations

4.12.1.1 SIP Telephony with a Headset

When no phone calls are active, the headset is dedicated to radio traffic. Select audio is heard in the headset earpiece and the headset mic is directed to a radio entity when a PTT is activated. When at least one telephone call is active, the headset is primarily dedicated to telephony operations. Select audio is directed to the select speaker and the dispatcher interacts with the focused telephone call using the headset in full duplex mode. That is, the dispatcher speaks into the headset mic and hears the telephone audio in the headset earpiece. There is no need to PTT to speak on the phone call. The dispatcher can still use the headset for radio communication by pressing a radio PTT. Headset mic audio is routed to the radio entity and muted to the phone call while the PTT is active. Once the PTT is released, the headset returns to full duplex operation with the phone call.

4.12.1.2 SIP Telephony without a Headset

When a headset is not attached and there is an active telephone call, the dispatcher interacts with the focused call using the select speaker and desk microphone. The dispatcher must press the Telephony PTT button while speaking into the desk microphone to speak on the telephone call. This is referred to as half duplex mode.

The Table 4-6 shows the comparison of audio routing when the console is in full duplex vs half duplex mode of operation during an active telephony call.

Table 4-6: Audio Routing During Telephony Calls

	HALF DUPLEX Desk Mic & Speaker Operations	FULL DUPLEX Headset Operations
Headset Hears:	N/A	Select Audio or Telephony Received Audio
Select Speaker	Telephony Received Audio and select audio	Select Audio
Dispatcher Speaks into the Headset microphone:	N/A	“Hands-free” (no PTT necessary)
Dispatcher speaks into the Desk microphone:	No Audio is transmitted	N/A
Dispatcher speaks into the Desk microphone while pressing Telephony PTT	Audio is transmitted on the active telephone call.	N/A

4.12.1.3 SIP Telephony with a Handset

The handset operation must be configured by the dispatch administrator. It replaces the headsets. The headset and handset cannot be installed at the same time.

The handset includes a hook switch, which detects when the handset is on-hook (not being used) and off-hook (in use).

While the handset is on-hook, audio is routed as described in Section 4.12.1.2.

While the handset is off-hook, it is used primarily for phone audio, and the desk mic is used for radio communications. In handset mode, all PTTs except for the PTT on the handset itself (mouse, foot switch, and desk mic), will key the desk mic only. Pressing the PTT on the handset will key the handset and send audio to the select radio entity.

Select radio audio is always sent to the Select Speaker even if the handset is off hook.

Table 4-7: Handset Mode Audio Routing with Handset Off-Hook

	NO ACTIVE PHONE CALL	ACTIVE PHONE CALL HEADSET OPERATIONS
Handset Hears:	Select Audio	Telephone Audio
Select Speaker	Select Audio	Select Audio
Dispatcher Speaks into the Handset microphone:	N/A	“Hands-free” to phone (no PTT necessary)
Dispatcher keys the Radio. Mouse PTT, Footswitch, or Desk Mic PTT	Desk Mic Audio is transmitted to the radio.	Desk Mic Audio is transmitted to the radio. Audio to the phone is muted.
Dispatcher presses the handset PTT	Handset Audio is transmitted to the radio.	Handset Audio is transmitted to the radio. Audio to the phone is muted.

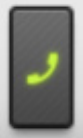

4.12.2 SIP TelePatch

A TelePatch connects a phone call to a radio entity such as a talkgroup, base station or radio patch. The dispatcher can join the TelePatch or just monitor it in the background. Audio routing for TelePatch has additional dependencies over local telephony including whether the call is focused or unfocused and whether the radio entity is selected or not. These dependencies are defined as:

- Focused: A call is focused if it is loaded in the Call Control Area and Compact Telephony Control and is not on hold. That is, the dispatcher is actively exchanging media with the call.
- Unfocused: The call is on hold. That is, the dispatcher is not actively exchanging media with the call. A call can be placed on hold manually by the dispatcher or automatically by Symphony when the dispatcher focuses a different call.
- Selected: The radio entity in a TelePatch is selected (i.e., assigned to the select module).
- Unselected: The radio entity in a TelePatch is not selected.

In some cases, the TelPTT (Section 3.1.2) must be used to talk to a phone call. The TelPTT takes the form of a button in these cases as shown in Table 4-8.

Table 4-8: TelPTT Transmit States

TELPTT	ACTION
	Press to talk on a phone call.
	Press to talk on a TelePatch.

4.12.2.1 Audio Routing with Headset

TelePatch routing with a headset depends on whether the call is focused or unfocused and whether the radio entity is selected or unselected. The four possible combinations are explained in the following sections.

4.12.2.1.1 Selected and Focused

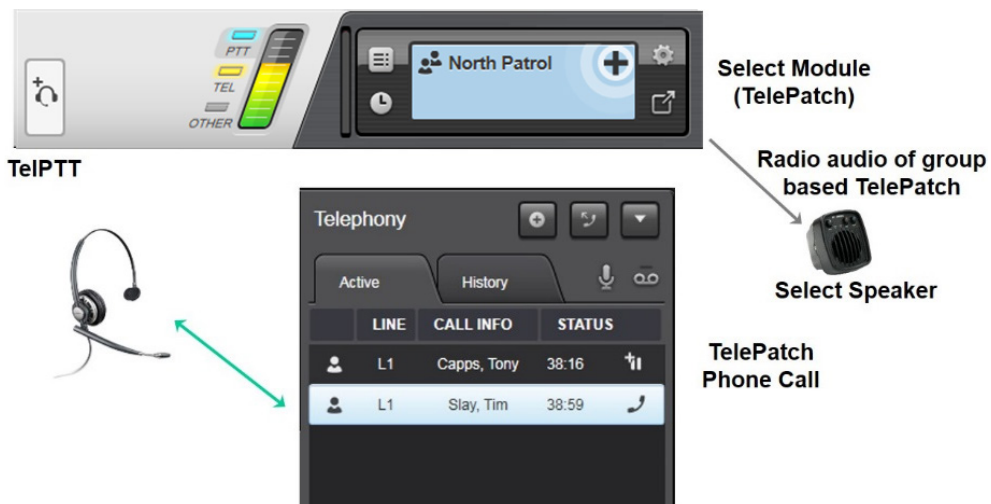
When the TelePatched entity is selected and the TelePatched phone call is focused (i.e., highlighted and not on hold), the dispatcher is actively engaged in the TelePatch interacting with the associated phone and radio entities via the headset in full duplex mode. Other select audio such as priority group and call history playback continue to be routed to the headset.



4.12.2.1.2 Selected and Unfocused

If the dispatcher places the TelePatch on hold by focusing another phone call, the TelePatch becomes unfocused. The headset is full duplex to that non-TelePatch phone call. The telephone leg of the TelePatch is not heard at the console and the dispatcher cannot PTT on the TelePatch. For TelePatches to a group, patch, or base station, there may be group traffic unrelated to TelePatch, so audio from the TelePatched radio entity is heard on the select speaker. Non-TelePatch audio streams that would normally go to the headset, such as priority group and call history playback, are also directed to the select speaker.

Placing a focused TelePatch on hold manually unfocuses the TelePatch, however this state has little value operationally.



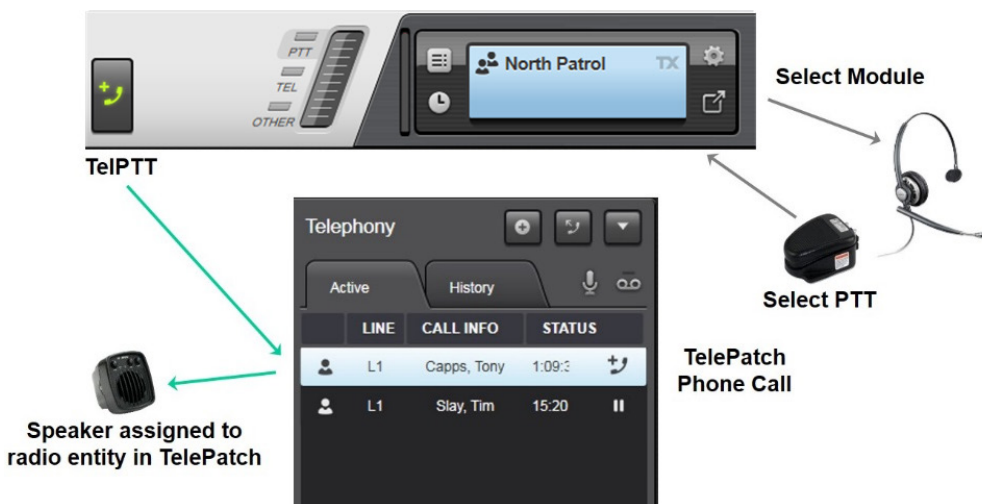
4.12.2.1.3 Unselected and Focused

When the TelePatched entity is unselected, the dispatcher is monitoring it in the background. The TelePatched phone and radio entity communicate with each other while the dispatcher monitors the conversation on the speaker assigned to the radio entity. Select audio is routed to the headset. The dispatcher must use the TelPTT to talk to the TelePatch and the Select PTT to talk to select entities.



NOTE

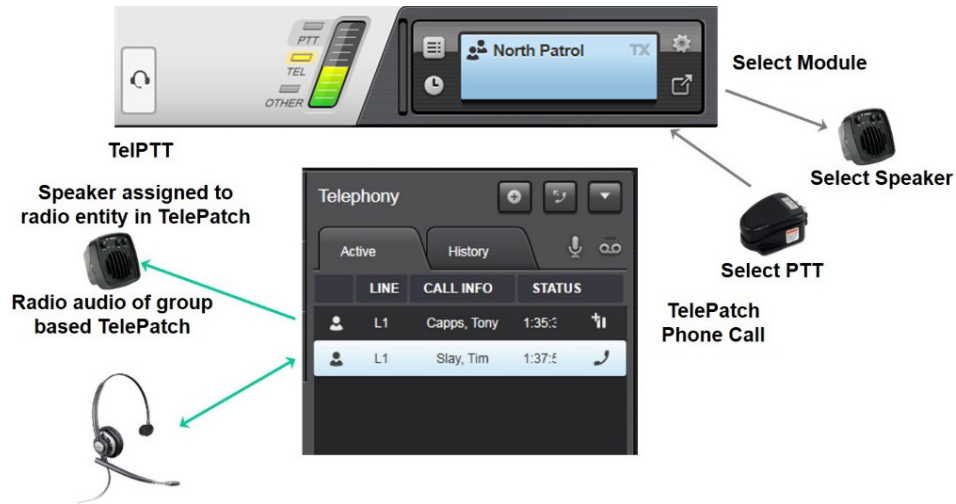
If the speaker assigned to the TelePatch radio entity is speaker 1 (i.e., a Priority Group), the Telepatch stays in the headset mixed with select audio. To change this behavior, change the speaker number of the Priority Group.



4.12.2.1.4 Unselected and Unfocused

If the dispatcher places the TelePatch on hold manually or by focusing another phone call, the TelePatch becomes unfocused. The telephone leg of the TelePatch is not heard at the console nor can the dispatcher PTT on the TelePatch. For TelePatches to a group, patch, or base station, there may be group traffic unrelated to TelePatch, so audio from the TelePatched radio entity is still heard on the speaker assigned to the radio entity.

As usual, select audio is heard in the headset unless a phone call is focused. If the dispatcher focuses a non-TelePatch phone call, the headset becomes full duplex to that non-TelePatch call and all other audio from the headset is pushed to the select speaker.

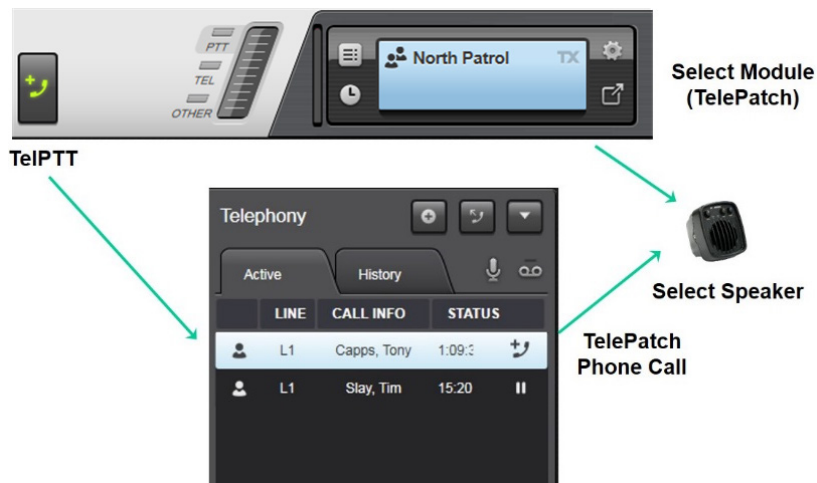


4.12.2.2 Audio Routing with No Headset

TelePatch routing without a headset depends on whether the call is focused or unfocused, and whether the radio entity is selected or unselected. The four possible combinations are explained below. Though supported, using SIP TelePatch without a headset is not recommended, because the limited number of audio paths complicates audio routing and telephony operations using a desk mic is cumbersome.

4.12.2.2.1 Selected and Focused

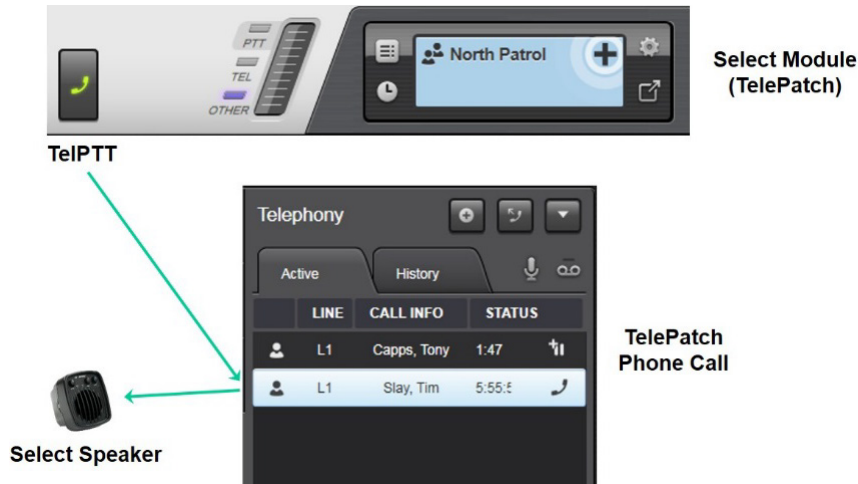
When the TelePatched entity is selected and the TelePatched phone call is focused (i.e., highlighted and not on hold), the dispatcher is actively engaged in the TelePatch talking half-duplex to the TelePatched phone and radio entity using the TelIPTT. The TelePatched audio is heard in the select speaker mixed with other select audio.



4.12.2.2 Selected and Unfocused

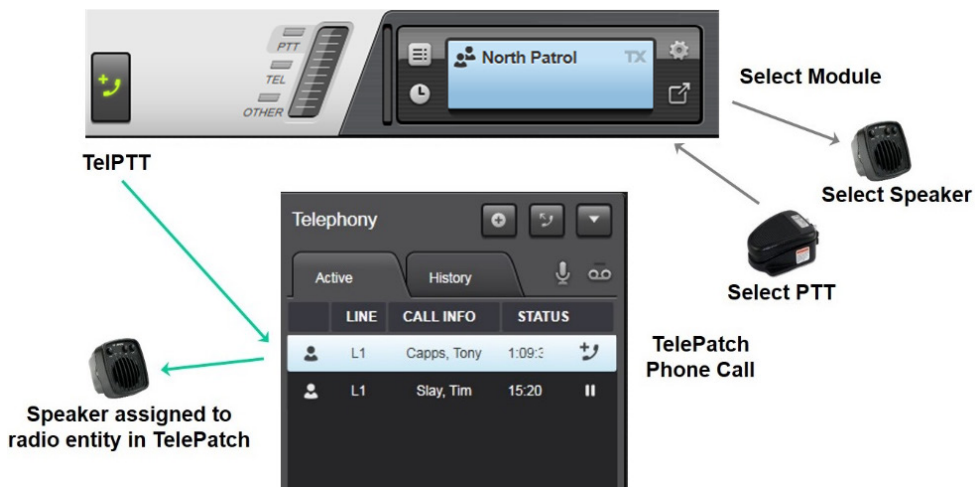
If the dispatcher places the TelePatch on hold manually or by focusing another phone call, the TelePatch becomes unfocused. The telephone leg of the TelePatch is not heard at the console nor can the dispatcher PTT on the TelePatch. For TelePatches to a group, patch, or base station, there may be group traffic unrelated to TelePatch, so audio from the TelePatched radio entity is still heard on the select speaker.

If the dispatcher focuses a different non-TelePatch phone call, the non-TelePatch call audio is routed to the select speaker and mixed with the other select audio. The TelPTT is used to talk half duplex to the non-TelePatch phone call. To talk to the TelePatch, the dispatcher must focus the TelePatch first and then use the TelPTT.



4.12.2.3 Unselected and Focused

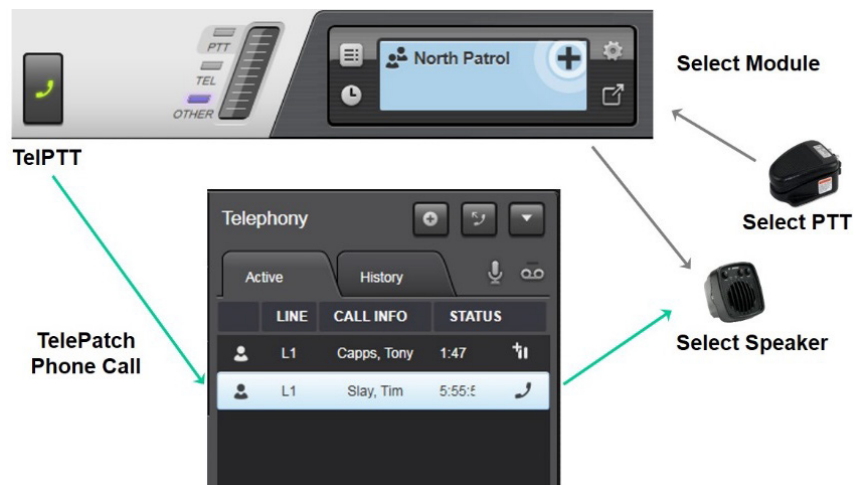
When the TelePatched entity is unselected, the dispatcher is monitoring it in the background. If the dispatcher focuses the TelePatch, the TelePatched phone and radio entity communicate with each other while the dispatcher monitors the conversation on the speaker assigned to the radio entity. If the radio entity is assigned to speaker 1 (i.e., a Priority group), the TelePatch conversation is sent to the select speaker. Select audio is also routed to the select speaker. The dispatcher must use the TelPTT to talk to the TelePatch and the Select PTT to talk to select entities.



4.12.2.2.4 Unselected and Unfocused

If the dispatcher places the TelePatch on hold manually or by focusing another phone call, the TelePatch becomes unfocused. The telephone leg of the TelePatch is not heard at the console nor can the dispatcher PTT on the TelePatch. For TelePatches to a group, patch, or base station, there may be group traffic unrelated to TelePatch, so audio from the TelePatched radio entity is still heard on the speaker assigned to the radio entity.

If a non-TelePatch phone call is focused, the non-TelePatch phone audio is routed to the select speaker and mixed with the other select audio. The dispatcher must use the TelPTT to talk to the non-TelePatch phone and the Select PTT to talk to select entities. To talk to the TelePatch, the dispatcher must focus the TelePatch first and then use the TelPTT.



4.12.2.3 Audio Routing with Telephony Style Handset

When using a telephone-style handset, TelePatch audio routing changes slightly. If the handset is on-hook, Symphony behaves as if no headset is inserted (see above). The following describes the behavior when the handset is off-hook during a TelePatch.

4.12.2.3.1 Selected and Focused

Like headset mode, while the TelePatched entity is selected and the TelePatched phone call is focused (i.e., highlighted and not on hold), the dispatcher is actively engaged in the TelePatch interacting with the associated phone and radio entities via the handset in full duplex mode.

Unlike headset mode, priority group and playback audio are sent to the select speaker. If the dispatcher PTTs to an unselected talkgroup, the desk mic will be keyed while the handset mic is muted to prevent the phone from picking up the radio audio.

4.12.2.3.2 Selected and Unfocused

If the dispatcher places the TelePatch on hold manually or by focusing another phone call, the TelePatch becomes unfocused. The telephone leg of the TelePatch is not heard at the console nor can the dispatcher PTT on the TelePatch. For TelePatches to a group, patch, or base station, there may be group traffic unrelated to the TelePatch, so audio from the TelePatched radio entity is still heard on the select speaker.

If the dispatcher focuses a different non-TelePatch phone call, the non-TelePatch call audio is routed to the select speaker and mixed with the other select audio. The TelPTT is used to talk half duplex to the non-TelePatch phone call. To talk to the TelePatch, the dispatcher must focus the TelePatch first and then use the TelPTT.

4.12.2.3.3 Unselected and Focused

The telephone leg of the TelePatch is not heard at the console nor can the dispatcher PTT on the TelePatch. For TelePatches to a group, patch, or base station, there may be group traffic unrelated to TelePatch, so audio from the TelePatched radio entity is still heard on the speaker assigned to the radio entity.

If a non-TelePatch phone call is focused, the non-TelePatch phone audio is routed to the handset speaker. The dispatcher must use the TelPTT to talk to the non-TelePatch phone and the Select PTT to talk to select entities. To talk to the TelePatch, the dispatcher must focus the TelePatch first and then use the TelPTT.

4.12.2.3.4 Unselected and Unfocused

If the dispatcher places the TelePatch on hold manually or by focusing another phone call, the TelePatch becomes unfocused. The telephone leg of the TelePatch is not heard at the console nor can the dispatcher PTT on the TelePatch. For TelePatches to a group, patch, or base station, there may be group traffic unrelated to the TelePatch, so audio from the TelePatched radio entity is still heard on the speaker assigned to the radio entity.

If a non-TelePatch phone call is focused, the non-TelePatch phone audio is routed to the select speaker and mixed with the other select audio. The dispatcher must use the TelPTT to talk to the non-TelePatch phone and the Select PTT to talk to select entities. To talk to the TelePatch, the dispatcher must focus the TelePatch first and then use the TelPTT.

4.12.2.4 **TelePatch Mute Controls**

A TelePatch can be muted by several controls as shown in Figure 4-3.

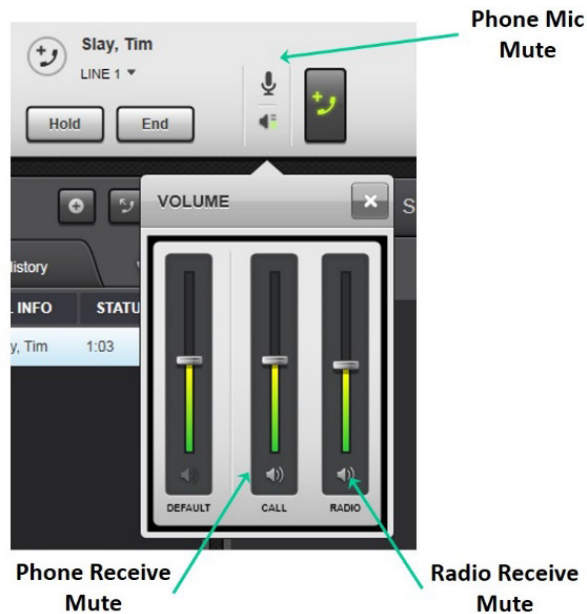


Figure 4-3: TelePatch Mute Overview

These mute controls have the following functionality:

- **Phone Receive Mute:** Audio from the phone leg of the TelePatch is muted to the dispatcher. The dispatcher continues to hear radio traffic from the TelePatch. The radio entity and phone call continue to hear each other, and both continue to hear the dispatcher mic audio.
- **Phone Mic Mute:** Audio from the dispatcher mic is muted to the phone and radio entity. The dispatcher continues to hear the radio and phone audio. The radio entity and phone call continue to hear each other.
- **Radio Receive Mute:** Audio from the radio entity is muted to the dispatcher. The dispatcher continues to hear the phone audio. The radio entity and phone call continue to hear each other, and both can hear the dispatcher.

5 BASIC OPERATION

5.1 COMMUNICATION BASICS

5.1.1 Transmission

A transmission is a dispatcher-initiated call to communication entities such as radios, consoles, or base stations. Transmissions include voice calls, alert tones, Marker Tones, and pages. Transmissions can be sent in the foreground or background. Foreground transmissions use the dispatcher's User ID and only one foreground call can be transmitted at a time. Background transmissions use different IDs from the dispatcher's User ID and can occur simultaneously with foreground transmissions and other background transmissions, as long as the calls do not interfere with each other. Depending on how your console is licensed and configured, you may be able to initiate multiple transmissions simultaneously.

Unless noted otherwise, dispatcher calls are sent in the foreground. Features that run, or can be configured to run, in the background are noted in subsequent sections of this manual. Background transmissions can improve dispatch efficiency and save precious moments, but interactions can become complex. Symphony monitors call interactions and either prevents the new transmissions or preempts a current transmission depending on priorities. The details of handling potential transmit interactions are noted with each feature description.

5.1.2 Reception

Symphony receives calls from multiple sources including radios, base stations, and consoles. Symphony can receive multiple calls simultaneously. Each call is routed to a speaker or headset based on configuration and routing rules that are detailed in later sections. The important concept to understand at this point is how Symphony limits the number of received calls.

Each console is licensed for a number of FlexPaths. Each FlexPath allows you to hear all traffic related to a communications entity such as a group. In most cases, a FlexPath equates to one call. However, there are cases where a FlexPath may support multiple calls for a given entity. For example, if two radios on different sites key on the same group at the same time, VIDA will declare one to be the winner and one to be the loser. All radios in the system will hear the winner, but Symphony will hear both the winner and loser. The latter is referred to as "loser audio." If a supervisor radio then preempts the winner and loser, then the supervisor becomes the new winner and the old winner becomes loser audio. All radios in the system hear the new winner, while Symphony hears the new winner and both losers. Therefore, the FlexPath in this case carries three calls.

If Symphony receives calls from more entities than it has licensed FlexPaths (e.g., calls from six entities with only five licensed FlexPaths), Symphony cannot process all the calls. To ensure the most important calls are received, Symphony chooses the highest priority calls that conform to the licensing limitations. Examples of the highest priority calls are Selected Entities and Emergencies, while muted calls are the lowest priority. Calls that could not be processed are logged in the call history and tagged with a special icon. See Section 5.5 for details. To determine the number of licensed FlexPaths, click the main menu and select **About**. To determine your FlexPath usage, see the Network Status dialog described in Section 3.2.2.



Symphony manages FlexPaths (the number of entities that can be monitored) while C3 Maestro^{IP} manages talkpaths (the total number of calls). C3 Maestro^{IP} users will find the Symphony approach is more flexible.


5.2 COMMUNICATION MODULES



NOTE

In cases where it is desired not to have any talkgroups selected, a Receive-Only talkgroup (“NONE-SELECTED”) can be added to the UAS and as a module in the console. Selecting this module will cause all RX calls to route to unselect speaker(s). Note that the dispatcher must select a valid talkgroup before transmitting.

5.2.1 Pick a Module



Some features require that the user “pick” a module. Picking a module identifies a module or modules to which a feature will be applied, such as SimulSelects or Patches. The  icon appears on the module when it can be picked.

5.2.2 Select an Entity

The “select” feature provides a quick and easy method to identify one entity as the dispatcher’s central point of focus. This “selected entity” receives special treatment including prominent placement in the select module, special audio routing to the headset or the select speaker and association with special transmit controls like the footswitch. Any entity can be chosen as the selected entity.

Given the importance of the selected entity, a valid entity must always be selected, so there is no “unselect” operation which can lead to ambiguity concerning which entity to select next. Instead of unselecting an entity, select a new entity to replace the previous entry. This ensures that a valid entry is always assigned the select status. In the rare event that a select entity becomes invalid or there is a question about which entity to select, Symphony uses the Default Select Entity defined in the next section.

To select an entity:

- Click the **Select** button on a normal size Communication Module or popped out module.
- Or
- Click  button in the Select Bar to open the Select Menu and select the entity from the list. See Section 3.1.5.2 for more information on the Select Menu.
- Or
- On any small or mini module, click  and click **Select**.

Any module associated with the selected entity is outlined with a green border.

Notes on selected modules:

- The selected entity is displayed in the Select Bar in the Full Screen display and Baton.
- The selected entity is the primary dispatch group.
- Only one entity at a time can be selected (exceptions are Patch and SimulSelect).
- Audio for the selected entity is routed to the Select speaker.
- A new selection cancels the old selection.
- The selected entity does not change when changing tabs but may change when switching setups depending on configuration.

5.2.3 Default Select Entity

The Default Select Entity (DSE) feature allows you to define an entity that will automatically be selected when there is no other module selected, e.g., on initial start after installation.

The Default Select Entity is defined in the Directory (see Section 3.5.1).

Rules for the DSE feature are as follows:

- The DSE is selected when the console is first started and when the dispatcher changes setups. The DSE is saved per-setup.
- If the previously selected entity cannot be selected, the Default Select Entity is automatically selected after a selected local or remote Patch or a local SimulSelect is deactivated. This may happen if the previous entity was un-programmed or there was no previously selected entity.
- A selected remote Patch is unselected if modified to remove the DSE. The previously selected entity or DSE will be reselected as appropriate.

5.2.4 Transmit to Selected Entities

Whenever transmitting from the console, always wait for channel access before speaking into the microphone. In most cases, another caller should not be pre-empted by the console. Before transmitting, always observe the module indications to verify that no other unit or console is calling the selected module.

While the console is transmitting to the selected entity, a transmit indication is displayed on the module's Smart Button. When the open microphone is spoken into, the console's VU meter indicates transmit audio level. The console can also generate tones to audibly indicate the transmit status. These tones are referred to as "confirmation tones," and are discussed in Section 5.3.

1. Press and hold the Smart Button on the module (touch with finger if using a touch screen monitor or left mouse-click).
Or
Press the desk microphone transmit button.
Or
Press the footswitch.
Or
Use the "Transmit on Select" Quick Action key (see Section 3.6).
2. Speak into the microphone.
3. Release the button or footswitch when transmission is complete. Be sure you have finished speaking before releasing PTT.

5.2.5 Transmit to Unselected Modules

Whenever transmitting from the console, always wait for channel access before speaking into the microphone. In most cases, another caller should not be pre-empted by the console. Before transmitting, always observe the module indications to verify that no other unit or console is calling the unselected module.

While the console is transmitting to the unselected module, a transmit indication is displayed on the Smart Button of the module. When the open microphone is spoken into, the console's VU meter indicates transmit audio level. The console can also generate tones to audibly indicate the transmit status. These tones are referred to as "confirmation tones," and are discussed in Section 5.3. Audible confirmation tones are still generated, if they are enabled.

To transmit to an unselected module:

1. Press and hold the Smart Button on the module.
2. Speak into the microphone.
3. Release the button when the transmission is complete.

5.2.6 Receive Calls on Communication Modules

When a radio unit or another console in the system transmits on an entity that is programmed into a Communication Module, indications of the incoming call appear on the screen in one or more areas:

- On the Full Screen Display:
 - When calls are received via Communication Modules on the displayed Workspace tab, call indications appear on the respective module's Smart Button on the Workspace tab.
 - When calls are received via Communication Modules on a Workspace tab other than the one currently displayed, these incoming calls are indicated on the respective Workspace tab.
 - The call is indicated on the appropriate Speaker Button.
 - The call is listed in the call history.
- On the Baton:
 - When calls are received via Communication Modules that are visible on the Baton, a call indication appears on the module.
 - When calls are received via Communication Modules that are not visible on the Baton, but the module is in a Workspace that is visible on the Baton, a call indication appears on the Workspace List button.
 - The call is indicated on the appropriate Speaker Button if visible on the Baton.
 - The call is listed in the call history, but this is only visible if the call history panel is displayed.

5.2.7 Mute Entity



To mute an entity, click the speaker icon , available on the entity's module volume control. The module will appear as in Figure 5-1 when muted. All modules containing the entity are muted.



Figure 5-1: Module Muted

The ability to mute an entity is configurable by the administrator via the “Can Mute/Unmute entity” setting.

To mute all unselected entities, click  in the top right of the main display. The icon changes to show a countdown. If you do not manually unmute the modules, mute times out based on an administrator-defined time. Click the button again to manually unmute.

5.3 PROGRESS TONES

Symphony provides progress tones to indicate the success or failure of user-initiated operations.

5.3.1 Transmit Confirmation Tones

The following list identifies the transmit confirmation tones heard just after a PTT action at the console. These tones are audible indications of the status of each call throughout the radio system. They also provide “OK” or “not OK” signals to speak into the microphone. Typically, only the grant tone sounds upon each PTT. In rare cases, transmit confirmation tones and other error/message type tones are turned off by the system or dispatch administration personnel (Settings “Enable Grant Tone” and “Enable Denied Tone”). In such situations, no tones are heard during a console PTT sequence.

- Grant Tone

This short-duration medium-frequency beep signifies that it is OK to speak into the microphone. It is usually heard immediately after a console PTT. Technically, the tone signifies that a communication radio channel in the radio system has been assigned for the call; therefore, it is sometimes referred to as the “channel access tone.” In most circumstances, do not begin speaking into the microphone until this tone is heard.

- Access Denied Tone

This low-frequency beep signifies that the transmission request cannot be granted by the radio system. If this tone sounds upon repeated PTT attempts, consult the system or dispatch administrator for possible causes and/or corrective actions.

- Unkey Beep

If enabled, the Unkey Beep feature causes a tone to be generated by the console when a radio-originated transmission on an un-muted, selected module unkeys. This beep provides an audible indication to the console operator that a console originated transmission can be initiated without causing a console preempt.

5.3.2 Error Tones

An error tone is generated whenever an invalid or unauthorized action is attempted by the user. Check the System History for an error message explaining the error tone.

5.4 INDIVIDUAL CALL



Since Individual Call is a feature-licensed option, this feature is only available if the console is equipped with this software license.

5.4.1 Overview

In a trunked radio system, the individual call feature provides personal unit-to-unit communications. Calls of this type are sometimes referred to as “unit calls,” “select calls,” or “I-Calls.” Unlike talkgroup calls, individual calls usually involve communications between two trunked units only. Since the dispatch console is considered a unit in the trunked radio system per its unique Unit ID number, it has the ability to send and receive individual calls like most trunked radio units in the trunked radio system. Individual calls are initiated by one unit—the caller—and received by another unit—the callee.

Symphony supports two kinds of I-Calls: unacknowledged and acknowledged. Unacknowledged I-Calls behave just like group calls. PTT and start talking to the individual you are calling. Acknowledged I-Calls must be accepted by the callee before audio can be passed. In this case, the first transmit on the I-Call is a message sent to the callee. The callee can accept or reject the call as required. Once the callee accepts the I-Call, audio is passed between the units on the next PTT. OpenSky only supports Acknowledged I-Calls. Other systems like P25 can be configured to support either of both call types. Consult your systems administrator for details on your system’s configuration.

Symphony always accepts the preferred I-Call type specified by the caller and saves this information for each unit. This allows Symphony to place future calls with the unit’s preferred type.

If Symphony does not know the preferred I-Call type for a unit, it initiates an I-Call based on the Default VIDA I-Call Session Type configuration parameter. Typically, only one type of I-Call is used in a system and this parameter is set to match the system configuration. In the rare case that the callee cannot support Symphony's default I-Call type, the call will typically go through, but may behave differently due to the mismatch. If the call fails, try sending a Call Alert to the unit to prompt a call back. In either mismatch case, Symphony updates the unit's preferred I-Call type when the unit calls back so future calls will behave normally.

5.4.2 Acknowledged Individual Calls

When the dispatcher starts an individual call operation to a radio unit, the console generates an acknowledged individual call request to the radio unit. Normal individual call transmit operations apply once the acknowledged individual call request has been answered.


Radio units can also send acknowledged individual call requests to the console. Normal I-Call transmit operations apply once the acknowledged individual call request has been answered.



5.4.3 Transmit Acknowledged Individual Calls

If an individual configured for Acknowledged Individual Call is programmed to a Communication Module on the Workspace, Baton, or In Progress tab of the I-Call panel, click the module's Smart Button to initiate the call. Otherwise, use one of the following methods to initiate an acknowledged individual call from the I-Call panel:



NOTE

On the Baton, click  to open the I-Call Sidebar Panel.

- Choose an entry from the **Log** tab in the I-Call Sidebar Panel and click **Start** (see Section 3.4.3.5).
- Click the  button on the I-Call Sidebar Panel and enter the Unit ID. After entering the Unit ID, click **Start** (see Section 3.4.3.2).
- Click the  button on the I-Call Sidebar Panel. Choose the desired contact from the list and click **Start** (see Section 3.4.3.3).

A Smart Button is added to the **In Progress** tab of the I-Call Sidebar Panel.

Once the call is started, the Smart Button shows the call progress as noted for the specific systems below:

For OpenSky systems:

The Smart Button shows an hourglass icon in the upper right corner and a countdown timer is shown on the left of the second line. On mini modules, the first and second lines alternate. Once the callee answers, the hourglass is replaced with a TX icon and the countdown timer disappears. If the countdown timer expires, the call ends.

For non-OpenSky systems that support I-Call:

The Smart Button shows a bell icon in the upper right corner and a countdown timer is shown on the left of the second line as shown in Figure 5-2. On mini modules, the first and second lines alternate. A ring, like a telephone, is also sounded. Once the callee answers, the ringing stops, the bell is replaced with a TX icon, and the countdown timer disappears. If the countdown timer expires, the ringing stops and the call ends.

Once the callee answers, the Smart Button is in the active call state where pressing the button transmits to the callee and received PTTs from the callee are indicated on the button.




Figure 5-2: Transmitting Acknowledged I-Call

5.4.4 Receive Acknowledged Individual Calls

When receiving an acknowledged Individual Call, the call is displayed on the **In Progress** tab of the I-Call Sidebar Panel. The tab of the I-Call Sidebar Panel and the I-Call button of the Baton display a bell icon (see Figure 5-3) until you respond to the call or dismiss the call. If the caller is programmed to a Communication Module on the Workspace or Baton, a ringing bell icon appears on the module's Smart Button. The console rings until the call is accepted or the call times out. If the call is not accepted by the console and times out, the bell is replaced by the number of calls awaiting response.



NOTE

Click  on the Baton to open the I-Call Sidebar Panel.

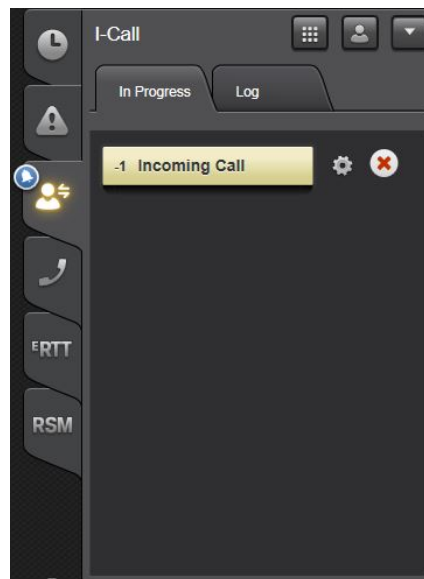







Figure 5-3: Incoming Acknowledged I-Call

- After the inbound Acknowledged I-Call is accepted, the system requires that the recipient transmit first before the initiator can transmit. If you continue to hold down the smart button after the I-Call is accepted, Symphony will begin transmitting automatically. Otherwise, press the Smart Button again to satisfy the first talker requirement.
- To dismiss the call, click the  button next to the Communication Module in the I-Call Sidebar Panel, or click  and choose **Decline** from the communications module's context menu. To accept the call, click the Communication Module's Smart Button and "TX" appears. Click and hold the Smart Button to transmit.
- To end the call, click the  button next to the Communication Module in the I-Call Sidebar Panel or click  and choose **End Call** from the communications module's context menu.
- To clear the call from the I-Call Sidebar Panel, click the  next to the Communication Module in the I-Call Sidebar Panel.

5.4.5 Unacknowledged Individual Calls

Unacknowledged Individual Calls function like a group call to one radio.



5.4.5.1 Transmit Unacknowledged Individual Calls

If an individual is programmed to a Communication Module on the Workspace, Baton or In Progress tab of the I-Call panel, click the module's Smart Button to initiate the I-Call. Otherwise, use one of the following methods to initiate an unacknowledged individual call:



NOTE

Click  on the Baton to open the I-Call Sidebar Panel.

- Choose an entry from the **Log** tab of the I-Call Sidebar Panel and click **Load** (see Section 3.4.3.5).
or
- Click the  button on the I-Call Sidebar Panel and enter the Unit ID. After entering the Unit ID, click **Start/Load** (see Section 3.4.3.2).
or
- Click the  button on the I-Call Sidebar Panel. Choose the desired contact from the list and click the **Load** button (see Section 3.4.3.3).

After clicking **Load**, a Smart Button is added to the **In Progress** tab of the I-Call Sidebar Panel. Click the Smart button to initiate the call and begin transmission to the callee.


Unacknowledged I-Calls do not require an explicit call end action; they time out.

Though OpenSky I-Calls (also known as Selective Calls) are inherently acknowledged, Symphony can place an unacknowledged call to an OpenSky radio, but the behavior will be different from that described above. This situation only arises in mixed systems with P25 and OpenSky radios where Symphony is configured to use unacknowledged I-Calls to accommodate the P25 radios.

Unacknowledged OpenSky I-Calls:

1. Click the Smart Button to start the call. The Smart Button shows an hourglass icon in the upper right corner and a countdown timer is shown on the left of the second line. On mini modules, the first and second lines alternate.
2. Once the callee answers, the hourglass is replaced with a TX icon and the countdown timer disappears. If the countdown timer expires, the call ends.
3. After the callee answers, the Smart Button is in the active call state, where pressing the button transmits to the callee and received PTTs from the callee are indicated on the button.

5.4.5.2 Receive Unacknowledged Individual Calls

When receiving an unacknowledged I-Call, the caller is added to the **In Progress** tab of the I-Call Sidebar Panel. A badge on the I-Call Sidebar Panel's tab and the Baton's I-Call button indicates the number of unacknowledged I-Calls awaiting response. Press the Smart Button to respond to the caller or click  to clear the call from the **In Progress** tab.

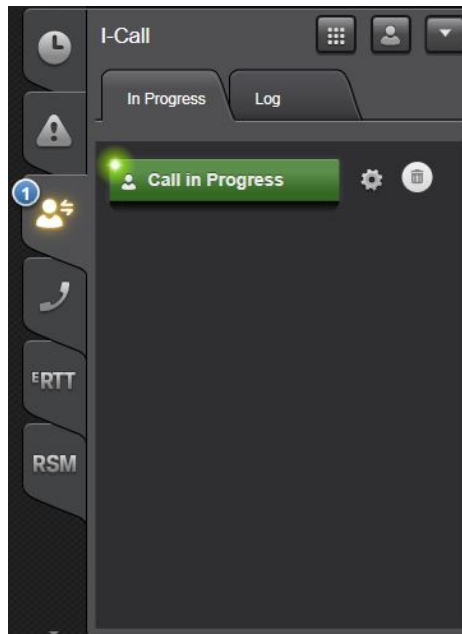


Figure 5-4: Incoming Unacknowledged I-Call

5.4.6 Instantaneous Individual Calls

A third type of I-Call included in the VIDA system is the instantaneous I-Call. This functions similar to an unacknowledged I-Call and is typically used only on BeOn[®]-enabled systems. It must be noted, however, that BeOn handles I-Calls differently “under-the-hood” than the other types of radio communication systems.

Like unacknowledged call, instantaneous I-Call does not need to be dismissed; it times out. However, BeOn's different message handling provides some differences in the indications and timing of the events the dispatcher receives at the console. Most significant of these is the way BeOn calls display a call history record when an I-Call occurs. BeOn uses a "hang-timer" of approximately 30 seconds when it establishes an I-Call session. For this reason, a BeOn I-Call session is always at least 30 seconds long even if the actual conversation (talk-spurt) is much shorter. Should a new BeOn I-Call begin before this hang-timer has expired, it may not show a new entry in the call history. In addition, this new talk-spurt resets the 30 second hang-timer which can cause further delay in call history updates.

This hang-timer delay can also affect the new I-Call indication in the I-Call Sidebar Panel in some instances. For example, if a console receives an instantaneous I-Call from a BeOn unit, the console may respond to it via the BeOn unit's mini-module in the I-Call Sidebar Panel. If the dispatcher then dismisses the mini-module from the panel and the BeOn unit in the field places another I-Call to the console before the hang timer expires, that call is not indicated in the I-Call Sidebar Panel. The system treats the new call as part of the same session so no new indication of the establishment of a new I-Call is sent to the console (only the voice data and RX information). The console therefore does not know that it needs to indicate the new call, so the indications are not shown.

5.4.7 Missed I-Call Reminder

If the Missed I-Call Reminder Delay is configured by the administrator, the console will start ringing the configured number of seconds after an I-Call is missed. A call is considered missed when the I-Call, acknowledged or unacknowledged, times out and is not answered with a response PTT to the caller. A mini module for the missed I-Call remains in the I-Call Panel for quick access. Use one of the following methods to stop the ringing:

- Initiate an I-Call to the missed caller.
- Dismiss the missed I-Call from the Log.
- Press the Mute Tone button on the Select Bar.

5.5 CALL HISTORY

There are multiple sources for call history information.

- The History tab of each Communication Module (see Section 3.3.7).
- The History Sidebar Panel (see Section 3.4.1).
- The Extended Call History dialog (see Section 3.5.2).
- The Select History dialog (see Section 3.1.5.3).
- Speaker History (see Section 3.1.6).

Audio from incoming calls received by the console is saved for playback and is available from all history dialogs (see Section 5.6). Audio from transmitted calls is also saved for playback and is available from the History Sidebar Panel, the Extended Call History Dialog, the Select History Dialog, and the History tab of each Communication Module, if configured to show TX calls. To preserve system performance, recording is limited to the first 10 minutes of the call. This limitation does not apply to phone call recording.

Table 5-1 describes the icons displayed in all Call History dialogs. A call may have more than one of these properties. For example, a call from another console may be muted. The history will display only one icon. The icons towards the top of the table are higher priority and will be displayed.

Table 5-1: Call History Visual Indicators





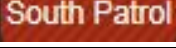








INDICATOR	DESCRIPTION
	The call was dropped by the console. This will occur when the console runs out of FlexPaths and must drop a call to carry a more important call. (See the section in the Installation and Configuration manual about call priority.)
	Some or all of the call was "Losing Audio". These calls were heard at the consoles, but not by the radios. This occurs when there are two or more talkers active on a group at the same time. Only one of the talkers can be delivered to the radios.
	The call was received, but the module was muted for some or all of the call, so the dispatcher did not hear it.
	Indicates an emergency declaration. Note: It is not a call; therefore, there is no audio to playback.
	Indicates a call while group is in emergency (see Figure 5-5) or an emergency declaration.
	Problem with encryption. It is possible that the console does not have the necessary key to decode the call. In which case, the console plays silence. This may also indicate that the encryption parameters received with the call were not valid or consistent. This can be caused by poor coverage or interference. When this is the case, the console may or may not hear the call.
	Call was on the select entity.
	Call was to or from a base station.
	Call was on a Patch rather than a talkgroup.
	Call was on a Remote SimulSelect.
	Call was to or from a Unit.
	Call was from another console.
	The No Subscriber Warning was indicated for this call. Transmissions only.



Figure 5-5: Call while Group is in Emergency

5.6 AUDIO PLAYBACK

Audio from all calls transmitted or received by the console is saved for playback. The playback function is accessed through the history dialogs. The history dialog contains three tabs: Radio Calls, Phone Calls, and ABR Calls. Select the appropriate tab to display the corresponding audio recordings.

Highlight one or more calls from the list to activate the playback controls.

- Click to play the highlighted call(s). The button changes to .
- While playing back a call, starts playing the next call in the list. When not playing back a call, highlights the next call in the list.
- After one half second of playback, the following functions are available. Pressing during playback of call restarts the current call from the beginning. Double-click during playback of call to start playing the previous call in the list. When not playing back a call, it highlights the previous call in the list.
- At any time, use the volume pop out to adjust the volume of the playback audio.
- By default, playback audio is directed to the current select audio device. This is typically the headset or select speaker (if no headset is inserted, or a phone call is active). The administrator can configure a different audio device for playback, such as one of the unselect speakers. For information about configuring playback devices, consult the *Symphony Installation and Configuration Manual*.

Instant replay, Figure 3-24, is available on each Communication Module and plays the last received call.

5.7 PATCH AND SIMULSELECT COMPARISON

The Patch and SimulSelect features provide the ability to operate simultaneously on multiple trunked talkgroups. Table 5-2 summarizes the main similarities and differences between Patch and SimulSelect. The rest of this chapter describes each feature in detail.

Table 5-2: Comparison of Patch and SimulSelect

PATCH	SIMULSELECT
Patch ties trunked talkgroups together as one common communication entity (basically, as one talkgroup) throughout the radio system. It modifies the operation of terminal-originated calls.	SimulSelect joins trunked talkgroups and/or individual trunked radio units together only at the console that created the SimulSelect. From the terminal user's perspective, the SimulSelect does not exist because SimulSelected groups and users remain separate communication entities.
Console receives all patched group transmissions in the select speaker if the Patch is selected and in an unselect speaker if the Patch is unselected.	Console receives all SimulSelected group and radio unit transmissions <u>in the select speaker</u> .
All radio units on the groups in the Patch hear the console when it transmits on the Patch.	All radio units on groups/units in the SimulSelect hear the console when it transmits on the SimulSelect.
All radio units on groups in a Patch <u>can</u> communicate across group boundaries.	All radio units on groups in a SimulSelect <u>cannot</u> communicate across group boundaries.
A Patch can be selected and unselected.	All groups and users in an active SimulSelect are always simultaneously in the selected state. A SimulSelect will be deactivated if any module not in the SimulSelect is selected.
The number of Patch activations is controlled by licensing. Symphony may be allowed to have up to 16 active patches at one time. Check with your administrator for information about your licensed limit.	The number of SimulSelect activations is controlled by licensing. Symphony may have up to 16 active SimulSelects at one time. Check with your administrator for information about your licensed limit.
Other consoles in the system cannot activate a Patch or SimulSelect with groups that are in an active Patch.	Other consoles in the system cannot activate a Patch or SimulSelect with groups/units that are already in an active SimulSelect.
Consoles <u>can</u> see a Patch activated by another console in the Patches Sidebar Panel and by indications on the respective Communication Modules.	Consoles <u>can</u> see a SimulSelect activated by another console in the SimulSelect Sidebar Panel and by indications on the respective Communication Modules.
A Patch cannot be activated if one of the groups is in an emergency.	A SimulSelect can be created if one of the groups is in an emergency. A unit with an active unit alert can also be SimulSelected.
An active Patch <u>cannot</u> be deactivated if an emergency exists on the Patch. The emergency condition on the Patch must be cleared before the Patch can be deactivated. This ensures that radio units logged onto different groups within the Patch do not lose communications before the emergency condition is cleared.	An active SimulSelect <u>can</u> be deactivated if an emergency exists on one or more of the SimulSelected groups.
In VIDA Proprietary Patch/Simul Mode, if an emergency is declared on a group within a Patch, emergency indications appear in <u>all of the "patched" Communication Modules</u> . The Patch itself is considered to be in an emergency condition. These indications emphasize the fact that all groups within the Patch are in emergency since at least one of the patched groups is in emergency condition.	If an emergency is declared on a SimulSelected group, emergency indications appear <u>only in the respective group-programmed Communication Module</u> .

PATCH	SIMULSELECT
A Patch from another console will be selected if a common entity on the local console is selected. All common (overlapping) modules will be selected.	SimulSelect transmit audio from another console will be selected if a common entity on the local console is selected. There is no select indication on other overlapping modules. Remote SimulSelects cannot be selected.
Volume levels on patched modules are all set to the <u>highest module level when the Patch is activated.</u> Volume levels are restored when the Patch is deactivated.	There are no volume level changes.

5.8 PATCH OPERATION



Since Patch and SimulSelect are feature-licensed options, these features are only available if the console is equipped with this software license.

The console’s Patch feature allows trunked talkgroups and/or conventional channels to be connected as one common communication entity. When groups and/or channels are patched, they are considered to be “in” an active Patch. Patched groups essentially become one talkgroup when the Patch is active. Typically, a Patch is active only on a temporary basis when common communications across normal group/channel boundaries are required.

Any group can be patched. However, if an Announcement Group is patched, talkgroups associated with that Announcement Group cannot be included in another Patch. Conventional channels can be patched.

Symphony can be licensed for up to 16 patches active at one time. Check with your administrator for your licensed limit. Modules included in a Patch are identified via colored tags on the top of each module (see Figure 5-6). Each Patch can be created, activated when needed, and deactivated. Each Patch can consist of up to fifteen patchable entities. If a group or a conventional channel is currently in an active Patch, it cannot be placed in another Patch by the original console or by any other console in the system.



Figure 5-6: Patched Communication Modules

All groups and/or conventional channels in a Patch are connected for common communications when the Patch is active. For example, if the group “POLICE 2” and the group “FIRE 1” are patched, any radio unit or console transmitting to “POLICE 2” is heard by units monitoring either of the two groups. Likewise, any radio unit or console calling on “FIRE 1” is heard by units monitoring either group.

Using the Patch Modify feature, active patches can be modified. The feature allows the dispatcher to add or remove members of the patch while it is active. Before a modified patch is deactivated, the changes can be persisted using the Save and Save As options. If the changes are not saved, the patch is restored to its previous definition upon deactivation.

When a Patch is deactivated, its groups/channels are disconnected from common communications. The console retains inactive patches in a Patch memory so that each can be quickly reactivated when necessary if it had been created from the Patches Sidebar Panel or had been saved when created via the Ad-hoc Patch button. The active/inactive status of each Patch and the entities comprising the Patch can be viewed via the Patch Sidebar Panel (see Figure 5-10).

At the console, an active Patch can be selected and unselected like any single module; select one of the patched Communication Modules to select the Patch. When selected, all patched modules indicate the selected state via the modules’ highlighted border. Dispatch communications on a selected Patch are performed per standard selected module receive and transmit methods. If a Patch is not selected, all of the corresponding Communication Modules remain unselected and dispatch communications on the Patch are performed per standard unselect module receive and transmit methods.

Patches can be pre-defined for quick-and-easy activation. The console can be licensed to store up to 32 Patch definitions. Check with your administrator for your licensed limit.

To summarize:

- A Patch ties talkgroups and/or conventional channels together as one common communication entity. Consoles and individual radio units cannot be patched.
- All radio units on the groups/channels within the Patch can hear the console when it transmits on the Patch.
- All radio units on groups/channels within the Patch can communicate with each other across group/channel boundaries.
- A Patch can be selected and unselected.
- Each console can be licensed for up to 16 patches active at one time. Check with your administrator for your licensed limit.
- An entity can only be in one active Patch at a time.
- Other consoles in the system can see a Patch activated at another console via indications in the respective communications modules, if the patched groups/channels are programmed. This is indicated on the module with a colored tab labeled with an “R:” followed by the name of the remote Patch. The full-size Patch label is only available on modules in the Workspace. See Section 5.8.8 for details.
- Other consoles in the system cannot activate a Patch with groups/channels that are within an active Patch.
- Patches can only be deactivated by the console that activated the Patch or by the RNM.



- The Patch Modify feature supports the adding and removal of qualifying entities (groups and base stations) to or from a patch while it is active.

Any Patch created and activated at a dispatch console is considered to be a “local” Patch to the console. Any Patch created and activated at another console in the system is considered a “remote” Patch to all other consoles in the system. Throughout this manual, all operational descriptions of the Patch feature are assumed to be for local patches, unless otherwise noted.

5.8.1 Create Patches

There are two methods for creating a Patch. Section 5.8.1.1 defines how to create an Ad-hoc Patch from the Workspace (Full Screen Only). Section 5.8.1.2 defines how to create a Patch using the **NEW/EDIT PATCH** dialog.

5.8.1.1 Ad-hoc Patches

1. Click the **Patch** button in the top right corner of the console display.
2. Click the  icon to pick each module that you want to include in the Patch. Click the  icon again to unpick a module.



Patches can include Communication Modules from multiple Workspace tabs.

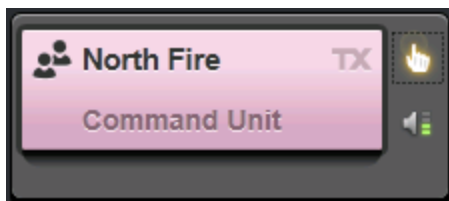


Figure 5-7: Picked Module

3. Click **Patch now** to activate the Patch.

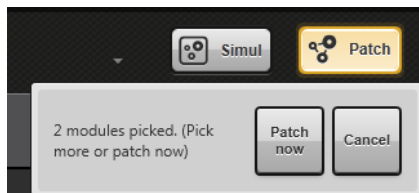


Figure 5-8: Activate Patch




If the system administrator has configured this console for Auto Save Patch/SimulSelect, then the newly created Ad-hoc patch will be saved as “Patch (1-8),” whichever alias is next available – starting with 1 and going to 8. If this setting is not enabled by the administrator, then the Ad-hoc patch will not be saved upon creation.

4. Modules included in the Patch are identified with a tag that includes the name of the Patch.



Figure 5-9: Patched Modules

5.8.1.2 Using the New/Edit Patch Dialog

1. Click the **Harris** logo and choose **Create Patch**, or click  in the top right corner of the Patch Sidebar Panel and choose **Create Patch**.
2. The **NEW/EDIT PATCH** dialog opens (see Section 3.4.7.5).
3. Enter a name for the Patch.
4. Choose the entities from the list on the left and click > to add to the Patch member list on the right.
5. Specify whether or not to activate the Patch (**Activate on Save & Close**) and to select the Patch (**Select on Save & Close**).
6. Click **Save & Close**.

See Section 3.4.7 for more information on the Patch Sidebar Panel.

5.8.2 View Patches

View and interact with Patches via the Patch Sidebar Panel (see Figure 5-10). Choose the desired filtering option from the drop-down. Available options include: All, Local Active, Local Inactive, and Remote Active.

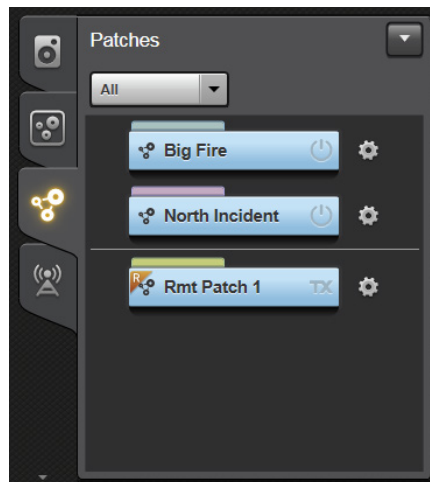



Figure 5-10: Patch Sidebar Panel

5.8.3 Delete a Patch


1. From the Patch Sidebar Panel, click the context menu button  on the Patch you want to delete.
2. Choose **Delete Patch**.



NOTE

The Patch must be saved and deactivated for this option to be available.


5.8.4 Edit a Patch

1. From the Patch Sidebar Panel, click the context menu button  on the Patch you want to edit.
2. Choose **Edit Patch**. This selection is not available if the Patch is active.
3. The **NEW/EDIT PATCH** dialog opens. Make the desired changes to the Patch and click **Save & Close**.

5.8.5 Select/Unselect a Patch

At the console, an active Patch can be selected and unselected like any single module. Since a Patch is essentially one common communication entity, selecting one module currently in an active Patch automatically selects all modules in the Patch. Dispatch communications on a selected Patch are performed per standard select entity receive and transmit methods. If an active Patch is not selected, all of the corresponding Communication Modules remain unselected and dispatch communications on the Patch are performed per standard unselect entity receive and transmit methods.

Use one of the following methods to select a Patch:

- Click the **Select** button on any normal sized module included in the Patch or on the Patch module.
- From the Patch Sidebar Panel or Workspace, click the context menu button  on the Patch module and choose **Select**.
- Check the **Select on Save** box on the New/Edit Patch dialog (Section 3.4.7.5) after Patch creation or edit.

5.8.6 Receive Calls on a Patch

When a radio unit or another console in the system transmits on a group or conventional channel that is in an active Patch, the transmission is broadcast on all groups and/or channels in the Patch. If the call originates from a trunked radio unit, the caller's unit name or Unit ID number displays in all of the patched modules at the console. If the call originates from a conventional radio, the respective conventional channel's name or ID number displays in all of the patched modules. Basically, in each patched Communication Module, incoming call indications are identical to those that appear in non-patched modules. Incoming calls received on a Patch are also indicated in the call history panel. However, instead of indicating the group/channel being called on, the Patch's name is indicated as the "callee." When a call is received on a Patch, the volume level is equalized to the highest volume of any module in the Patch.

5.8.7 Transmit on a Patch

Transmits can be performed on both selected and unselected patches via standard transmit actions (see Sections 5.2.4 and 5.2.5). Transmitting from any module in the Patch transmits on the Patch.

5.8.8 Remote Patches

Other consoles in the system can also create patches. These patches are referred to as "remote" patches. Throughout this manual, all operational descriptions of the Patch feature are assumed to be for locally created and activated patches, unless otherwise noted. A special indicator is displayed on a Communication Module if another console in the system has placed the module in a Patch (see Figure 5-11 and Figure 5-12).



Figure 5-11: Workspace - Remote Patch

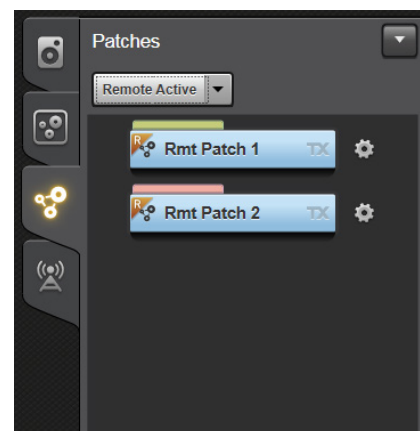


Figure 5-12: Patch Sidebar Panel - Remote Patches

5.8.8.1 Receive and Transmit on Remote Patches at the Local Console

Remote Patch receive and transmit operations are identical to receive and transmit operations performed on a locally created Patch. A remote Patch appears in the Patch Sidebar Panel even if none of the remote patches' entities are currently monitored in the console.

Like a local Patch, a remote Patch can be selected by clicking the **Select** button on one of the patched modules. Likewise, a remote Patch can be unselected by selecting a module outside of the remote Patch. When a remote Patch is selected, all of the respective modules' borders are highlighted in green.

Incoming calls on a remote Patch are indicated in the respective patched Communication Modules. Like a call received on a local Patch, a call received on a remote Patch is indicated in all of the modules within the Patch.

If the local console transmits on a remote Patch, the transmission is broadcast on all groups/channels in the Patch, even if they are not all programmed at the local console.

5.8.8.2 Remote Patch Restrictions

The following list identifies several restrictions applicable to remote Patch operations:

- A remote Patch cannot be deactivated or cleared at the local console. It must be deactivated or cleared by the remote console that created it, or by the system or dispatch administration personnel.
- The local console cannot activate a pre-defined Patch or SimulSelect that contains a group or conventional channel currently in a remote Patch.
- The local console cannot create an Ad-hoc Patch or SimulSelect that contains a group or conventional channel currently in a remote Patch.
- The local console MAY add a group or channel to a Patch or SimulSelect definition even if it is included in a local or remote Patch.

5.8.9 Emergencies on a Patch

Information in this section only applies to group emergencies, not unit alerts.

- The console operates in one of two ways, depending on how the administrator configured the console; L3Harris mode or Motorola® Compatibility mode. L3Harris radios support emergencies on patches while Motorola radios do not.
- You cannot activate a Patch while a group is in emergency.
- You cannot deactivate Patch while Patch is in emergency.


5.8.9.1 Moto Mode Behavior

- In Moto Mode, patches CANNOT be put in emergency.
- The console cannot declare an emergency on a Patch. If you use a Patch module to declare an emergency, you will get an error message at the bottom of the display.
- The console operator must declare an emergency from a group module.
- When a radio or console declares an emergency on a group that is in an active Patch, the group is automatically removed from the active Patch. If there are only two groups in the Patch, then the Patch is deactivated and the group goes into emergency.
- When the emergency is cleared, the Patch remains unchanged. The dispatcher must deactivate and reactivate the Patch to return it to the way it was before the emergency.
- The group is NOT removed from the Patch definition, if so defined – just the active Patch.


5.8.9.2 L3Harris Mode Behavior

- Patches can be in emergency.
- From the console, if you try to declare an emergency from the group module in a Patch, the Patch goes into emergency.
- If a radio in a patched group declares an emergency, the emergency is declared on the Patch.
- In extraordinary situations, a radio or other system component may declare an emergency on a group in an active Patch. If this occurs, the group is automatically removed from the active Patch. If there are only two groups in the Patch, then the Patch is deactivated and the group goes into emergency (same as Moto Mode).

5.8.10 Save a Patch

Click the context menu button  on the Patch module in the Patch Sidebar Panel and choose **Save Patch**.

5.8.11 Deactivate a Patch

Click the  button on a Communication Module included in the Patch or the Patch module (on the Patch Sidebar Panel or the Workspace) and choose **Deactivate Patch**.

5.8.12 Modify a Patch

Use the Patch Modify feature to modify a local patch while it is active. This lets you add or remove groups or base stations to and from an active patch. The feature also supports saving changes made to a modified local patch or saving a copy of the patch under a new name.



NOTE

Patch Modify is only supported for local active patches. Administrator defined patches can also be modified but the changes cannot be saved. However, a copy of the modified administrator-defined patches can be created. If the patch is deactivated without being saved first, it reverts to its previous definition the next time it is activated.

1. Activate a patch by either clicking on its Smart Button or selecting **Activate Patch** from its context menu.
2. Use one of the following methods to start modifying the patch.
 - Click the **Patch** Button under the clock panel. This opens a popup that displays a list of active patches. Pick the patch you wish to modify.



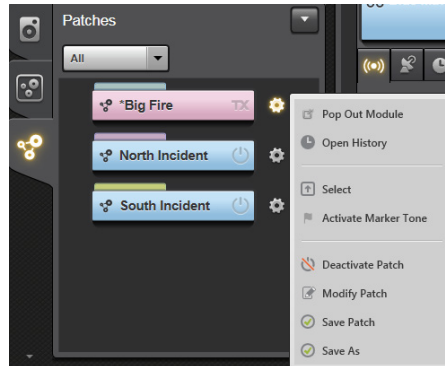
- On the Patches panel, pick Modify Patch from context menu of the patch you wish to modify.
 - On the Group or Base Station module for any of the members of the patch, pick **Modify Patch** from its context menu.
3. The icon displays on all modules that can be added to the active patch, and the illuminated icon displays on all modules currently in the active patch. The popup under the Patch button displays an **Update** (disabled) and a **Cancel** button.



4. Click the icon to add eligible members to the patch or click the icon to unpick a module. After making a change, the **Update** button is enabled.



5. Click the **Update** button to commit the changes. The Patch will now have an asterisk character (*) preceding its name. In addition, the patch adorning on all members of the modified patch will also have the asterisk character.
6. To save the modified local patch, click the context menu on the patch's Smart Button in the patch panel.



7. Click **Save Patch** to save the changes made to the patch or click **Save As** to save it under a new name. In the second case, the New Patch dialog displays.



NOTE

Ad-hoc patches can also be modified and saved. However, they do not support the option to Save As.

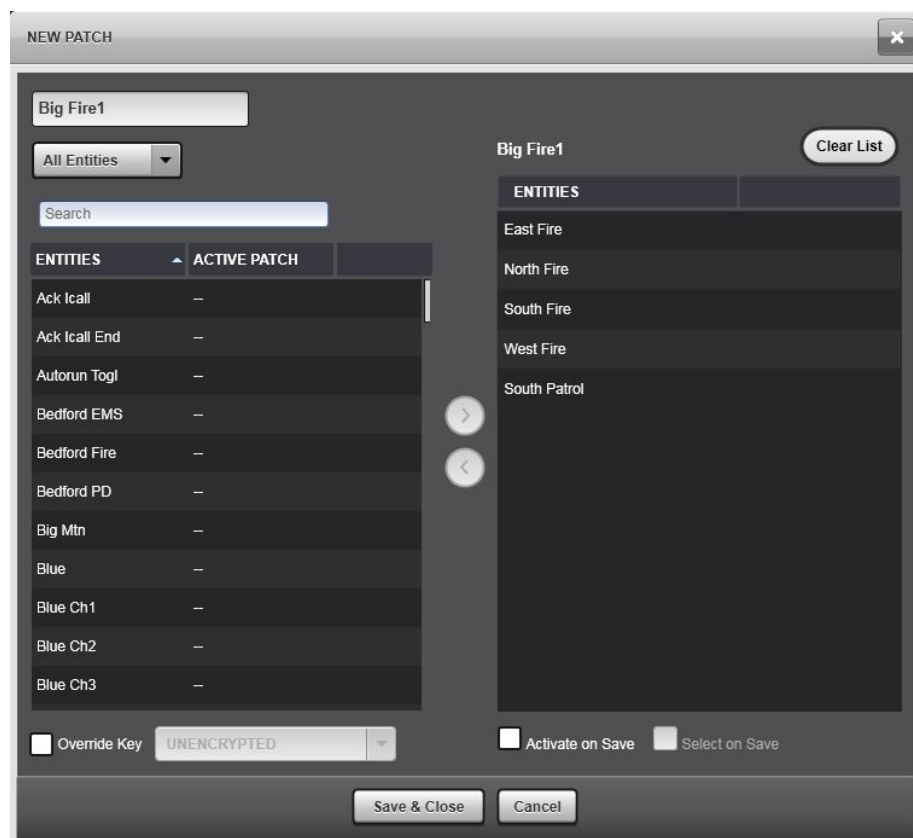


Figure 5-13: Save As

8. Refer to Section 5.8.4 for instructions on how to use this dialog to make any further modifications to the active patch, then click **Save & Close** to persist the newly defined patch. The new patch is added to the Patch Panel but is not activated.

5.9 SIMULSELECT OPERATION



Since SimulSelect is a feature-licensed option, this feature is only available if the console is equipped with this software license.

The console's "SimulSelect" feature allows simultaneous selection of up to fifteen trunked talkgroups, individual trunked radio units, and/or conventional channels. Typically, this feature is used when common broadcasts must be made from the console on two or more groups/units/channels, such as an "all-points bulletin." SimulSelect is also often used when a dispatcher wishes to simultaneously monitor incoming calls on two or more groups/units/channels in the select speaker. This behavior can also be achieved with priority groups (see Section 4.6). Typically, a SimulSelect is active (that is, in use by the console) only on a temporary basis. However, a SimulSelect may remain active for long periods of time, if necessary.

As its name implies, all Communication Modules within an active SimulSelect are simultaneously selected; therefore, incoming calls on the respective groups/units/channels are heard in the select speaker or headset. Also, a selected transmission from the console is simultaneously broadcast on all groups/units/channels in the active SimulSelect.

Like Patch operations, any group can be SimulSelected. However, if an Announcement Group is SimulSelected, talkgroups associated with that Announcement Group cannot be included in another SimulSelect. Any conventional channel can be SimulSelected. Before a group, unit, or conventional channel can be SimulSelected by the console, it must first be programmed into a Communication Module.

From a radio user's point-of-view, the SimulSelect does not exist because the groups/units/channels in the SimulSelect remain separate communication pathways. For example, radio units logged into "RESCUE 1" do not receive calls from a radio unit transmitting on "RESCUE 3" when these two groups are SimulSelected.

From the dispatcher's point-of-view, SimulSelected groups/units/channels are connected together for common dispatch communications. Other than slight incoming call indication differences in the SimulSelected modules and in the call history panel, SimulSelect acts like a selected Patch from the dispatcher's point-of-view. However, a dispatcher should always keep in mind that, unlike Patch, SimulSelected groups/channels remain separate communication pathways on the radio user's side.

The console can activate only one SimulSelect at one time. However, the console can be licensed to store up to 32 SimulSelects in its SimulSelect memory for quick-and-easy reactivation when required. Check with your administrator for your licensed limit.

Up to 15 modules can be SimulSelected together. However, SimulSelects of eight or more modules should be avoided whenever possible due to the audio summation that occurs in the select speaker. Also, an entity currently in an active SimulSelect at one console cannot be SimulSelected or patched at other consoles in the system.

The SimulSelect created and currently active at the dispatch console is considered to be a "local" SimulSelect. Any SimulSelect created and activated at another console in the system is considered a "remote" SimulSelect to all other consoles in the system.

Using the SimulSelect Modify feature, active SimulSelects can be modified. The feature allows the dispatcher to add or remove members of the SimulSelect while it is active. Before a modified SimulSelect is deactivated, the changes can be persisted using the Save and Save As options. If the changes are not saved, the SimulSelect is restored to its previous definition upon deactivation.

5.9.1 View SimulSelects

View and interact with SimulSelects via the SimulSelect Sidebar Panel. Choose the desired filtering option from the drop-down. Available options include: All, Local Active, Local Inactive, and Remote Active.

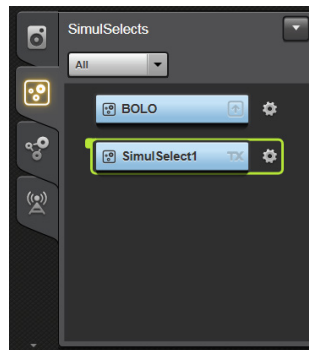




Figure 5-14: SimulSelect Sidebar Panel

5.9.2 Create a SimulSelect

There are two methods for creating a SimulSelect. Section 5.9.2.1 defines how to create an Ad-hoc SimulSelect from the Workspace (Full Screen Only). Section 5.9.2.2 defines how to create a SimulSelect using the **NEW/EDIT SIMULSELECT** dialog.

5.9.2.1 Ad-hoc SimulSelects

1. Click the **Simul** button in the top right corner of the console display.
2. Click the  icon to pick each module that you want to include in the SimulSelect. Click the  icon again to unpick a module.



SimulSelects can include Communication Modules from multiple Workspace tabs.

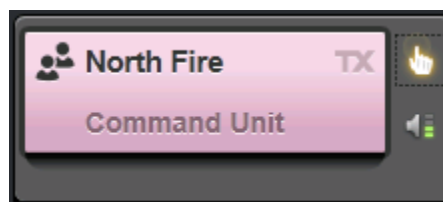


Figure 5-15: Picked Module

3. Click **SimulSelect now** to activate the SimulSelect.

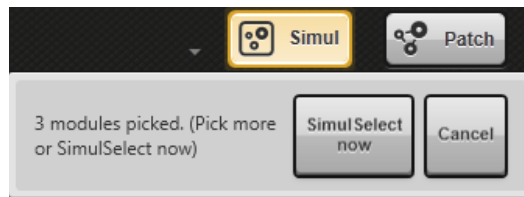


Figure 5-16: Activate SimulSelect




If the system administrator has configured this console for Auto Save Patch/SimulSelect, then the newly created Ad-hoc SimulSelect will be saved as “SimulSelect(1-8),” whichever alias is next available – starting with 1 and going to 8. If this setting is not enabled by the administrator, then the Ad-hoc SimulSelect will not be saved upon creation.

4. Modules included in the SimulSelect are identified with an “S” tag.



Figure 5-17: SimulSelected Modules


5.9.2.2 Using the New/Edit SimulSelect Dialog

1. Click the **Harris** logo or click  in the top right corner of the SimulSelect Sidebar Panel.
2. Choose **Create SimulSelect**. The **NEW/EDIT SIMULSELECT** dialog opens (see Section 3.4.8.5).
3. Enter a name for the SimulSelect.
4. Choose the entities from the list on the left and click **>** to add the SimulSelect member list on the right.
5. Specify whether or not to activate the SimulSelect (**Activate on Save & Close**).
6. Click **Save & Close**.

See Section 3.4.8 for more information on the SimulSelect Sidebar Panel.

5.9.3 Delete a Saved SimulSelect

This procedure applies to SimulSelects that have been saved and deactivated.

1. From the SimulSelect Sidebar Panel, click the context menu button  on the SimulSelect you want to delete.
2. Choose **Delete SimulSelect**.

5.9.4 Deactivate the Current SimulSelect

To deactivate the current SimulSelect, select a new entity as defined in Section 5.2.2.

5.9.5 Edit a SimulSelect

1. From the SimulSelect Sidebar Panel, click the context menu on the SimulSelect you want to edit.
2. Choose **Edit SimulSelect**. The **Edit SimulSelect** option is only available when the SimulSelect is inactive.
3. The **NEW/EDIT SIMULSELECT** dialog opens. Make the desired changes to the SimulSelect and click **Save & Close**.

5.9.6 Receive Calls on a SimulSelect

When a radio unit or another console in the system transmits on a SimulSelected trunked talkgroup or conventional channel, indications of the call appear in the respective Communication Module. If the call originates from a trunked radio unit, the caller's unit name or Unit ID number displays in the module. If the call originates from a conventional radio unit, the respective conventional channel's name or ID number is displayed in the module. Basically, incoming call indications in SimulSelected Communication Modules are identical to incoming call indications that appear in non-SimulSelected modules. Incoming calls received on a SimulSelect are also indicated in the call history.

5.9.7 Transmit on a SimulSelect

Since all modules in an active SimulSelect are always selected, a console-originated transmission on all SimulSelected groups/units/channels can be performed via a single selected transmit PTT action. Refer to Section 5.2.4 for more information on select transmitting.

In most cases, the console should not pre-empt another caller. Therefore, before transmitting, verify that no other radio unit or console is calling on a group or channel in the SimulSelect. Proceed with the PTT action only if the SimulSelected modules do not display any incoming call indications.

5.9.8 Remote SimulSelects

Other consoles in the system can also create SimulSelects. These SimulSelects are usually referred to as "remote" SimulSelects. Throughout this manual, all operational descriptions of the SimulSelect feature are assumed to be for locally-created and activated SimulSelects, unless otherwise noted.

A special SimulSelect indicator is displayed on a Communication Module if another console in the system has placed the module in a SimulSelect (see Figure 5-18 and Figure 5-19). The incoming audio is routed to the select speaker if any common module on the local SimulSelect is selected.

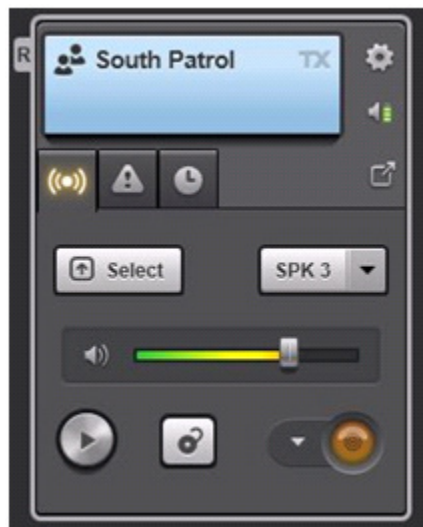


Figure 5-18: Workspace – Remote SimulSelect

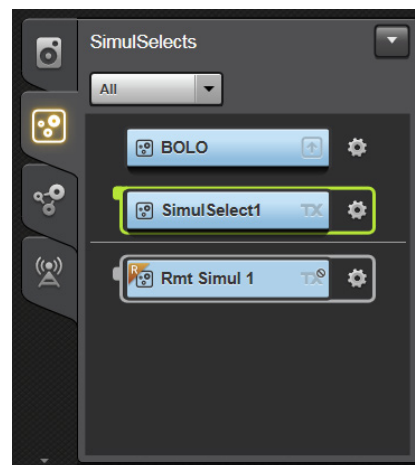


Figure 5-19: SimulSelect Sidebar Panel – Remote SimulSelect

The following identifies several operational restrictions applicable to remote SimulSelect operations:

- A remote SimulSelect cannot be deactivated or cleared at the local console. It must be deactivated or cleared by the remote console that created it, or by the system or dispatch administration personnel.
- The definition of a local Patch or SimulSelect may be modified while a group, unit or conventional channel is currently in a remote SimulSelect.
- A local Patch or SimulSelect definition that contains a group, unit, or conventional channel currently in a remote SimulSelect cannot be activated at the local console. This includes creating Ad-hoc patches and SimulSelects.

5.9.9 Emergencies on a SimulSelect

The following information applies to emergency operations on SimulSelected groups. Refer to Section 5.11 for additional information on emergency operations.

- Since SimulSelected groups/channels remain separate communication pathways (unlike patched groups/channels), an emergency declaration on one SimulSelected group does not cause the other groups within the SimulSelect to enter an emergency condition. This statement applies to all SimulSelects, both local and remote.
- Accordingly, if an emergency is declared on a SimulSelected group, indications of the emergency appear in the respective group-programmed Communication Module, not all of the SimulSelected modules. The group emergency is also indicated in the call history panel's select history list. In addition, the tab of any Workspace containing the module in emergency indicates the emergency via a badge.
- Any console in the system that has a SimulSelected emergency group programmed may clear an emergency on the SimulSelected group. A supervisory trunked radio unit that has access to the SimulSelected emergency group can also clear an emergency on the SimulSelected group.
- An active SimulSelect can be deactivated if an emergency exists on one or more of the SimulSelected groups.
- If an emergency is declared on a SimulSelected group, emergency indications appear only in the respective group-programmed Communication Module.

5.9.10 Mixed Mode Patch and SimulSelect

The "Enable Patch/SS to be created with different Vcoders" option in the Symphony Configuration Utility determines which groups, or which groups and units, can be included in a Patch or SimulSelect. This option can be changed only by the administrator. When the option is enabled, any group or any group or unit can be included in a Patch or SimulSelect regardless of its vocoder. When disabled, all groups or all groups and units must have the same vocoder to be included in a Patch or SimulSelect. The Symphony user interface enforces this by removing possible groups and units from the selection process when creating and editing patches and SimulSelects.

When multiple vocoders are included in a Patch or SimulSelect, Symphony automatically chooses the vocoder for the Patch/SS. For details about this process, see the Installation and Configuration manual.

5.9.11 Modify a SimulSelect

Use the SimulSelect Modify feature to modify a local SimulSelect while it is active. This lets you add or remove groups, users, or base stations to and from an active SimulSelect. The feature also supports saving changes made to a modified local SimulSelect or saving a copy of the SimulSelect under a new name.





NOTE



SimulSelect Modify is only supported for local active SimulSelects. Administrator defined SimulSelects can only be modified but the changes cannot be saved. However, a copy of the modified administrator-defined SimulSelects can be created. If the SimulSelect is deactivated without being saved first, it reverts to its previous definition the next time it is activated.

1. Activate a SimulSelect by either clicking on its Smart Button or clicking **Select** from its context menu.
2. Use one of the following methods to start modifying a SimulSelect.
 - Click the **Simul** button under the clock panel. This opens a popup that displays the name of the active SimulSelect. Pick the SimulSelect to modify it.



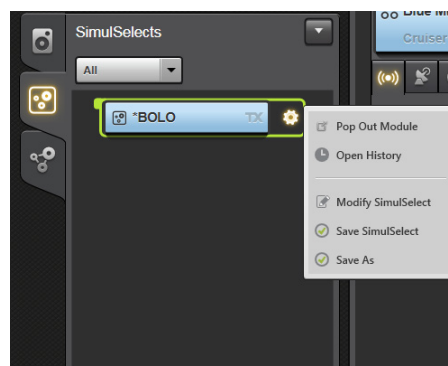
- On the SimulSelect panel, click **Modify SimulSelect** from context menu of the SimulSelect.
 - On the Group, User, or Base Station module for any of the members of the SimulSelect, click **Modify SimulSelect** from its context menu.
3. The  icon displays on all modules that can be added to the active SimulSelect, and the illuminated  icon displays on all modules currently in the active SimulSelect. The popup under the **Simul** button displays an **Update** (disabled) and a **Cancel** button.



4. Click the  icon to add eligible members to the SimulSelect or click the  icon to unpick a module. After making a change, the **Update** button is enabled.



5. Click the **Update** button to commit the changes. The SimulSelect will now have an asterisk character (*) preceding its name.
6. To save the modified local SimulSelect, click the context menu on the SimulSelect's Smart Button in the SimulSelect panel.



- Click **Save SimulSelect** to save the changes made to the SimulSelect or click **Save As** to save it under a new name. In the second case, the New SimulSelect dialog displays.



NOTE

Ad-Hoc SimulSelects can also be modified and saved. However, they do not support the **Save As** option.

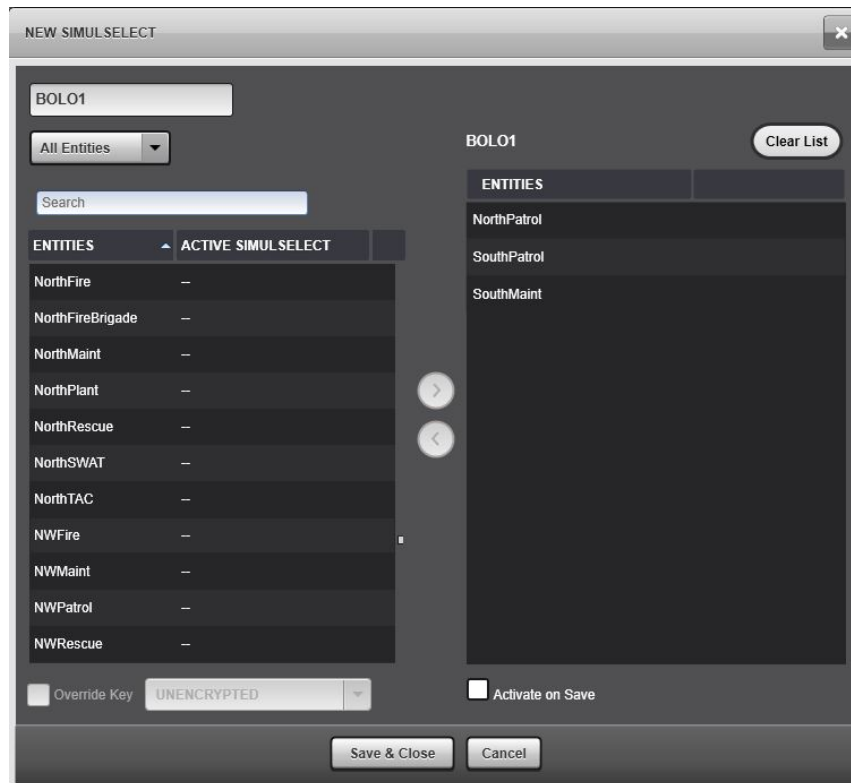


Figure 5-20: Save As

- Refer to Section 5.9.5 for instructions on how to use this dialog to make any further modifications to the active SimulSelect, then click **Save & Close** to persist the newly defined SimulSelect. The new SimulSelect is added to the SimulSelect Panel but is not activated.

5.10 ENCRYPTED VOICE DISPATCH



NOTE

Since Encrypted Voice is a feature-licensed option, this feature is only available if the console is equipped with this software license.

Some dispatch centers are configured for encrypted voice dispatch. If your dispatch center is not configured for encrypted voice dispatch, the information in this section does not pertain to your console and can be ignored.

Encrypted voice dispatch allows the console to communicate with radio units and consoles in the system that can transmit and receive encrypted voice signals. For the trunked radio system in general, these signals can provide great voice security for sensitive communications by making unauthorized monitoring of the radio frequency channels nearly impossible.

Encryption requires a special encryption key. If encryption is used, this key must match at both the transmitting and receiving units/consoles before encrypted communications are possible. For consoles in the radio system, the encryption key is stored in the console and all encryption encoding and decoding is done within the console.



When the console indicates it is transmitting an encrypted call, this means the call from the console to the system is encrypted. However, depending on the system configuration some call legs may be decrypted before being sent over the air. For example, conventional stations do not support over the air encryption, so the call must be decrypted at the Interoperability Gateway. Consult with your system administrator to understand when calls may not be encrypted end-to-end on all call legs in your system.

5.10.1 Encryption Algorithms

The Symphony Dispatch console supports both DES (Data Encryption Standard) and AES (Advanced Encryption Standard) encryption algorithms. The algorithm(s) used in a given dispatch center or system is determined by the radio system's administrator and is configured/maintained by the system's crypto-officer(s). As such, the selection of algorithm should be transparent to the dispatcher who does not need to know which type of encryption is being used for any given encrypted call. It should be noted, however that the DES encryption algorithm is only supported for group calls and base stations. It cannot be used to encrypt I-Calls, Patches, or SimulSelects.



FIPS-140 compliance prohibits the use of DES. Therefore, FIPS-140 compliance must be disabled to use DES. See the Installation Manual for details.

5.10.2 Receiving Encrypted Voice Calls

This section describes the general behavior for receiving encrypted calls of any type and specifically the behavior of receiving group calls. Deviations from this behavior is noted in sections describing other call types.

For received calls, switching between encrypted voice mode and clear voice mode is automatically handled by the console. ***The console does not control the voice mode for calls being received.*** This automatic switching is based upon the voice mode of the incoming call. In other words, if an incoming call is an encrypted voice ("private") transmission and the encryption key matches, the call is automatically decoded by the console. Whereas an unencrypted incoming call does not require decoding.

Figure 5-21 shows a received encrypted call. The receive indicator in the top left of the Smart Button glows orange for encrypted calls (the indicator is green for unencrypted calls) and the callee icon on the bottom left of the Smart Button includes a lock symbol. If encryption is being used and the console cannot find the proper key, audio is not heard or may be garbled and the receive indicator in the top left of the Smart Button glows red. The callee icon also has a red padlock.



Figure 5-21: Group Module Receiving Encrypted Call

5.10.3 Transmit Encrypted Voice Calls

This section describes the general behavior for transmitting encrypted calls of any type and specifically the behavior of transmitting group calls. Deviations from this behavior is noted in sections describing other call types.

For console transmit operations, encrypted voice transmit mode is entered by manually switching a module from unencrypted (clear voice) transmit mode to the encrypted voice transmit mode (private). This is accomplished via the Encryption Button on the module as shown in Figure 5-22. When in private transmit mode, the Encryption Button glows orange and an encryption badge typically appears in the top right corner of the Smart Button. The appearance of the badge depends on Nominal Encryption Configuration Setting. A badge can be configured to appear on encrypted modules only, unencrypted modules only, or both. The badge for encrypted modules shows a padlock in the locked position and the unencrypted badge shows a padlock in the unlocked position. Consult your System Administrator to determine which badge behavior to expect.

It is the responsibility of the dispatcher and the radio user to utilize encrypted (private voice) transmit mode when required to guarantee his/her voice security throughout the radio system.

To transmit an encrypted call to a group, switch the respective group module to private transmit mode using the Encryption Button, and then use standard select or instant transmission procedures. When the console is transmitting a call of this type, the encryption badge remains in the module per the previous transmit mode switching action.

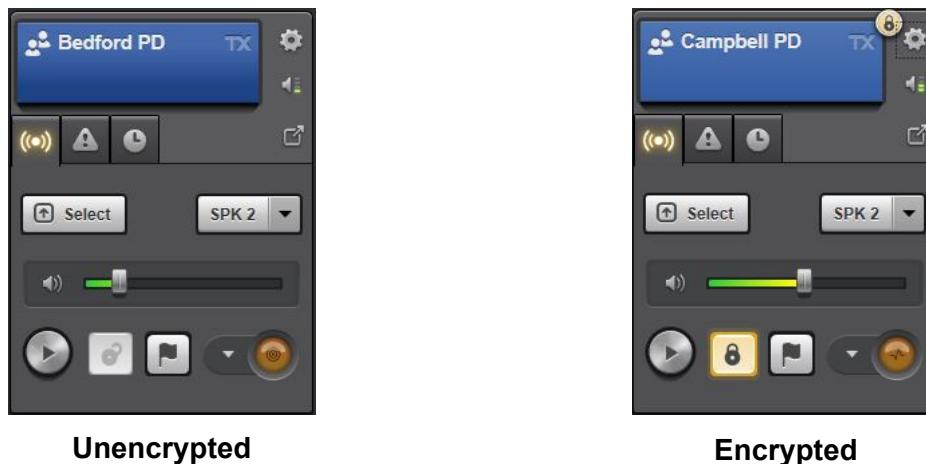


Figure 5-22: Group Module Encrypted Transmission States

5.10.4 Encrypted I-Calls (P25 and AES Only)

I-Call encryption follows the general encryption behaviors noted above with a few exceptions. Unit modules include an Encryption Button to control the transmit state of I-Calls to the associated unit. All encrypted I-Calls use the System key. The Encryption button on unit modules will be grayed out if encryption is not licensed or if a system key is not configured.

Unlike group calls, I-Calls have an associated session. I-Calls starts with the caller proposing parameters including the Vocoder type and encryption state. Once the callee accepts the proposed parameters, the I-Call parameters typically remain the same for the I-Call session.



The console prohibits changing the encryption state during the I-Call; however, it is possible the other member of the call could change their transmit state. If this happens, you will see a received call indication that is different from the current indicated I-call state (green light on an encrypted I call, or orange on unencrypted). The audio is played out of the speaker or headset as normal, unless you do not have the proper encryption key.

When receiving an I-Call, the caller specifies the encryption state of the call in the call request. If the console can support the requested call, it accepts the call irrespective of the encryption settings on the unit module. If the call cannot be supported, it is rejected and logged in the System History. For example, if encryption is requested, the console accepts the encryption request even if the unit module is unencrypted.

The encryption button and badge on the unit module change to match the state of the I-Call. When the call ends, the unit module reverts to its pre-call state. That is, the received I-Call does not permanently change the state of the unit module. Acknowledged I-Calls can be ended by explicit user action or timeout while unacknowledged I-Calls only end by timeout. Be aware of the timeout as the encryption state may appear to change spontaneously, but it is just reverting to the pre-call state when the timeout expires.

When initiating an encrypted I-Call from the console, set the encryption button to the encrypted position (depressed and glowing) and place the call as usual. If the callee can support the call, it is accepted. If not, it is rejected.

Though BeOn® generally follows P25 conventions, BeOn I-Calls are an exception. BeOn uses a form of instant unacknowledged I-Call where the I-Call session is only as long as the talk spurt, so each talk spurt is independent. If the console and radio are in the same encryption state, the I-Call behaves like the unacknowledged I-Call described above. If the encryption states are different, the console's encryption state indicators flip during each received call spurt.

5.10.5 **Encrypted Dispatch with Patches (P25 and AES Only)**

The Symphony Dispatch Console supports encryption of patches with limitations based on your system's configuration. The determination of when a Patch may be encrypted can be complex, especially in systems that support multiple technologies such as P25, OpenSky, and analog conventional. Fortunately, the Symphony Dispatch Console simplifies this process by preventing the Patch from being encrypted when the system cannot support it. The following general rules will help you understand the basic process:

- Consoles can encrypt a Patch if (1) the console has been provisioned with the P25 system key and (2) the Patch's vocoder is a P25 vocoder. The console determines the vocoder for a Patch based on the vocoder types of the group being patched. The most common is used for the Patch.
- Patches are encrypted with the P25 system key by default. The system key can be overridden using the New/Edit Patch Dialog. Only those units with the key used by the Patch can hear the encrypted calls.
- If ALL groups are encrypted when a Patch is activated, and the console can encrypt the Patch; the Patch will be encrypted.
- If the Patch cannot be encrypted, the Encryption Button on all modules in the Patch will be grayed out.
- Pre-defined and saved Patches use the encryption state of the previous activation when they are re-activated. That is, if the Patch was unencrypted the last time it was active, it stays unencrypted when it is reactivated regardless of current encryption state of each entity in the Patch. The previous encryption state is preserved through a reboot of the Symphony application.
- If the Patch can be encrypted, toggle the Patch between encrypted and unencrypted modes by pressing the Encryption Button on any module in the Patch. The Encryption Button on each module in the Patch reflects the encryption state of the Patch. When the Patch is deactivated, each module will revert to its encryption state before the Patch was activated.



NOTE

If your system comprises multiple technologies such as P25, OpenSky, and analog conventional, an encrypted Patch may not be encrypted for all radios in the Patch. Conventional radios generally do not support encryption, so the call is encrypted over the VIDA network, but it is unencrypted over-the-air. OpenSky radios may not be encrypted depending on your system configuration. Consult your system administrator for information on your system.

5.10.6 Encrypted Dispatch with SimulSelect (P25 and AES Only)

Symphony supports encryption of SimulSelects with limitations based on your system's configuration. The determination of when a SimulSelect may be encrypted can be complex, especially in systems that support multiple technologies such as P25, OpenSky, and analog conventional. Fortunately, Symphony simplifies this process by preventing the SimulSelects from being encrypted when the system cannot support it. The following general rules will help you understand the basic process.

- Consoles may encrypt a SimulSelect if (1) the console has been provisioned with the P25 system key and (2) the SimulSelect's vocoder is a P25 vocoder. The console determines the vocoder for a SimulSelect based on the vocoder types of the group being SimulSelected. The most common is used for the SimulSelect.
- SimulSelects are encrypted with the P25 system key by default. The system key can be overridden using the New/Edit SimulSelects Dialog. Only those units with the key used by the SimulSelects can hear the encrypted calls.
- If ALL groups are encrypted when a SimulSelect is activated, and the console can encrypt the SimulSelect, the SimulSelect will be encrypted. The encryption state of units is ignored when determining the SimulSelect encryption state. If only units are SimulSelected, the SimulSelect is unencrypted regardless of the unit's encryption state.
- Pre-defined and saved SimulSelects use the encryption state of the previous activation when they re-activated. That is, if the SimulSelects was unencrypted the last time it was active, it will stay unencrypted when it is reactivated regardless of current encryption state of each entity in the SimulSelects. The previous encryption state is preserved through a reboot of the Symphony application.
- If the SimulSelect can be encrypted, toggle the SimulSelect between encrypted and unencrypted by pressing the Encryption Button on the Select Module. Note that this is different from the Patch behavior. Unlike patches, groups and units within an active SimulSelect remain separate communication pathways. The Encryption Button on each module in the SimulSelect shows the state of the module, not the SimulSelect. Therefore, the Encryption button on the Select Module is used to switch the SimulSelect between encrypted transmit mode and unencrypted transmit mode.



If your system comprises multiple technologies such as P25, OpenSky, and analog conventional, an encrypted SimulSelect may not be encrypted for all radios in the SimulSelect. Conventional radios generally do not support encryption, so the call is encrypted over the VIDA network, but it is unencrypted over-the-air. OpenSky radios may not be encrypted depending on your system configuration. Consult your system administrator for information on your system.

5.10.7 Recommendations and Limitations of Digital Voice Encrypted Communications

The following list identifies recommendations and limitations of encrypted voice dispatch communications. Consult with the system or dispatch administration personnel for additional information.

- Utilize encrypted transmit mode only when instructed by the system or dispatch administration personnel.
- In some cases when attempting to set a group, Patch, SimulSelect, or unit to encrypted mode by pressing the Encryption Button, the mode will not toggle. Typically, this occurs when the console does not have the necessary encryption key. Contact the system or dispatch administration personnel to have them remedy the problem.
- If encryption is being used and garbled audio is heard when a call is received, there is a key mismatch. If this continues, correlate the noise with call indications on modules and then contact the system or dispatch administration personnel with the information. The key for that entity needs to be programmed into console, the database corrected, or the entity should be removed from the console.
- If the console cannot find the proper key when receiving an encrypted call, the receive indicator glows red and the lock on the caller icon is red as well.
- Symphony only supports AES encryption on base stations. All channels, represented to the VIDA system as trunked groups, must be configured for AES encryption. Encryption is disabled for the base station if any of its mapped channels (trunked groups) are configured for DES encryption or are not configured for AES encryption.

5.11 EMERGENCY OPERATION

Symphony provides two features for managing emergency situations: group emergency and unit alert. Group emergencies are declared on a talkgroup and change the priority of the group in the system. Unit Alerts only apply to the unit sending the alert and do not affect a group. Radios can be programmed to send either a group emergency or unit alert or both. Consult your system administrator to determine how your system is configured.

5.11.1 Group Emergency

A group emergency can be declared on any trunked talkgroup. Group emergencies are not available from base stations. Trunked radio units and dispatch consoles within the system can declare an emergency on a group. This feature provides a quick and easy means of identifying special circumstances, such as a life-threatening situation, that must be acted upon immediately by system personnel logged into or monitoring the group. If a group is in an emergency condition, it is sometimes referred to as an “emergency group.” Voice transmission to an emergency group can be performed from radio units and dispatch consoles via standard transmission procedures.

Trunked radio units monitoring an emergency group typically display a visual indication of the emergency. In some cases, a radio unit may also be programmed to sound an alert tone (“beep”) to audibly indicate the emergency. Like trunked radio units, the Symphony dispatch console indicates an emergency group via visual indications and, in most cases, audible alarms. These indications are described in detail later in this section.

If a radio unit declares an emergency on a group, the emergency is considered a “radio-originated emergency.” Likewise, if a console declares an emergency on a group, the emergency is considered a “console-originated emergency.” The Symphony dispatch console displays the identity of originating radio unit or dispatch console.

After declaration and subsequent voice transmissions, an emergency condition on a group can be cleared from either a dispatch console or a supervisory trunked radio unit. The emergency’s originator need not be the one who clears the emergency condition on the group. Consoles can be set up so that the local emergency alarm must be acknowledged before the emergency can be cleared via the console. Acknowledging the alarm silences it locally at the console. This provides extra safety against inappropriately cleared emergencies. In addition, acknowledging the alarm allows the dispatcher to recognize and manage multiple emergency declarations on the same talkgroup. Emergencies can be acknowledged or cleared by any console monitoring the emergency. When an emergency is cleared, all consoles reflect the new emergency state. When a group emergency is acknowledged, all consoles where the group/patch is unselected will reflect the new state. Consoles where the group/patch is selected, must manually acknowledge the emergency.

Dispatchers wishing to turn off the emergency tone without affecting the emergency state should use the TONE OFF button in the select bar.

Calls on emergency groups are given a higher priority throughout the trunked radio system than non-emergency calls. This ensures that these calls are less likely to be queued by the system during busy periods.

Multiple emergencies can be declared on the same group. When subsequent emergencies are declared, the unit name or ID of the declaring radio or console is displayed until the emergency declaration call (hot mic) or any subsequent receive call ends. The module will then display the unit name or ID of the first emergency declarer.

5.11.2 Receive Emergency Group Calls

When an emergency call is received, there are both visual and audible indications at the console. The visual indications displayed for an incoming emergency group call include:


- The Communication Module is highlighted red and  flashes on the module’s Smart Button as shown in Figure 5-23. The declarer is shown in red text and no icon on the left where the caller is normally shown on the applicable Smart Button size. For example, North Dispatch declared the emergency in Figure 5-23. Cruiser 17 is a caller as denoted by the black text and icon on the left.
- Normal size modules include an emergency tab to manage the emergency on this group. The tab lists the uncleared emergencies on this group and provides a Declare button to declare a group emergency from the console and an ack/clear control to acknowledge all the declarers and clear the group emergency. The **Ack/Clear** control includes a badge with the count of uncleared group emergencies.



Figure 5-23: Communication Modules in Emergency

- The Emergency Sidebar Panel is populated with a mini module for each group in emergency and an **Ack/Clear** control with a badge as shown in Figure 5-24. The **Ack/Clear** control includes a badge with the count of uncleared group emergencies. The Emergency Sidebar Panel tab also has a badge indicating the number of uncleared group emergencies, plus uncleared unit alerts.



NOTE

If radios are not programmed to send unit alerts, the badge is the number of uncleared group emergencies.



Figure 5-24: Emergency Sidebar Panel

- An emergency badge indicating the number of emergencies plus the number of unit alerts is displayed on the Workspace tab of the Full Screen display or the emergency button of the Baton.



NOTE

If radios are not programmed to send unit alerts, the badge is the number of uncleared group emergencies.

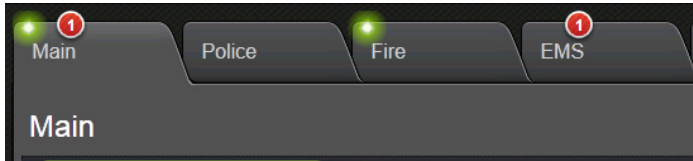


Figure 5-25: Emergency Badges – Full Screen



Figure 5-26: Emergency Badge – Baton

- The Workspace Module List button turns red if a module in the associated Workspace is in emergency, but not displayed on the Baton.



Figure 5-27: Emergency Indication – Baton

The audible emergency indications at the console are heard as an alarm that begins sounding upon declaration of the emergency. During the emergency, this alarm can be silenced locally at the console without clearing the emergency by clicking the **TONE** button in the Select bar.

5.11.3 Transmit on an Emergency Group

During an emergency condition, the console's transmit functions operate the same as during non-emergency conditions. This includes operations on the groups within the emergency and on other entities programmed into other Communication Modules. Both selected transmits and unselected transmits can be performed per standard PTT actions. **Great care should be exercised when transmitting on an emergency group. For example, a radio unit's incoming emergency declaration transmission cannot be pre-empted by a dispatch console's transmission. Consult with system and/or dispatch administration personnel for specific instructions.**

5.11.4 Declare an Emergency

The procedure listed below describes how to declare an emergency on a trunked talkgroup. **Use great caution, as this action declares the emergency on the group throughout the entire radio system where access to the group is available.**

- If the module on which the emergency is to be declared is not on the currently displayed Workspace tab, switch to the appropriate Workspace tab or use the Directory dialog to access the module.
- Display the emergency tab on the Communication Module (pop-out mini and small modules to access the emergency tab).
- Click and hold the **Declare** button for one second. The delay prevents accidental emergency declaration. After the emergency is declared, release the button.

4. Once the emergency is declared, continuing to hold down the button transmits on the talkgroup allowing the dispatcher to make an announcement.
5. Once the announcement is completed, unkey the **Declare** button to end the call.

Some radio systems require that a call is placed at the same time as emergency declaration to propagate the emergency to the field units. Symphony will always place a short call (about ½ second long) when declaring an emergency even if the dispatcher unkeys the **Declare** button quickly.




After an emergency is declared on a group, receive and transmit operations will follow the same procedures as previously described in this chapter, using the Smart Button and other PTT sources. Pressing the **Declare** button will re-declare the emergency on the talkgroup.

5.11.5 Acknowledge/Clear an Emergency



The clear action clears the emergency on the group throughout the entire radio system. The acknowledge action affects all consoles on the system where the group is not selected.



From a group module:

1. If the emergency group module is not on the currently displayed on the Workspace, switch to the appropriate Workspace.
2. On the emergency tab, click  at the bottom of the tab to acknowledge all emergencies on the group, or click  to acknowledge a specific declarer.
3. Click  at the bottom of the tab to clear the emergency in the system.



The **Clear** button will only appear once all the declarers have been acknowledged.

From the Emergency Sidebar Panel:

1. Click  under the group mini module to acknowledge all emergencies on the group. To individually acknowledge each declarer, use the emergency tab procedure described above.
2. Click  under the group mini module to clear the emergency in the system.



The **Clear** button only appears after all the declarers have been acknowledged.

5.11.6 Transmit Alert Tones

Typically, tone transmissions are utilized during emergency operations to signal radio units of various pre-defined emergency conditions, with voice transmissions following. When a tone is transmitted from the console, it is simultaneously heard in a speaker at the console. To prevent voice interference, all console microphone audio is muted during a tone transmission.

Five separate distinctive tones can be transmitted from the console to signal emergency conditions:

- Alert Tone (A) - A continuous medium-frequency tone that sounds while the respective button is depressed.
- Warble Tone (W) - A repeating sequence of a medium-frequency tone followed by a higher frequency tone that sounds while the respective button is depressed.
- Pulse Tone (P) - A medium-frequency tone that turns on and off repeatedly while the respective button is depressed.
- Custom1 Tone (C1) - A custom1 tone that sounds while the respective button is depressed.
- Custom2 Tone (C2) - A custom2 tone that sounds while the respective button is depressed.

To transmit an alert tone:

1. Choose the desired tone type from the drop-down on the audio tab of the Communication Module if you want to change the setting (Figure 5-28).



Figure 5-28: Alert Tone Selection

- Click and hold the orange button to transmit the tone. The button will light during tone transmission. Release the button to stop tone transmission.

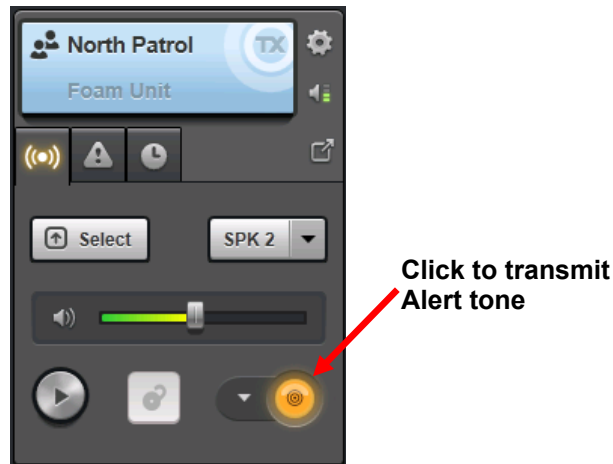


Figure 5-29: Alert Tone Transmit

5.11.7 Unit Alert

Unit alerts are initiated by radios and are managed at the console. The console cannot declare a unit alert. The definitive unit alert state is maintained by the radio and mirrored in the system. When the console acknowledges or clears a unit alert, these actions only apply to the system and do not change the state in the radio. The radio user must clear the unit alert in the radio. However, changes in the system unit alert state are seen by all consoles. That is, if one console acknowledges a unit alert, all other consoles showing the alert will see the acknowledge. The console that acknowledges the unit alert takes ownership of the alert and therefore is the only console that can clear the alert.

Unit alerts appear in the Emergency Sidebar Panel as shown in Figure 5-30 and an alarm tone sounds. A badge appears on the Emergency Sidebar Panel tab indicating the number of group emergencies plus the active unit alerts.



NOTE

If radios are not programmed to send group emergencies, the badge is the number of uncleared unit alerts.

The declarer is listed along with a button to acknowledge/clear the unit alert. Unit Alerts can be associated with a “talkback” group in which case a group mini-module is included for that group. This group is the selected group on the radio when the unit alert was declared. This group does not update when the radio selection is changed. Note that the group is not in emergency. The group is associated with the unit alert and provides a way to talk to group members associated with and who may render aid to the declarer. Use the mini-module to communicate with the associated group or click the pop-out icon to pop-out an individual module and place an I-Call to the declarer.

A single user may be logged into multiple radios on OpenSky and VIDA ID-enabled systems. If multiple unit alerts are declared by the same user ID from multiple radios, the console displays each unit alert separately. On the emergency Panel, the User Alias will be preceded by a “P-” or “M-” indicating the user’s portable or mobile. Administrators may configure the “P-” or “M-” indicators so they may be different on your system.

Click the **Ack** button to acknowledge the unit alert and silence the alarm tone. This also takes ownership of the unit alert so only this console can clear it. The **Ack** button changes to a **Clear** button. Click the **Clear** button to clear the unit alert if you acknowledged the unit alert. Each unit alert must be individually acknowledged and cleared. If the unit alert is acknowledged by another console, the **Ack** button changes to **Owned** so the unit alert cannot be cleared from this console.



Figure 5-30: Unit Alert in Emergency Sidebar Panel






Figure 5-31: Multiple Unit Alerts from one User in Emergency Sidebar Panel

Unit alerts are also indicated in the emergency tab of individual modules as shown in Figure 5-32. This tab provides information about the unit alert, including the declaration time and GPS location if available (OpenSky systems only). It also provides an indicator of the radio type initiating the alert if the user is registered to more than one radio (VIDA ID and OpenSky Units). In addition, it includes an Ack/Clear icon and a mini-module for the associated group. To place an I-Call to the individual, key on the module Smart Button at the top. To place a group call to the associated group, key the group mini-module at the bottom of the tab.



NOTE

GPS information is available on OpenSky systems only. Unit alerts from P25 and conventional units will show no GPS location.

To acknowledge the unit alert and silence the alert tone, click the  icon. The icon changes to . Click  to clear the unit alert if you acknowledged it.

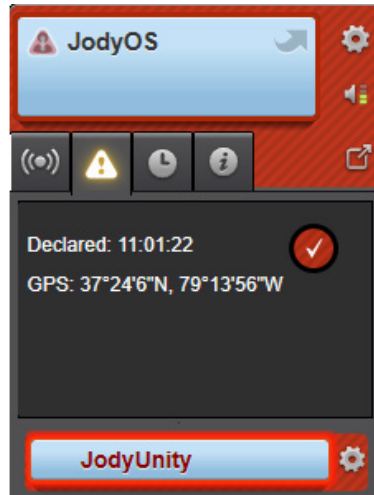


Figure 5-32: Unit Alert in Individual Module

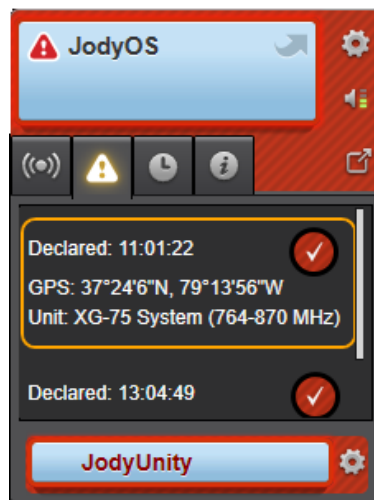


Figure 5-33: Multiple Unit Alerts in Individual Module

5.11.8 Mixed Emergency Operation

Radios can be programmed to send both a group emergency and a unit alert. In this case, the group and individual modules handle the group emergency and unit alert respectively as defined above. However, the Sidebar Panel integrates the two emergency types as shown in Figure 5-34.



Figure 5-34: Unit Alert and Group Emergencies in Emergency Sidebar Panel


The radio declares an emergency on a group and associates the group with its unit alert. Figure 5-34 shows an example of two radios declaring group emergencies and unit alerts on the same group “Roanoke PD.” The emergency panel includes, from top to bottom, a mini module for the group in emergency (which is also the talkback group for the unit alerts), the ack/clear control for the group emergency, a list of each unit alert (in this case, two unit alerts). Unit alerts are distinguished by the person icon.

Manage the group emergency and unit alerts as described above. The panel organizes them by the common group they share (i.e., emergency group and unit alert talkback group). The badge on the Emergency Sidebar Panel tab shows the total count of groups in emergency and units in alert. In this example, there is one group in emergency and two units in alert for a total of three as shown on the badge.

5.12 CROSS-MUTE OTHER CONSOLES

The Symphony Dispatch Console can cross-mute other consoles. The cross-mute function:

- Reduces unnecessary receive audio at the local console by preventing transmissions from other consoles from being heard.
- Prevents audio feedback problems when two or more consoles are placed in proximity with each other and at least one is equipped with speakers. This audio problem is typically described as a “squeal” in the consoles’ speakers and/or headsets. It occurs when an open microphone at one console picks up a speaker at another console connected to the same entity.

When another console in the system is cross-muted, none of the cross-muted console’s transmit audio is heard in the local console’s speakers or headset when the cross-muted console transmits. However, the cross-mute icon  appears in the respective module to visually indicate that the cross-muted (remote) console is transmitting. The cross-muted console’s name may also be indicated in the respective module.

Cross-Mute functionality is enabled by the administrator using the Symphony Configuration Utility. However, depending on the mode of Cross-Mute enabled, a dispatcher may or may not be able to dynamically cross-mute other consoles while operating their console. Symphony supports two modes of Cross-Mute: Static and Dynamic.

5.12.1 Static Cross-Mute

Static Cross-Mute is configured using the Symphony Configuration Utility. This utility enables an administrator to create a list of all the consoles in the system, then specify which consoles are to be cross-muted and which are not. In this mode, a dispatcher cannot change the list of consoles predefined for Cross-Mute by the administrator. Any change to this list will require intervention by the system administrator. Refer to the *Symphony Dispatch Console Installation and Configuration Manual* for details.

5.12.2 Dynamic Cross-Mute

Dynamic Cross-Mute is also configured using the Symphony Configuration Utility. However, this feature requires a license to be enabled in Symphony for this capability to work. In this mode, dispatchers are given the ability to enable and disable cross-mute for a list of all consoles predetermined by the administrator to operate on the network. Refer to the *Symphony Dispatch Console Installation and Configuration Manual* for details on licensing and configuring this feature.

If Symphony is licensed and configured for Dynamic Cross-Mute, the Cross Mute Sidebar Panel can be displayed. If Dynamic Cross-Mute is not licensed, but is configured in the Symphony Configuration Utility, the Cross Mute Sidebar Panel is displayed with no Alias entries but will have the text “Feature not licensed” displayed on the panel.

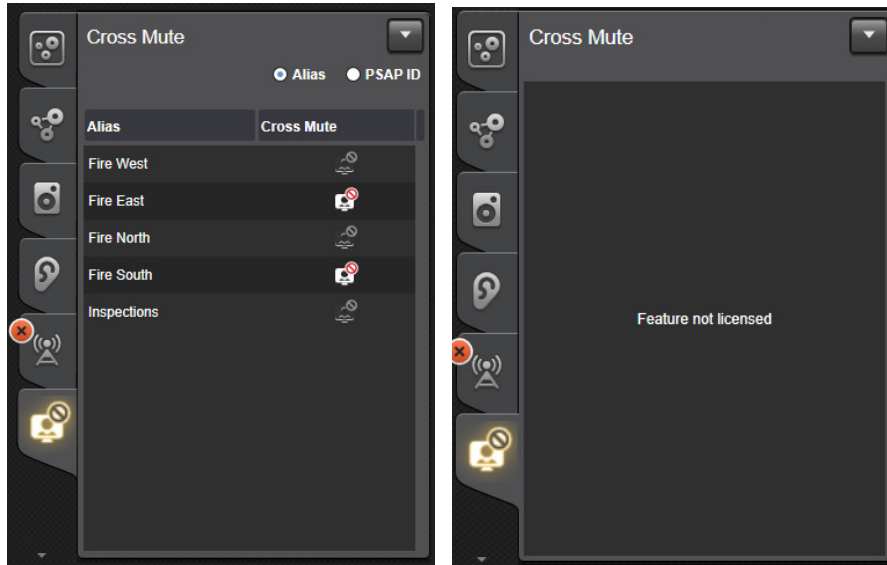


Figure 5-35: Cross Mute Sidebar Panel

The panel lists all consoles that the administrator has previously defined. These consoles are either identified by their Alias or PSAP ID (PSAP ID specifies the physical hardware rather than the User). At the top right of the panel is an Options Menu button (Open Directory or Open History) and two radio buttons that allow the dispatcher to dynamically toggle between list by Alias or list by PSAP ID.

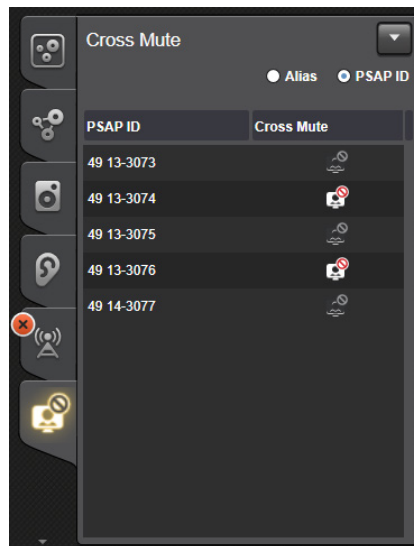


Figure 5-36: Cross Mute Sidebar Panel (List by PSAP ID)

In the right column of the console list, beside each console identifier, is one of two icons indicating whether or not the corresponding console is actively cross-muted.



The 'active' icon indicates that the corresponding console is actively cross-muted.



The 'ghosted' icon indicates that the console is not cross-muted.

Clicking the icon for the applicable console toggles the console from one state to the other.

5.13 SETUPS

The Symphony Dispatch Console is capable of storing up to 16 setups. Each setup can be unique in terms of Communication Module programming, as well as various other settings and parameters. Setups allow customization of the console for different situations and different dispatch operators.

To switch to a new setup:

1. Click the **Harris** logo in the top left of the display.
2. Click **Setups** → **Switch Setups**. The **SWITCH SETUPS** dialog opens as shown in Figure 5-37. Choose the desired setup from the list.



Figure 5-37: Switch Setups Dialog



NOTE

If the setup is password protected, an **Enter Password** field is displayed on the **SWITCH SETUPS** dialog. The correct password must be entered before switching to this setup.

3. Click **Switch**.

If the current setup is not password protected, then it is restored on restart. If it is password protected, the console restores the Default setup to preserve the password protection.

6 TELEPHONY OPERATION

Symphony supports two kinds of telephony:

- **Call Director:** This feature connects a physical phone to the console as if the console were a headset for the phone. All telephony functions are controlled with the physical phone.
- **SIP VOIP Telephony:** This feature uses Voice over IP (VOIP) and Session Initiation Protocol (SIP) to integrate telephony features directly into the Symphony user interface. With this feature, Symphony replaces the physical phone and provides access to all telephony features including call transfer and conferencing.

Symphony currently only supports one telephony mode at a time. The user interface for Call Director and SIP telephony are similar, but SIP VOIP telephony provides many more features. Details of each interface are provided in the appropriate sections of this manual.

6.1 CALL DIRECTOR OPERATION



NOTE

Since Call Director is a feature-licensed option, this feature is only available if the console is equipped with this software license.

The Symphony Console can be connected to a telephone for telephone-interconnect operations. This enables the dispatcher to access standard telephone lines for standard telephone call operations while maintaining normal dispatch communications. The Call Director (CD) feature manages the communications with the telephone and coordinates interactions with radio dispatch features.

This manual describes the typical Call Director configuration where Symphony interfaces to a telephone via the headset jack on the phone. Due to differences in phone systems, you may have additional equipment specifically designed for your installation. Consult with your administrator for details on your system.

Symphony provides a Call Director Control which indicates the current on/off-hook status and serves as a telephone PTT when the headset is not used. The term on-hook means the phone is inactive. When you hang up the phone, it is on-hook. The term off-hook means the phone is active. When you answer a call or receive dial tone to place a call, the phone is off-hook. Figure 6-1 shows the on/off hook states of the Call Director Control for the Full Screen and Baton. The Baton off-hook example includes a "TX" label. When this label is present, the CD Control becomes a button which is used to PTT to the phone. The "TX" is only present when a headset is not plugged in as described below. The telephone PTT function is available on the Full Screen and Baton.



Figure 6-1: Call Director Control

6.1.1 Local Call Director Operations

Connecting a phone to the Symphony Call Director feature is referred to as “local” since no radio units are involved. During local CD operations, normal dispatch operations can continue as normal. For example, if a module is selected, selected transmits may be performed via the Select Module Smart Button or the footswitch’s PTT switch.

During local CD operations, monitoring of the incoming CD telephone audio and incoming audio from programmed entities can occur. Also, transmissions to programmed entities and speaking to the CD telephone callee/caller can be performed separately.

6.1.1.1 Using the CD Volume Slider

The volume of the audio from the Call Director telephone to the select speaker/headset can be controlled using the CD Volume Slider control. This control is displayed adjacent to the Call Director Control and is enabled when Call Director is off hook.

To adjust the volume:

1. Click on the Volume Slider icon to display the Volume Slider popup.
2. Adjust the volume by moving the slider as required. The changes in the volume level can be heard on the connected headset or speaker while the slider is being moved.

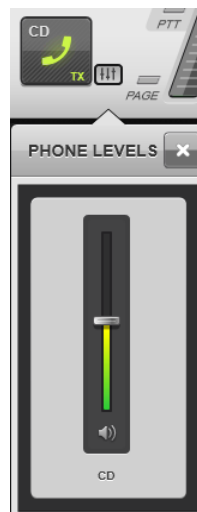


Figure 6-2: Call Director Volume Slider Control

6.1.1.2 Placing a CD Telephone Call

6.1.1.2.1 Using the Desk Microphone and Speaker

The following procedure details the operations required to place a telephone call with the Call Director, but *without* a headset at the console. Normal dispatch operations can be performed at any time during this procedure.

1. Take the phone off-hook by lifting the handset or pressing the headset button. A dial-tone sounds in the select speaker and the CD Control indicates off-hook. Incoming select audio will also be heard in the select speaker.
2. At the phone, dial the desired telephone number. Tone/Pulse digits sound in the select speaker as they are dialed. When the called telephone begins ringing, the rings sound in the select speaker along with incoming select audio, if any.
3. When the telephone user answers, their voice is heard in the select speaker along with incoming select audio, if any. Adjust the speaker's volume control as necessary. Module volume adjustments along with other normal dispatch operations may be performed at any time during this procedure.
4. To speak to the telephone user, PTT via the CD Control button. Speak into the open mic during the PTT.
5. Release CD Control to listen to the reply. Incoming audio from the CD telephone is heard in the select speaker along with incoming select audio, if any.
6. Re-PTT the console and monitor replies as necessary.
7. To end the telephone call, hang up the phone. The CD Control indicates the on-hook condition.

6.1.1.2.2 Using the Headset

The following procedure details operations required to place a telephone call with the Call Director while using a headset at the console. No console PTT action is necessary when speaking to the CD telephone callee since CD operation is full-duplex. When the CD is off-hook and a headset is plugged-in, any incoming select audio (for example, on a selected trunked talkgroup) is heard in the select speaker instead of the headset. Select audio returns to the headset as soon as the CD telephone is in the on-hook condition. Normal dispatch operations can be performed at any time during this procedure:

1. Take the phone off-hook. A dial-tone sounds in the headset and the CD Control indicates the off-hook condition.
2. At the phone, dial the desired telephone number. Tone/Pulse digits sound in the headset as they are dialed. When the called telephone begins ringing, the rings sound in the headset.
3. When the telephone user answers, their voice is heard in the headset. Adjust the headset's volume control as necessary. Normal dispatch operations can be performed at any time during this procedure.
4. No PTT action is required to speak to the telephone user; speak into the headset's microphone. Incoming audio is heard in the headset. Incoming select audio is heard in the select speaker.
5. To end the telephone call, hang up the phone. The CD Control indicates the on-hook condition. Also, select audio returns to the headset.

If, during the above procedure, a select or unselect (“instant”) transmission is performed, console headset microphone audio temporarily switches from the CD telephone to the selected/unselected entity. In other words, the CD telephone user does not hear the dispatcher speak during a select or instant console PTT.


6.1.2 Hang up a CD Telephone Call

To hang-up (end) a Call Director telephone call, hang up the phone. The CD Control will indicate the on-hook condition.

6.1.3 Answering a CD Telephone Call

When the phone rings, answer the call on the phone. This is usually done by pressing a headset button on the phone. The CD Control will show the off-hook condition and the caller will be heard in the select speaker or the headset if the headset is plugged in. Manage the call as described above depending on whether you have the headset plugged in or not.

6.1.4 TelePatch a CD Telephone Call

When the Call Director telephone call is established, the operator can patch the phone call to a Talkgroup, radio Patch, or a Base Station by using the  button on the Telephony Sidebar Panel Active Tab. Pressing the More button opens up the Call Director TelePatch panel. From the panel, the call can be TelePatched to the current Select entity by selecting “TelePatch Select” or the TelePatch dialog can be opened by selecting “TelePatch.”

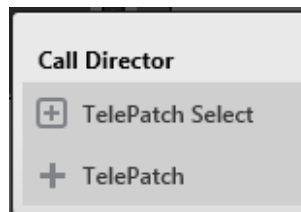


Figure 6-3: Call Directory TelePatch

When selecting “TelePatch Select,” the outside phone will be patched to the current Select entity. The Smart button in the Select bar will now show the TelePatch cross icon as well as the active call radiating waves.

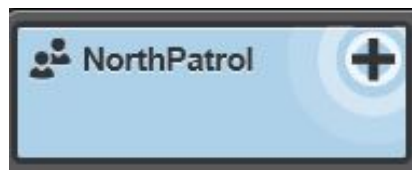


Figure 6-4: Module in TelePatch

From the TelePatch Dialog, the desired entity to patch the phone call to can be selected. Once selected, the **Patch now** button will be enabled if the entity is available to patch. Pressing the **Patch now** button will create a connection from the outside phone line to the patched entity and the dispatcher operator. If the **Patch now** button is grayed out, then the selected entity cannot be TelePatched. Hover the mouse pointer over the grayed-out **Patch now** button to display the possible reasons why the entity cannot be patched.



Figure 6-5: TelePatch Dialog

6.2 SIP TELEPHONY OPERATION

6.2.1 Overview



NOTE

Since SIP Telephony is a feature-licensed option, this feature is only available if the console is equipped with this software license. In addition, various system and console-related configuration changes must be made before this feature is operational. Consult with system and/or dispatch administration personnel as necessary.

The Telephony Call feature of the Symphony console provides the capability to perform telephony functions such as making and receiving calls to phones within the organization and also to phones in the public switched Telephone network (PSTN). The Symphony console registers with a local Private Branch Exchange (PBX) and uses the local PBX to make and receive telephone calls.

The Symphony Console provides the ability to have multiple telephone lines and also provides call controls and displays to manage telephone calls. Basic Telephony features can be accessed from the console such as Dial, End Call, Hold, Transfer, Forward, Voicemail, etc. The Symphony Console also provides quick access to Phonebooks and Speed Dial Lists to assist in making telephone calls.

SIP Telephony is managed through a telephony panel and the compact telephony control. The former provides access to all SIP telephony functions while the latter focuses on managing active calls. The compact telephony control requires limited screen real estate, but handles the typical call taking controls.

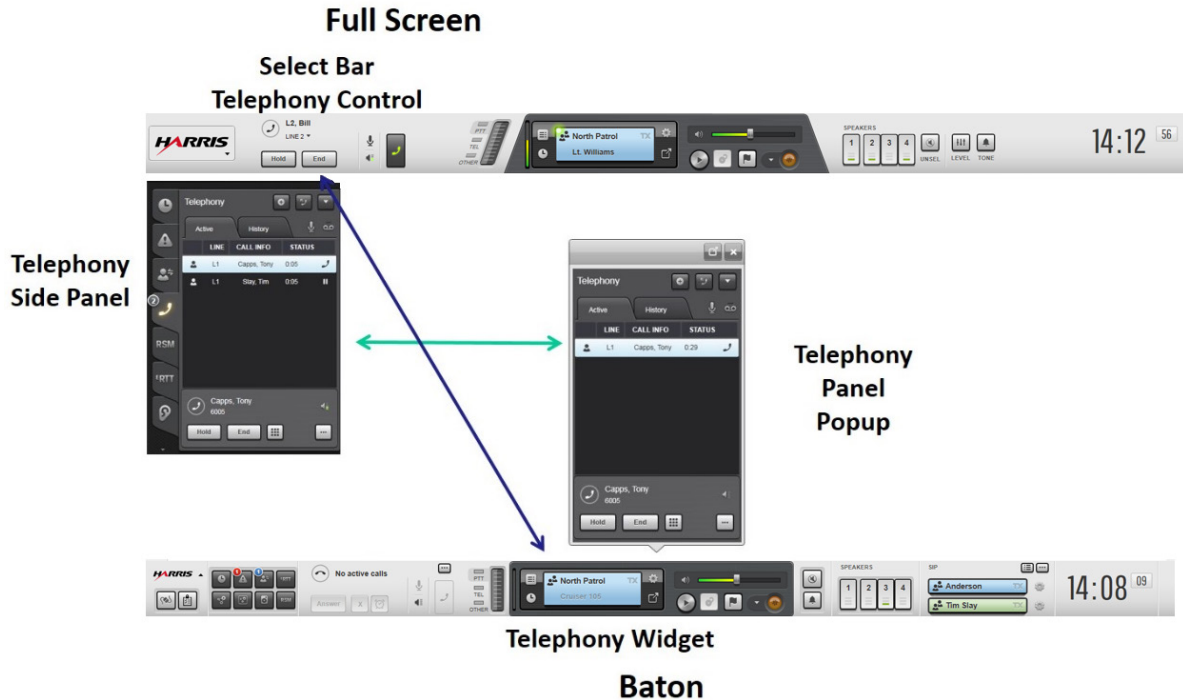


Figure 6-6: SIP Telephony Full Screen Display vs Baton

Figure 6-6 shows how these controls are used in the Full Screen Display and the Baton. In the Full Screen Display, the compact telephony control is placed on the Select Bar and is referred to as the Select Bar Telephony Control. The Telephony Sidebar Panel is a typical Sidebar Panel and is referred to as the Telephony Sidebar Panel. The Baton includes the compact telephony control on the main toolbar and provides a button to pop out the telephony panel when it is needed. The compact telephony control is known as the Telephony Widget while the telephony panel is known as the Telephony Panel Popup.

The following conventions are used in the subsequent step by step instructions:



- Only one version of the Telephony panel is shown since the Telephony panel is the same for the Full Screen Display and Baton, just packaged differently.
- Depending on the number of lines configured, some controls like the Line Selector and Message Waiting Indicator will behave differently. The following scenarios assume multiple lines have been configured unless otherwise stated. Refer to section 3.1 and 3.4.5 for details on these controls.

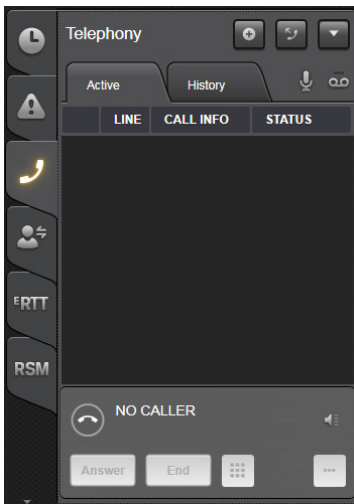
6.2.2 Performing an Outgoing Telephony Call

There are numerous ways of performing an outgoing Telephone call from the Symphony Console. These are described in the following sections.

6.2.2.1 Using the Dialpad


Telephone calls can be made to another telephone party by manually dialing the telephone number of the other party using a Dial Pad. Use the following method to initiate a telephone call using the Dial Pad:

1. Select the Telephony Panel by clicking the Telephony Sidebar Panel Tab  when using the Symphony Full Screen Application or by clicking the More Button  on the Baton Telephony Widget when using the Baton.



2. Click the New Call button .



3. If not selected, click the Dial pad button .
4. Enter the desired phone number with a mouse and the numeric keypad on the screen or enter the number via the keyboard.

- Specify the desired line by clicking the Line Selector button  and choosing the desired line.






- Click the **Dial** button.

The new telephone call appears on the Active Call List.

6.2.2.2 Using the Contacts List


Telephone calls can be initiated by selecting a contact from a preconfigured contact list. The Symphony contact list provides a quick way to call to another telephone without remembering the actual phone number. Contact list entries are populated using the Symphony Configuration Utility or added locally using the Directory dialog.

Use the following method to initiate a telephone call using the Contact List:

- Open the Telephony Panel by clicking the Telephony Sidebar Panel Tab  when using the Symphony Full Screen Application or by clicking the More Button  on the Baton Telephony Widget when using the Baton.
- Click the New Call button .






- If not selected, click the Contacts List button .

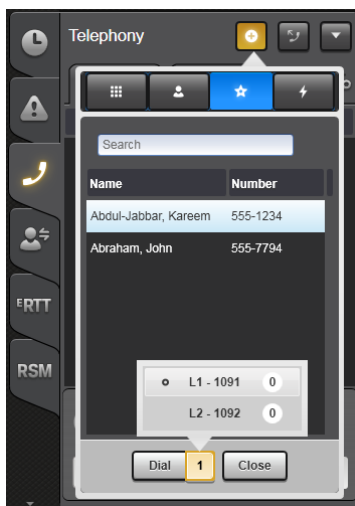
4. Specify the desired line by clicking the Line Selector button  and choosing the desired line.
5. Highlight the desired contact in the Contacts List and Press the **Dial** button or double-click the desired contact.



The new telephone call appears on the Active Call List.

6.2.2.3 Using the Favorites List

Telephone calls can be initiated by selecting a contact from a configured Favorites List. The Symphony Favorites List is a special type of Speed Dial List which is displayed as a separate list to standard Speed Dial List. Use the following method to initiate a telephone call using the Favorites List.

1. Select the Telephony Panel by clicking the Telephony Sidebar Panel Tab  when using the Symphony Full Screen Application or by clicking the More Button  on the Baton Telephony Widget when using the Baton.
2. Select the New Call button .





3. If not selected, click the Favorites List button .
4. Specify the desired line by clicking the Line Selector button  and choosing the desired line.
5. Highlight the desired contact in the Favorites List and press the **Dial** button or double-click the desired contact.

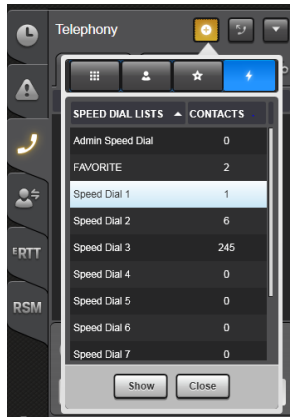
The new telephone call appears on the Active Call List.



6.2.2.4 Using the Speed Dial List

Telephone calls can be initiated by choosing a contact from a preconfigured Speed Dial List. The Symphony Speed Dial List displays a subset of the contact list grouped into smaller groups. Use the following method to initiate a telephone call using the Speed Dial List:

1. Open the Telephony Panel by clicking the Telephony Sidebar Panel Tab  when using the Symphony Full Screen Application or by clicking the More Button  on the Baton Telephony Widget when using the Baton.

- Click the New Call button .





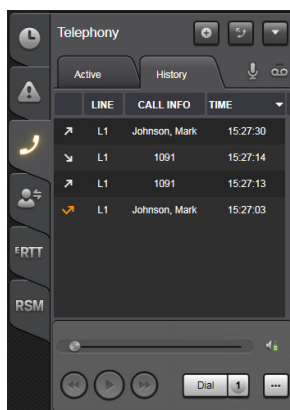
- If not selected, click the Speed Dial List button .
- Highlight a Speed Dial List and click the **Show** button. This reveals the contact list for the selected Speed Dial List.
- Specify the desired line by clicking the Line Selector button  and choosing the desired line.
- Highlight the desired contact in the Speed Dial List and Press the **Dial** button or double-click the desired contact.


The new telephone call will appear on the Active Call List.

6.2.2.5 Using Call History

The Call history section displays all the previous incoming and outgoing telephony calls at the Symphony Console. Telephone calls can be initiated by selecting from an entry in the Call History List. Use the following method to initiate a telephone call from the Call history list:

- Open the Telephony Panel by clicking the Telephony Sidebar Panel Tab  when using the Symphony Full Screen Application or by clicking the More Button  on the Baton Telephony Widget when using the Baton.
- Click the History Tab.



- Specify the desired line by clicking the Line Selector button  and choosing the desired line.
- Highlight an entry in the Call History List and Select Dial.

The new telephone call appears in the Active Call List and is also shown as a new call on the History List.

6.2.2.6 Using Speed Dial Hotkeys

Hotkeys can be used to initiate outgoing calls. Hotkeys are programmed in the Symphony Configuration Utility. The Hotkeys (Alt + 0-9 Number Pad) are assigned to a specific phone number.

To quickly perform a call to a desired contact, press the contact's corresponding hotkey (Alt + 0-9 Number Pad).

The new telephone call appears on the Active Call List.

6.2.2.7 Using the Contacts Directory

The Contacts Directory displays all the preconfigured Telephony Contacts.


1. Telephone calls can be initiated by selecting from an entry in the Contacts Directory.
2. Choose the appropriate number (Work, Mobile, Home, or Other).
3. Choose the desired line.
4. Click the **Dial** button.

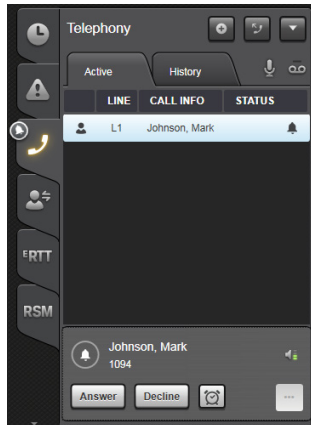
The screenshot shows the 'DIRECTORY' application window. The main area contains a table of contacts with columns: LAST, FIRST, WORK, MOBILE, HOME, OTHER, AFFILIATION, LOCAL, and HOTKEY. The contact 'Abdul-Jabbar, Kareem' is selected. To the right, an 'Info' panel displays details for this contact, including Work Number (555-1234), Mobile Number (555-4848), Home Number (876-1954), and Other Number (980-1234). At the bottom of the Info panel, there is a 'Number Type Selector' (Home dropdown), a 'Dial' button, and a 'Line Selector' (1). Three yellow arrows point from labels on the right to these three elements.

LAST	FIRST	WORK	MOBILE	HOME	OTHER	AFFILIATION	LOCAL	HOTKEY
Abdul-Jabbar	Kareem	555-1234	555-4848	876-1954	980-1234			
Abraham	John	555-7794	567-1593	595-2356	1114			
ACSHACK	UZQFIEU	585-2356	585-2356	585-2356	585-2356		*	
ACSXCZL	YSUQQDW	585-2356	585-2356	585-2356	585-2356			
ACYDQIT	JMTKUPK	239-0740	239-0740	239-0740	239-0740			
ADEEYF	WPRZGBG	867-6558	867-6558	867-6558	867-6558			
ADPKWYL	ZVCRADO	239-0740	239-0740	239-0740	239-0740		*	
AENGJJK	NXFSGO	585-2356	585-2356	585-2356	585-2356		*	
AEQCJPH	LAAYNXG	239-0740	239-0740	239-0740	239-0740		*	
AEVKHAC	JBRMBUA	239-0740	239-0740	239-0740	239-0740		*	
AEYCAHU	AKLVITE	867-6558	867-6558	867-6558	867-6558		*	
AFFFLN	QNEQBP	867-6558	867-6558	867-6558	867-6558		*	
AGDDJSQ	DEKUPQW	867-6558	867-6558	867-6558	867-6558		*	
AGDXVRT	OEVKTYN	867-6558	867-6558	867-6558	867-6558		*	
AGETVRO	VVMUOCY	585-2356	585-2356	585-2356	585-2356		*	

6.2.3 Receiving an Incoming Telephony Call

When an incoming Telephone Call is received, there are multiple incoming call indications that are presented including:

- Audible Indication (Ring) presented on the Select Audio Output. The ringtone may be unique for the caller based on configuration. If a call is active (i.e., not on hold) the ring is a soft boop to indicate the new call without disturbing the current call audio.
- Visual Indication on the Sidebar Panel .
- Visual indication on the Telephony Sidebar Panel → Active Tab → Active Call List.



- Visual indication on the Select Bar Telephony Control Area.



- Visual indication on the Baton Telephony Widget.



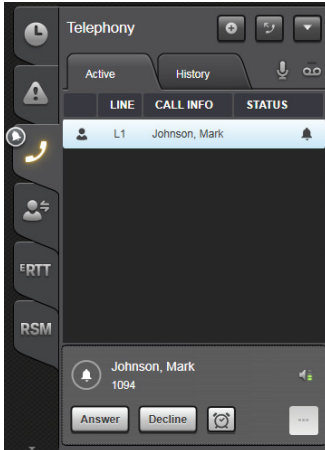
- Visual indication on the Baton → Telephony Widget → More → Active Tab → Active Call List.

The audible ring and visual indications are presented until the call has been cancelled by the calling party, has reached the incoming call timeout, the call has been manually answered, or the call has been declined.

6.2.3.1 Answering Incoming Calls

The following are several ways to answer an incoming telephone call:

- Using the Answer button on the **Telephony Sidebar Panel → Active Tab → Call Control Area**. To answer an incoming telephone call, highlight the incoming call from the Active Call List and click the **Answer** button on the Call Control Area of the Telephony Sidebar Panel.

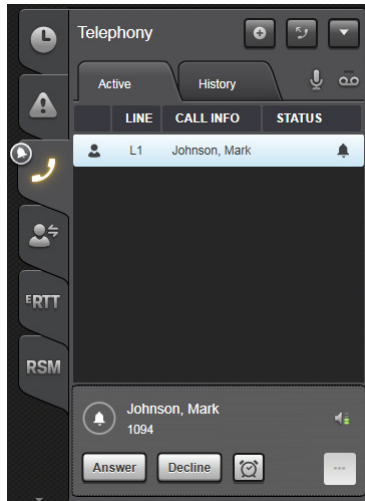



- Clicking the **Answer** Button on the Select Bar Telephony Control Area. If no calls are in the active call list, the new call appears in the Select Bar Telephony Control. Click the **Answer** button on the SBTC to answer the incoming call.
- Double-clicking the incoming call on the Telephony Sidebar Active Call List.
- Using the Answer hotkey (Alt + A). If there are multiple incoming calls, the Answer hotkey answers the earliest incoming call.
- Clicking the **Answer** button on the Baton Telephony widget. If no calls are in the active call list, the new call appears in the Baton Telephony Widget. Click the **Answer** button on the Baton Telephony widget to answer the incoming call.
- Clicking the **Answer** button on the **Baton → Telephony Widget → More → Active Tab → Call Control Area**. Highlight the incoming call from the Active Call List and click the **Answer** button on the Call Control Area.
- Double-clicking the incoming call on the **Baton → Telephony Widget → More → Active Tab → Active Call List**. When an incoming call entry in the active call list is double-clicked, this action answers the incoming call.

6.2.3.2 Declining Incoming Calls


Incoming telephony calls can be declined. When a call is declined, a busy message is sent back to the calling party. The following are ways to decline an incoming telephone call:

- Using the **Decline** button on the Telephony Sidebar Call Control Area. Highlight the unwanted call from the **Telephony Sidebar Panel** → **Active call list** and click the **Decline** button on the Call Control Area.




- Using the **Decline** button on the Select Bar Telephony Control Area. Click the Decline button, , on the SBTC to decline the incoming call.

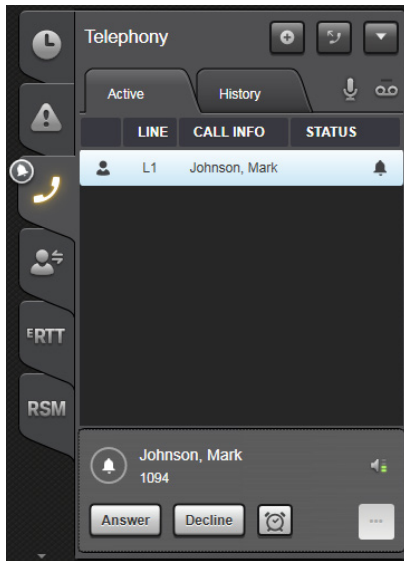



- Using the End hotkey (Alt + E). Press the End hotkey to decline the incoming call. If there are multiple incoming calls, the End hotkey declines the earliest incoming call.
- Using the Decline button on the Baton Telephony Widget. Click the Decline button  on the Baton Telephony Widget to decline the incoming call.
- Using the Decline button on the **Baton** → **Telephony Widget** → **More** → **Active Tab** → **Call Control Area**. Highlight the unwanted call from the Active call list and click the Decline button on the Call Control Area.

6.2.3.3 Activating Call Back Reminder



Incoming telephony calls can be declined and also have a call back reminder. The call back reminder duration is configured in the Symphony Configuration Utility. The following are ways to decline an incoming call and activate the Call Back Reminder:

- Using the Call Back Reminder Button on the Telephony Sidebar Panel. To decline an incoming telephone call and activate a call back reminder, highlight the unwanted call from the Active call list and press the Call Back Reminder button  on the Call Control Area of the Telephony Sidebar Panel.


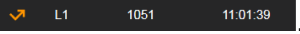


- Using the Call Back Reminder button on the Select Bar Telephony Control Area. To decline an incoming telephone call and activate a call back reminder, click the Call Back Reminder button  on the Select Bar Telephony Control Area.



- Using the Call Back Reminder Hotkey (Alt +B). Press the Call Back Reminder hotkey to decline the incoming call and activate a call back reminder. If there are multiple incoming calls, the Call back reminder hotkey declines the earliest incoming call.
- Using the Call Back Reminder Button on the Baton Telephony Widget. To decline an incoming telephone call and activate a call back reminder, click the Call Back Reminder button  on the Baton Telephony Widget.
- Using the Call Back Reminder Button on the **Baton → Telephony Widget → More → Active Tab → Call Control Area**. To decline an incoming telephone call and activate a call back reminder, highlight the unwanted call from the Active call list and click the Call Back Reminder button  on the Call Control Area.

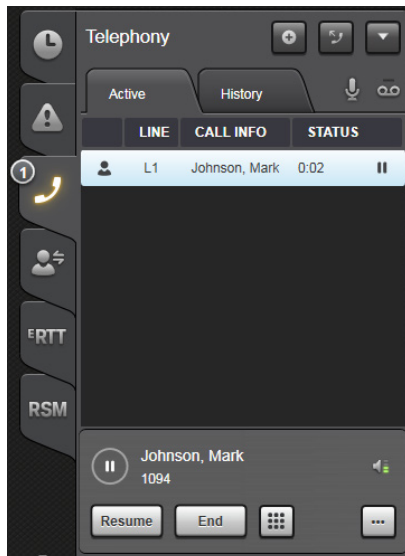
6.2.3.4 Missed Call

When an incoming Telephony Call is not answered, the telephone call is considered to be a “Missed Call” event. This is indicated by a number on the upper right corner of the Call History Tab  and also indicated as a missed call in the Call History List . The number indicated on the Call History Tab indicates how many missed calls have been received since the history list was last checked.

6.2.4 Placing Active Telephony Calls On/Off Hold

The Call Hold feature provides the ability to place active calls on Hold temporarily. When a telephone call is placed on hold, other actions such as consulting with other colleagues or making another telephone call on another line can be performed. Symphony automatically places the current call on hold when a new call is started or a different call is focused. Calls can be placed on/off hold in the following ways:

- Using the Hold/Resume Button on the Telephony Sidebar Panel. Select the call from the Active Call list and click the **Hold** or **Resume** button to put the call on and off hold, respectively.



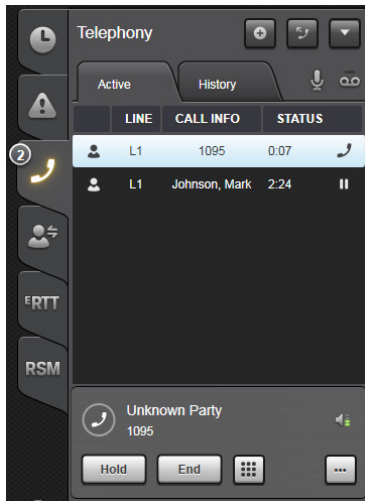
- Using the Hold/Resume Button on the Select Bar Telephony Control. Select the Active Call List Popup, highlight the call from the Active Call List, and click the **Hold** or **Resume** button to put the call on and off hold, respectively.



- Double-clicking the call on any the Active Call List. Using the mouse to double-click an entry in any Active Call List puts a call On/Off Hold.
- Using the **Hold/Resume** button on the Baton Call Control Area. Click the **Hold** or **Resume** button to put the highlighted call on and off hold, respectively.

6.2.5 Placing Multiple Telephony Calls

The Symphony console allows for multiple telephone calls to be established simultaneously, but only one call is active at one time, except when calls are conferenced or TelePatched. When an additional call is made while there is an active call, the previous active call is placed on Hold automatically. To make another call while an active call is already present, follow Section 6.2.2. All calls are listed in the Active Call List.




Symphony only allows one call to be active at a time. Other calls on the Active Call list are automatically placed on hold. Perform the following to switch among calls on the Active Call List.

1. Clicking the **Resume** button on a call that is currently on Hold.
2. Double-clicking on a Call on the Active Call list to resume the call.

6.2.6 Using Overdial on Active Telephony Calls

The Symphony console provides the ability to send Dual Tone Multi Frequency (DTMF) over an active telephone call. DTMF provides additional call control in some telephone calls such as navigating through Interactive Voice Response (IVR) menus in a voicemail system.

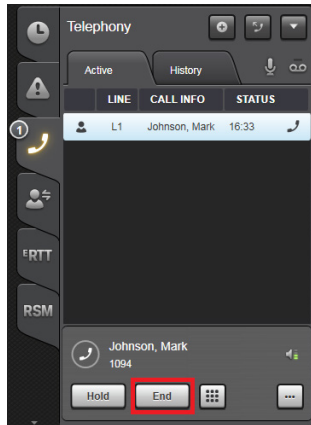
To send Overdial tones on an active telephone call, press the Overdial function button  on the Call Control Area, then use the numeric keypad. Local DTMF tones are generated to confirm each DTMF tone is being sent; however, Symphony does not play received DTMF tones.



6.2.7 Ending Active Telephony Calls

Active Telephone calls can be ended from the Console or by the remote end. When a call is ended by the remote end, the call automatically disappears from the Active List. This signifies that the call has already been ended. To terminate a call from the Symphony Console, select the call from the Active List and click the End Button. This can be done from the following:

- Telephony Sidebar Panel → Active Tab → Call Control Area.



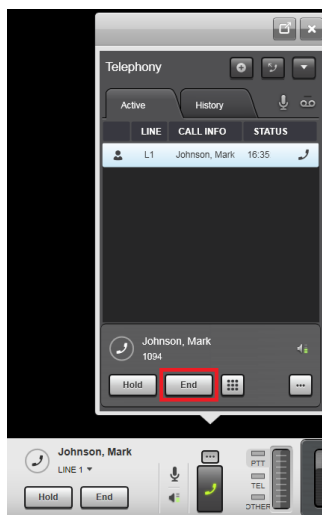
- Select Bar Telephony Control Area.



- Baton Telephony Widget



- Baton → Telephony Widget → More → Active Tab → Call Control Area



6.2.8 Performing Call Transfers

Active Calls can be transferred to another telephony party. Once the transfer is complete, the parties involved in the transfer are removed from the Active Call List. Symphony supports three kinds of call transfers: Blind, Consultative, and Attended transfer.

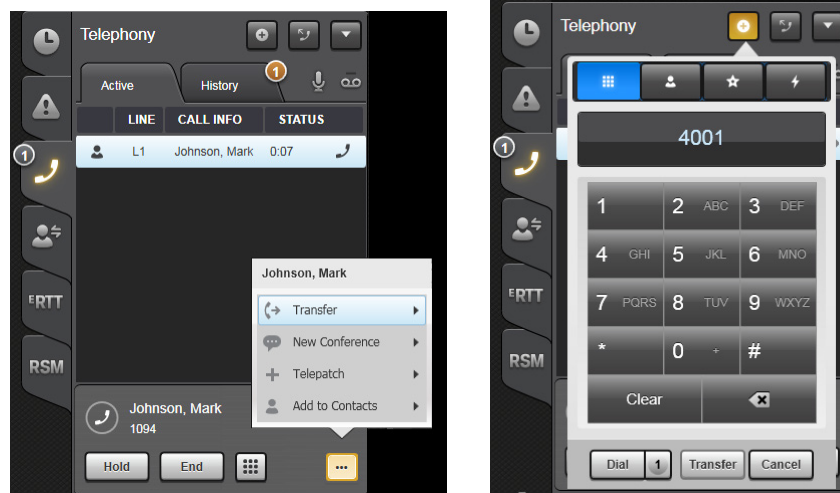
- **Blind:** In a blind transfer, the call is transferred directly to the target party without talking to the target party.
- **Consultative:** In a consultative transfer, the transfer target is first consulted to explain the transfer before the transfer is completed.
- **Attended.** In an attended transfer all parties enter a 3-way call before the transfer is completed.

The terms for call transfer are as follows:

- **Transferor** – This is the person making the transfer.
- **Transferee** – This is the person being transferred.
- **Transfer Target** – This is the person that to whom the transferee will be transferred.

6.2.8.1 Blind Transfer

Blind Transfer is a procedure to transfer a call to a different number without previously calling/consulting with the other party.



To perform a blind transfer:

1. Highlight the call on the Active Call List (Transferee).
2. Click **More Menu** → **Transfer** → **To Number** to manually dial the phone number of the Transfer Target.

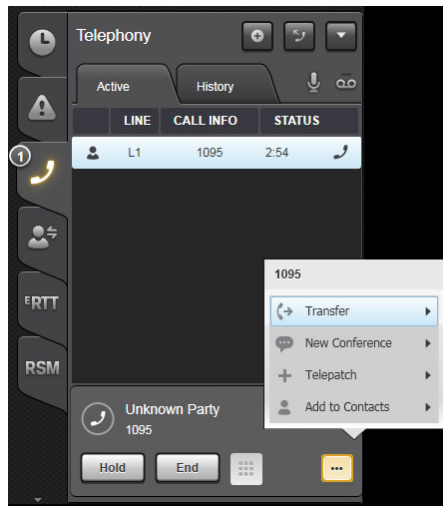
Or

Click **More Menu** → **Transfer** → **To Contact** to select a contact from the Contact List.

3. Dial the phone number or select the contact from one of the call lists (Transfer Target).
4. Click the **Transfer** button.

6.2.8.2 Consultative Transfer


Consultative Transfer is a procedure to transfer a call to a different telephone number but consulting the other party first if they would accept the call to be transferred to them.



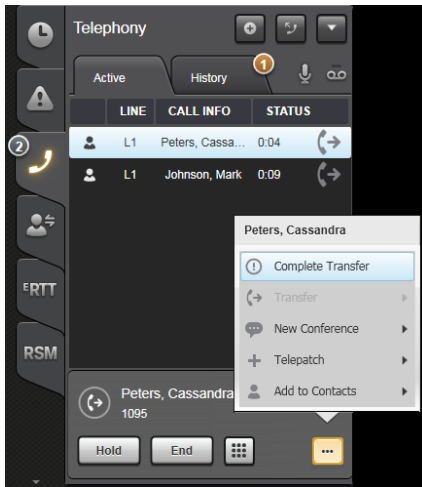
There are multiple ways of performing Consultative Transfer:

- Performing Consultative transfer from one call:
 1. Highlight the call on the Active Call List (Transferee).
 2. Click **More Menu** → **Transfer** → **To Number** to manually dial the phone number of the Transfer Target.

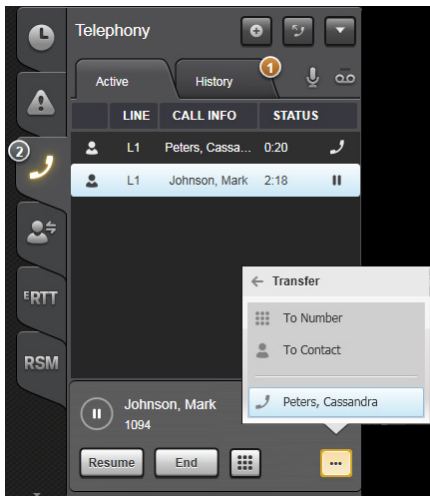
Or

 - Click **More Menu** → **Transfer** → **To Contact** to select a contact from the Contact List.
 3. Dial the phone number or select the contact from one of the Contact Lists.
 4. Specify the desired line by clicking the Line Selector button  and choosing the desired line.
 5. Click the **Dial** button.
 6. Consult with the other person on the second call (Transfer Target).

7. Click the **More Menu** → **Complete Transfer**.



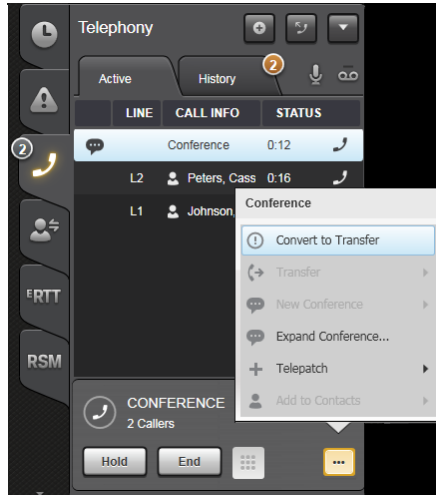
- Performing Consultative transfer between two calls.
 1. From having 2 or more calls on the Active Call List, Highlight one of the calls from the Active Call List (Transferee).
 2. Click the **More Menu** → **Active call** to whom to transfer the transferee (Transfer Target).



6.2.8.3 Attended Transfer

Attended Transfer is performing a transfer from a 3-way conference.

1. From an established 3-way conference call, highlight the Conference Call.
2. Select the **More Menu** → **Convert to Transfer**.



6.2.9 Configuring Call Forwarding

The symphony console provides the ability to automatically forward incoming calls to a preconfigured phone number. Call forward can be enabled on a per line basis or be applied to all SIP lines of the console. Perform the following to enable Call Forwarding.

1. Open the Phone Preferences Dialog:

Click the **Harris Logo** → **Preferences** → **Phone Preferences** or use the menu in the **Telephony Panel** → **Phone Preferences** to open the Phone Preferences dialog.



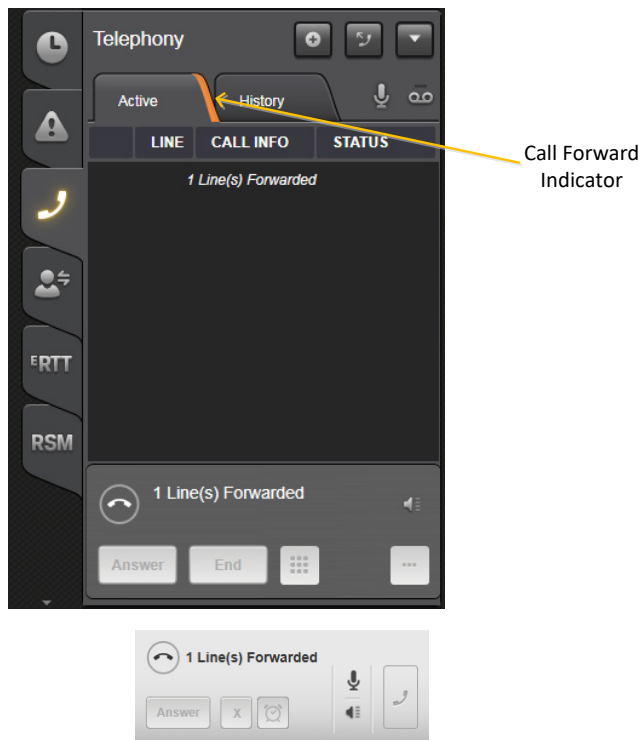
The Phone Preferences Dialog displays all configured SIP lines which has corresponding input boxes and checkboxes to configure and activate call forwarding function of the console.

2. There are two ways to enable Call Forwarding:
 - Call Forward on a per line basis:
 - a. Enter the desired phone number into the textbox corresponding to the SIP line you want to enable Call Forwarding.
 - b. Enable the checkbox corresponding to the SIP line you want to enable Call forwarding.
 - Call Forward on all SIP lines:
 - a. Enter the desired phone number into the textbox of "All Lines."
 - b. Enable the checkbox of "All Lines"



Call Forward all lines overrides the settings configured on a per line basis.

3. The Symphony console has clear indications that Call forwarding is currently active.



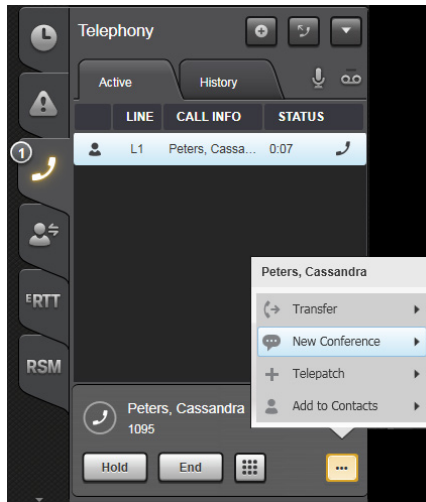
6.2.10 N-Way Conference

Symphony console allows the hosting of an N-way conference. The N-way conference is an Ad-hoc conference that is established and controlled from a Symphony Console which combines multiple telephony calls together into one conference. Symphony supports one conference with a maximum of 6 members at a time.

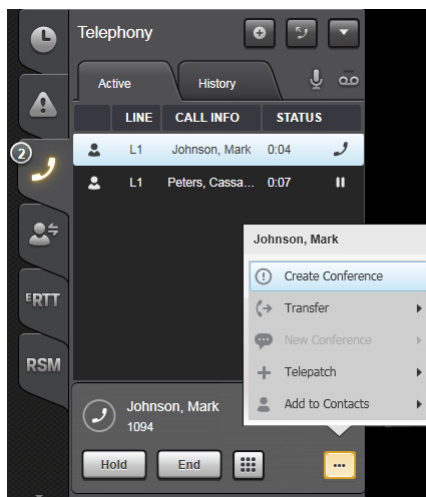
6.2.10.1 Creating Three-way Conferences

There are two ways to establish a Three-way conference from the symphony console.

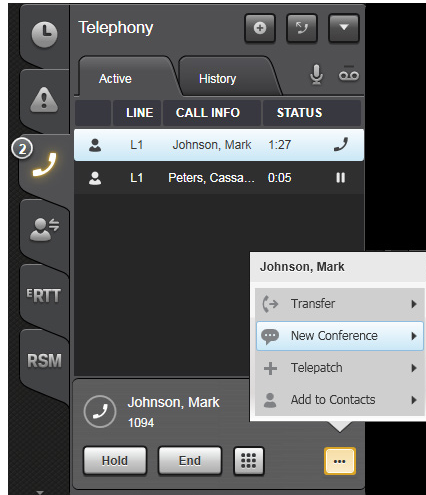
- Three-way conference from a single call:
 1. Highlight one of the conference candidates from the Active Call List.
 2. Select the **More Menu** → **New Conference**.



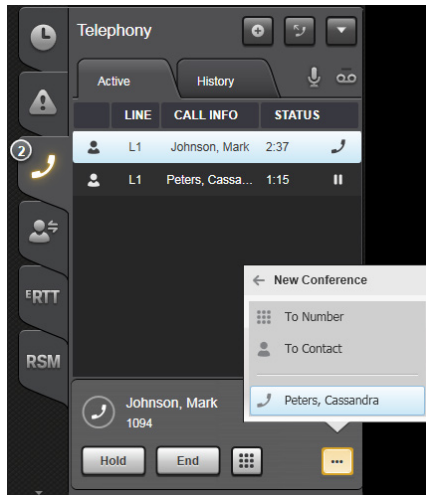
3. Select **To Number** to dial a number manually or select **To Contact** to select from the Contact list.
4. Dial the phone number or select the contact from one of the call lists.
5. Select Dial.
6. Click the **More Menu** → **Create Conference**.



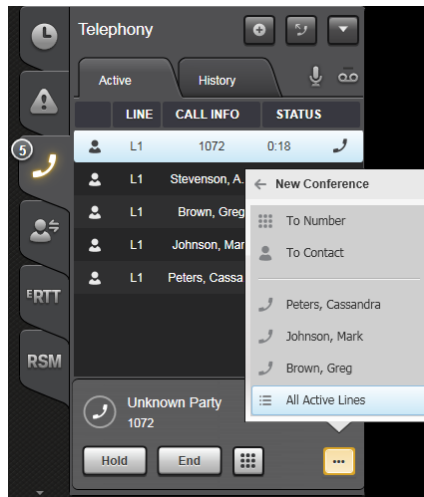
- Three-way conference from Two Calls:
 1. From having two or more calls on the Active call list, highlight one of the calls from the Active Call List.
 2. Click the **More Menu** → **New Conference**.



3. The next menu offers the option to Conference to the other active lines. If there are less than four conference targets, the target lines are listed in the More menu as shown below. Otherwise, the menu offers the option All Active Lines.



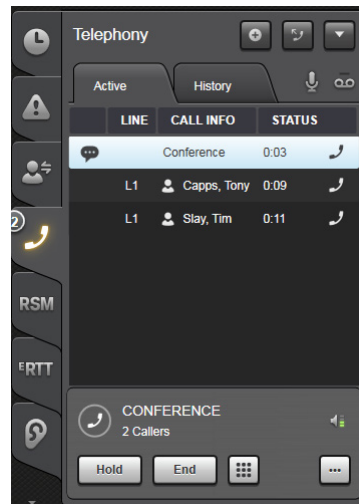
- If there are less than four targets, select the target from the menu and the console completes the conference. Otherwise, chooses the option **All Active Lines**. A line selection panel pops up.



- Select the other call to be part of the three-way conference from the list, then click **Done**.



The conference is created and is shown in the Active Call List. The conference is listed on one line and the members are listed below the conference.

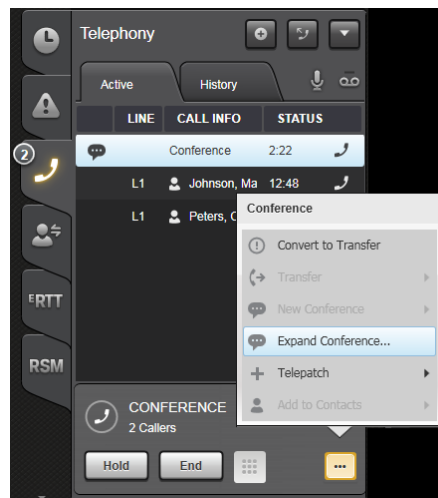


6.2.10.2 Creating N-Way Conference

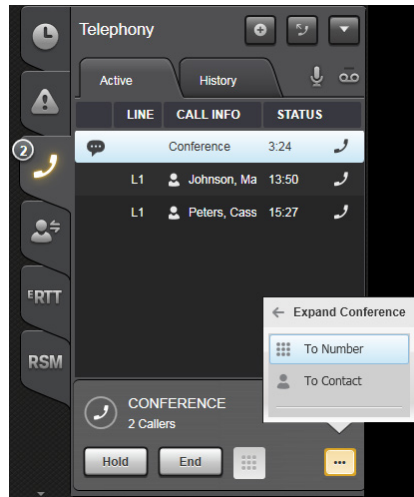
An N-Way conference is an expanded three-way conference. Additional members can be added to an already existing three-way conference.

There are multiple ways to expand an existing three-Way conference to an N-way Conference.

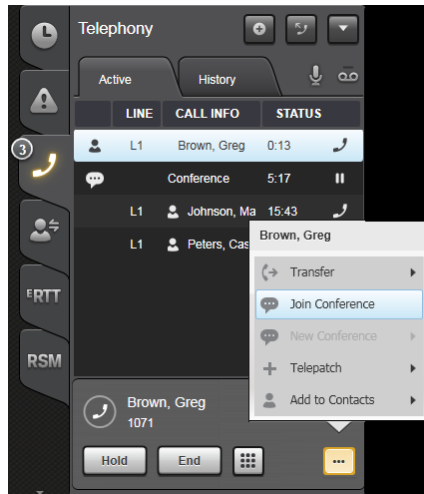
- Using Expand Conference to make a new call to the intended member:
 1. Highlight the existing three-way Conference.
 2. Click the **More Menu** → **Expand Conference**.



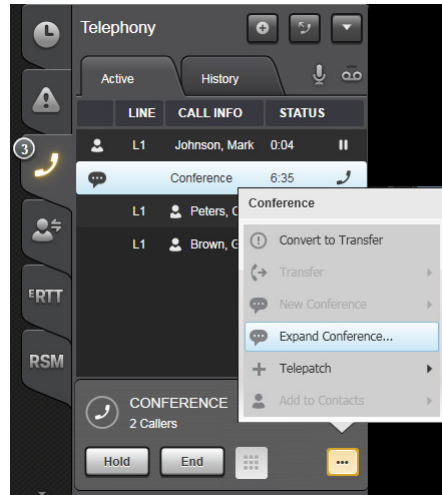
3. Select **To Number** or **To Contact** from the popup dialog to place a new call.



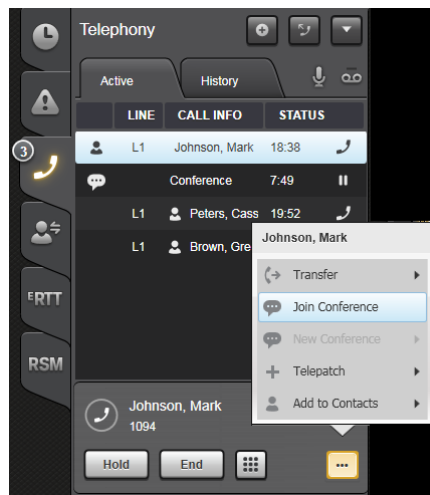
4. Perform a new call by using the Dialpad or by selecting a contact from the Contact List. The new call is then presented on the Active Call List as a new Call.
5. Highlight the new call on the Active call list and click the **More Menu** → **Select Join Conference** to add the highlighted call to the existing three-way conference.



- Using Expand Conference to select an already active call:
 1. Highlight the existing three-way Conference.
 2. Select the **More Menu** → **Expand Conference**.

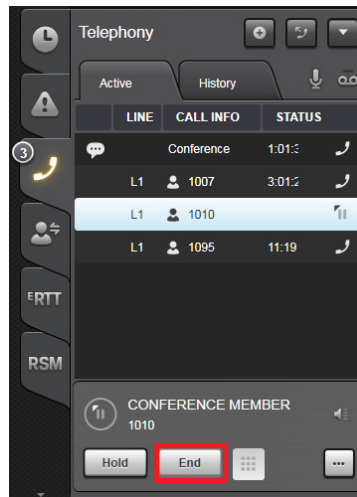


3. The next menu offers the option to add other active calls as a new member to the existing Conference. If there are less than four conference targets, the target lines are listed in the More menu as shown below. Otherwise, the menu offers the option All Active Lines.
 4. If there are less than four targets, the target can be selected from the menu and the console completes the conference. Otherwise, choose the option **All Active Lines** similar to the procedure on creating three-way conferences.
- Using Join Conference:
 1. While having an existing Conference, highlight an existing call on the Active call list.
 2. Click the **More Menu** → **Join Conference**.



6.2.10.3 Removing Members of an N-Way Conference

To remove a member from the N-Way conference, highlight the member that is part of the N-Way conference from the Active Call List, and click **End** in the Call Control Area.



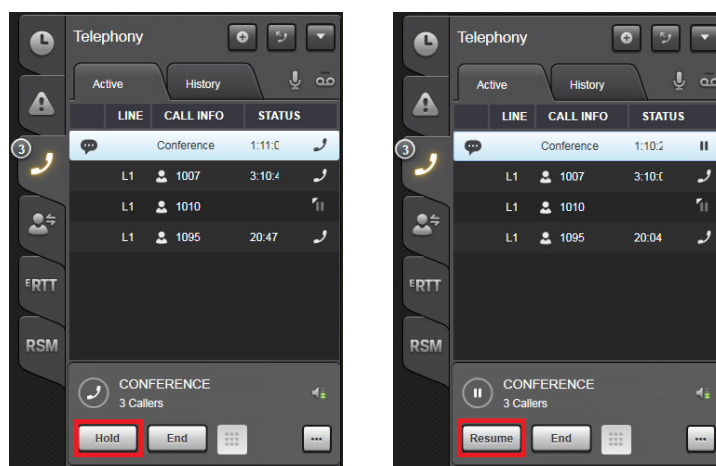
When a member is removed from the Conference, the call is not permanently ended. It remains in the Active Call List but is automatically placed on hold. The removed party from the conference can then be resumed or ended. When there is only one member left in a conference, i.e., all other members have been removed or have ended the call, the Conference is torn down automatically.

6.2.10.4 Putting the N-Way Conference on Hold

An N-Way Conference can be placed on Hold. This means that receive or transmit audio to and from the Conference is not heard at the console. All other members of the conference are still able to communicate amongst themselves.

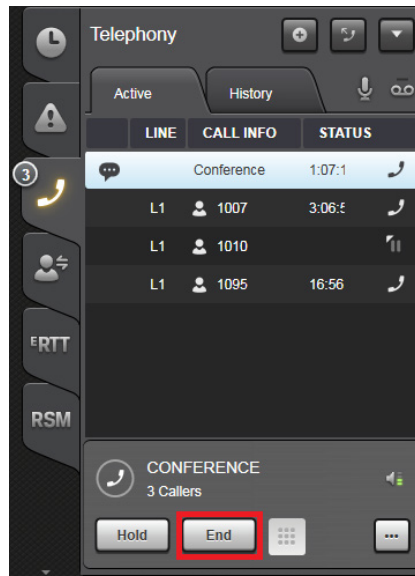
To Place an N-Way Conference on or off hold, highlight the Conference in the Active Call List and click the **Hold** or **Resume** buttons respectively in the Call Control Area. The **Hold/Resume** button that is presented depends on the state of the Conference.

Double-clicking the Conference in the Active Call list also switches between putting the Conference on Hold and Off Hold.



6.2.10.5 Ending an N-Way Conference

To end an N-Way Conference, highlight the Conference from the Active Call List and click the **End** button in the Call Control Area.



All Conference Members are automatically placed on hold when a conference is ended.

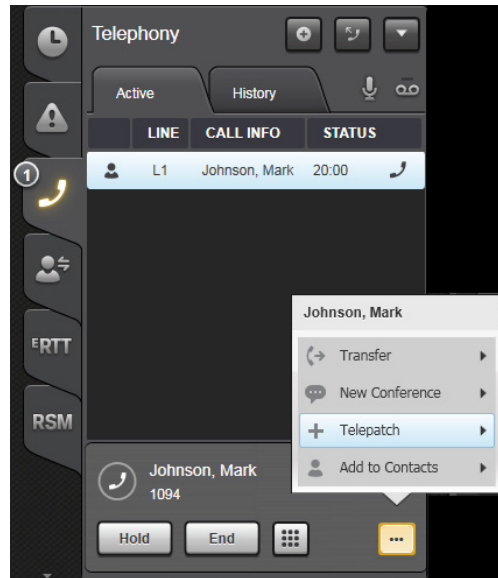
6.2.11 TelePatch

Symphony provides the ability to connect Telephony Calls to a “radio” entity, thus allowing a party on the phone system to communicate directly with radio end users in the field. This feature is known as TelePatch. The types of entities which may be TelePatched to the phone in this manner are Talkgroups, Base Station channels, and active radio Patches. The entity being TelePatched to the phone may be selected or unselected. During TelePatch operations, normal dispatch operations can continue as usual. For example, while TelePatched to an unselect entity, if a module for a different entity is selected, transmits can be performed via the Select Module Smart Button, the footswitch, or the desk mic’s PTT switch while communication with the TelePatch is accomplished using the TelPTT button in the Select Bar. See Section 4.11.2 for details on audio routing with TelePatch.

A TelePatch can be established by performing one of the following procedures.

6.2.11.1 TelePatch a Single Party SIP Call

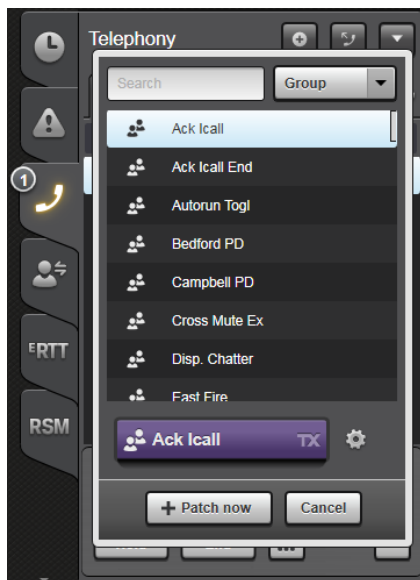
1. Highlight the call to be TelePatched from the Active Call List.
2. Click the **More Menu** → **TelePatch**.



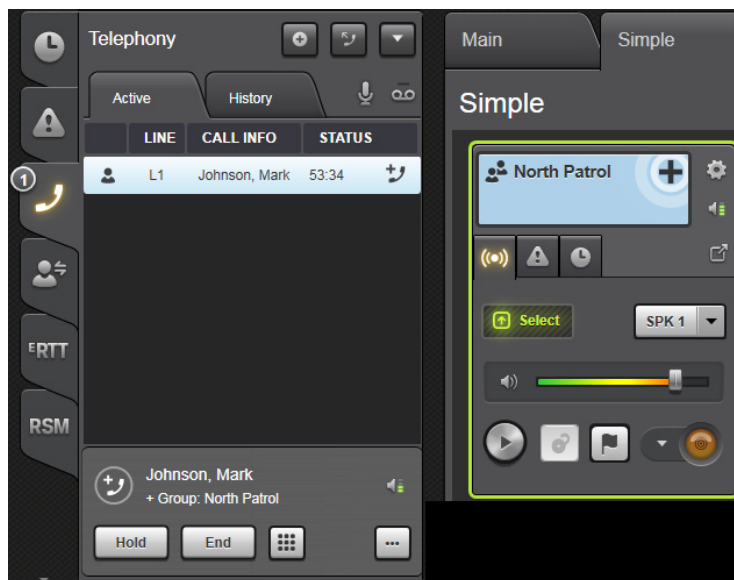
3. The next menu offers the option to perform a TelePatch to the current Selected Radio or to launch the TelePatch Dialog to select which Radio or Talkgroup is to be TelePatched to the Telephony Call.
4. If the TelePatch Select is selected, the icon of the Call on the Active Call List displays the TelePatch icon and the Select Radio or Talkgroup goes into Transmit mode. The Select entity must be inactive (i.e., not transmitting or receiving) to initiate the TelePatch. If the Select entity is active, the TelePatch fails and an error message appears in the CCA. Wait for the Select entity to be inactive and try again.



- If the TelePatch menu item is chosen, the TelePatch dialog is presented. The dialog shows all entities regardless of their “patch-ability.”



- The filter is set to Group by default. The filter can be changed to select the desired filter. Filter options include: Group, Patch, and Station.
- The mini module populates with the chosen entity. The aspects of the entity can be changed, such as encryption state or select, using the mini module context menu. The mini module dynamically shows the state of the module. If the mini module cannot be placed in a TelePatch, the **Patch Now** button is grayed out. The **Patch Now** button dynamically shows whether or not the TelePatch can be initiated.
- If the **Patch Now** button is not grayed out, press **Patch Now** to close the dialog and create the TelePatch.
- The console creates the TelePatch and updates the Active Tab. The TelePatched phone shows the TelePatch icon in the ACL and the CCA is updated to show the TelePatched entity as shown below. All legs of the call will be taken off hold and unmuted.

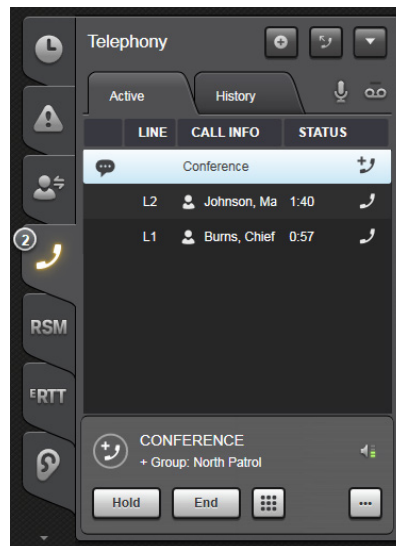


10. If the TelePatch fails within the system, the second line in the CCA contains the error icon and the message “TelePatch Failed” for 5 seconds. The error is added to the system history with more detail. After the timeout, the CCA returns to the appropriate state for the highlighted call. Highlighting another call in the ACL, clears the timeout and error info in the CCA.
11. During the TelePatch call, the volume control in the CCA only controls the volume and mute of the phone call leg of the TelePatch. The radio portion can be controlled from the radio module or from the Select Bar Telephony Control.
12. Once the TelePatch is established, switching to another call in the ACL places the TelePatch on hold. This state is referred to as unfocused with respect to the TelePatch. The TelePatched phone and radio entity continue to communicate. See Section 4.12.2 for details.

6.2.11.2 TelePatch a Conference

TelePatching a conference is similar to the procedure of TelePatching a single call.

1. Highlight the existing Conference.
2. Click the **More Menu** → **TelePatch**.
3. Once the TelePatched Conference is created, all legs of the call are taken off hold and unmuted. The Conference line in the ACL changes to the TelePatch icon and the CCA change as shown below:



4. While the conference is TelePatched, the More menu item **Convert to Transfer** is not available. The TelePatch must be ended first.
5. Parties can be added to the conference as usual.
6. To end the Conference and TelePatch, highlight the conference line and click the **End** button. This also means that if you end a TelePatched conference, the TelePatch is ended as well.
7. To end the TelePatch, use the More menu as usual. Ending the TelePatch does not end the conference. The conference reverts to its normal GUI treatment in the ACL and CCA.

8. To remove a party from the conference, highlight the appropriate party and press **End**. Removing a party from the conference does not end the TelePatch. If the conference is still active, the conference remains TelePatched. If the conference reverts to a regular single party call, then the single call remains TelePatched. The rules in this step also apply if a party leaves the conference.

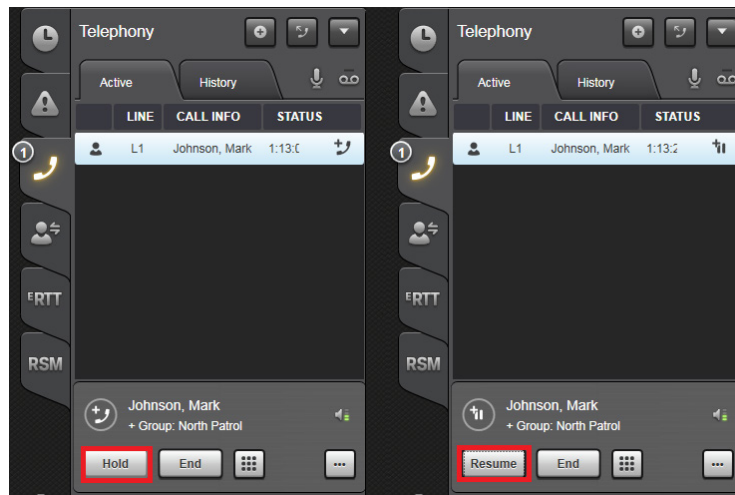
6.2.11.3 Conference a TelePatch

A TelePatch can be conferenced with other active telephony calls. The TelePatched line is treated like a regular line for the purposes of conferencing. This scenario assumes the line to conference is already active in the ACL.

1. Highlight the existing TelePatch.
2. Click the **More Menu** → **New Conference** and choose the desired line to conference.
3. The conference is created in the TelePatched state.

6.2.11.4 Placing TelePatch on and off Hold

A TelePatch can be placed On Hold or Off Hold. This is also known as Focused and Unfocused, respectively. This is similar to how a normal conference behaves when a conference is placed On hold and Off hold, i.e., when a TelePatch is On Hold, TelePatch audio is not heard and microphone audio is not transmitted to the TelePatch. The other members involved in the TelePatch continue to communicate with each other.



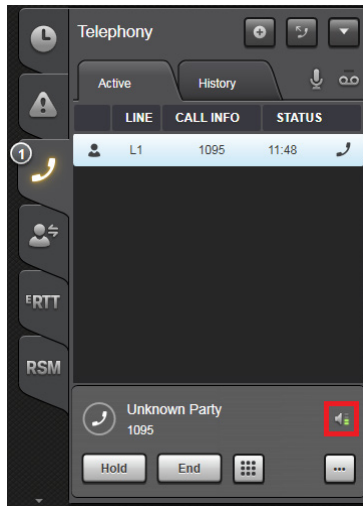
6.2.12 Audio Control

Audio control on telephony calls such as received volume control and mute microphone control can be performed.

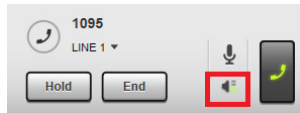
6.2.12.1 Volume Control

Symphony console provides individual receive volume control for each individual Telephony call. Controls for receive volume are located in the following sections.

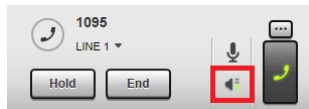
- Call Control Area




- Select Bar Telephony Control Area



- Baton Telephony Widget



Select the volume icon  to display the volume control slider, which controls receive audio level of the currently highlighted call. Use the volume control slider to increase or decrease the volume.



6.2.12.2 Mute Receive Audio

Telephony calls can be muted completely. When a telephony call is muted, receive audio from the telephony call is not heard. To mute a telephony call, highlight the call to be muted on the Active Call List, then press the speaker icon on the volume control slider. This mutes/unmutes the receive audio of a call.






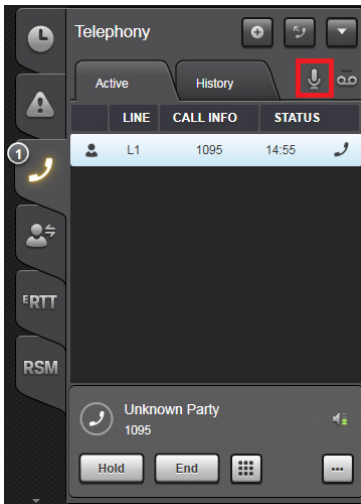
NOTE

Clicking the speaker icon on the volume control slider is different from placing the call on HOLD. Completely muting the receive audio does not automatically mean that the Microphone is muted. The microphone may still be live on the call.

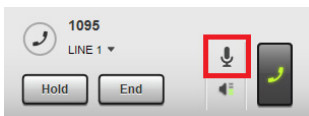
6.2.12.3 Mute Microphone


The microphone can be completely muted. When the microphone is muted, the receive audio on the telephony call can still be heard, but the microphone is not heard by the person on the other end of the call. This is the typical mute function you would expect on a desk phone. The following are the ways to mute the microphone on the currently active call:

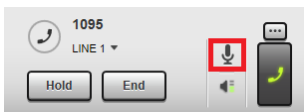
- Clicking the Mic icon  on the Telephony Sidebar Panel.








- Clicking the Mic icon  on the SBTC.



- Clicking the Mic icon  on the Baton Telephony Widget.



When the microphone is muted, the following indications are presented on the console:

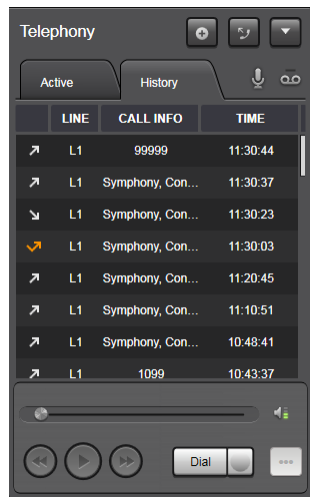
- A microphone symbol with a slash  is displayed on the Telephony Sidebar Panel.
- A microphone symbol with a Red Slash  is displayed on the call on the Active Call List.
- A microphone symbol with a Red Slash  is displayed on the Call Control Area.
- A microphone symbol with a Red Circle with a slash  is displayed on the Select Bar Call Control Area.
- A microphone symbol with a Red Circle with a slash  is displayed on the Baton Call Control Area.

To unmute the microphone, click the Mic icon again. The microphone mute indication is no longer displayed. The microphone is automatically unmuted if the current highlighted call is placed on hold, a new call is highlighted, or a new call is started.

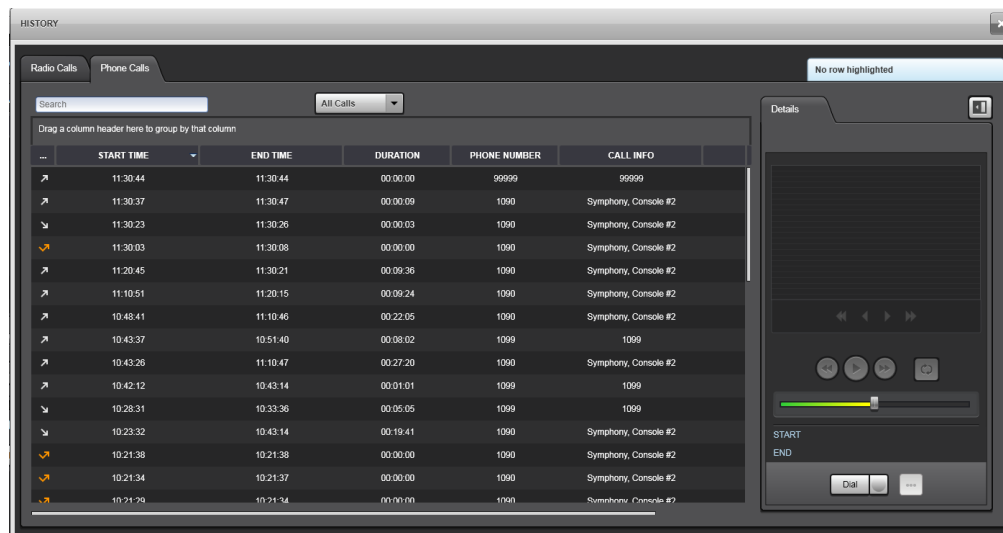
6.2.13 Call History

There are two places to find the Telephone Call History of a Symphony Console:

- **Telephony Sidebar Panel History Tab:** The History Tab on the Telephony Sidebar Panel shows a summarized version of the call history.



- **Extended History Dialog:** The Extended History Dialog shows complete call information on all calls performed and received by the Symphony Console. Click the **Harris Logo** → **Open History** → **Phone Calls** to display the Extended History Dialog.



See Section 3.5.2.2 for more details

6.2.14 SIP Call Recording and Playback

See Section 5.6 for detailed instructions to play back SIP telephony call recordings.

6.2.15 Managing Contacts List

The local contact list can be managed locally using the Symphony Application if enabled by your Systems Administrator. Administrator Contacts, which are the contacts managed by the Systems Administrator, cannot be modified or deleted using the Symphony Application.

6.2.15.1 Adding New Contacts

Use the following method to add a new contact to the contact list:

1. Click the **Harris Logo** → **Open Directory** → **Contacts**.

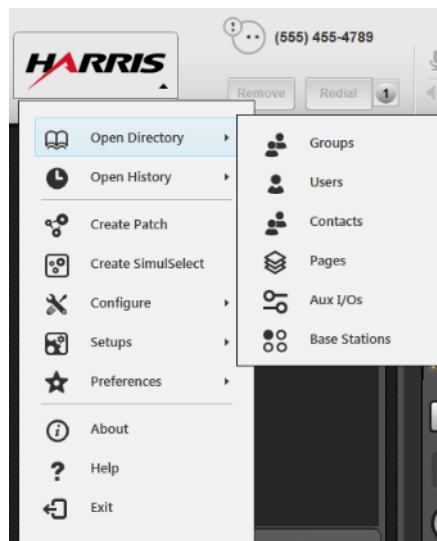


Figure 6-7: Contact List Menu

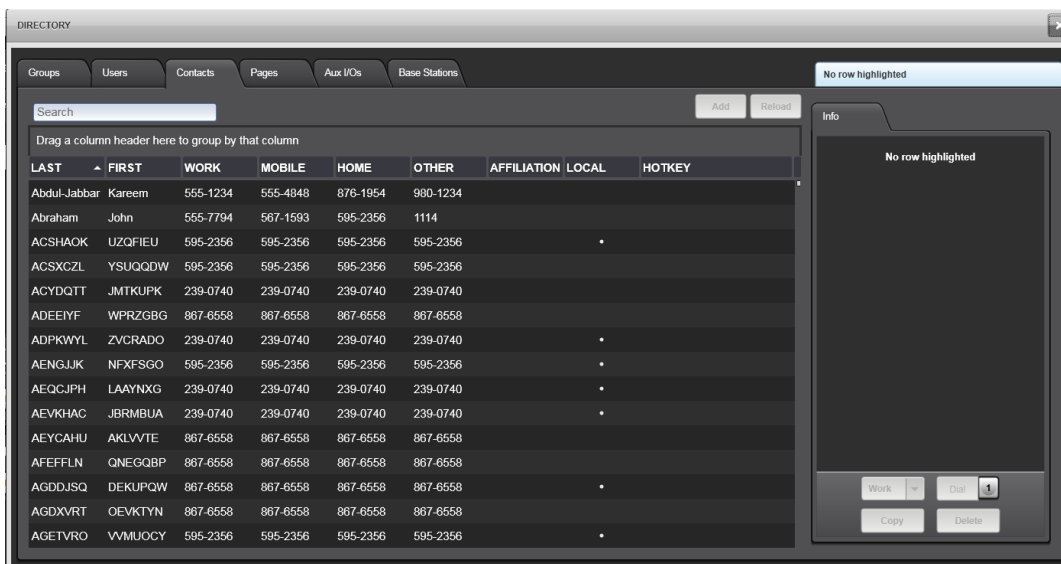


Figure 6-8: Contact List

2. Press the **Add** button.

3. The New Contact window is displayed.

4. Enter the Contact Information:

- **First:** First name of the contact.
- **Last:** Last name of the contact (mandatory).
- **Affiliation:** The affiliation of the contact, such as department or company.
- **Work Number:** Work Phone number of the contact.
- **Work Ringtone:** The selected ringtone played when a call is received from the Work number.
- **Mobile Number:** Mobile Phone number of the contact.
- **Mobile Ringtone:** The selected ringtone played when a call is received from the mobile number.
- **Home Number:** Home Phone number of the contact.
- **Home Ringtone:** The selected ringtone played when a call is received from the home number.
- **Other Number:** Other Phone number of the contact.
- **Other Ringtone:** The selected ringtone played when a call is received from the other number.
- **Primary:** Selects the primary Phone number. The selected primary number becomes a mandatory parameter.

The phone numbers specified for a Contact can be configured to automatically provide Overdial (DTMF) digits. Pause “,” characters can be added after a phone number followed by the Overdial digits to automatically send Overdial digits once a call is established. A Pause “,” character symbolizes 1 second of delay before Overdial digits are sent.

Example: Other Number: 5551234,,5793,1

With this example, a call is made to 5551234. Once a call is established, the console waits for 3 seconds, since there are three pause characters, then send over dial 5, 7, 9, and 3, then waits for 1 more second, then sends over dial 1.

This can be used to call the voicemail mailbox and automatically send the mailbox PIN number without manually dialing the PIN when retrieving voicemail messages.

5. Click the **Save** button to save the contact, the **Save/New** button to save the contact and prepare the New Contact Dialog for another entry, or the **Cancel** button to close the New Contact Dialog without saving the data.

The newly created contact is displayed in the Contact list.

6.2.15.2 Updating Existing Contacts

Use the following method to edit an existing contact in the local contact list. Admin Contacts (contacts created using the Configuration Utility) cannot be edited using the Symphony Application.

1. Click the **Harris Logo → Open Directory → Contacts**.
2. Highlight the contact to be edited.
3. Click the **Edit** button.
4. Update the details that require modification.
5. Click the **Update** button once finished.

The newly updated contact is automatically updated on the Contact list.

6.2.15.3 Copying Admin Contacts

Admin Contacts cannot be edited using the Symphony Application. A local copy of the Admin contact can be used instead if updates are needed to the Admin contact. Use the following method to copy an Admin contact into the Local Contact List:

1. Click the **Harris Logo → Open Directory → Contacts**.
2. Highlight the Admin Contact to be copied.
3. Click the **Copy** button.
4. The New Contact dialog will be displayed, prepopulated with the details of the Admin Contact.
5. Update the details that requires modification.
6. Click the **Save** button once finished.

6.2.15.4 Deleting Existing Contacts

Use the following method to delete an existing local contact in the contact list. Admin Contacts (contacts created using the Configuration Utility) cannot be deleted.

1. Click the **Harris Logo → Open Directory → Contacts**.
2. Highlight the contact to be deleted.
3. Click the **Delete** button.

The deleted contact will be automatically removed from the Contact list.

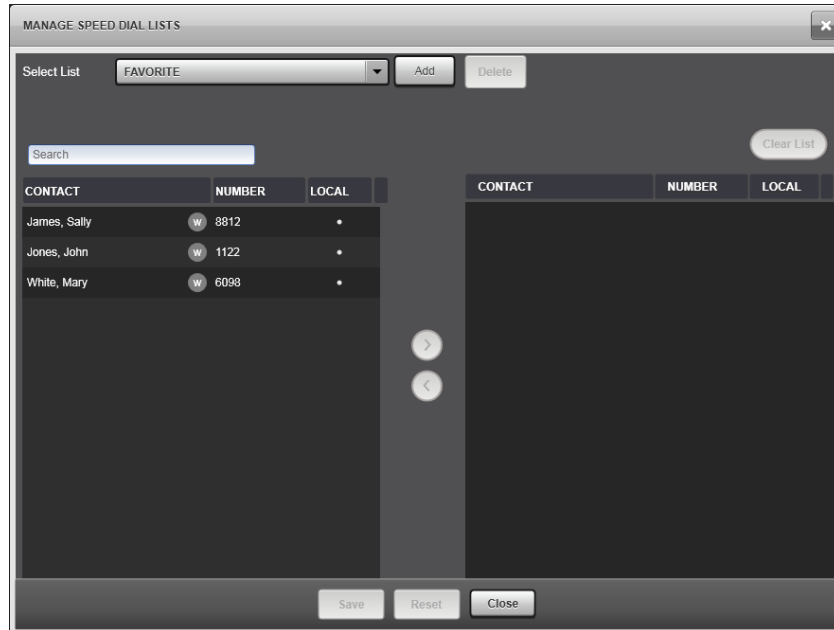
6.2.16 Managing Speed Dial Lists

Local Speed Dial Lists can be created, edited, and deleted using the Symphony Application. Admin Speed Dial Lists which are created using the Configuration Utility Application cannot be modified or deleted.

6.2.16.1 Adding New Speed Dial Lists

Use the following method to add a new Speed Dial List:


1. Click the **Harris Logo** → **Configure** → **Manage Speed Dial Lists** to open the Manage Speed Dial Lists Dialog.

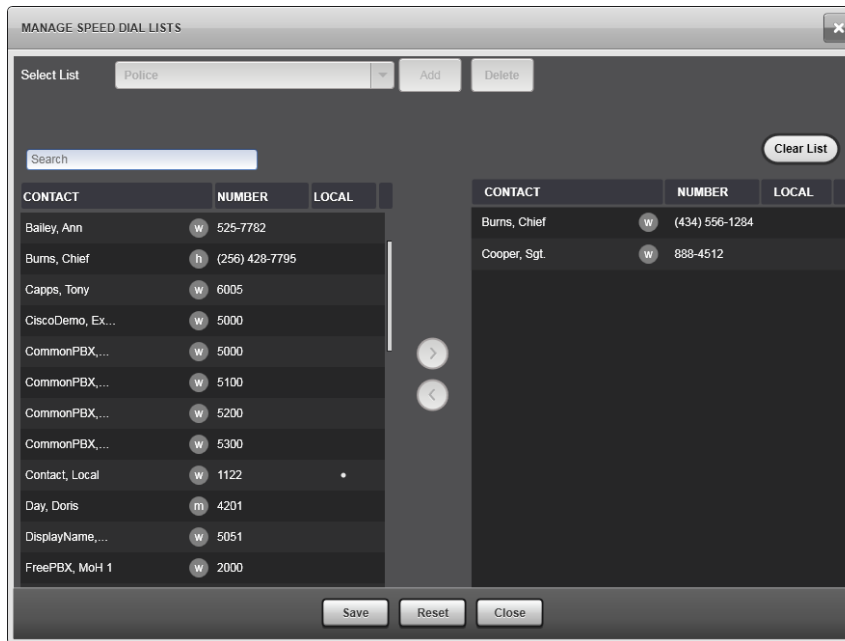


2. Click the **Add** button.
3. Enter the Name of the Speed Dial List in the Name Field.



4. Click the **Save** button to save the Speed Dial List name.

5. Select Contact(s) to add to the list and press the Right Arrow button .

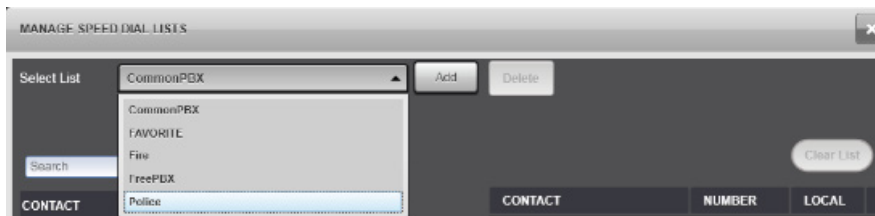




6. Click the **Clear List** button to remove all selected Contacts from the Selected Panel (right panel).
7. Click the **Save** button to save the list.

6.2.16.2 Updating Existing Speed Dial Lists

Use the following method to edit an existing Speed Dial List:

1. Click the **Harris Logo → Configure → Manage Speed Dial Lists**.
2. Choose a speed dial list using the drop-down arrow.

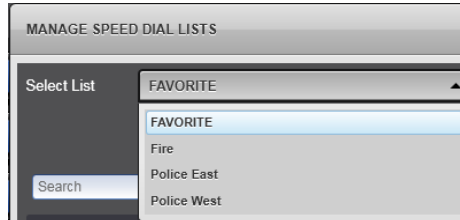


3. Add and remove entries on the speed dial list by using the left  and right  arrow buttons.
4. Click the **Reset** button to undo all recent unsaved changes.
5. Once all changes are finished, click the **Save** button.

6.2.16.3 Deleting Existing Speed Dial Lists

Use the following method to edit an existing Speed Dial List:

1. Click the **Harris Logo** → **Configure** → **Manage Speed Dial Lists**.
2. Choose a speed dial list using the drop-down arrow.



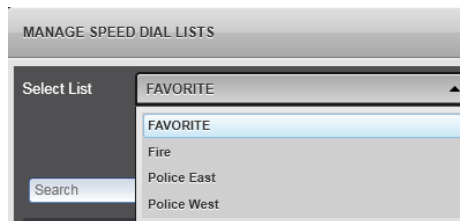
3. Click the **Delete** button. Symphony displays a prompt to confirm deletion of the Speed dial list.



6.2.17 Managing Favorites list

Managing Favorites List is the same procedure as managing the Speed Dial List. The Favorites List is a special Speed Dial List which cannot be deleted.

Use the following method to edit a Favorites List:

1. Click the **Harris Logo** → **Configure** → **Manage Speed Dial Lists**.
2. Choose the **FAVORITE** speed dial list using the drop-down arrow.



3. Add and remove entries on the Favorites List by using the left  and right  arrow buttons.
4. Once all changes are finished, click the **Save** button.

6.2.18 Voicemail and Message Waiting Indicator



NOTE

Voicemail behavior is highly dependent on the configuration of your PBX. For example, the Message Waiting Indicator may not be provided by the PBX after every Symphony reboot. Consult your System Administrator for details on your configuration. The scenario described below assumes a typical voicemail configuration.

Whenever a call is not answered, after a preconfigured time, the call is forwarded to voicemail where the caller can leave a voicemail message. When a new message is left in the mailbox, the console provides a Message Waiting Indicator that indicates that there is a new voicemail waiting to be retrieved in the mailbox. A green indication is displayed on the console as shown in Figure 7-2.

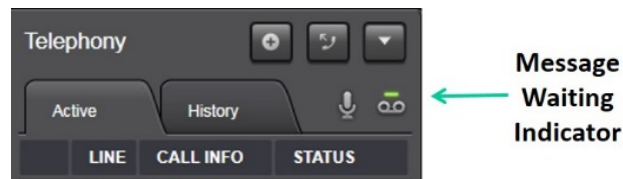


Figure 6-9 – Message Waiting Indicator

Clicking the Message Waiting Indicator on the Telephony Sidebar Panel displays a more detailed display of MWI on a per line basis as shown in Figure 6-10.

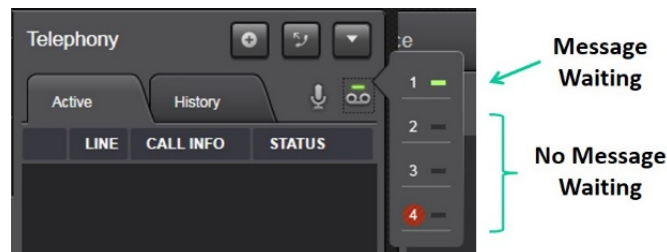


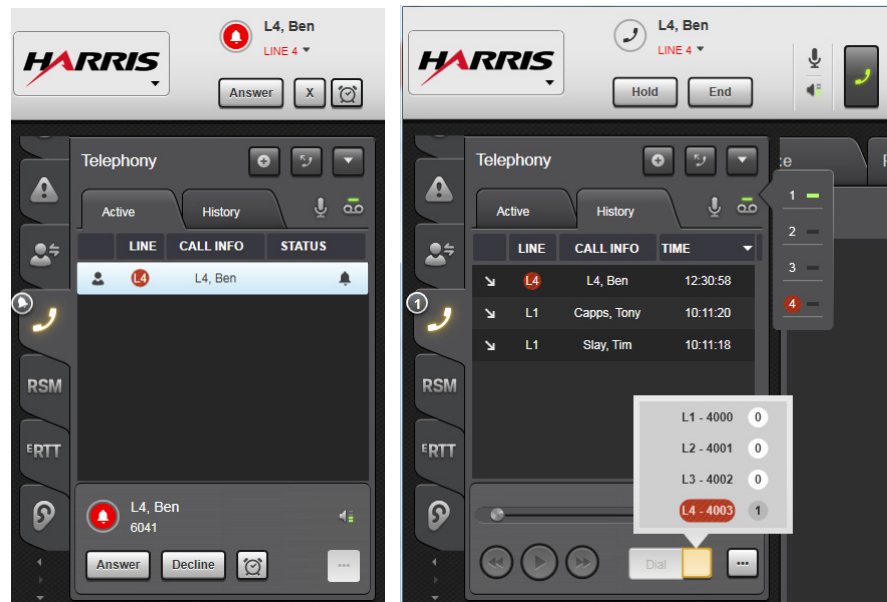
Figure 6-10 – Message Waiting Indicator All Lines

Further clicking the Message Waiting Indicator Icon of the corresponding SIP line automatically calls the configured voicemail number to retrieve voicemail messages. You can also manually dial the voicemail number to retrieve voicemail messages.

Once a call to the voicemail mailbox is performed, listen to the voicemail attendant's instructions and use the over dial function to navigate through the voicemail mailbox options.

6.2.19 Primary Line Designator

The primary line is an administratively designated SIP line that has red highlighted icons, text, and indicators. This is typically the main line used to answer incoming calls. The special treatments for the primary line help you identify and prioritize primary line calls. The figures below show the primary line designator in various locations including the Select Bar Telephony Control, Active Call List, Call Control Area, Line Menu, Call History List, and Voicemail Menu.



6.2.20 Light Pole

Symphony console local Aux Outputs that can be configured to drive light poles when a phone call is active. This functionality applies to both Call Director and SIP Telephony feature. This is configured using the Symphony Configuration Utility application by the system administrator.

The output is activated in the following manner:

- Inbound Ringing: Turn output on and off each second while any call on the console is ringing.
- Active call no ringing: Turn the output on if no inbound call is ringing and at least one call is in the ACL (active or on hold) or an outbound call is ringing. That is, the light pole does not flash for outbound ringing.
- Turn the output off otherwise.

7 CONVENTIONAL OPERATIONS

7.1 PAGING



Since the Paging capability at the console is a feature-licensed option, this feature is only available if the console is equipped with this software license. In addition, various system and console-related configuration changes must be made before this feature will be operational. Consult with system and/or dispatch administration personnel as necessary.

The Symphony Dispatch Console supports single sequence paging, stacked paging, “One Touch” paging, and Composite paging. A paging sequence is a sequence of tones sent to a single group, conventional channel, or unit. The sequence is designed to only set off a select set of pagers such as a fire station or a collection of volunteer firemen. A paging stack is a collection of related paging sequences that are issued together from a single paging module. The “One Touch” paging feature allows the operator to send a page or stacked page, open a SimulSelect, and send the optional alert with one touch of the paging module. Composite paging provides the ability to build a complex page on the fly using existing page sequences and stacks. The composite page can also include an alert and automatic SimulSelect at the end. Auto selecting all of the paged entities allows the dispatcher to effortlessly speak to everyone involved in the incident once they have been paged.

7.1.1 Paging Modules

Symphony supports three sizes of paging modules as shown in Figure 7-1. The mini module contains a single line Smart Button, the small module contains a two-line Smart Button, and the Normal module extends the small module with a single information tab. The Baton supports paging modules that are virtually identical to mini modules.

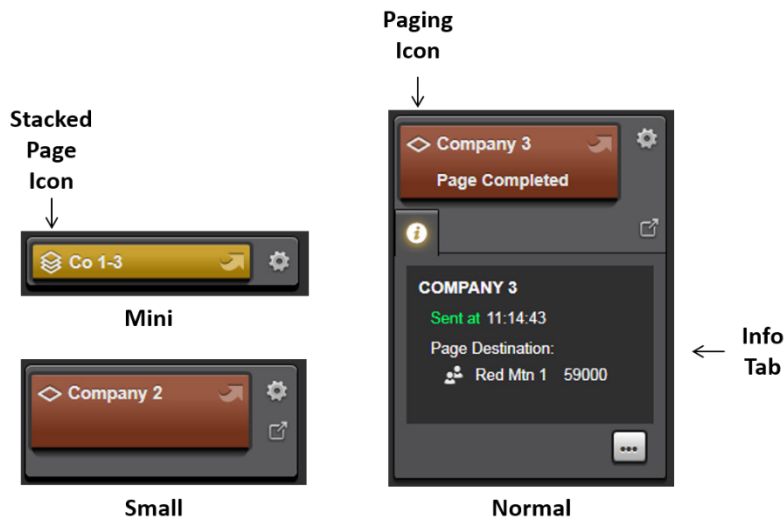


Figure 7-1: Paging Modules

The first line of the Smart Button contains the Paging Icon and the name of the page associated with the module. Note that the stacked page icon is a stacked version of the sequence page icon. Two-line modules use the second line to provide page progress information. The information tab in the normal module lists the sequence(s) and destination(s) of the page or stacked page. It also shows the status of the last page sent, including the time the page was sent and its success or failure. Remember that you can always pop-out a Normal module from a mini or small module using the Context Menu (gear icon).

The Context Menu includes a paging unique entry “Send without SimulSelect.” When pages are defined by your administrator, they may be configured to select all entities in the page or stack. You can override this behavior and page without the select using the “Send without SimulSelect” menu option.

7.1.2 Paging Directory Tab

The Paging Directory Tab provides a summary of all paging parameters as shown in Figure 7-2. See Section 3.5.1 for details on basic directory usage. This section focuses on the Pages tab. Table 7-1 describes each column in the main dialog list.

ALIAS	PAGE DESTINATION ENTITY	ALERT TONE	AUTO SELECT	VALIDITY	PAGING CODE PLAN
Company 3	Red Mtn 1		•	Valid	User Defined
Company 1	Red Mtn 1		•	Valid	User Defined
Company 2	Red Mtn 1		•	Valid	User Defined
Rescue 1	Purple Mtn 1		•	Valid	User Defined
Rescue 2	Purple Mtn 1		•	Valid	User Defined
Rescue 3	Purple Mtn 1		•	Valid	User Defined
no reply	No Reply 20			Valid	User Defined
DTMF Seq 1	Current Select			Valid	User Defined
Co 1-3	Red Mtn 1		•	Valid	User Defined
All	Multiple Destination Entities	Alert1	•	Valid	User Defined

Right sidebar details for 'All':

- Alert Name: Alert1
- Sequences:
 - Company 1
 - Company 2
 - Company 3
 - Rescue 1
 - Rescue 2
 - Rescue 3
- Page Destination:
 - Red Mtn 1 59000
 - Purple Mtn 1 59100

Figure 7-2: Directory Dialog Paging Tab

Table 7-1: Directory Columns – Paging Tab

COLUMN	DESCRIPTION
ALIAS	The alias for the page.
PAGE DESTINATION	Destination of the page. The destination can be a group, conventional channel, unit or multiple destinations. In the latter case, see the Info tab for a complete list of the destinations.
ALERT TONE	Contains the name of the alert if an alert tone is included with the page.
AUTO SELECT	Contains a dot if Symphony will auto select all the destinations when the page completes.
VALIDITY	“Valid” if the page is properly defined, “Invalid” with explanation otherwise. The validity of a page can change over time if for example a destination base station is down or group definitions are changed in the system.
PAGING CODE PLAN	Symphony currently supports “User Defined” pages.

The panel on the right side of the Directory Dialog provides additional Tabs that apply to the highlighted page in the main dialog list.

Info Tab: This tab provides detailed information about the page that cannot fit in the columns in the main dialog list including the sequence(s) and destination(s) of the page.

Module Tab: This tab provides the paging module for the highlighted page in the main dialog list.

7.1.3 Paging Operation

7.1.3.1 **Foreground vs. Background Paging**

Pages can be sent in the foreground or background based on configuration. Consult with your system administrator to determine how your console is configured. Foreground pages are sent from the consoles User ID and prevent simultaneous voice transmissions while the page is active. In other words, only a voice call or a page can be sent at one time, but not both at the same time. Background pages use a specially configured ID and allow simultaneous transmission on entities that will not interfere with the page. Therefore, you can start a page and then transmit on a group that is not involved in the page. If you attempt to transmit on an entity that could cause the page to fail, the transmission is denied to protect the page. The reason for denying the transmission is noted in the System History.



If background paging is enabled, but the background paging ID is not configured, all paging modules are grayed out.

Abort an active page as described below to initiate a voice transmission. A page is considered active until all tones and alerts have been sent and the AutoSelect has expired.

7.1.3.2 Page Types

Symphony supports two types of pages. Analog pages are sent as normal calls to their destination Talk group(s), unit(s) or channel(s) the paging tones are transmitted as part of the call. Digital pages may only be sent to a single destination unit and may only contain Standard DTMF or KNOX™ DTMF tones. Digital pages are sent to the destination units as a digital command and do not require a working channel.

7.1.3.3 Sending Analog Pages

Analog paging sequences, stacks, and “One Touch” pages are defined by the administrator and assigned to modules on the Workspace or Baton. To send a page, press the Smart Button on the appropriate Page Module. If the page definition is valid, Symphony begins sending paging tones to the specified destination(s). The icon on the upper right of the Smart Button changes to a bell to show tones are being sent and the tones are played out the speaker assigned to the current destination. If the module contains a paging sequence, the second line of the Smart Button, if available, displays “Page in Progress” while the tones are being transmitted and “Page Completed” when the sequence is complete. If the module contains a paging stack, the second line of the Smart Button displays a progress count “3/7 Page in Progress” while the tones are being transmitted and “Page Completed” when all the sequences have completed. The progress count increments as each sequence in the stack is sent. If a paging module X contains a sequence Y that is part of a stack, the Smart Button on module X will behave as if it had been pressed when the stack executes the sequence Y. This allows you to follow the progression of the stack or composite page.

If auto select is enabled in the page definition, Symphony automatically selects the destination(s) in the page. If the page is a sequence, the single destination is selected. If the page is a stack, the destinations specified in the stack definition will be SimulSelected unless the destinations are all in a Patch, in which case the Patch is selected. For simplicity, the term AutoSelect refers to both cases. The AutoSelect will remain in effect for approximately five seconds or until you transmit on the Select Module. Once the Select Module transmission is complete or the timeout expires, whichever is latest, the previously selected entity will be re-selected. The last AutoSelect can be recalled using the **Page** button as described in Section 7.1.3.6. If an alert is included in the page definition, the alert will be sent on the AutoSelect.

Once the page, auto SimulSelect, and alert complete, the Information tab updates with the time and status of the page.

7.1.3.4 Sending Digital DTMF Pages

The Symphony administrator may configure page modules to send digital pages. These pages do not create a call and instead send a digital message to the radio instructing it to produce the desired DTMF tones.

Digital DTMF pages may be sent as a single sequence, as part of a stack or composite page and they may be “One Touch” pages.

There are some important differences to be aware of when working with Digital DTMF pages:

- Digital pages do not produce a tone at the console when they are sent. They are visually confirmed on the user interface.
- If the system is unable to deliver the page command to the radio, the module will display “Page Denied” or a similar error message.
- Except when using an Alert or AutoSelect, digital pages are considered complete as soon as the digital command is delivered by the system. Typically, this is very quick.
- When a digital DTMF page has an AutoSelect or Alert (or is part of a composite or stack which has one), the console will pause to allow the radio to completely play out the DTMF tone before proceeding. During this time, the paging module will display “Page in Progress” message but will not play the DTMF digits out of the speaker. This allows time for the radio to finish before the alert tone and/or announcement is sent.

7.1.3.5 Paging Anomalies

Pages generally proceed without incident if the pages are properly defined and tested in advance. However paging anomalies can occur when:

- Changes to the system invalidate a page definition; for example, a paging definition includes group X as a destination and group X is deleted from the system.
- Paging resources are temporarily unavailable; for example, a paging destination base station is busy or off line.

Before sending a page, Symphony determines if the Page is valid. This determination is fairly complex and includes checking to see that all destination base stations are on line, making sure the auto SimulSelect will be valid and verifying that no paging rules will be violated. If the page is invalid, an error tone sounds, a message appears in the System History on the Status Bar, and “Page Denied” appears for a few seconds on the second line of the module Smart Button if available.

If a page is denied, examine the message in the System History to determine the denial reason. Contact your system administrator to correct the anomaly. In the meantime, build a composite page as defined in Section 7.1.3.6 to page the destinations that are available. For example, if a stack comprises sequences A, B, and C, and sequence C has caused the anomaly, create a composite page with sequences A and B. The Info Tab of the paging module or Directory lists the sequences in the stack.

If the page is valid, it may still fail while it is in progress if a destination is busy or becomes unavailable during the page. In this case, the page will stop, an error tone will sound, the failure will be logged in the System History on the Status Bar, and “Page Failed” will appear in the second line of the Smart Button for a few seconds.

If the page failed due to a transient condition such as a busy channel, try the page again by pressing the Smart Button. If the page fails multiple times, consult your systems administrator and build a composite page as described above for a denied page.

7.1.3.6 Send a Composite Page

A composite page provides the ability to define a page on the fly from existing paging modules. To build a composite page, press the **Page** button in the upper right corner of the Workspace as shown in Figure 7-3.

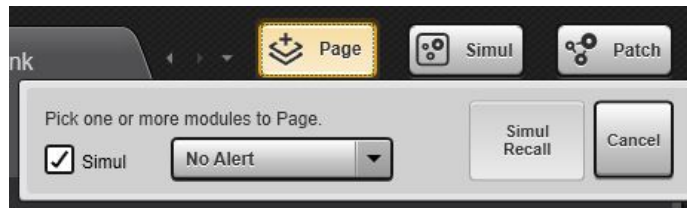


Figure 7-3: Composite Paging Dialog

The composite page dialog box appears as shown in Figure 7-3 and the context menu icon (gear) on each paging module is replaced by a hand icon as shown in Figure 7-4. Click on the hand icon of the modules you want to include in the composite page. If the module you chose is invalid or will make the composite page invalid, an error tone will sound and a message will appear in the System History to explain why the module cannot be added to the composite page.

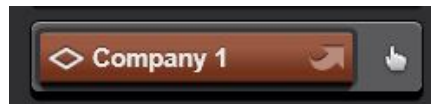


Figure 7-4: Paging Module with Hand Icon

Check the “Simul” check box to AutoSelect all the destinations of the picked pages that have AutoSelect enabled. That is, if a picked page has auto select disabled, its destination is not included in the AutoSelect.

Select an alert from the drop-down list. The composite page alert always overrides the alert of its constituent pages. If no composite page alert is chosen, no alert will be sent. If a composite page alert is chosen, the composite page alert is sent.

AutoSelect works as defined for pages in Section 7.1.1. To recall the last AutoSelect, press the Simul Recall button in the composite page dialog. Once recalled, the AutoSelect will remain in effect until a new entity is selected; that is, the AutoSelect does not time out.

For long paging sequences, it is best to keep the Composite Page Dialog open to abort the page if necessary. However, you can close the dialog by pressing the Page button. To reopen the dialog, press the Page button again.



NOTE

Composite paging is not available from the Baton.

7.1.3.7 Abort a Page

To abort a pre-defined page in a paging module, do one of the following:

1. Open the context menu on the page module.
2. Click the “Abort Page” menu item.

Or

Press the **Abort** button just below the Select Bar in the upper right corner of the screen as shown in Figure 7-5. This is the Composite Page button that turns into an Abort button while any pre-defined page is active.

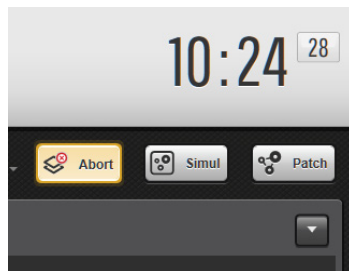


Figure 7-5: Abort Page Button

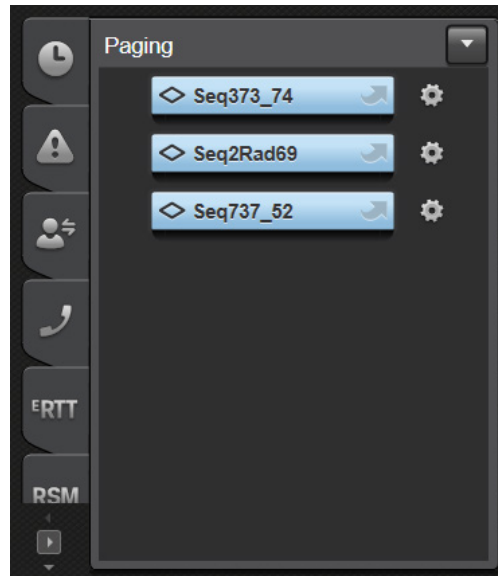
To abort a composite page, do the following:

1. Press the **Page** button to open the Composite Page Dialog if it is not already open.
2. Press the **Abort** button on the dialog.
3. Press the **Cancel** button on the dialog to close out the dialog.

7.1.4 Advanced Paging

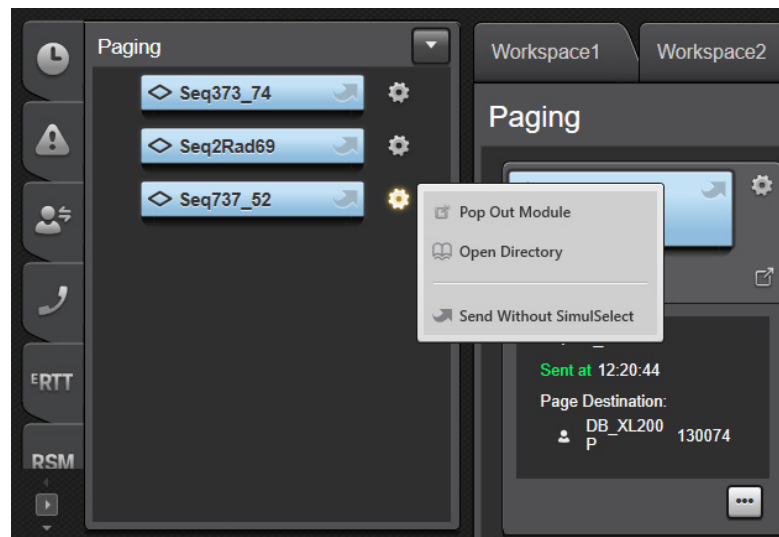
Advanced Paging allows you to attempt all Page Sequences in a Page Stack or Composite Page Stack before stopping. If a sequence fails, it keeps a record of the failure but proceeds to the next sequence until the Stack has completed. The Advanced Paging feature must be licensed and enabled via the Symphony Configuration Utility Paging Tab before it can be used.

This feature is accessed via the Advanced Paging Sidebar Panel and displays all the sequences that were attempted during the last Page Stack or Composite Page Stack, displaying a Mini Smart Button for each in the order they were attempted. It is a convenient way to retry any sequences that were not sent successfully. The System History Panel will identify any failed sequences and explain why they failed, so any issues can be resolved before retrying. This Sidebar Panel allows you to retry one of the sequences individually or to retry multiple sequences with one action, as explained in the following sections.



The Advanced Paging Sidebar Panel includes the following elements:

- **Mini Smart Button:** identifies each sequence in the recently attempted Stack Page or Ad-hoc Stack Page. These buttons are displayed top to bottom in the order they were attempted.
- **Gear Icon:** beside each Mini Smart Button to pop out a context menu of options that can be used in conjunction with each page sequence on the side bar panel.

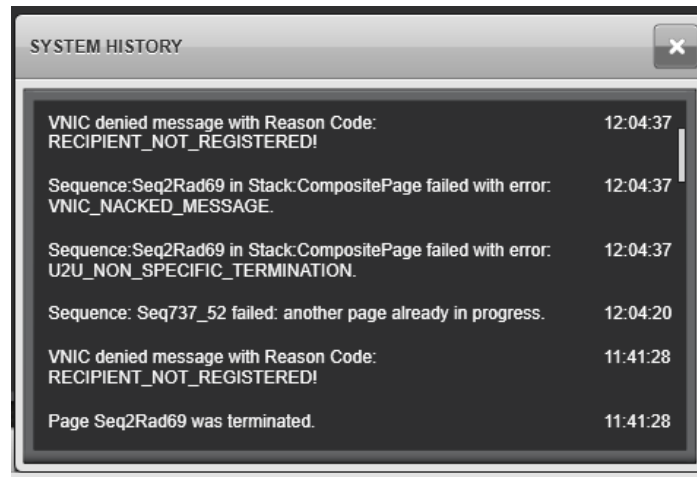


The following options can be accessed from this context menu:

- **Pop Out Module:** pop out a full-size paging module for the corresponding sequence.
- **Open Directory:** display a directory listing of all pre-defined sequences and stacks.
- **Send Without SimulSelect:** send the corresponding page but do not SimulSelect the targets of all the sequences listed on the side panel.

7.1.4.1 Determine Why A Sequence Failed

Use the expanded System History Panel to determine why one or more sequences failed. The System History will display one or more entries to identify each failed sequence and indicate why it failed. In the following example, the System History reports that 'Seq2Rad69' failed because its target entity is not registered with the system. Make sure the device is powered on and is registered with the system. Then use the Mini Smart Button on the Paging Side Bar Panel to retry the page.



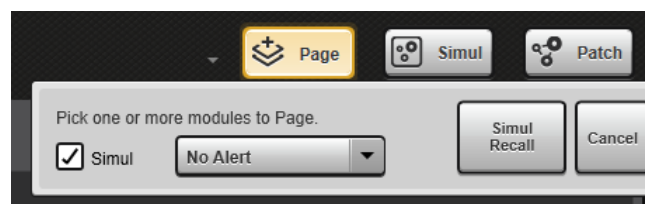
7.1.4.2 Retrying a Single Sequence

After resolving the issue(s) that caused a sequence to fail, retry sending it by clicking or pressing its **Mini Smart Button** one time. If the retry is successful, no new error messages are displayed in the System History panel. If additional issues cause the retry to fail, a new error message in the System History panel will identify it.

7.1.4.3 Retrying Multiple Sequences

To retry multiple sequences, create a Composite Page Stack with those sequences picked.

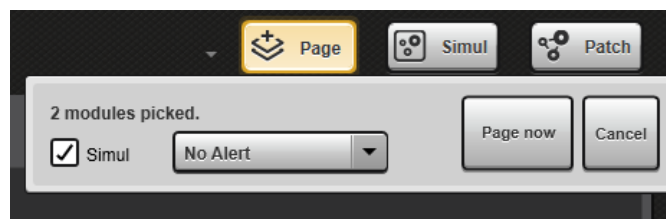
1. Click the **Page** button located below the clock. The Ad-hoc Page dialog displays. Note that the **Simul Recall** button will be replaced by the **Send Now** button after the first sequence is picked for retry.



- On the Advanced Paging Sidebar Panel, the Gear icons are replaced by Pick icons. Click on the Pick icons beside the Mini Smart Buttons for the sequences to be retried. The picked icons now glow.



- After resolving the issues that caused the sequences to fail, pick all the sequences to retry and click the **Page now** button on the Ad-hoc Page dialog. The picked sequences will be sent in the order they are listed on the Advanced Paging Sidebar Panel.



- If the retries are successful, no new error messages are displayed in the System History Panel. If additional issues cause any of the retries to fail, a new error message in the System History panel will identify each.



Consult Section 7.1 on the standard paging feature for a more extensive explanation of all the features associated with Advanced Paging.

7.2 BASE STATION CONTROL



Since Base Station Control is a feature-licensed option, this feature is only available if the console is equipped with this software license. In addition, various system and console-related configuration changes must be made before this feature will be operational. Consult with system and/or dispatch administration personnel as necessary.

The Symphony Dispatch Console supports conventional base station control with special modules. This allows dispatch communications with the conventional radio units operating on the respective conventional base station equipment interconnected to the VIDA system via the Interoperability Gateway. Communications include receiving calls from and transmitting calls to the conventional radio units as well as emergencies declared by conventional radios using MDC-1200 signaling.

In addition to the basic ability to transmit on conventional channels, conventional base station control functions provided by the Symphony Dispatch Console includes standard remote control functions normally performed by a conventional base station remote controller. These functions are listed below and described in detail in the sections that follow:

- Enable/Disable Monitor
- Toggle Repeat Mode
- Enable/Disable Second Receiver
- Enable/Disable Protected Mode
- Wildcard 1 and 2
- Digital/Analog
- Toggle Scan Mode
- Toggle Main Standby Site

Symphony currently supports the most common base station type known as TnRn. This type of base station supports one transmit/receive channel pair at a time but can switch between n channel pairs where n is up to 16. Channel pairs may be based on different frequencies and/or different base station configuration settings such as Continuous Tone-Coded Squelch System (CTCSS) value.

7.2.1 Base Station Modules

The Symphony conventional station modules look and act similarly to normal trunked talkgroup modules. This is because the Interoperability Gateway (conventional gateway) and the Symphony console map normal trunked talkgroups to conventional channels.

For example, a talkgroup such as “Red Mtn 2” is mapped to a frequency on a conventional base station. When the conventional station receives traffic on the frequency, the Interoperability Gateway generates a call on the corresponding talkgroup; trunked units and consoles on that talkgroup hear the call. Similarly, when the Interoperability Gateway detects a call on a talkgroup, it will change the conventional station to the matching frequency and transmit the audio to the conventional units.

A conventional base station can be equipped with a CTCSS or Digital Code Squelch (DCS) decoder; these tones may also be mapped to trunked talkgroups.

The TnRn base station modules are shown in Figure 7-6. The TnRn module is available in the normal and small sizes. Base station modules are similar to regular communications modules except that they have a banner at the top, a different set of tabs in the normal size and slight differences in the Smart Button, and standard tabs.



Figure 7-6: Base Station Modules

The banner contains the base station name and the channel change controls. Channels can be changed one at a time using the up or down arrows or selected directly from the channel change menu. The up/down arrow will disappear when you reach the top or bottom of the channel list respectively. The list does not roll over.

The Smart Button has a unique icon to indicate a base station. The top line shows the current channel and the bottom line shows the caller. Note that only base stations that support MDC-1200 can identify specific callers. Base stations without MDC-1200 show the user ID of the Unified Audio Card (UAC).

The conventional station module includes the Audio and History tabs, which are similar to trunked groups, and a Base Station Control tab that is unique to conventional operation. Conventional base stations do not support group emergencies, so there is no Emergency tab.

The Base Station Control tab, shown in Figure 7-7, contains the base station controls that are available for the associated base station. The tab only includes the controls supported by the specific base station, so the controls shown may vary between base stations.



Figure 7-7: Base Station Control Tab

The history tab, shown in Figure 7-8, is similar to that of trunked communications modules except that each entry indicates the caller and called channel separated by ">."



Figure 7-8: History Tab

Base Station modules are available on the Baton as shown in Figure 7-9. Space limitations force the module to be a single line Smart Button which will behave like a mini module. Where space is available, a small banner with the base station name is included.

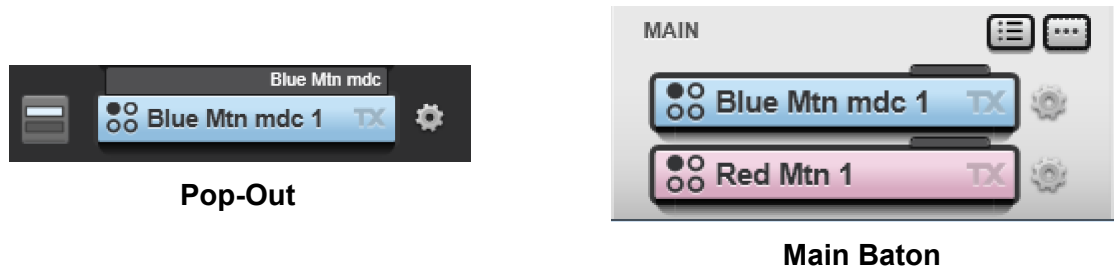


Figure 7-9: Baton Base Station Modules

7.2.2 Base Station Directory Tab

The Base Station Directory Tab provides a summary of all base station parameters as shown in Figure 7-10. See Section 3.5.1 for details on basic directory usage. This section focuses on the Base Station tab. Table 7-2 describes each column in the main dialog list.

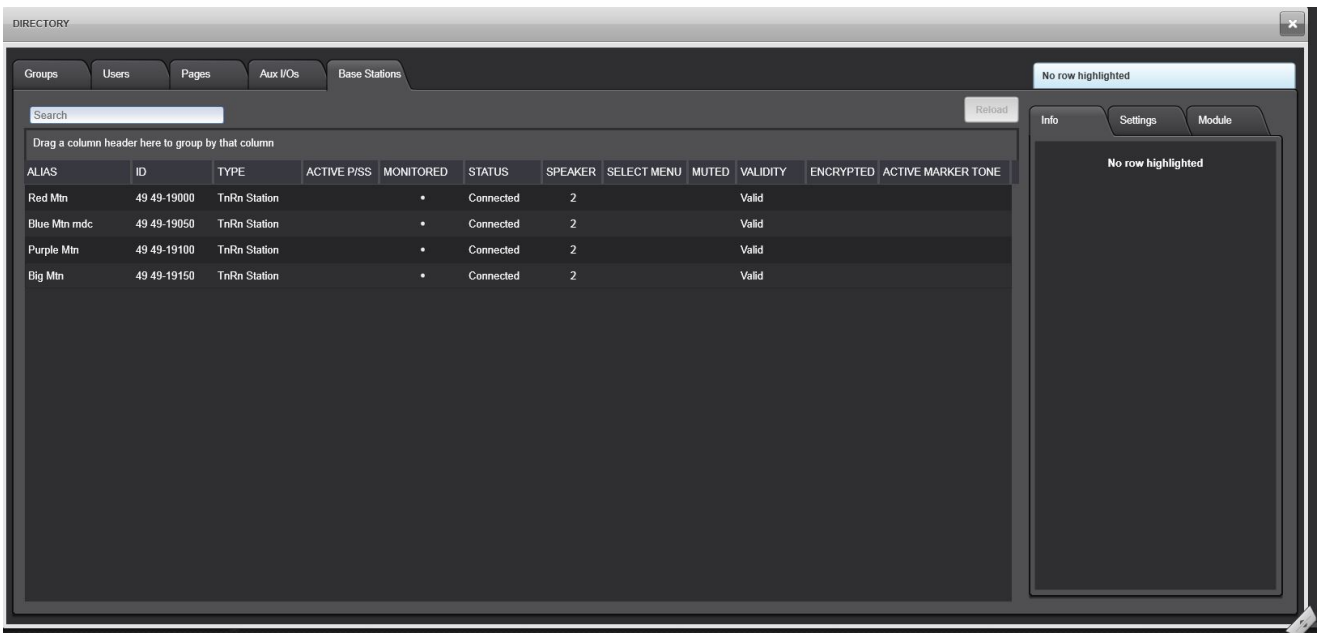





Figure 7-10: Directory Dialog Base Station Tab

Table 7-2: Directory Columns – Base Station Tab

COLUMN	DESCRIPTION
ALIAS	The alias for the base station.
ID	Site ID assigned to the base station.
TYPE	Base station type (i.e., TnRn).
ACTIVE P/SS	Indicates if the base station is part of an active Patch or SimulSelect.
MONITORED	Console receives calls on this base station.
STATUS	Connection status of base station: Connected or Disconnected.

COLUMN	DESCRIPTION
SPEAKER	Shows the currently assigned speaker.  indicates selected.
MUTED	Indicates the current mute state for the base station.
SELECT MENU	Indicates that the base station is included in the Select Menu.  indicates the base station is the Default Select Entity.  indicates the base station is a Preferred Select Entity.
VALIDITY	“Valid” if the UAC for the base station has a valid user ID, “Invalid User” otherwise. The ID may be invalid because an ID was not programmed in the UAC or because the Filter by Agency feature is enabled and the ID is not in a configured Agency.
ENCRYPTED	“Encrypted” indicates the connection to the UAC is encrypted. Note: this does not mean that transmissions over-the-air are encrypted. That is only possible if the base station provides some form of analog encryption.
ACTIVE MARKER TONE	Indicates if the group has an Active Marker Tone.



NOTE

If you intend to set up encryption on a base station and any of its channels, represented to the VIDA system as trunked groups, you must satisfy these requirements:

- Symphony supports AES encryption ONLY on base stations.
- All channels (trunked groups) must be in an AES crypto net that includes the Symphony dispatcher. Encryption is disabled if any of the channels are in a DES crypto net or are not included in an AES crypto net.

The panel on the right side of the Directory Dialog provides additional tabs that apply to the highlighted base station in the main dialog list. Note: selecting multiple base stations only applies to the Settings Tab. Other tabs will be grayed out if multiple tabs are highlighted.

Info Tab: This tab provides a textual summary of key columns in the main dialog list for the selected base station.

Settings Tab: The Directory dialog supports choosing multiple entities for changes on the **Settings** tab. After making changes on the **Settings** tab, click **APPLY**. The following options are available from the Settings tab:

- **CONSOLE:** Monitor base station with console - When this option is checked, the console user can monitor and interact with this base station even if it is not programmed to a Communication Module on the Workspace.
- **UNSELECT SPEAKER:** Specify the speaker assigned to the highlighted base station(s).
- **SELECT MENU:** Specify the Select List for assignment (Not in Select, In Preferred Select, or As Default Select). The maximum number of entities allowed in the Preferred Select list is configured by the administrator. When the maximum is reached, the last one in the list is removed (i.e., oldest entry is removed).
- **VOLUME:** Adjusts volume for all selected entities.

Module Tab: This tab provides the base station module for the highlighted base station in the main dialog list.

7.2.3 Conventional Operation

Conventional modules operate very much like trunked communications modules in most cases. See Section 3.3 to review basic module operation. This section highlights operations unique to base stations. Base station channels are mapped to VIDA groups; so, when working with channels, you are in essence working with VIDA groups.

7.2.4 Changing Channels

Unlike trunked modules, conventional station modules can communicate on multiple channels. Ensure that the base station is set to the proper channel so that the desired calls are heard, and transmissions go to the proper channel.

To change the base station channel:

1. Click the “Up” or “Down” arrows in the module banner.

Or

Click the Channel Change Menu in the module banner and select the desired channel.

Or

Click the Context Menu (gear icon), select **Channels**, and then select the desired channel.

2. Symphony sounds the channel change tone and updates the channel name on the Smart Button to indicate the channel has changed.

7.2.5 Selecting Conventional Stations

Conventional stations can be selected, i.e., placed, in the selected module as show in Figure 7-11. Note that you are selecting the station, not just the current channel of the station. Any calls received from or transmitted to the base station are treated as selected. Changing channels on the base station does not affect the select state of the station.

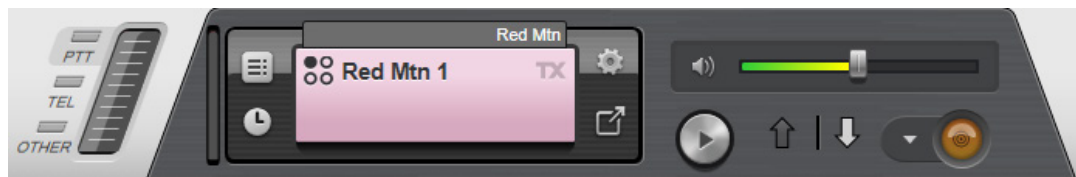


Figure 7-11: Base Station in Select Module

As with trunked modules, selecting a station places it in the Select Module, highlights associated modules in the Workspace with green, routes received audio from the station to the select speaker, and allows you to transmit on the current station channel with the footswitch, desk mic, Quick Action keys, or Smart button on the GUI. Base station modules use the same selection methods as trunked modules. See Section 5.2 for details on module selection.

In addition, selecting a conventional station module allows you to use the **Monitor** button on the desk mic to activate or deactivate the monitor function.

The Select Module indicates the base station name in a small banner and includes the up/down arrows to change the station channels. The Context Menu includes **Refresh** and **Channel Change** options.

7.2.6 Patches and SimulSelects

See Section 5.7 for the general procedure to create patches and SimulSelects. With conventional control modules, the procedure is largely unchanged with one notable exception: patching/SimulSelecting a base station patches/SimulSelects all the channels on the base station. Therefore, a Patch/SimulSelect may fill up sooner than you would expect with trunked groups.

The number of groups/channels supported in a Patch or SimulSelect depends on the Mode (L3Harris or Motorola), but is at most is limited to 15 members. Patching a base station with 8 channels uses up 8 of the 15-member slots of a Patch. Symphony will clearly indicate when the Patch or SimulSelect is full as described in Section 5.8.1.



Symphony requires version UAC version R4A08 (MPC860 GWB with Encryption) and requires that the station Patch feature is enabled on the UAC.

7.2.7 Conventional Controls

Symphony supports multiple base station control functions on the Base Station Control Tab as shown in Figure 7-12. These controls are used to send standard remote control tones to conventional stations. Control tones are generated by the Interoperability Gateway (IG or Gateway) at the command of the console. The Gateway also informs the console of the station's current status. Not all functions are available for all stations. The Gateway is configured to support the tones that are applicable for the type of station. When a function is not configured at the Gateway, the function slider will not appear in the Base Station Control Tab.

Unless otherwise stated, click the feature slider to toggle the position from left to right or right to left. Mode features are enabled when the slider is on the right side with a green background as shown in Figure 7-12 and are disabled when the slider is on the left side with a gray background. Features with a "/" in the name such as "Main/Standby" are in the first state when the slider is on the left side with a gray background (i.e., Main) and in the second state when the is on the right side with a green background (i.e., Standby).



Figure 7-12: Base Station Control Tab

7.2.7.1 Monitor Enable/Disable

A conventional base station can be equipped with a CTCSS or DCSS decoder. In some cases, it may be desirable to monitor the conventional channel for any and all activity on the inbound (receive) radio frequency. Typically, a monitor action is performed by a dispatcher just prior to transmission from the console to ensure that a conventional radio unit that is transmitting an improper tone/digital code on the base station's inbound radio frequency is not preempted ("stepped-on") by the dispatcher. When the monitor function is enabled, the conventional channel's base station receiver passes all received signals to the Symphony console, regardless of the originating conventional radio unit's transmitted tone/digital code. If the base station is a repeater, this action normally does not affect repeat operation, as the valid tone/digital code is still necessary to key the repeater.

The Monitor function is available on the base station control tab of the station module as shown in Figure 7-13.

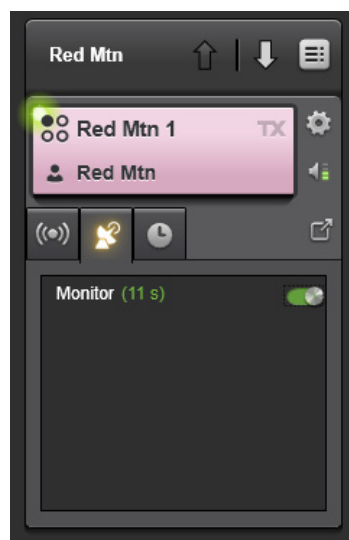


Figure 7-13: Monitor Tab with Count-Down Timer

To enable monitor:

1. Click on the Monitor slider.

Or

If the module is selected, press the "Monitor" button on a compatible desk mic.

2. When monitor is active, the Gateway places a call on the current group with any activity on the current receive frequency irrespective of any coded squelch such as CTCSS. If the conventional station is configured with a monitor timeout, a countdown timer is shown on the monitor line (see Figure 7-13). When the timer reaches 0, Symphony automatically drops out of monitor. If monitor timeout is not configured, the monitor must be disabled manually.

The procedure to toggle monitor mode off is identical to that above. The **Monitor** slider toggles to "Off" state.

7.2.7.2 Toggle Repeat Mode

A conventional base station may be capable of operating as a repeater. When a base station's repeat function is enabled, it re-transmits radio-originated audio that it has received on its inbound (receive) conventional radio frequency. This re-transmission occurs on the base station's outbound (transmit) conventional radio frequency. Together, both frequencies form the conventional channel. The re-transmission extends the coverage range of the originating conventional radio unit.

The "Repeat Mode" slider enables the repeater when moved to the right and disables the repeater when moved to the left.

7.2.7.3 Enable/Disable Second Receiver

Enable/disable a station's auxiliary receiver which is usually configured to receive direct mode traffic. This slider enables the second receiver when moved to the right and disables the second receiver when moved to the left.

7.2.7.4 Protected Mode

A conventional base station's receiver may be equipped with a continuous tone-coded squelch system (CTCSS), such as Channel Guard™ (CG), or a digital-coded squelch system (DCSS) such as Digital Channel Guard™ (DCG). These CTCSS/DCSS electronic decoder circuits at the base station help prevent extraneous signals from "getting into" the base station—typically a repeater—and they allow selective access to only those radio units transmitting the proper continuous low-frequency tone or digital code.

Protected mode may also refer to encryption functionality provided by the base station. Check with your system administrator to determine the meaning for protected mode in your system.

The Protected Mode slider enables the protected mode when moved to the right and disables the protected mode when moved to the left.

7.2.7.5 Wildcard 1 and 2

A conventional base station's receiver may be equipped with any number of non-standard tone-controlled features. The two Wildcard buttons may be used to toggle these features.

The slider enables the associated wildcard when moved to the right and disables the associated wildcard mode when moved to the left.

7.2.7.6 Digital/Analog

A conventional station may be capable of transmitting Analog or Digital signals. This slider toggles the base station between "Digital" and "Analog" modes.

7.2.7.7 Toggle Scan Mode

A multi-channel remotely controlled conventional base station may be equipped with channel scan capability. The **Scan Mode** slider enables the scan mode when moved to the right and disables scan mode when moved to the left.

Making changes to the scan capability can potentially affect communications between conventional radio units and other consoles in the system. Therefore, consultation with system or dispatch administration personnel is recommended prior to making any changes.

7.2.7.8 Main and Standby Site Toggling

For remote control operations, conventional systems may be equipped with hardware that allows toggling between a main set of conventional base stations, typically located at one radio site, and a standby set of conventional base stations typically located at a different radio site.

Depending upon system wiring, this console operational procedure can potentially affect communications between conventional radio units and other consoles in the system. Therefore, consultation with system or dispatch administration personnel for main/standby “site” toggle approval is recommended prior to making these changes.

The **Main/Standby** slider enables the standby when moved to the right and enables the main station when moved to the left. Base station modules in Standby mode provide a yellow band in the banner as shown in Figure 7-14.



Figure 7-14: Base Station Module in Standby

7.2.7.9 Refresh Controls

The Refresh control provides a way to fix mismatches in the indicated remote control state and the actual state of the conventional station. For example, the Symphony console may indicate that Scan Mode is off, but the station is receiving calls as if Scan Mode is enabled.

This may happen for several reasons, including the base station restarting after maintenance or missing a remote control tone.

To resynchronize the base station:

1. Open the Context Menu and select **Refresh**.
2. Observe the behavior of the base station to see if it matches the indicated state. For example, observe that the Monitor state indicator matches the actual behavior of the conventional station.
3. If the behavior does not match, refresh again. Repeat 2-4 times until the station is synchronized. Wait 10 seconds between each retry.

7.2.7.10 Base Station Down

If Symphony loses connectivity with a base station due to network or hardware issues, the module will be shaded as shown in Figure 7-15. Symphony will periodically attempt to reestablish communications with the base station, but you can manually check the base station status by pressing the **Refresh** button. When Symphony determines the base station is back up (automatically or manually), the module returns to its normal configuration.



Figure 7-15: Base Station Down Post-boot

The base station module is treated slightly differently if Symphony cannot establish communications with a base station during the boot sequence. In this case, Symphony does not know what channel the base station is on, so the Smart Button is left blank, as shown in Figure 7-16, until the base station comes back up.



Figure 7-16: Base Station Down Pre-boot

7.2.7.11 Base Station Call History

Base station call history is available from the same sources as trunking communications modules. See Sections 3.3.7, 3.4.1, and 3.5.2 for details. However, the call back feature works slightly differently for base stations. Figure 7-17 shows the History panel with a base station call highlighted. Note that the call was received on Red Mtn 1, but the call back module shows Red Mtn 2. The call back module shows the current channel of the base station, not necessarily the channel on which the highlighted call was received. Before using the call back feature, make sure the base station is on the desired channel. See Section 7.2.4 for details on changing the base station channel.



Figure 7-17: Base Station in History Panel

7.2.7.12 Emergencies

Conventional stations do not support group emergencies like trunked sites. Dispatchers cannot declare an emergency on a conventional channel since conventional stations do not support the feature. Therefore, the emergency tab is not available on a conventional station module.

Stations equipped with MDC-1200 can support unit emergencies declared by radios. These emergencies are mapped into VIDA as Unit Alerts and are handled as described in Section 5.11. Figure 7-18 shows a Unit Alert on a base station in the Emergency Sidebar Panel.



Figure 7-18: Unit Alert on Base Station with MDC-1200

7.3 PRIORITY MARKER TONE (PMT) AND SIGNAL TONE



NOTE

Since Priority Marker Tone is a feature-licensed option, this feature is only available if the console is equipped with this software license.

A Priority Marker Tone (PMT) is an audio tone that is sent over an RF channel (i.e., a talkgroup, patch or conventional channel) after a period of inactivity on that channel and is usually used to indicate an emergency or special condition on that channel. Typically, this is a short duration tone intended to be heard by mobile units. The console is responsible for sending the tone. The PMT feature is not associated with the console's existing Priority Group feature.

The Signal Tone is a short audio tone, similar to the PMT. However, it is sent at regular intervals on the channel.

PMT and Signal Tone are generically referred to as Marker Tone (MT). In the following discussion, Marker Tone will be used when the information applies to both PMT and Signal Tone.

Marker Tones are activated by the Marker Tone Button shown in Figure 7-19 and Figure 7-20. If your console is licensed for Marker Tones, the Marker Tone Button will appear on all normal size modules where Marker Tones are supported. Marker Tone is not supported on individual, intercom, or SimulSelect. The Marker Tone button will not appear on any module if your console is not licensed for the feature.



Figure 7-19: Marker Tone Button on Normal Module

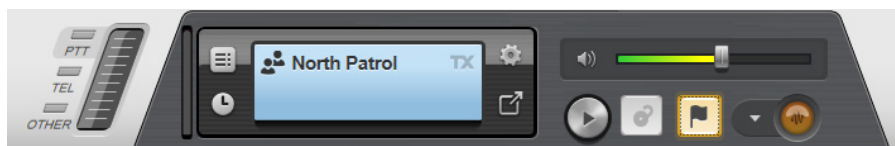


Figure 7-20: Active Marker Tone Button on Select Module

To enable Marker Tone on a group or channel, press the Marker Tone button on the associated entity's module or chose "Activate Marker Tone" from the context menu. The Marker Tone button will glow to indicate Marker Tone is enabled and the Marker Tone will be sent as soon as possible after the button is pressed and the group/channel is inactive. Subsequent tones will be sent based on the Marker Tone configuration. Consult with your System Administrator for details.

The console supports enabling Marker Tone on up to 10 entities simultaneously; however, the tones will be transmitted one at a time. That is if multiple Marker Tones need to be transmitted at the same time, they will be sent one at a time, not all simultaneously. This ensures the network is not overburdened with low priority traffic.

Marker Tones enabled by your console are considered local while those enabled by other consoles are considered remote. Remote Marker Tones are indicated by a glowing MT button with an “R” in the upper left corner as shown in Figure 7-21. When transmitting local Marker Tones or receiving remote Marker Tones, the tone is also played in the speaker assigned to the entity.



Figure 7-21: Active Remote Marker Tone Indication

To disable Marker Tone, press the Marker Tone Button again or chose “Deactivate Marker Tone” from the context menu. The button will stop glowing to indicate Marker Tone is disabled. If a Marker Tone is being transmitted when Marker Tone is disabled, the tone will terminate immediately. Any dispatcher can disable a remote Marker Tone as noted above.



It may take a few seconds to disable the remote Marker Tone. The Marker Tone button will return to the disabled state when the disabling process completes.

Marker Tones are sent in the background which means you can continue to dispatch while the tones are being sent. That is, you can transmit voice to one group while the Marker Tone is sent to another group. To minimize contention with voice traffic, the console delays transmission of the Marker Tone if the group/channel is active. When the group becomes inactive, the console transmits the Marker Tone with the lowest applicable priority on the system. Marker Tones are sent from a unique ID specifically configured for this purpose. That is, the primary dispatcher ID is not used for Marker Tones. If this ID is not configured, Marker Tones cannot be initiated. Consult with your system administrator to resolve this configuration issue.



On OpenSky and Conventional systems, the lowest applicable priority is below that of a radio, so inbound radio traffic will pre-empt the Marker Tone. On systems including P25 or P25C or BeOn, the lowest applicable priority is the same as a radio, so radios cannot pre-empt the Marker Tone. In the latter case, the tone is limited to a short duration (200 ms) and the recommended repetition time is greater than 20 seconds. This minimizes the time a Marker Tone can interfere with an inbound radio call.

By default, Marker Tones are not shown in the Call History Panel or Extended Dialog even if the Show Outbound box is checked. This prevents the call history from filling up with low priority Marker Tone call records. However, this behavior can be changed through configuration. See the *Symphony Installation and Configuration Manual* (14221-3100-4100) for details.

The directory includes useful information about Marker Tones as shown in Figure 7-22. The Active Marker Tone column shows the entities with active Marker Tones in the Groups and Base Station tabs. The Info tab provides detailed Marker Tone information including who enabled/disabled the tone. The System History, on the bottom of the full screen, also provides information about Marker Tones including Marker Tone activation, deactivation, and error messages.

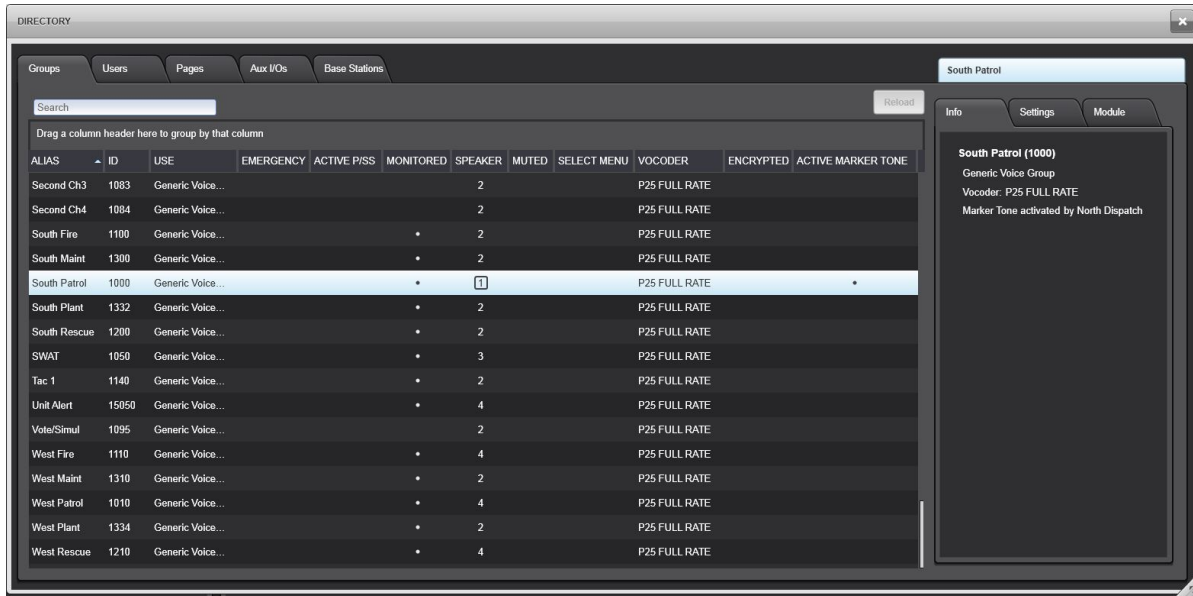


Figure 7-22: Marker Tone in the Directory

Marker Tone interacts with other features in the following ways:

- **Conventional:** Enabling MT on a conventional channel prevents channel changes on the associated base station until MT is disabled. The channel change arrows and menu are grayed out during this time. If the base station goes down, local or remote MTs on the base station are disabled. MTs must be enabled manually when the base station is back online.
- **Patch:** MTs must be disabled on any group/channel before it can be patched. MTs on a patch must be disabled before the patch can be torn down.
- **SimulSelects:** MT is not supported on SimulSelects; however, a MT can be enabled on a group/channel in SimulSelects.
- **Un-monitoring:** Local MTs must be disabled before a group/channel can be un-monitored.
- **Un-programming:** Local MTs must be disabled before a module containing the entity with the MT can be removed from the workspace.
- **Change Setups, user logout, or application shutdown:** Local MTs are disabled automatically when the user logs out, the setup is changed, or the console application is shutdown.

- Connectivity loss: Local MTs are disabled automatically when the connection to the VNIC is lost. Remote MTs are disabled on the user interface since the actual state is unknown. Once connectivity is restored, the remote MT states will be updated as the information becomes available. Local MTs must be re-enabled manually.
- Emergency: Group emergency and MT are compatible and independent features. That is, they work together and do not limit each other.
- Motorola Mode Emergency on a Patch: Motorola mode emergencies declared on a patch require changes to the patch, so the MT is automatically disabled to modify the patch. The MT must be restarted manually.
- Receive Only: Local MT must be disabled before making a module Receive Only. Receive Only modules cannot enable/disable MT on the associated module. The MT button will reflect the current Marker Tone state, but the button cannot be pressed.
- Paging: During a page/stack and subsequent auto SimulSelect, local MTs are temporarily disabled on the entities in the page/stack. This prevents a low priority MT from interfering with a higher priority paging tone.
- Marker Tone via EMG: The console supports Marker Tone on EDACS systems through an EDACS Migration Gateway (EMG) with the following restrictions:
 - MT tones generated from VIDA are heard on EDACS as a standard group call. There is no indication of the MT state of the VIDA group on EDACS.
 - MT tones generated from EDACS are heard on VIDA as a standard group call. There is no indication of the MT state of the EDACS group.
 - VIDA consoles cannot enable or disable MTs on the EDACS system.
 - EDACS consoles cannot enable or disable MTs on the VIDA system.

8 ADVANCED BACKUP RADIO (ABR)

The Advanced Backup Radio feature allows dispatchers to control the backup radio directly from the Symphony application. Dispatchers can seamlessly continue dispatch operations via the radio such as changing talkgroups, etc., in the event of a network connectivity loss to the Symphony application. The Symphony GUI essentially acts as a control head for the radio.



Since Advanced Backup Radio Control is a feature-licensed option, this feature is only available if the console is equipped with this software license. In addition, various system and console-related configuration changes must be made before this feature will be operational. Consult with system and/or dispatch administration personnel as necessary.

The Symphony Dispatch Console supports the advanced backup radio with special modules. This allows dispatch communications with the ABR modules operating on the respective advanced backup radio equipment without requiring a VIDA network connection.

Communications include:

- Receiving calls from P25 Trunked radio units.
- Transmitting calls to P25 Trunked radio units.
- Emergencies declared by P25 Trunked radios and itself.

Symphony currently supports the most common advanced backup radio type known as TnRn. This type of advanced backup radio supports one transmit/receive talkgroup pair at a time and can switch between n talkgroups, where n is up to 50.

In addition to the basic ability to transmit on talkgroups, the ABR has advanced features to maintain communication during emergency situations. These functions are listed below and are described in detail in the sections that follow:

- Emergency
- VNIC Disconnect
- Auto Select
- Select Shadowing
- Auto Mute

8.1 ADVANCED BACKUP RADIO MODULES

The Symphony advanced backup radio modules look and act similarly to normal trunked talkgroup modules and features a banner similar to that of the base station module. The TnRn advanced backup radio module is shown in Figure 8-1. This module is only available in the normal size. ABR modules are similar to regular communications modules except that they have a banner at the top and slight differences in the Smart Button and standard tabs. More importantly, ABR modules remain functional even after a VNIC disconnect.

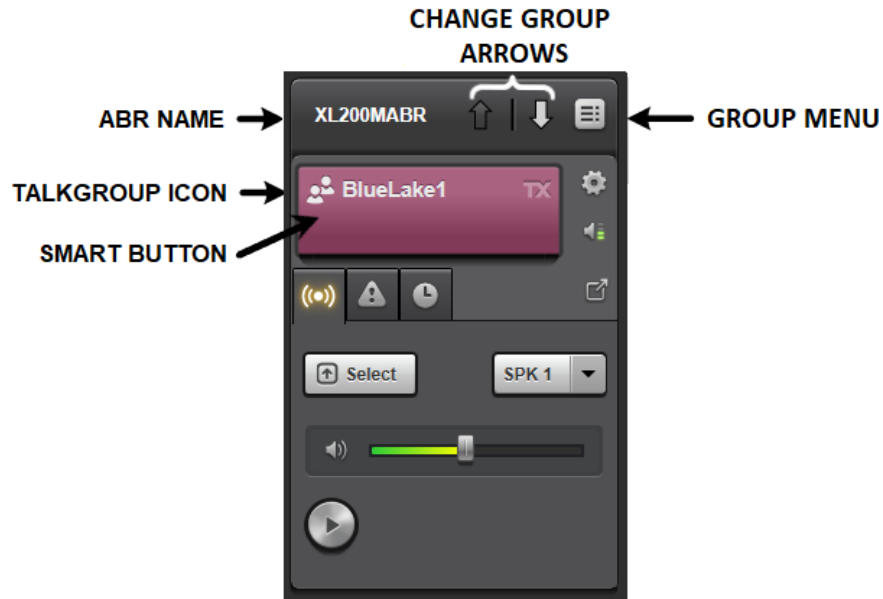


Figure 8-1: Advanced Backup Radio Module

The banner contains the radio name and the talkgroup change controls. Groups can be changed one at a time using the up or down arrows or selected directly from the group menu. The up/down arrows grey out when you reach the top or bottom of the group list respectively; the list does not roll over. The Smart Button depicts a talkgroup icon. The top line shows the current talkgroup alias or ID, and the bottom line shows the caller.

ABR Emergency and History tabs are similar to that of trunked communication modules. More information about ABR emergencies can be found in Section 8.7. The ABR History tab features the caller's name and time.

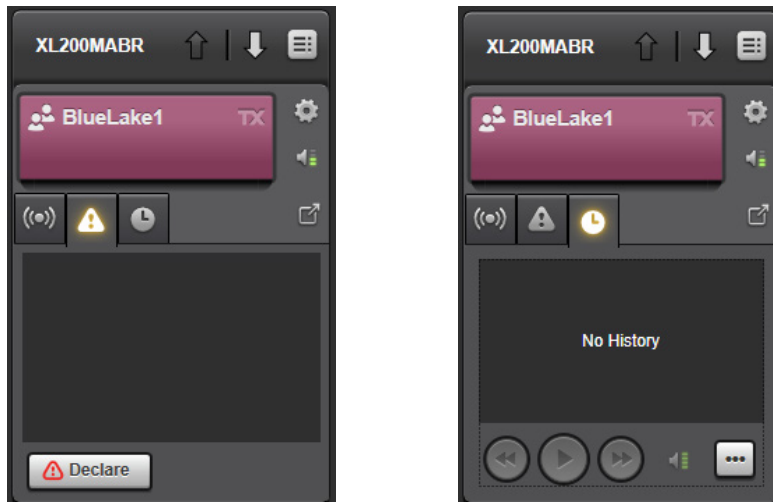


Figure 8-2: ABR Emergency Tab and History Tab

Advanced backup radio modules are available on the Baton as shown in Figure 2-1. Space limitations force the module to be a single line Smart Button which behaves like a mini module. Where space is available, a small banner with the advanced backup radio name is included.

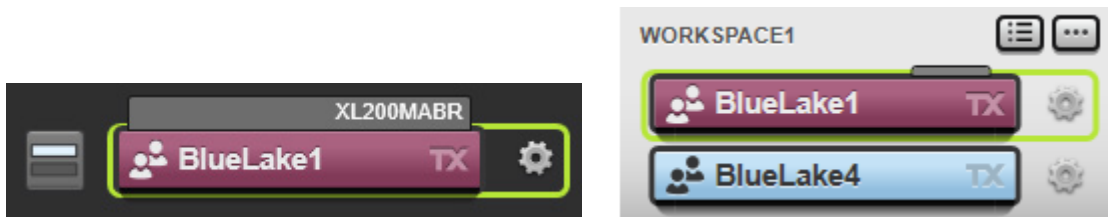


Figure 8-3: Baton ABR Modules

8.2 CHANGING TALKGROUPS

Unlike trunked modules, ABR modules can communicate on multiple talkgroups. Ensure that the advanced backup radio is set to the proper talkgroup so that the desired calls are heard and transmitted to.

To change the selected group:

1. Click the “Up” or “Down” arrows in the module banner.
 Or
 Click the Group Menu in the module banner and select the desired talkgroup.
 Or
 Click the Context Menu (gear icon), select **Groups**, and then select the desired talkgroup.
2. Symphony updates the talkgroup name on the Smart Button to indicate the change.

8.3 SELECTING ABR MODULES

ABR modules can be selected, i.e., placed in the selected module as show in Figure 8-4. You are selecting the advanced backup radio, not just the current group of the ABR. Any calls received from, or transmitted to, the advanced backup radio are treated as selected. Changing talkgroups on the advanced backup radio updates the smart button but does not affect the select state of the advanced backup radio.



Figure 8-4: ABR in Select Module

As with trunked modules, selecting an advanced backup radio module places it in the Select Module, routes received audio from the station to the select speaker, and allows you to transmit on the current talkgroup with the footswitch, desk mic, Quick Action keys, or Smart button on the GUI. ABR modules use the same selection methods as trunked modules. See Section 5.2 for details on module selection.

The Select Module indicates the advanced backup radio name in a small banner and includes the up/down arrows to change the talkgroups. The Context Menu includes change **Group** options.

8.4 ABR SELECT SHADOWING

When the ABR is unselected but one of its corresponding talkgroups is selected, the ABR will update itself to match the selected talkgroup. This change is visible on the Smart Button. This ensures that if the dispatcher was using a talkgroup that existed on the ABR, they can easily continue that conversation on the ABR if the VNIC connection drops.

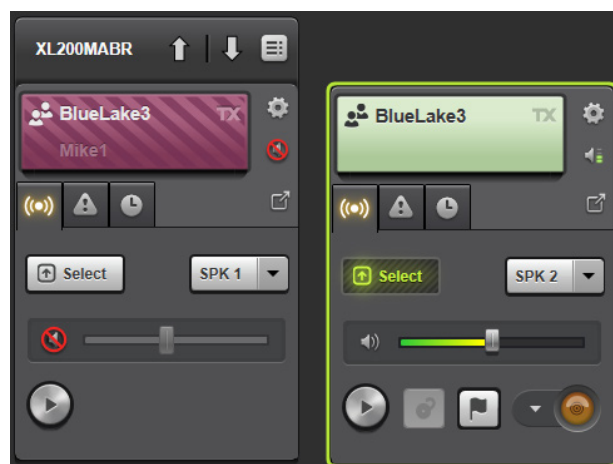


Figure 8-5: ABR Select Shadowing

8.5 PATCHES AND SIMULSELECTS

See Sections 5.8 and 5.9 for the procedures to create patches and SimulSelects. An advanced backup radio cannot be added to a patch or SimulSelect, but its corresponding talkgroups can be added. If one or more ABR programmed talkgroups are added to a patch and then that patch is selected while select shadowing is enabled, the ABR will shadow one of the talkgroups.

8.6 ON SYMPHONY STARTUP

If the VNIC is disconnected during Symphony Startup, the dispatcher gets the option to continue, which allows the Symphony application to run using only the ABR. In this case, the ABR only refers to talkgroups through their IDs.

8.7 EMERGENCY

The ABR module supports group emergencies via the Emergency tab, shown in Figure 8-6. Dispatchers can declare an emergency from the ABR module or the corresponding group module. Regardless of where the emergency is declared, both modules will enter the emergency state. When declared from the ABR, the notification appears on the emergency side panel (shown in Figure 8-7).

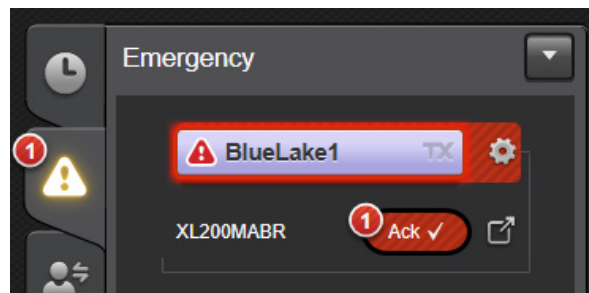


Figure 8-6: Emergency Panel during ABR Emergency

In an emergency, the ABR module does not need to be acknowledged and only features the **Clear** button. The ABR's emergency declare tone is heard only during a VNIC disconnect. To silence the tone, press the tone button in the top right corner.

An emergency can only be cleared from the ABR module when the VNIC is disconnected. During normal operation, the ABR emergency is cleared when the corresponding talkgroup emergency is cleared. More information about emergencies can be found in Section 5.11.

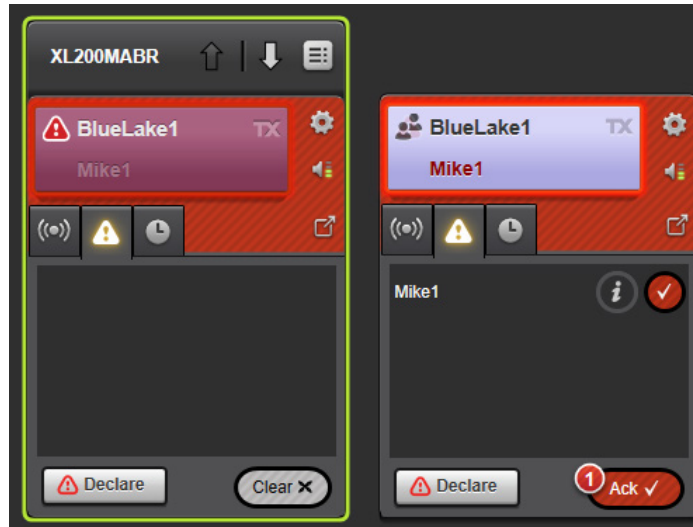


Figure 8-7: ABR and Talkgroup in Emergency

8.8 VNIC DISCONNECT

The ABR connection is maintained through a separate network connection that stays up in case of a VNIC network failure/shutdown. The backup radio uses RF to communicate to the site and ensures that the dispatcher consistently has a way to communicate.



Figure 8-8: ABR and talkgroups after VNIC Disconnect

8.9 ABR AUTO SELECT

During a VNIC failover event, every module on the symphony console becomes inactive except for the advanced backup radio. The ABR Auto Select feature ensures that if the VNIC goes down, the ABR module is automatically selected and can immediately be put into use by the dispatcher.

8.10 ABR AUTO MUTE

While unselected, the advanced backup radio automatically mutes to prevent duplicate audio streams. When selected, the ABR automatically unmutes.

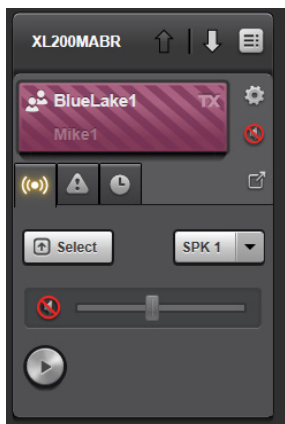


Figure 8-9: Muted ABR module

8.11 ABR DOWN

If Symphony loses connectivity with the advanced backup radio due to hardware issues, the module is shaded as shown in Figure 8-10. The ABR module automatically attempts to reconnect every 15 seconds after disconnecting. Ensure that the ABR is fully connected and powered on before pressing the **Refresh** button. Verify this by opening Network Settings and checking that the ABR is connected.



While the Advanced Backup Radio is reconnecting, the Dispatcher may hear a short ring tone in the operator headset.

When Symphony determines that the advanced backup radio is back up, the module returns to its normal configuration. The advanced backup radio module is treated slightly differently if Symphony cannot establish communications with the advanced backup radio during the boot sequence. In this case, Symphony does not know what talkgroup the advanced backup radio is on, so the Smart Button is left blank until the advanced backup radio comes back up.



Figure 8-10: ABR Down Post-Boot and Pre-Boot

9 ADVANCED OPERATIONS

9.1 AUX I/O



Since auxiliary I/O monitoring and control capability at the console is a feature-licensed option, this feature is only available if the console is equipped with this software license. In addition, various system and console-related configuration changes must be made before this feature will be operational. Consult with system and/or dispatch administration personnel as necessary.

Using special modules referred to as “Auxiliary I/O Modules,” two-state auxiliary inputs and outputs can be monitored and controlled at the console. For example, one auxiliary input module might be utilized to monitor the state of a door (either open or closed) via a sensor switch at the door. An auxiliary output module might be utilized to turn a relay-controlled siren on and off via a button press or keystroke action at the console.

Symphony monitors remote I/O points in the VIDA system as well as local I/O points on the Symphony Dispatch Platform. The Aux I/O controls defined in this section support local and remote I/O points. No distinction is made between the two I/O types on the user interface.

9.1.1 Aux I/O Modules

Figure 9-1 shows several types of Aux I/O modules. Each module provides a summary of the Aux I/O point in a Smart Display for inputs or Smart Button for outputs. Smart Displays and Smart Buttons contain the following information:

- Aux I/O Icon in the top left corner.
- Aux I/O Name on the top line for small modules.
- Aux I/O Type Icon in the top right corner.
- Aux I/O State Text on the bottom line for small modules. The text states the current state, e.g., “Siren On” or “Door Closed.”
- Aux I/O State Indication on the right portion of the Smart Button or Display. This indication turns white when the I/O point is active.

- Aux I/O Context Menu: This menu provides Aux I/O specific functions including:
 Open Directory: The function opens the Directory Aux I/O tab with the module I/O point highlighted.

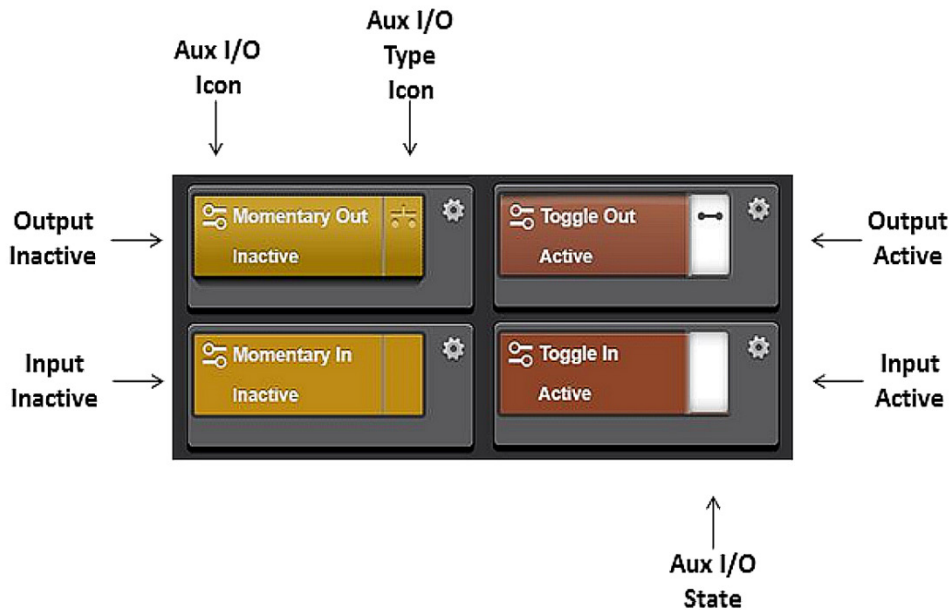


Figure 9-1: Summary of Aux I/O Module Types

Smart Buttons on Aux I/O modules can be pressed to change the state of an output as described below.

Aux I/O modules are available on the Baton. The modules are smaller as shown in Figure 9-2. The Smart Button and Display are compressed to a single line. This single line module contains all the information of the larger Aux I/O modules except for the Text name of the current state. However, the state can be determined from the Aux I/O State Indication.

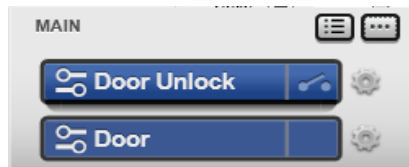


Figure 9-2: Baton Aux I/O Module

9.1.1.1 Auxiliary Input Modules

9.1.1.1.1 Standard Operation

Auxiliary input modules that are set up for standard input operation and provide monitoring functions for the respective auxiliary inputs. No dispatcher response is required at each auxiliary input state change. When an input changes state, the Aux I/O State Text and Indication change to reflect the state of the input as shown in Figure 9-1.

9.1.1.1.2 Acknowledge Operation

Auxiliary input modules that are set up for acknowledge operation require a dispatcher to acknowledge an inactive-to-active state change. On the inactive-to-active state change, the state text portion of the Auxiliary input module changes to its active text, a warning icon is shown, and an alarm tone is heard on the speaker configured to play Aux I/O alarm tones. The warning icon continues to be displayed on the module and the alarm tone continues to play until the Auxiliary input is acknowledged or it becomes inactive. When the input is acknowledged, the warning icon disappears and a checked icon appears to indicate the module is acknowledged. However, the state text continues to show the active state text until the input returns to its inactive state.

An active Aux I/O input alarm can also be acknowledged by a dispatcher on a different console. When this happens, the checked icon on the module has a triangular badge to indicate remote acknowledgement (Figure 9-3).



Figure 9-3: Auxiliary Input Acknowledged Module

On the Baton, Auxiliary input acknowledged modules are smaller (Figure 9-4). The Smart Button is limited to a single line. This single line module contains all of the information of the larger Aux I/O modules except for the text name of the current state. However, the state can be determined from the Aux I/O State Indication. When the input is active, the Baton module flips between the input name and the text name of the current state. It continues to do that for as long as the input is active.

The point name text and active text with a warning icon is shown periodically when the point changes from inactive to active state. When the input is acknowledged, the warning icon disappears and a checked icon appears to indicate the module is acknowledged. The checked icon with a triangular badge appears when the point is acknowledged by a different console.



Figure 9-4: Auxiliary Input Acknowledged Module on Baton

An Input Alarm badge indicating the number of active unacknowledged Aux I/O inputs is displayed on the Workspace tab of the Full Screen display.



Figure 9-5: Aux I/O Alarm Badges

9.1.1.2 Auxiliary Output Modules

9.1.1.2.1 Momentary Operation

Auxiliary output modules that are set up for momentary operation allow momentary activation of auxiliary outputs. In other words, the output changes to the active state while the Smart Button is depressed. The Aux I/O State Text and Indication (on the Smart Button) change to reflect the state of the output as shown in Figure 9-6. In this example, the Momentary Out I/O point is looped back to the Momentary In I/O point. When the Momentary Out button is not depressed, the output state is inactive so the Momentary In is inactive as well. This is shown in the left side of the figure. When the Momentary Out button is depressed, the output becomes active and the Momentary In state changes to active as well. This is shown in the right side of the figure.

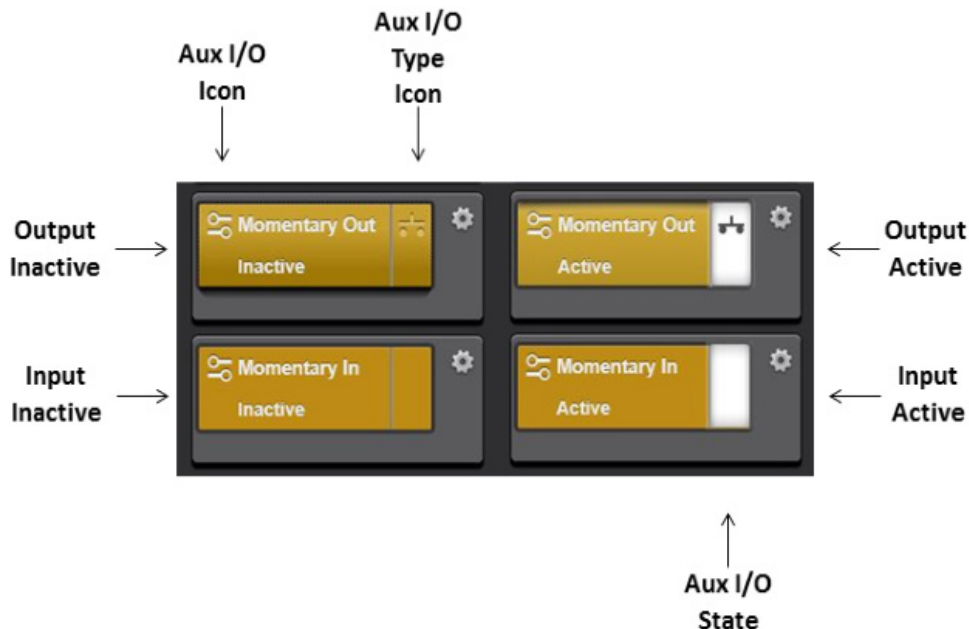


Figure 9-6: Aux I/O Momentary States

In some cases, other (“remote”) consoles in the system may also be set up to control the same auxiliary outputs as the local console. In these cases, the corresponding auxiliary output modules’ state indicators may change state when no button or keystroke action occurs at the local console. This updating process is by design; it guarantees that the state indicators at all consoles within the system always indicate the current states of the respective auxiliary outputs.

9.1.1.2.2 Toggle Operation

Auxiliary output modules that are set up for toggle operation provide steady-state toggling between two possible output states. In other words, via a Smart Button press and release action, the output can be changed to one state and it will remain in that state until another similar or identical button action is performed. The Aux I/O State Text and Indication (on the Smart Button) change to reflect the state of the output as shown in Figure 9-7. In this example, the Toggle Out I/O point is looped back to the Toggle In I/O point. When the Toggle Out button is not depressed, the output state is inactive so the Toggle In is inactive as well. This is shown in the left side of the figure. When the Toggle Out button is depressed, the output becomes active and the Toggle In state changes to active as well. This is shown in the right side of the figure.

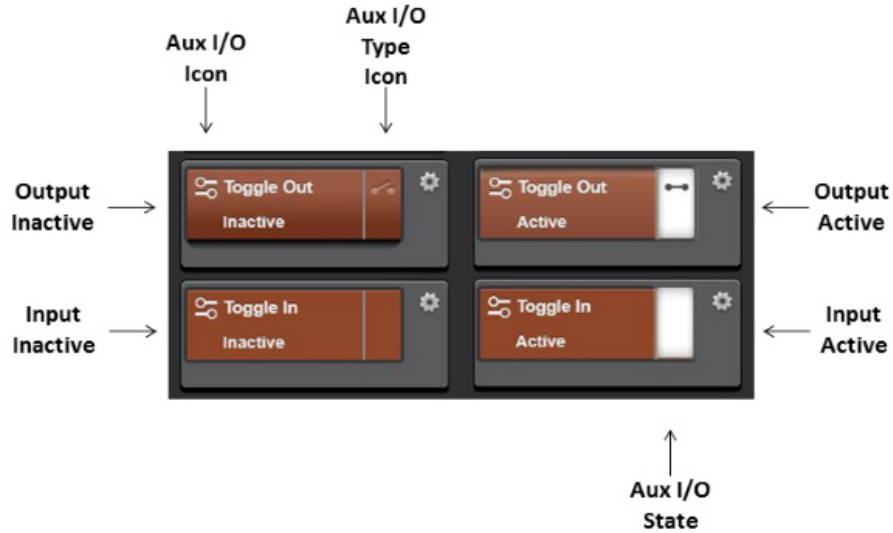


Figure 9-7: Aux I/O Toggle States

In some cases, other (“remote”) consoles in the system may also be set up to control the same auxiliary outputs as the local console. In these cases, the corresponding auxiliary output modules’ state indicators may change state when no button action occurs at the local console. This updating process is by design; it guarantees that the state indicators at all consoles within the system always indicate the current states of the respective auxiliary outputs.

9.1.2 Aux I/O Directory Tab

The Directory dialog’s Aux I/O Tab provides a summary of all Aux I/O points as shown in Figure 9-8. See Section 3.5.1 for details on basic directory usage. This section focuses on the Aux I/O tab. Table 9-1 describes each column in the main dialog list.

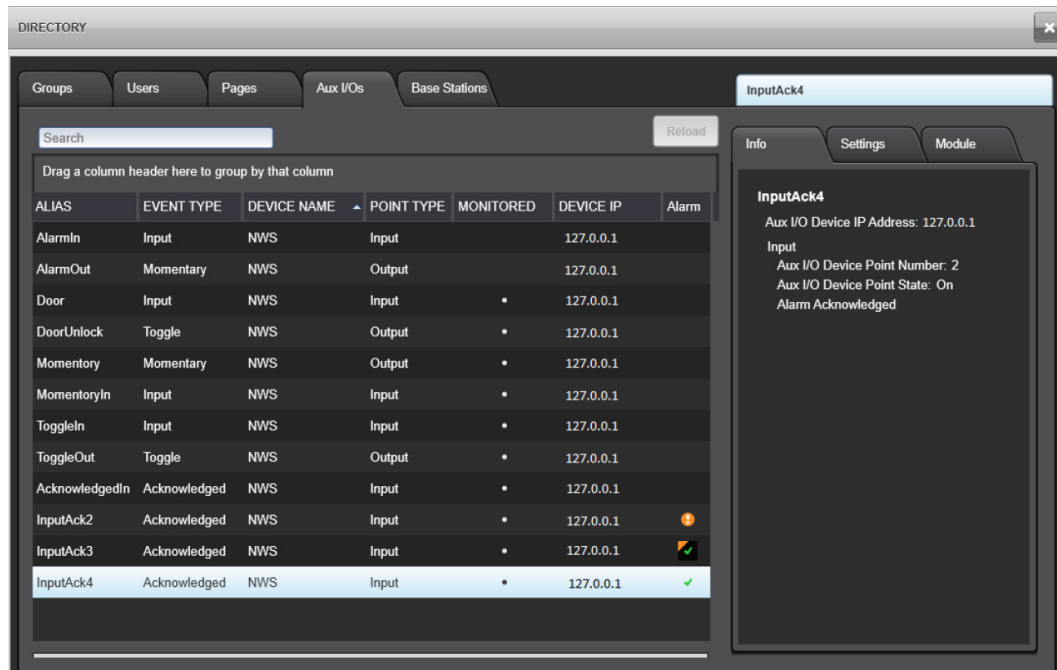


Figure 9-8: Directory Dialog Aux I/O Tab

Table 9-1: Directory Columns – Aux I/O Tab

COLUMN	DESCRIPTION
ALIAS	The alias for the I/O point.
POINT TYPE	Input or Output.
DEVICE NAME	Name of the Network Sentry or Local Aux I/O that supports the physical I/O point.
EVENT TYPE	For outputs, this can be “Toggle,” “Momentary,” or “Special Event.” For inputs, this is “Input.” Toggled outputs remain in the active state until the user initiates a change to the inactive state. Momentary outputs only stay in the active state while the user is initiating the activation control (i.e., pressing the output activation button). Special Event outputs are internal Symphony events that drive a light pole such as “Console on Air.”
MONITORED	Console actively monitors the I/O Point and reports changes in state.
DEVICE IP	IP Address of the Network Sentry that supports the physical I/O point.
ALARM	Shows a warning icon when the I/O Point has an active alarm or has changed from inactive to active state; a checked icon when the I/O point is acknowledged; or a checked icon with an orange badge when acknowledged from a different console.

The panel on the right side of the Directory Dialog provides additional tabs that apply to the highlighted Aux I/O point(s) in the I/O point list.



NOTE

Selecting multiple I/O points only applies to the Settings tab.

- Info Tab:** This tab provides detailed information about the highlighted I/O point including:
- Aux I/O Device Point Number: The I/O point number on the Network Sentry that corresponds with the physical I/O point.
 - Aux I/O Device Point State: Text description of the current I/O state.
- Settings Tab:** This tab enables or disables the monitoring of the highlighted Aux I/O points. Check the “Monitor Aux I/O Point with console” checkbox to enable monitoring.
- Module Tab:** This tab provides the Aux I/O module for the highlighted Aux I/O point.

9.2 RADIO STATUS MONITORING (RSM)



NOTE

Since Radio Status Monitoring is a feature-licensed option, this feature is only available if the console is equipped with this software license.

The Radio Status Monitor feature (also known as Status Message Monitor) allows visual monitoring of status messages transmitted by trunked radio units. For example, one unit could transmit an “en route” status and another unit could transmit an “at scene” status. At the console, each unit is identified by the status that it transmitted, its unit name or unit ID number, and the time at which the status transmission was received by the console.

For a given trunked radio unit, only one status message is valid during any given instant. For example, a unit could not be “en route” and “at scene” at the same time. The last transmitted status message received by the console for a particular unit is considered the currently valid status. However, several different trunked radio units can transmit identical status messages and therefore they would be in the same status during any given instant. For example, several units could be “en route” at the same time. It is the responsibility of a trunked radio unit user to transmit an appropriate status message when necessary to update his/her status. Units that have never transmitted any status message or that do not support status messaging have no status from the radio system’s point-of-view.

9.2.1 Radio Status Message Modules

A special console module called a “Radio Status Message (RSM) module” provides radio status message indications at the console. Figure 9-9 shows an RSM module.



Figure 9-9: Radio Status Message (RSM) Module

The RSM module comprises the following:

- Banner at the top which includes from left to right:
 - Module Type: An indication this is an RSM module (“RSM”).
 - Status Code: The status represented by the module.
 - Entry Count: The total count of radios in this state.
 - Context Menu: Gear icon to access context menu items including Pop-out the module.
 - Module Color: Color swatch assigned to the RSM module (on bottom of banner). This helps the dispatcher quickly distinguish one RSM module from another.
- Radio List: A list of all the radios with this status including the radio alias and the time the status was sent. The newest entry is placed at the top of the list.
- Mini Modules: Mini modules are provided to return an I-Call or group call to the radio highlighted in the list. If the group information is not available, the group mini module is grayed out. Highlight the radio of interest and the mini modules update with the appropriate info for the chosen radio.

If the “route all status messages” option is disabled (the default), it prevents the console from receiving status message transmissions from groups that are not monitored at the console. To determine if this configuration option is enabled or disabled, consult with the system or dispatch administration personnel. Radio unit users should be informed if this option is disabled, as they will have to log into a different group (a group programmed at the respective dispatch console) before transmitting a status message. To determine if a talkgroup is monitored, consult the “Settings” tab in the Group directory.

If a status code is assigned to an RSM module, the status message is added to the module. If the code is not assigned to an RSM module, it is not shown anywhere. If a radio sends a status message that does not have a corresponding module, but is defined in the Symphony configuration tool, it will be removed from its current RSM module.

RSM modules are not available on the Symphony Baton. This feature must be used from the Full Screen application.



NOTE

The RSM Side Bar Panel can be opened using the **RSM** Baton Widget button.

9.2.2 Recommendations and Limitations of Status Messages

The following recommendations and limitations should be noted:

- All status history is lost whenever a login occurs. This means that the information developed over time can be lost when a link failure occurs between the console and the VNIC, when the setup is changed, or when the console is re-started.
- It is the responsibility of a trunked radio unit user to transmit an appropriate status message when necessary to update his/her status.
- It is the System Administrator’s responsibility to properly configure radios to send the proper status message. See the *Symphony Installation and Configuration Manual* (14221-3100-4100) for details.

- Units that have never transmitted any status message or that do not support status messaging have no status from the radio system's point-of-view.
- A status message is transmitted on a radio frequency control channel, not on a radio frequency voice ("working") channel. Therefore, status message communications are more efficient than standard voice transmissions. Consequently, if the radio system and the respective console are adequately equipped, trunked radio unit users should use this feature whenever possible instead of standard voice transmissions, especially during busy periods. Transmitting status messages instead of standard voice transmissions may help reduce call queuing during busy periods.
- By design, no transmit or receive audio is associated with the status message operations. If voice audio communications are necessary to a unit that has transmitted a particular status, communicate via group or individual call operations.

9.3 ENHANCED REQUEST TO TALK (ERTT)



NOTE

Enhanced Request to Talk is a feature-licensed option. This feature is only available if the console is equipped with this software license.

The Enhanced Request-To-Talk (ERTT) feature provides an orderly way to manage dispatch communications in a busy dispatch center. Rather than having field units place voice calls to the dispatcher requesting service, the unit sends a short control message known as a Request-to-Talk (RTT) message or RTT. RTT messages have a unique code that indicates the type of service required such as "Need tow truck" or "License plate check". RTT messages are also associated with the radio's current talkgroup. The console can be configured to only receive RTT messages on certain groups. The exact codes and group associations are configured by the System Administrator. See the *Symphony Installation and Configuration Manual* (14221-3100-4100) for details or consult your Systems Administrator for more information.

The ERTT feature organizes the RTT messages into modules associated with each RTT code. This allows you to quickly see how many units have pending requests for each RTT code. The modules also allow you to quickly respond to an RTT message using a group or individual call.

This mode of operations has many advantages including:

- RTT messages are short control channel messages. No voice resources are required which frees up system capacity.
- The ERTT modules save and organize RTT messages so there is no need to remember verbal requests for service. Scan the ERTT modules to see all the pending requests.
- RTT messages can be prioritized within an ERTT module making it easy to find the highest priority RTT messages.
- Processing time is reduced with the Next Call feature that picks the next highest priority unanswered RTT in the ERTT module and optionally auto-selects the associated RTT group.
- RTT messages can trigger alarms to ensure new RTTs are not missed.
- Dispatchers can share a pool of RTT messages further enhancing the efficiency of dispatch operations. Actions from one dispatcher will be reflected on the consoles of other dispatchers.

The ERTT feature is similar to the Radio Status Message feature in that both use short messages to convey information about a unit. However, RSM tracks the unit status (i.e., On Duty, In Pursuit) while ERTT tracks requests for service. A radio can only have one RSM status at a time but may have multiple RTTs at once depending on configuration. Both features can be used simultaneously.

- The following terms are used to describe the ERTT feature:
 - ERTT: This refers to the Enhanced Request to Talk feature that manages RTT messages.
 - RTT Message or RTT: The Request to Talk message sent by a unit to the console.
 - RTT code: Each RTT message contains a code to identify the type of service being requested such as “Need tow truck” or “Set up patch.”
 - Service: The service being requested based on the RTT code.
 - RTT associated group. Each RTT message includes the current talkgroup of the issuing unit. This information is used to control RTT access, determine the RTT priority and to reply to the issuing unit with a group call.
 - ERTT Module: Each ERTT Module shows the pending RTT messages with a specific RTT code. For example, one ERTT module may handle the “Need tow truck” while another handles “Set up patch.”
 - RTT Reply: The dispatcher replies to an RTT by placing an I-Call or group call to the issuing unit using the ERTT mini modules. Calling the unit from non ERTT group or unit modules does not constitute an RTT reply.
 - Unanswered RTT: An RTT message in an ERTT Module to which the dispatcher has not replied using the ERTT mini modules.
 - Unique RTT: When comparing RTTs from the same unit, the RTTs are unique if they have a different RTT code and associated talkgroup.
 - New RTT: An RTT is new when it is Unique or when it has previously been deleted and it is received again.
 - Duplicate RTT: When comparing RTTs from the same unit, the RTTs are duplicates if they have the same RTT code or associated talkgroup.
- For C3 Maestro users, the Symphony ERTT feature combines the C3 Maestro RTT and ERTT functionality into a single feature.

9.3.1 ERTT Module

The ERTT Module, shown in Figure 9-10, is the primary interface element for the ERTT feature. It organizes and manages all the RTT messages with the RTT code assigned to the module.

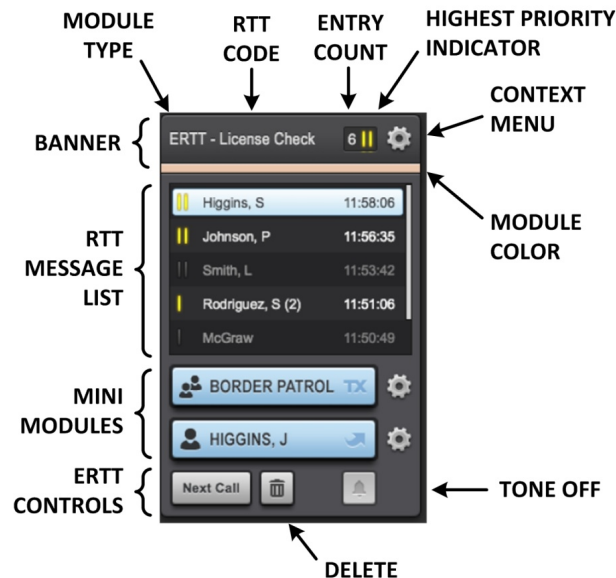


Figure 9-10: Enhanced Request to Talk (ERTT) Module

The ERTT module comprises the following controls and indications:

- Banner: The banner at the top provides the following summary information:
 - Module Type: An indication of the module type (ERTT).
 - RTT Code: The RTT code or service managed by the module (i.e., “License Check”).
 - Entry Count: The total number of unhandled list entries waiting for this service.
 - Highest Priority Indicator: This indicates the highest priority unanswered entry. Two yellow vertical lines indicate a High priority, and one yellow vertical line indicates a Normal priority. No lines are shown if all RTTs have been answered.
 - Context Menu: Gear icon to access menu items including:
 - Pop-out the module.
 - RTT Alarms On/Off. This menu item turns global RTT Alarms on or off. It contains an icon which shows the current status of the global RTT Alarm (the icon is enabled if the global RTT Alarm is enabled); as well as text showing what action is available (e.g., if the global RTT Alarm is enabled, the text will read “Disable Global RTT Alarm” and vice versa).
- Module Color: Color assigned to the RTT module. This helps quickly distinguish one RTT module from another.

- RTT Message List: The list of RTT messages waiting for this service. The exact RTT messages listed, and their order, are controlled by configuration as explained later in this section. The entry listing includes:
 - Priority Icon: Indicates the priority of the entry (High or Normal). This uses the same icons as the High Priority Indicator.
 - Alias for the unit that generated the RTT message. The alias can be configured to show the unit alias or RTT associated group alias. The default is unit alias. If the UI is configured to show IDs instead of aliases, then the alias is replaced by the appropriate ID. This setting is controlled by the console administrator.
 - Request count: A count of times the request has been sent and not answered. The count follows the alias in parentheses.
 - Time associated with the list entry. The timestamp can be configured to maintain the timestamp of the first RTT message or to show the time when the last duplicate RTT was received.
 - Answered calls are indicated with gray text and gray priority icon.
- Mini Modules: Mini modules return an I-Call or group call to the issuing radio of the highlighted entry. If the mini module is not valid for any reason, it will not be populated with a Smart button.
- ERTT Control buttons:
 - Next Call Button: The Next Call button chooses the highest priority unanswered RTT.
 - Delete Button: The Delete button removes the highlighted entry from the module.
 - Tone Off Button: The Tone Off button terminates the RTT Ringing tone(s) associated with this module for any existing entries. The ringing tone will return when a new RTT message arrives on the module.

9.3.1.1 RTT Message List Control

The messages that appear in the RTT Message List are configured by the System Administrator. This section describes the behavior of the different configuration settings. Consult your Systems Administrator to determine how your console is configured.

Depending on the system configuration, RTTs from a user can have different RTT codes and/or different associated talkgroups. RTTs with the same RTT code and talkgroup (known as duplicates) update the count associated with the current entry already in the RTT Message List. However, unique RTTs with a different code or talkgroup can lead to multiple entries and therefore require special handling.

The ERTT feature can be configured to handle multiple unique RTTs from a user in one of the following ways:

- **Show All Unique RTTs From User.** Symphony accepts all unique RTTs from the user and displays them accordingly. This can lead to multiple entries in the same RTT module (i.e., same code with different talkgroups) and/or different RTTs in different RTT modules (i.e., same talkgroup with different codes).
- **Show Most Recent RTT From User.** Symphony only allows the latest unique RTT from a user. The previous different RTT from that user is deleted.

- **Show Most Recent RTT From User On Each Talkgroup.** Symphony deletes the previous RTT from a user if it has the same talkgroup as the new RTT from that user. This allows a user to have multiple RTTs, but only one per talkgroup.

In the last two modes, existing RTTs can be removed when a new RTT is received for an RTT code that is defined in the Symphony Config, but not programmed into modules. This results in the old RTT being deleted with none taking its place (the new RTT is in a module that is not programmed on the screen).

9.3.1.2 RTT Message Priority

Each RTT message is assigned a priority based on the priority of the RTT associated group. The priority levels are: High, Normal, or None. RTT Message List entries are ordered with High priority entries ahead of Normal priority entries. Entries assigned the None priority are not displayed, thereby providing a means to control which consoles see which RTT messages. Within each priority, the RTT list has the oldest message at the top of the list. The overall order is oldest High priority to newest High priority, followed by oldest Normal priority to newest Normal priority. If only a single priority is used, the order is oldest to newest.

The priority of each entry is indicated by the Priority Icon. Two yellow vertical lines indicate High priority, and one yellow vertical line indicates Normal priority.

RTT group priorities are set in the Directory as shown in Figure 9-11. The default priority is Normal. To change the group priority:

1. Highlight the desired group (or groups using shift+click) and choose the **Settings** tab on the right side of the Directory dialog.
2. Choose the desired priority from the RTT Priority Menu.

The group RTT priority assignments are stored as part of the module programming information of the current console setup. When the setup is changed or the console application is restarted, the priority levels are set to the last saved state for the setup. Changes to a group's RTT priority level are not reflected in the ERTT modules until the next RTT message associated with the changed group is received.

RTT priority changes do not affect older RTT messages. They are applied to new RTTs only. In the case that a new RTT is a duplicate, the old RTT is deleted and the new RTT, with the new priority, is added as if the RTT were just received. It is placed at the bottom of the relevant priority section.

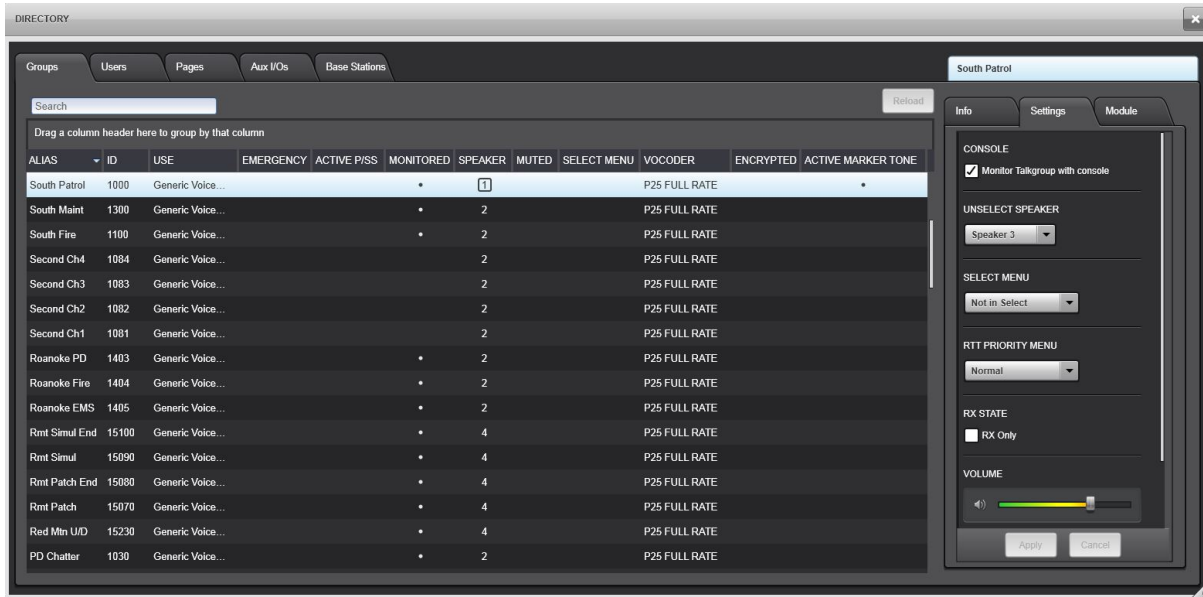


Figure 9-11: Setting RTT Group Priority

9.3.1.3 RTT Alarms



An alarm for a specific RTT message will be heard if the RTT Message Priority is set to High or Normal for the associated talkgroup (see Section 9.3.1.2), the corresponding ERTT Module is programmed on a Workspace tab for the current setup, and RTT Alarms are enabled.

The RTT Alarms option provides the user with an audible indication that an RTT message has been received at the console. Separate ring tones can be configured for High and Normal priority RTTs. Only one of the two tones is active at a time. The High priority tone pre-empts the Normal priority tone. The Normal priority tone is only generated if the High priority tone has been answered, deleted, or muted.

The feature is controlled by configuration which sets the initial state of RTT Alarms to on or off and determines whether the user can change the setting during normal operation. If the user has permission to change the setting, the RTT Alarms can be turned on or off from the Preferences submenu in the Main menu or from the context menu of any ERTT module. The specific menu item will be **Preferences** → **Disable Global RTT Alarms** or **Enable Global RTT Alarms** to turn the alarms on and off respectively. Changing the RTT Alarm setting applies only to the processing of subsequent RTT messages. If the RTT alarms are turned off while the RTT ringing tone is being generated, the tone continues until reset. If the user is not given permission to change the RTT Alarm setting, the menu items are not present.

If alarms are on, each ERTT module generates ringing demand based on its highest priority unanswered RTT that has not been silenced. Symphony generates one tone for the highest priority tone request from all the ERTT modules. Each time a tone is silenced, Symphony re-evaluates the pending demand (i.e., unanswered RTTs that have not been silenced) and sounds the highest pending priority tone.

Tones can be silenced with several levels of granularity:

- To silence a highlighted RTT tone, reply to the initiating unit via the ERTT mini modules press the Next Call button or delete the entry from the RTT Message List. Pressing the Next Call button with Auto-Select enabled will also silence the RTT chosen by the Next Call feature. Ringing may continue if there are other unanswered RTTs at the console that have not been silenced.
- To silence all current RTT tones in an ERTT module, press the Tone Off button on the appropriate ERTT module. Ringing may continue if there is pending demand in other ERTT modules. A new RTT to the silenced module restarts RTT ringing.
- To silence all current RTT tones on all ERTT modules and any other active alarm, press the Master Tone Off button on the Select Bar. New RTTs to any ERTT module will restart RTT ringing.
- To disable all future RTT tones, turn RTT Alarms off using the context menu on any ERTT module or the Preferences Menu in the Main Menu. Pending tones are not affected by this setting and will continue until all are silenced.

The RTT Alarm can be delayed for a configurable time after replying to or managing an RTT. Delay values in the range of 10 seconds provide an alarm free time to communicate with the RTT initiator. Longer values in the range of a minute or more serve as a reminder that RTTs remain to be serviced. The delay timer is started/restarted after replying with an RTT mini module or after the Next Call or Delete button is pressed. Pending alarms are silenced while the timer is active. When the timer expires an alarm will be sounded if there is pending demand.

9.3.1.4 Enhanced RTT Module Controls

The ERTT module provides the following control buttons to manage RTT messages.

Next Call Button: The Next Call button chooses the highest priority unanswered RTT in the RTT Message List. Any RTT alarm tone that is currently active in the associated RTT module is silenced until reminder delay timer expires (5 seconds by default). The feature can optionally be configured to automatically select the RTT associated group, if Auto-Select is enabled. Additionally, Next Call can be configured to remove the previously highlighted entry from the list if Next Call Delete is enabled.

Delete Button: The Delete button removes the highlighted entry from the RTT Message List. Deleting an unanswered RTT message silences the alarm associated with the deleted RTT.

Tone Off Button: The Tone Off button terminates the RTT Ringing tone(s) associated with this module. The button has two states: grayed out and active. In the grayed-out state, the button cannot be pressed. This indicates that alarms on the associated ERTT module have been silenced. In the active state, the button is raised and can be pressed. This indicates the associated ERTT module has active alarms. Pressing the button stops all alarms for the ERTT module and the button switches to the grayed-out state until a new alarm is generated on the ERTT module.

9.3.2 ERTT Message Operations

This section provides a few examples of how to use the ERTT feature to manage RTT Messages.

9.3.2.1 Service RTT Messages – None Unanswered

Use the following steps to manage an RTT with RTT Alarms on. This example assumes there are no unanswered RTTs and therefore no active RTT alarms:

1. When a new RTT is received, the console sounds an alarm tone based on the priority of the RTT.
2. Observe the ERTT module with the Priority Indicator set (no others should be set).
3. Press the **Next Call** button to highlight the entry and silence the tone. The mini modules populate based on the RTT Message data. If this is the first RTT in the module, it is already highlighted by default.
4. Press the group or I-Call mini module to reply to the unit initiating the RTT message.
5. Once service has been rendered, highlight the RTT (if not already highlighted) and press the Delete button to remove the RTT from the RTT Message List.

9.3.2.2 Service RTT Messages – Many Unanswered

When you have many unanswered RTTs pending in multiple RTT modules, it is difficult to know which RTT to process. ERTT helps you find and process the RTTs in priority order. The following scenario is one example of using ERTT functionality to simplify RTT operations.

Use the following steps to find the next RTT to process. This example assumes there are multiple unanswered RTTs and demand for multiple RTT alarms from the ERTT modules. It also assumes that all ERTT modules are on the same workspace.

1. Turn off alarms until the situation is under control. Once you are focused on servicing RTTs the alarms are a distraction.
2. Visually scan the Highest Priority Indicator of the ERTT modules to find a module containing High priority unanswered RTT Message(s).
3. Press the Next Call button to highlight the next unanswered, highest priority entry. The mini modules populate based on the RTT Message data.
4. Press the group or Unit module to reply to the unit initiating the RTT message.
5. Once service has been rendered, repeat steps 3 and 4 until all High priority RTTs have been serviced in this ERTT module.
6. Repeat steps 2-5 on each ERTT module with unanswered High priority RTT messages.
7. Repeat steps 2-5 on each ERTT module with unanswered Normal priority RTT messages.
8. Turn RTT Alarms back on once things calm down.

9.3.2.3 ERTT Module Operation for Remote Consoles

Symphony consoles share information about RTT processing so that dispatchers can optionally share responsibility for RTTs. If your local console is configured to process ERTT information from remote consoles, your local console responds to remote ERTT module controls as if you activated the controls locally.

If a remote console presses the Next Call Button, the matching RTT entry is deleted from the appropriate local ERTT module. The local ERTT module updates its Highest Priority Indicator and re-evaluates its RTT Tone demand based upon the priority of any remaining unanswered entries.

If a remote console presses the Tone Off Button, the local console behaves as if the Tone Off button on the local console was pressed for the appropriate ERTT module.

If a remote console replies to an RTT entry, the matching RTT message's state is set to answered in the appropriate local ERTT module. The local ERTT module updates its Highest Priority Indicator and re-evaluates its RTT Tone demand based upon the priority of any remaining unanswered entries in the ERTT module.

Be aware that your actions could impact remote consoles if they are configured to process remote ERTT information.

9.3.3 Recommendations and Limitations of ERTT

The following recommendations and limitations should be noted:

- It is the responsibility of a trunked radio unit user to transmit an appropriate message when necessary to update the RTT.
- Units that have never transmitted any message or that do not support RTT messaging have no status from the radio system's point-of-view.
- If an RTT module-initiated individual call must be transmitted in digital voice ("private") mode, the ERTT module's I-Call mini module must be placed in private transmit using the context menu.

9.4 RADIO TEXTLINK CLIENT

The Radio TextLink Client is an application that allows an End User to exchange text messages between combination of radios and clients on a ProVoice™, P25, or Enhanced Digital Access Communications System (EDACS) system. With Radio TextLink, text messages are received by the radio and displayed on the radio's user interface. Key Radio TextLink features include:

- Free form text sent to and displayed on portable and mobile radios.
- Two-way canned messages and responses.
- Message routing and logging.
- Console, Dispatch, CAD workstation, and/or third-party interface application support.
- IP-based data application.

The integrated Radio TextLink Client can be launched in the following two ways:

- Launch from Symphony.
- Launch from the Command line.

9.4.1 Launch from Symphony

The Radio TextLink client can be launched from Symphony's Main Menu. The Radio TextLink client interacts with the Radio TextLink server to get provisioned units. The Radio TextLink server provisions only those units which belong to the agencies configured in Symphony, with the default agency being that of the console. See the *Radio TextLink Feature Manual* (MM24165) for more information.

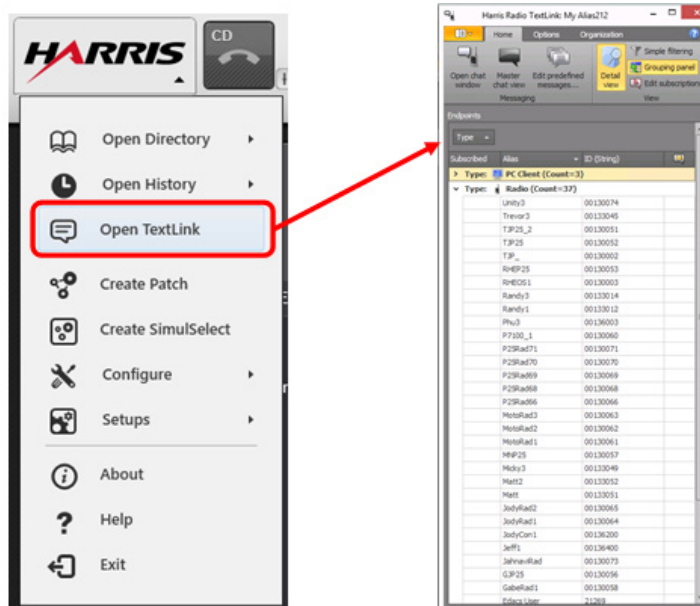


Figure 9-12: Launch Radio TextLink

9.4.2 Launch Radio TextLink Client from Command Line

To launch the Radio TextLink Client from the command line, change directory to the following location:

```
C:\Program Files (x86)\Harris Radio TextLink Client>
```

The following example shows starting the Radio TextLink Client from the command line in integration mode and filtering on agencies 13 and 88.

```
C:\Program Files (x86)\Harris Radio TextLink Client>Part1042.Presentation
LaunchingFrom:Symphony Agencies:13,88
```



NOTE

Symphony can bring the Radio TextLink Client instance to the foreground even if it is launched from the command line.

Table 9-2: Launch Radio TextLink Client by Command Line

COMMAND LINE ARGUMENTS	VALUE	DESCRIPTION	EXAMPLE
		Optional input that specifies the configuration file to use for loading. When the configuration file is not specified, then the application uses the default configuration file from the user's %AppData% location. The user can specify any file path that is a valid TextLink Client configuration file.	"C:\ProgramData\Harris\Harris Radio TextLink Client\Radio TextLink Client(Default Configuration).rtlconfig"
LaunchingFrom	Symphony	This is an input to the Radio TextLink client application to identify the launch mode (Symphony Integrated).	LaunchingFrom:Symphony
Agencies	0-1677	Agency list in the comma separated format. Server provisions units only for these agencies.	Agencies:13,14,15

The following tables list some of the valid command line invocations that can be executed from the directory C:\Program Files (x86)\Harris Radio TextLink Client>:

Table 9-3: Launch Radio TextLink Client in Standalone Mode

COMMAND	DESCRIPTION
Part1042.Presentation	Launches from the default configuration file which is stored in the %AppData%\Harris\Radio TextLink Client folder
Part1042.Presentation "C:\ProgramData\Harris\Harris Radio TextLink Client\Radio TextLink Client (Default Configuration).rtlconfig"	Launches from the specified configuration file

Table 9-4: Launch Radio TextLink Client in Integrated Mode

COMMAND	DESCRIPTION
Part1042.Presentation LaunchingFrom:Symphony Agencies:13,88	Launches from the default configuration file fetching the Units from Agencies 13 and 88
Part1042.Presentation "C:\ProgramData\Harris\Harris Radio TextLink Client\Radio TextLink Client (Default Configuration).rtlconfig" LaunchingFrom:Symphony Agencies:13	Launches from the specified configuration file fetching the Units from Agencies 13
Part1042.Presentation LaunchingFrom:Symphony	Launches from the default configuration file fetching all Units
Part1042.Presentation "C:\ProgramData\Harris\Harris Radio TextLink Client\Radio TextLink Client (Default Configuration).rtlconfig" LaunchingFrom:Symphony	Launches from the specified configuration file fetching all Units

9.5 DISCREET LISTENING



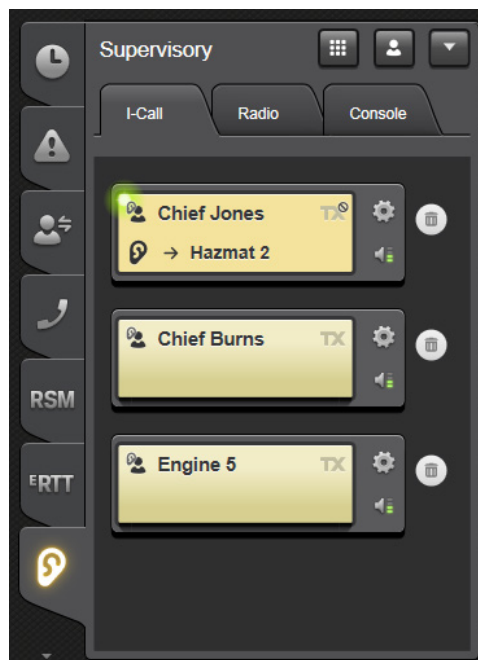
Since Discreet Listen is a feature-licensed option, this feature is only available if the console is equipped with this software license. The feature must also be enabled via configuration before it can be used.

The Discreet Listening feature enables the dispatcher to listen to active VIDA I-Calls between two other units. This dispatcher programs target units for Discreet Listening onto the I-Call tab on the Supervisory Sidebar Panel. Whenever any of those units are involved in an active VIDA I-Call, the call audio data is routed to the Symphony console. Information about the VIDA I-Call is indicated on the Supervisory I-Call tab.

The Supervisory Sidebar Panel and its I-Call Discreet Listening tab are covered in Section 3.4.11.5. VIDA I-calls are covered in Section 5.4.

To add a Discrete Listening target unit, follow these steps:

1. Select the I-Call tab in the Supervisory Sidebar Panel.
2. Choose a target unit using the Keyboard or the Contacts button. The Keyboard button pops up a dial-pad to enter the user ID. The Contacts button opens a list of units that are provisioned to the console.
3. Once the target is chosen, a unit module for the target is added to the I-Call tab. This module shows any I-Calls to or from the target unit.



4. Use the module volume, mute, and speaker controls to modify the Discrete Listening experience for each target. These changes apply until the I-Call session ends if made during an active I-Call, or until the module is removed from Discrete listening if made when the module is not in an active I-Call session.
5. Click the **Delete** icon next to the module to remove the module and terminate Discrete Listening of the associated target unit.

When the target participates in an I-Call, the associated unit module's Smart Button identifies the other endpoint on the bottom line. This button displays the Discreet Listening icon and indicates that the module is receive-only through the duration of the call. The button indicates the direction of the call, with an arrow pointing away from the caller and toward the callee.



Figure 9-13: Discreet Listen Module

Though the Supervisory Sidebar Panel is a convenient way to view all active Discrete Listening targets, any unit module assigned to a Discrete Listening target shows the Discrete Listening information shown above.

Additionally, the module displays buttons for accessing a context menu and a button for changing the volume of the call that is being listened to.



NOTE

Only seven units can be programmed into the I-Call tab at a time.

9.6 RADIO UNIT MONITOR (RUM)



NOTE

Since Radio Unit Monitor is a feature-licensed option, this feature is only available if the console is equipped with this software license. The feature must also be enabled via configuration before it can be used.

The Radio Unit Monitor (RUM) feature allows the console user to remotely put a P25 radio into a state of "Hot Mic." The radio transmits audio back to the console for a set period of time. This is typically done when the dispatcher suspects the radio has been stolen or an officer is in danger. The console can specify the duration for which the call should be active and if the call should be covert and/or encrypted. The console can also specify if the call should be an I-Call or a group call. The availability of runtime parameters is controlled by the administrator.

9.6.1 Radio Unit Monitor Operation

To initiate a RUM call, follow these steps:

1. Select the Radio Tab in the Supervisory Sidebar Panel. If no unit has been programmed for RUM, the tab says “No Target Picked.”

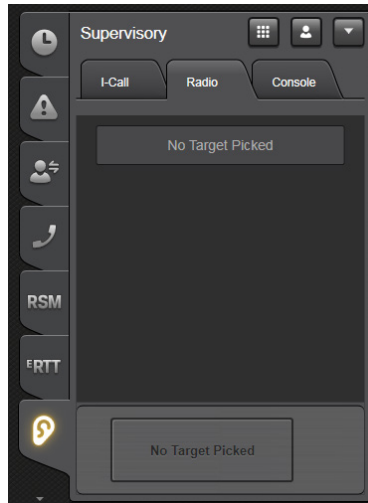


Figure 9-14: No Target Picked

2. Choose a target unit using the Keyboard or the Contacts button. The Keyboard button pops up a dial-pad to enter the user ID. The Contacts button opens a list of units that are provisioned to the console.



Symphony only supports Radio Unit Monitor on P25 Radios. I-Call Radio Unit Monitor is not supported for Motorola radios.

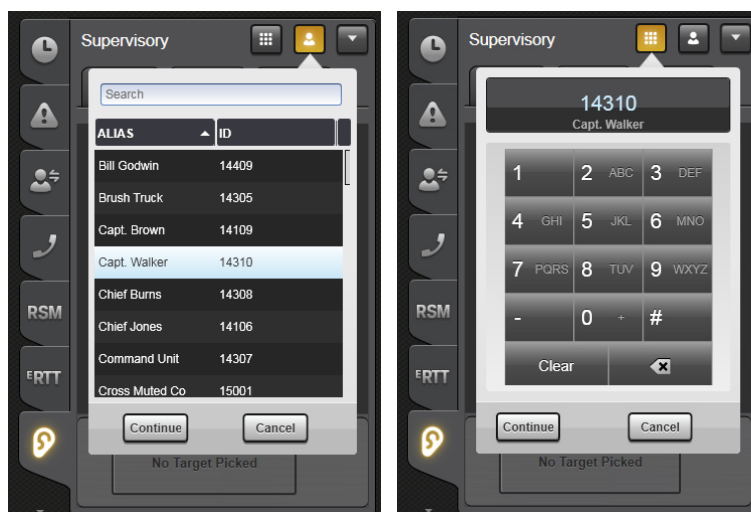


Figure 9-15: Initiate a Radio Unit Monitor Request

- Once the target is chosen, the User Alias (or ID, depending on GUI settings) will be shown with all of the radios that are registered to the chosen User ID listed in the list box at the top of the panel. Each will be identified by its radio model and Electronic Serial Number. If not using VIDA ID, only one radio is allowed for each User ID, so only the user alias will be displayed, as shown in Figure 9-16. After selecting the desired radio, the Radio Tab is populated with pre-configured values for the Call Length, Covert, and Encrypted fields. These fields are available for editing if the administrator has enabled runtime modification in the Symphony Configuration tool. In addition, the **Encrypted** checkbox can be disabled if the unit or group on which the Radio Unit Monitor call will be requested does not have encryption keys loaded. To determine the reason for a disabled field, hover over it with the mouse for a tooltip explanation.

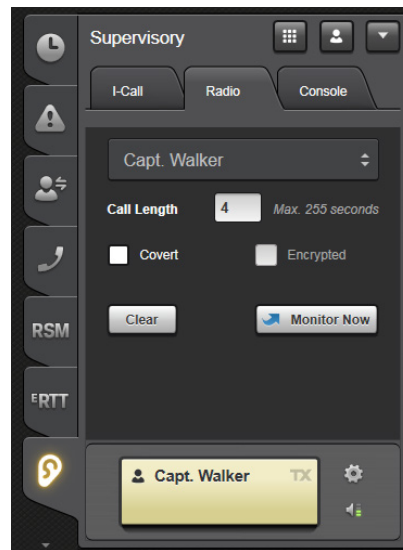


Figure 9-16: Set Radio Unit Monitor Parameters

- The bottom of the tab contains a unit or group module with the entity that will be used for the RUM call. If the RUM call is configured to be a VIDA I-Call or group, the module will be a unit or group module respectively.
- Once the **Monitor Now** button is pressed, the buttons in the Radio tab become disabled until the RUM call ends or the console times out waiting for the RUM call. During this period, another RUM request cannot be initiated, nor can the current request be cancelled. The RUM call is always heard in the headset or the select speaker if no headset is plugged in.
- While the RUM call is in progress, the RUM target module displays the call progress as shown in Figure 9-17. The clock in the upper right-hand corner of the smart button shows the time remaining with the granularity of one quarter of the call length. In addition, the RUM target module cannot be pre-empted when a RUM call is active on the module.



It is recommended that the RUM group be configured as receive-only to avoid accidental PTT on the module. See Section 3.5.1.2 for information on how to configure talkgroups to be receive-only.

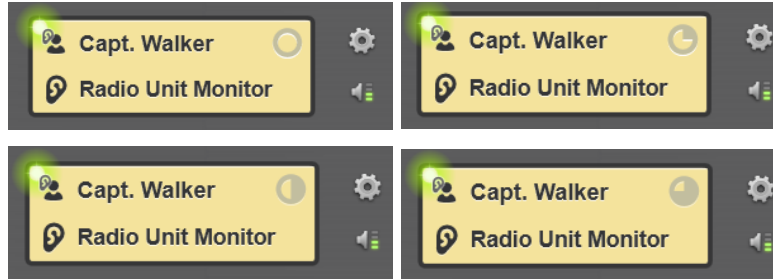


Figure 9-17: Unit Module States during a Radio Unit Monitor Call

7. Once the RUM call completes, press **Monitor Now** again to initiate another RUM call from the same unit or press the **Clear** button to remove the current target from the RUM tab.

Once the **Monitor Now** button is pressed, the buttons in the Radio tab become disabled until the RUM call ends or the console times out waiting for the RUM call. During this period, another RUM request cannot be initiated, nor can the current request be cancelled.

If the group that is configured for Radio Unit Monitor is unmonitored, the console auto-monitors the group and plays the RUM call when it starts. While the RUM call is in progress, the initiating console cannot transmit on the talkgroup or pre-empt the call in any way.



NOTE

Only one unit can be monitored at a time. While a Radio Unit Monitor call is active, another monitor request cannot be initiated until the call ends or the console times out waiting for the call.

9.6.2 Radio Unit Monitor on a Patch

If the console is configured to do a Radio Unit Monitor on a group, it is possible that the group is in an active patch when a RUM request is initiated. In this case, the RUM call comes on the patch, and all groups that are part of the patch will also hear the RUM call. It is strongly recommended that a patch not be deactivated while a RUM call is active on the patch. Doing so can prematurely end the RUM call. Also, if the RUM call is active on a group, it is recommended that the group not be patched while the call is still active.

9.7 CONSOLE ENABLE/DISABLE



NOTE

This feature is only available if the console is configured by the system administrator to support enabling/disabling consoles. Each console is configured with a predefined list of remote consoles it is allowed to enable/disable.

The Console Enable/Disable feature allows the console user to remotely disable (or alternatively re-enable) another Symphony or C3 Maestro^{IP} console that is currently logged in. Once a remote console is disabled, all modules and panels and select bar controls are essentially grayed out. The dispatcher at that time is no longer able to transmit or to hear incoming audio. All hardware peripherals (desk mic, foot switch, headset, etc.) on the disabled console are also disabled in a similar way. Once a remote console is re-enabled, all functionality is restored to its normal, non-disabled state.

The Supervisory Sidebar Panel and its console enable/disable tab are covered in Section 3.4.11.7.

9.7.1 Disabling a Remote Console

To disable a remote console, follow these steps:

1. Select the Console Tab in the Supervisory Sidebar Panel. If no console has been selected for disabling, the tab says “No Target Picked.”

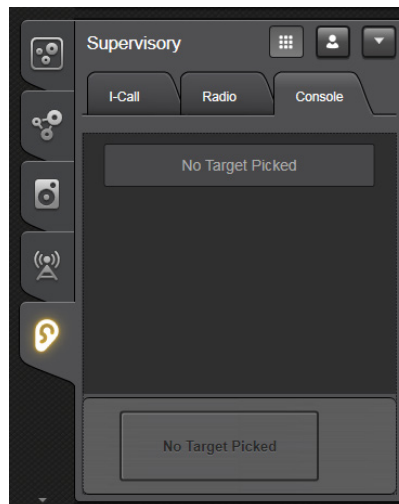


Figure 9-18: No Target Picked

2. Choose a target console using the Contacts button. The Contacts button opens a list of consoles that are allowed by the system administrator for enabling/disabling.

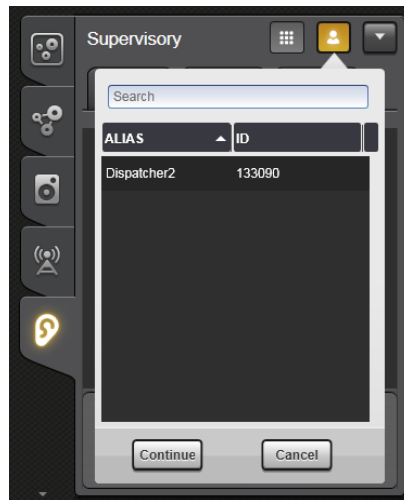


Figure 9-19: Select a Remote Console

3. Once the target is chosen, the Console Tab is populated with fields to support taking an action on the selected console. The bottom of the tab contains a unit module for the selected console.



Figure 9-20: Setting Command to Disable

4. Select the "Disable" radio button to prepare the command for disabling the selected remote console.

- Once the **Send** button is pressed, a command is sent to the remote console, and the nearby controls become disabled until the disabling command has completed. If the command was successful, the text below the **Send** button will read “Dispatcher *alias* Disabled” followed by the date and time of the disabling action. If the command was unsuccessful, the text below the **Send** button will contain an error message notifying the dispatcher about the failure.

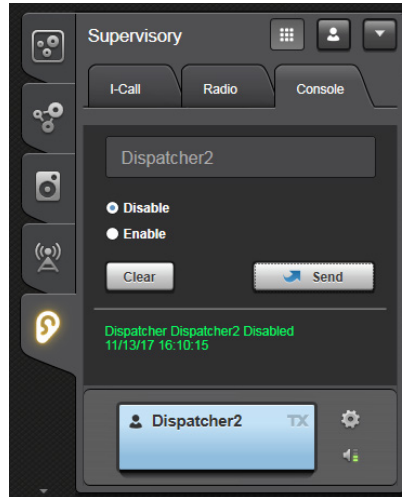


Figure 9-21: Successful Disabling of a Remote Console

9.7.2 Re-enabling a Remote Console

To re-enable a remote console, follow these steps:

- Select the Console Tab in the Supervisory Sidebar Panel. If no console has been selected for re-enabling, the tab says “No Target Picked.”

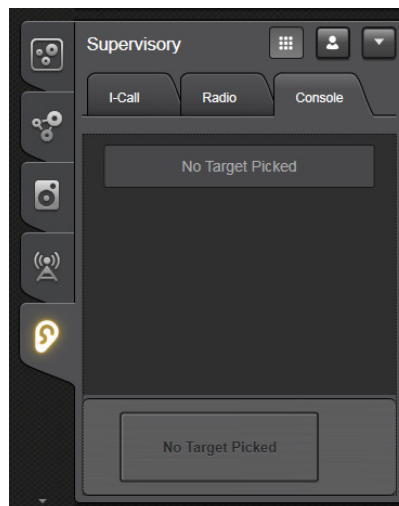


Figure 9-22: No Target Picked

2. Choose a target console using the Contacts button. The Contacts button opens a list of consoles that are allowed by the system administrator for enabling/disabling.



Figure 9-23: Select a Remote Console

3. Once the target is chosen, the Console Tab is populated with fields to support taking an action on the selected console. The bottom of the tab contains a unit module for the selected console.

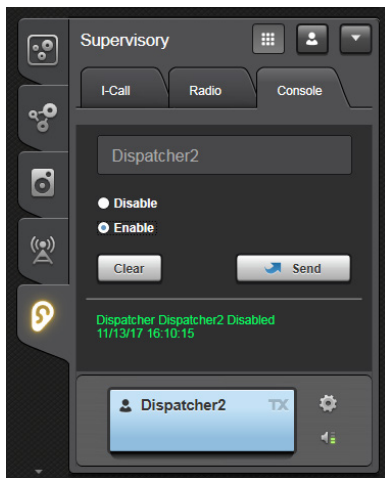


Figure 9-24: Setting Command to Re-enable

4. Select the "Enable" radio button to prepare the command for disabling the selected remote console.

- Once the **Send** button is pressed, a command is sent to the remote console, and the nearby controls become disabled until the re-enabling command has completed. If the command was successful, the text below the **Send** button will read “Dispatcher *alias* Enabled” followed by the date and time of the re-enabling action. If the command was unsuccessful, the text below the **Send** button will contain an error message notifying the dispatcher about the failure.

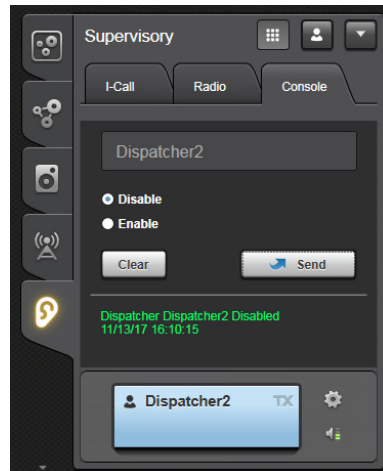


Figure 9-25: Successful Re-enabling of a Remote Console

9.7.3 Being Disabled by a Remote Console

When a dispatcher has been disabled by a remote console, the dispatcher will no longer be able to interact with the system (radio, telephony, etc.) in any meaningful way. The dispatcher will no longer be able to transmit, whether via mouse, footswitch, desk mic, headset, or any other means. The dispatcher will no longer hear audio from the console speakers. The utility of the Symphony application and screens will become limited with almost all controls, modules, and other clickable items becoming disabled. The dispatcher will receive a message in the System History notifying him/her that the console has been disabled. The white text “CONSOLE DISABLED” will also appear at the top of each workspace tab.



Figure 9-26: Disabled Symphony Console

9.7.4 Being Re-Enabled by a Remote Console

When a dispatcher has been re-enabled by a remote console, the console will be restored to normal operation. The dispatcher will again be able to interact with the system (radio, telephony, etc.) as normal. The dispatcher's ability to transmit, whether via mouse, footswitch, desk mic, headset, or any other means will be restored. The dispatcher will hear audio from the console speakers. The utility of Symphony application and screens will become fully restored. The dispatcher will receive a message in the System History notifying him/her that the console has been enabled.



Figure 9-27: Re-Enabled Symphony Console

10 WEB BROWSER WORKSPACE

If the Symphony is licensed and configured to support web browsing, one of the workspace tabs will make web browsing available.

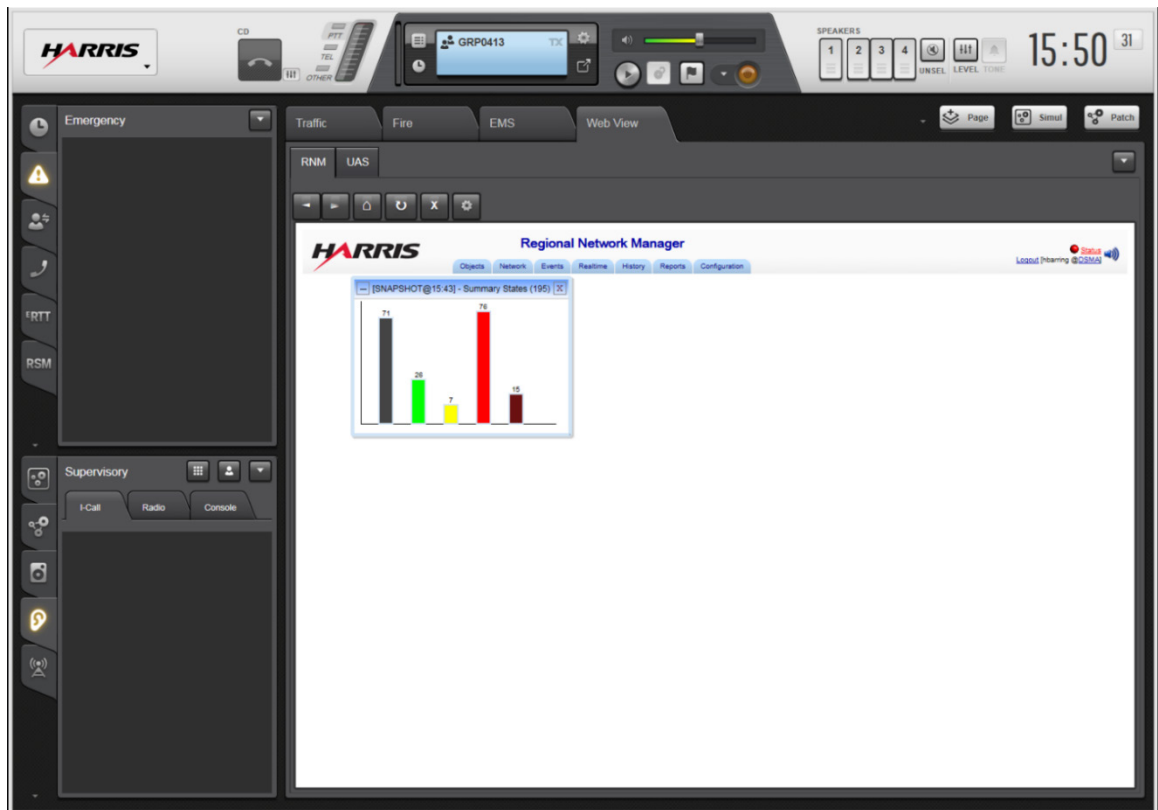


Figure 10-1: Web Browser Workspace

Symphony can support as many as four subtabs, each defined with a home URL. Figure 10-1 shows a Symphony with one browser workspace with two subtabs defined. The subtabs in this example are configured to access two VIDA components: UAS (Unified Administrative System) and RNM (Regional Network Manager).

Use the following navigation buttons as defined below:



The Forward and Back buttons are used to navigate sequentially back and forth through pages that have been visited previously.



The Home button is used to return the browser to the home URL for the current subtab.



The Refresh button is used to refresh the current web site.



The Stop button is used to halt navigation.



The Utility button is used to access an option to display the current URL.

11 CUSTOMER SERVICE

11.1 CUSTOMER CARE

If any part of the system equipment is damaged on arrival, contact the shipper to conduct an inspection and prepare a damage report. Save the shipping container and all packing materials until the inspection and the damage report are completed. In addition, contact the Customer Care center to make arrangements for replacement equipment. Do not return any part of the shipment until you receive detailed instructions from an L3Harris representative.

Contact the Customer Care center at <https://www.l3harris.com/all-capabilities/pspc-customer-care> or:

North America:

Phone Number: 1-800-368-3277

Fax Number: 1-321-409-4393

E-mail: PSPC_CustomerFocus@l3harris.com

International:

Phone Number: 1-434-455-6403

Fax Number: 1-321-409-4394

E-mail: PSPC_InternationalCustomerFocus@l3harris.com

11.2 TECHNICAL ASSISTANCE

The Technical Assistance Center's (TAC) resources are available to help with overall system operation, maintenance, upgrades and product support. TAC is the point of contact when answers are needed to technical questions.

Product specialists, with detailed knowledge of product operation, maintenance and repair provide technical support via a toll-free (in North America) telephone number. Support is also available through mail, fax and e-mail.

For more information about technical assistance services, contact your sales representative, or call the Technical Assistance Center at:

North America: 1-800-528-7711

International: 1-434-385-2400

Fax: 1-434-455-6712

E-mail: PSPC_tac@l3harris.com

12 STATUS AND ERROR MESSAGES

Table 12-1 describes the status and error messages that may be displayed on the splash screen during login or in the System History during runtime. Status messages provide information about normal completion of tasks. Error messages provide additional information about a failed operation. The notation “<value>” indicates information that will be filled in when the message is logged. For example, in the message “Setup successfully changed to <NAME>,” <NAME> will be replaced with the appropriate setup name. If the Recommended Solution column is blank or suggested solution fails to resolve the problem, contact your Systems Administrator or L3Harris TAC.

Table 12-1: Login Messages and Causes

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
Failed to login. Requested FlexPaths or Vcoders not allowed by VNIC.	The PSAP’s “Max Talk Paths” setting is less than the number of FlexPaths the console is licensed for.	Check Max Talk Paths in UAS.
Unable to log into Console.	Symphony’s ConsoleService is not running. Console does not have valid Symphony licenses.	Check Windows Task Manager > Services. Symphony Service should be running. Open Symphony Load Licenses and review status of licenses.
GENERIC INFORMATIONAL MESSAGES		
VNIC is now connected.	Status message. No error	N/A
SIP Server is now connected.	Status message. No error	N/A
Audio Box is now connected.	Status message. No error	N/A
Talkgroup provisioning has completed.	Status message. No error	N/A
User provisioning has completed.	Status message. No error	N/A
Site provisioning has completed.	Status message. No error	N/A
Symphony is now fully connected, provisioned, and ready.	Status message. No error.	N/A
Patch provisioning has completed.	Status message. No error	N/A
SimulSelect provisioning has completed.	Status message. No error	N/A
Emergency provisioning has completed.	Status message. No error	N/A
XRTT provisioning has completed.	Status message. No error	N/A
KMF is now connected.	Status message. No error	N/A
Loaded Page Sequences and Stacks from disk.	Status message. No error	N/A
Aux I/O Server is now connected.	Status message. No error	N/A
KMF Params loaded.	Status message. No error	N/A
KMF Bindings loaded.	Status message. No error	N/A
KMF Key Changeover completed.	Status message. No error	N/A
KMF Key Delete completed.	Status message. No error	N/A
KMF Zeroize completed.	Status message. No error	N/A
KMF Key Load completed.	Status message. No error	N/A
KMF Bindings Purge completed.	Status message. No error	N/A
KMF Reverse Warmstart requested.	Status message. No error	N/A
Manual Zeroize completed.	Status message. No error	N/A

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
Base Station <NAME> in service.	Status message. No error	N/A
Delta Talkgroup provision. <COUNT> Talkgroup(s) added, <COUNT> Talkgroup(s) deleted.	Status message. No error	N/A
Delta User provision. <COUNT> User(s) added, <COUNT> User(s) deleted.	Status message. No error	N/A
Delta Site provision. <COUNT> Site(s) added, <COUNT> Site(s) deleted.	Status message. No error	N/A
Setup successfully changed to <NAME>.	Status message. No error	N/A
Setup <NAME> successfully created.	Status message. No error	N/A
Setup successfully renamed to <NAME>.	Status message. No error	N/A
Setup <NAME> successfully deleted.	Status message. No error	N/A
Password successfully changed for setup <NAME>.	Status message. No error	N/A
Page <NAME> aborted.	Status message. No error	N/A
Aux I/O Alarm <NAME> acked remotely by User <NAME>.	Status message. No error	N/A
Successfully acked Aux I/O Alarm <NAME>.	Status message. No error	N/A
Aux I/O Alarm <NAME> set to ON.	Status message. No error	N/A
Aux I/O Alarm <NAME> set to OFF.	Status message. No error	N/A
The current Call Director phone call has exceeded the maximum time limit of <COUNT> minutes.	Status message. No error	N/A
Cannot ack Local Aux I/O Points.	Status message. No error	N/A
Cannot ack, Aux I/O Point <NAME> is already Acknowledged.	Status message. No error	N/A
Cannot ack, Aux I/O Point <NAME> is inactive.	Status message. No error	N/A
Call Alert failed on Unit <NAME>.	Call Alert could not be sent to unit <NAME>.	Retry sending the call alert. If it still fails, the radio could be turned off or the VNIC could be busy. Notify your network or system administrator.
Incoming Unit Call from <NAME> failed encryption negotiation.	Unit <NAME> attempted to make an encrypted I-Call with this console but the I-Call request was denied because this console does not have keys loaded or encryption licensed.	Notify your supervisor or system administrator to purchase a license for encryption feature or to load keys for the current console.
Outgoing Unit Call to <NAME> failed encryption negotiation.	Console attempted to make an encrypted I-Call with Unit <NAME> but the I-Call request was denied because the unit does not have keys loaded or encryption licensed.	Either make an unencrypted call with the unit or notify your supervisor or system administrator to load keys on the unit's radio or console.

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
Unit Call Recipient <NAME> requires Encryption.	Console attempted to make an unencrypted I-Call with Unit <NAME> but the I-Call request was denied because the unit is configured to only accept encrypted calls.	Notify your supervisor or system administrator to purchase a license for encryption feature or to load keys for the current console.
Encrypted unit call attempt. No System key present.	Console attempted to make an encrypted I-Call with Unit <NAME> but the I-Call request was denied because there was no system key loaded.	Notify your supervisor or system administrator to load keys for the current console.
Dispatcher is not authorized.	<ul style="list-style-type: none"> User ID is wrong or not in UAS. Password is incorrect. PSAP ID is incorrect or not in UAS 	Open Symphony Connection Manager and enter valid credentials for User ID, password, and/or PSAP ID. You may have to log into UAS and enter new User IDs and/or PSAP IDs.
VNIC protocol version is not supported!	The console is trying to login to a VNIC that does not support this console version.	Contact your network or system administrator to upgrade your VNIC to the correct version.
PSAP ID is undefined!	ID has not been entered or is invalid in the Connection Manager.	Open Symphony Connection Manager and enter valid credentials for PSAP ID. You may have to log into UAS and enter new PSAP IDs.
VNIC is not responding to boot messages.	<ul style="list-style-type: none"> IP address/Hostname of the VNIC is wrong in connection manager. The console is trying to login to a VNIC that does not support this console version. The VNIC is unavailable. 	Open Symphony Connection Manager and enter valid address for VNIC IP Address. If that does not work, there may be a network problem or the VNIC version does not support this console version.
VNIC is not responding to boot messages but Phone is available.	<ul style="list-style-type: none"> SIP Telephony is available. IP address/Hostname of the VNIC is wrong in Connection Manager. The console is trying to login to a VNIC that does not support this console version. The VNIC is unavailable. 	Open Symphony Connection Manager and enter valid address for VNIC IP Address. If that does not work, there may be a network problem or the VNIC version does not support this console version.
VNIC denied message!	VNIC denied a message sent by this console. This could be because of an invalid or disallowed operation.	Notify your network or system administrator to ensure that the VNIC is configured correctly. If the problem persists, logs should be submitted to TAC.
VNIC denied message with Reason Code: <REASON CODE>!	VNIC denied a message sent by this console. This could be because of an invalid or disallowed operation. The <REASON CODE> explains the reason for the deny.	Notify your network or system administrator to ensure that the VNIC is configured correctly. If the problem persists, logs should be submitted to TAC.
Failed to login! Requested FlexPaths or Vocoders not allowed by VNIC. Check Max FlexPaths in UAS.	The console is licensed for more FlexPaths than allowed by the UAS.	Notify your system or network administrator to configure the UAS to allow the appropriate number of talkpaths. If this is not possible, then the console will have to be licensed for fewer FlexPaths.
Unknown failure connecting to VNIC.	Many possible reasons.	Notify your network or system administrator to ensure that the VNIC is configured correctly. If the problem persists, logs should be submitted to TAC.

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
VNIC is not responding to message.	The VNIC is unavailable.	Notify your network or system administrator to ensure that the VNIC is configured correctly. If the problem persists, logs should be submitted to TAC.
Significant difference in Talkgroup database and cached version!	Provisioned data is vastly different from the previous data saved. This could be normal if the system configuration is changing, but it could also indicate a configuration problem.	If this is expected (because of recent system reconfiguration), then there is no action to take. If not, notify your system administrator to make sure there are no hidden problems in the system configuration.
VNIC is disconnected!	The VNIC is unavailable.	Notify your network or system administrator to ensure that the VNIC is configured correctly and is running. If the problem persists, logs should be submitted to TAC.
Audio Box is now disconnected!	The Audio Box is disconnected. This could be because the Audio box is not properly configured or it is rebooting.	Make sure that the audio box is configured properly. If it is configured correctly, and the problem persists past a few minutes, contact TAC with logs.
Call Denied!	VNIC denied an attempt by this console to make a call to/on an entity.	This could be because the entity is busy or invalid. Notify your network or system administrator to make sure the system is configured properly. If the problem persists, send logs to TAC.
No valid license found for console.	No license has been loaded or the license is invalid for this machine.	Contact your administrator to purchase/load a valid license.
Demo mode license has expired for console.	The Demo mode is a limited time license and the time has expired.	Contact your administrator to extend your demo license or purchase a new one.
Current application is not in approved list.	This is \when a non-Symphony and non-Baton try to connect, and we do not have that executable in our approved list.	Contact your administrator to add your application to the approved list.
Login rejected for invalid token.	Internal processing Error.	Restart Windows.
KMF is not responding to boot messages.	The KMF server is unavailable.	There may be a network problem that is affecting communications between the console and the KMF server, or the KMF server may be down temporarily. Notify your system or network administrator to fix the issue.
KMF is disconnected!	The KMF server is unavailable.	There may be a network problem that is affecting communications between the console and the KMF server, or the KMF server may be down temporarily. Notify your system or network administrator to fix the issue.
Aux I/O Server is disconnected!	The Aux I/O server is unavailable.	There may be a network problem that is affecting communications between the console and the Aux I/O server, or the Aux I/O server may be down temporarily. Notify your system or network administrator to fix the issue.
Call preempted by higher priority call.	An ongoing call was pre-empted by a higher priority call, for example, a supervisor call.	N/A

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
Unsupported vocoder for <NAME>.	Attempt was made to transmit on a group that does not have a supported vocoder.	Notify your supervisor to configure the group in UAS to use one of the supported vocoders.
Talkgroup <NAME> does not exist.	Attempt was made to transmit on a group that does not exist anymore.	You may have to login to UAS to re-add the talkgroup. If this status message is in error, send logs to TAC.
Exceeded maximum number of licensed speakers.	More speakers are connected to the console's hardware than is licensed. No audio will be heard on the extra speakers.	There is nothing you need to do for this unless you want to be able to use the extra speakers. In that case, contact your administrator to purchase a license that supports more speakers.
Max calls exceeded, dropping call to <NAME>.	The console had to drop one or more calls because more calls were received than there were supported FlexPaths for.	Notify your administrator to purchase a license that supports more FlexPaths, so as not to miss calls.
Setup change failed: setup <NAME> not found.	Console attempted to switch to a setup that no longer exists.	Notify your administrator to configure your setup correctly.
Setup rename failed: setup <NAME> not found.	Console attempted to rename a setup that does not exist anymore.	Notify your administrator to configure your setup correctly.
Setup delete failed: setup <NAME> not found.	Console attempted to delete a setup that does not exist anymore.	Notify your administrator to configure your setup correctly.
Password change failed: setup <NAME> not found.	Console attempted to change password for a setup that does not exist anymore.	Notify your administrator to configure your setup correctly.
Undefined error while changing to setup <NAME>.	There was an unknown error while trying to switch setups. This could be a Windows problem.	Send logs to TAC.
Undefined error while creating setup <NAME>.	There was an unknown error while trying to create a new setup. This could be a Windows problem.	Check to make sure that the setup was not already created by looking at the Switch Setups dialog. If there is no setup created with name <NAME>, then send logs to TAC.
Undefined error while renaming setup <NAME>.	There was an unknown error while trying to rename a setup. This could be a Windows problem.	Check to make sure that the setup <NAME> has not already been renamed by looking at the Switch Setups dialog. If the rename operation was not successful, send logs to TAC.
Undefined error while deleting setup <NAME>.	There was an unknown error while trying to delete a setup. This could be a Windows problem.	Check to make sure that the setup was not already deleted by looking at the Switch Setups dialog. If the setup <NAME> still exists, then send logs to TAC.
Undefined error while changing password for setup <NAME>.	There was an unknown error while trying to change password for a setup. This could be a Windows problem.	Check to make sure that the setup password has not already been successfully updated by trying to switch to that setup through the Switch Setups dialog. If that fails, then send logs to TAC.
Undefined error while activating Patch.	There was an unknown error while trying to activate a patch.	Send logs to TAC.
Undefined error while activating SimulSelect.	There was an unknown error while trying to activate a SimulSelect.	Send logs to TAC.
Unable to load Page Sequences. Sequence List has been corrupted.	The Page Sequences data has been corrupted.	Notify your administrator to reconfigure your page sequences using the Symphony Configuration tool.

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
Unable to load Page Stacks. Stack List has been corrupted.	The Page Stacks data has been corrupted.	Notify your administrator to reconfigure your page stacks using the Symphony Configuration tool.
Failed to restore previous select entity after Paging.	The console could not restore the previously selected entity after a page was sent. This could be because the previously selected entity does not exist anymore.	If your previous select was a Patch, then check to make sure that the Patch still exists. If your previous select was a SimulSelect, then check to make sure that the entities within the SimulSelect are all valid and are not in another Patch or SimulSelect. For other entity types, check the directory to make sure that the entity still exists and has not been deleted.
Cannot transmit on RX-only Talkgroup <NAME>.	An attempt was made to transmit on a talkgroup that is Rx-only.	Your administrator may have configured this talkgroup to be Rx-only. If it is necessary to transmit on this group, then open the Directory, navigate to this talkgroup, and uncheck the Rx-only setting in the Settings tab. Check with your administrator before changing this setting, if necessary.
Cannot unmonitor Talkgroup <NAME> while it is in emergency.	An attempt was made to unmonitor a talkgroup that is in emergency.	The emergency has to be cleared before the talkgroup can be unmonitored.
Cannot transmit during the interkey time.	An attempt was made to transmit too quickly after the last transmit ended.	Wait for the configured quick-key duration before attempting to transmit again. Contact your administrator to find out this quick-key duration.
Unable to preempt higher priority call.	Attempt was made to transmit on an entity that already has a higher-priority call in progress.	Wait for the call to end before transmitting.
Setup change to <NAME> failed: missing password.	Attempt was made to switch to a new setup without providing a password.	Enter the setup password before switching to the setup.
Setup change to <NAME> failed: invalid password.	Attempt was made to switch to a new setup using an invalid password.	Enter the correct setup password before switching to the setup.
Invalid operation requested on default console setup.	An attempt was made to rename, delete, or add password for the "Default" setup, OR to add new setup with the name "Default".	The name "Default" cannot be used for a new setup. The Default setup cannot be renamed, deleted, or supplied a password.
Cannot change to setup <NAME>: One or more Patches still active.	Change to a new setup failed because one or more patches are still active.	Deactivate all active patches before switching setups.
Cannot change to setup <NAME>: One or more I-Call sessions still active.	Change to a new setup failed because one or more I-Call sessions is still active in the I-Call tab.	End the active sessions and try switching setups again.
Invalid attempt to transmit on inactive I-Call session.	An attempt was made to transmit on a unit without initiating an I-Call session.	If you see this error, try pressing the Smart Button of the unit to initiate an I-Call session. If the problem persists, send logs to TAC.
Invalid attempt to end an active unacknowledged I-Call session.	An attempt was made to end an active unacknowledged VIDA I-Call session. Unacknowledged I-Call sessions cannot be ended by user. They are ended automatically.	N/A

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
Invalid attempt to play alert tone to <NAME>. Tones cannot be played to acknowledged Users.	An attempt was made to send alert tones to an acknowledged unit. Alert tones can be sent only to unacknowledged units.	N/A
Invalid attempt to play alert tone to <NAME>. Tones cannot be played to unacknowledged Users while session is not active.	An attempt was made to send alert tones to a unit that is in an active session.	Wait for the session to end or time out before retrying.
Activation failed: Patch <NAME> is already active.	An attempt was made to activate a patch that was already active.	N/A
Console not allowed to patch due to system configuration.	The system has been configured not to allow this console to activate patches.	Notify your supervisor or system administrator if you believe this is in error.
Patch <NAME> activation failed! At least one of its Talkgroups is in emergency.	An attempt was made to activate a patch in which at least one talkgroup is in emergency.	Wait for the emergency to clear before retrying.
Patch <NAME> activation failed! At least one of its Talkgroups is an All-Call group.	An attempt was made to activate a patch in which at least one talkgroup is an All-Call group. All-Call groups cannot be patched.	Try again after removing the All-Call group from the patch. Contact your supervisor if you do not have permissions to modify the patch.
Patch <NAME> activation failed! At least one of its Talkgroups is in an active Patch/SimulSelect.	An attempt was made to activate a patch in which at least one talkgroup or base station is part of another Patch or SimulSelect.	Wait for the other Patch or SimulSelect to be deactivated before retrying. Or modify the patch to remove the talkgroup or base station from the definition. Contact your supervisor if you do not have permissions to modify the patch.
Activation failed! Exceeded maximum number of active local Patches.	The number of allowed active local patches is licensed.	If you need to active more patches, notify your supervisor or administrator to purchase a higher license.
Activation failed! At least one of the Talkgroups has an active marker tone.	An attempt was made to activate a patch in which at least one talkgroup or base station has an active marker tone.	Wait for the marker tone to be deactivated before retrying. Or modify the patch to remove the talkgroup or base station from the definition. Contact your supervisor if you do not have permissions to modify the patch.
Deactivation failed! Patch <NAME> is in emergency.	An attempt was made to deactivate a patch that is in emergency.	Wait for the emergency to be cleared before deactivating the patch.
Deactivation failed! Patch <NAME> has an active marker tone.	An attempt was made to deactivate a patch that has an active marker tone.	Disable the marker tone or wait for it to be disabled before deactivating the patch.
Transmit not allowed! Patch <NAME> contains RX-only Talkgroup(s).	Transmit cannot be done on a patch that is Rx-only or contains Rx-only entities.	Deactivate the patch and make all the entities in the patch non-Rx-only, then reactivate the patch to transmit on it.
Transmit not allowed! Patch <NAME> is inactive.	Transmit cannot be done on a patch that is inactive.	Activate the patch to transmit on it.
Cannot declare emergency on Patch in Motorola Mode.	Console is currently configured to be in Motorola mode, so an emergency cannot be declared on patches in this mode.	Declare the emergency on one or more talkgroups within the patch. If it is necessary to declare an emergency on the patch itself, notify your supervisor.
Emergency operation failed! Patch is not active.	An emergency cannot be declared on a patch that is not active.	Activate the patch before declaring an emergency on it.
Activation failed! SimulSelect <NAME> is already active.	An attempt was made to activate a SimulSelect that was already active.	N/A

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
SimulSelect <NAME> activation failed! VNIC denied request.	The system has been configured not to allow this console to activate SimulSelects.	Notify your supervisor or system administrator if you believe this is in error.
SimulSelect <NAME> activation failed! At least one of its Talkgroups is an All-Call group.	An attempt was made to activate a SimulSelect in which at least one talkgroup is an All-Call group. All-Call groups cannot be SimulSelected.	Try again after removing the All-Call group from the SimulSelect. Contact your supervisor if you do not have permissions to modify the SimulSelect.
SimulSelect <NAME> activation failed! At least one of its Talkgroups is in an active Patch/SimulSelect.	An attempt was made to activate a SimulSelect in which at least one talkgroup or base station is part of another Patch or SimulSelect.	Wait for the other Patch or SimulSelect to be deactivated before retrying. Or modify the SimulSelect to remove the talkgroup or base station from the definition. Contact your supervisor if you do not have permissions to modify the SimulSelect.
SimulSelect <NAME> activation failed! At least one of its Users is in an active SimulSelect.	An attempt was made to activate a SimulSelect in which at least one user is part of another SimulSelect.	Wait for the other SimulSelect to be deactivated before retrying. Or modify the SimulSelect to remove the user from the definition. Contact your supervisor if you do not have permissions to modify the SimulSelect.
Transmit not allowed! SimulSelect <NAME> contains RX-only Talkgroup(s).	Transmit cannot be done on a SimulSelect that is Rx-only or contains Rx-only entities.	Deactivate the SimulSelect and make all the entities in the SimulSelect non-Rx-only, then reactivate the SimulSelect to transmit on it.
Cannot transmit on remote SimulSelects.	Transmit cannot be done on a remote SimulSelect.	N/A
Cannot transmit on inactive SimulSelect <NAME>.	Transmit cannot be done on an inactive SimulSelect.	Activate the SimulSelect before transmitting on it.
Cannot mute SimulSelects directly.	An attempt was made to mute a SimulSelect. SimulSelects cannot be muted but you can mute the entities inside the SimulSelect.	N/A
A conflict occurred while activating <NAME>.	There was an issue while trying to activate a Patch or SimulSelect. It is possible that one or more entities in this Patch or SimulSelect was added to another Patch or SimulSelect at the same time.	Try again and if it fails, look at the Patch/SimulSelect definition to find the offending entity. Reactivate the Patch or SimulSelect without that offending entity in the definition.
Add to Patch failed! Vocoder mismatch.	This entity cannot be added to the patch because its vocoder is not compatible with the other entities in the patch. It is possible that the console is configured not to allow entities with mixed vocoders to be added to a Patch.	Notify your supervisor if that setting needs to be changed. Otherwise, create the patch without that entity.
Add to Patch failed! Talkgroup is in emergency.	This entity cannot be added to the patch because it is in emergency.	Wait for the emergency to be cleared before retrying. Or create the patch without that entity.
Add to Patch failed! Talkgroup is an All-Call group.	This entity cannot be added to the patch because it an All-Call group.	N/A
Add to Patch failed! Talkgroup is in an active Patch or SimulSelect.	This entity cannot be added to the patch because it is part of another Patch or SimulSelect.	Wait for the other Patch or SimulSelect to be deactivated before retrying. Or create the patch without that entity.
Add to Patch failed! Talkgroup limit reached.	More talkgroups cannot be added to this patch because the entity limit has been reached already or will be crossed once this talkgroup gets added.	N/A

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
Add to Patch failed! Undefined Error.	There was an unknown error while trying to add this talkgroup to the patch.	Try again later (there may have been a network issue) and if it still fails, send logs to TAC.
Patch Limit reached, cannot add more Patches.	The predefined patches limit has already been reached.	<ul style="list-style-type: none"> • Delete one or more pre-defined patches. • Create an Ad-hoc patch. • Contact your administrator to purchase a license to allow for more pre-defined patches.
Add to SimulSelect failed! Vocoder mismatch.	This entity cannot be added to the SimulSelect because its vocoder is not compatible with the other entities in the SimulSelect. It is possible that the console is configured not to allow entities with mixed vocoders to be added to a SimulSelect.	Notify your supervisor if that setting needs to be changed. Otherwise, create the SimulSelect without that entity.
Add to SimulSelect failed! Talkgroup is an All-Call group.	This entity cannot be added to the SimulSelect because it is an All-Call group.	N/A
Add to SimulSelect failed! Talkgroup is in an active Patch or SimulSelect.	This entity cannot be added to the SimulSelect because it is part of another Patch or SimulSelect.	Wait for the other Patch or SimulSelect to be deactivated before retrying. Or create the SimulSelect without that entity.
Add to SimulSelect failed! User is in an active SimulSelect.	This entity cannot be added to the SimulSelect because it is part of another Patch or SimulSelect.	Wait for the other Patch or SimulSelect to be deactivated before retrying. Or create the SimulSelect without that entity.
Add to SimulSelect failed! Entity limit reached.	More talkgroups or units cannot be added to this SimulSelect because the entity limit has been reached already or will be crossed once this entity gets added.	N/A
Add to SimulSelect failed! Undefined error.	There was an unknown error while trying to add this entity to the SimulSelect.	Try again later (there may have been a network issue) and if it still fails, send logs to TAC.
SimulSelect Limit reached, cannot add more SimulSelects.	The predefined SimulSelects limit has already been reached.	<ul style="list-style-type: none"> • Delete one or more pre-defined SimulSelects. • Create an Ad-Hoc SimulSelect. • Contact your administrator to purchase a license to allow for more pre-defined SimulSelect.
Cannot declare emergency on RX-only entity.	Emergency cannot be declared on an entity that is Rx-only.	Notify your supervisor, if needed, or go to the directory and make the entity non-Rx-only.
Cannot ack emergency on RX-only entity.	Emergency cannot be acked on an entity that is Rx-only.	Notify your supervisor, if needed, or go to the directory and make the entity non-Rx-only.
Cannot clear emergency on RX-only entity.	Emergency cannot be cleared on an entity that is Rx-only.	Notify your supervisor, if needed, or go to the directory and make the entity non-Rx-only.
Cannot declare emergency on Talkgroup 0.	Talkgroup 0 is an invalid group that gets added to console for various reasons. Cannot declare emergency on it.	N/A
User ID is not defined.	ID has not been entered or is invalid in the Connection Manager	Open Symphony Connection Manager and enter valid credentials for User ID and password. You may have to log into UAS and enter new User IDs.

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
Cannot transmit while page is in progress.	There is an active Page in progress and you cannot transmit until the Page is complete. This happens when the console is configured for foreground paging and an attempt is made to transmit while the page is in progress.	<ul style="list-style-type: none"> • Wait until the page is over before trying to transmit. • You can also configure your console for Background Paging, to be able to transmit on a different entity while a page is in progress.
Cannot transmit while page to a unit is in progress.	There is an active Page in progress to a unit and you cannot transmit until that Page is complete.	N/A
Cannot transmit on entities that are part of ongoing page.	An attempt was made to transmit on an entity that is currently involved in an ongoing page.	Wait for the page to complete before attempting to transmit on any entity involved in the page.
Page <NAME> was terminated.	A page was terminated for some reason.	Check the system history for more information – this may help addressing the actual reason for page termination. Retry sending the page, and if it still fails, send logs to TAC.
Page <NAME> is invalid.	An attempt was made to send or abort an invalid page. Check the Directory dialog for more information on why the page is invalid.	N/A
Definition not found for Page <NAME>.	An attempt was made to send or abort a page that does not exist. You should almost never see this error message.	N/A
Page <NAME> not configured correctly.	This page is invalid.	Check the Directory dialog for more information on why the page is invalid. Sometimes it is possible to make the page valid again while it is running, other times, the page definition will have to be corrected and the console restarted. Notify your supervisor.
Page <NAME> not active.	This page is not active and does not need to be aborted.	N/A
Cannot declare emergency while page is in progress.	A page is active and there an emergency cannot be declared while it is in progress.	Wait for the page to complete or abort the page before declaring emergency.
Invalid attempt to send page to <NAME>. Pages cannot be sent to acknowledged Users.	A page cannot be sent to an acknowledged user.	N/A
Cannot drive unmonitored Auxiliary Outputs.	This Auxiliary output is unmonitored and its state cannot be changed.	Monitor the auxiliary output before trying to change its state.
Alert does not exist for Page.	This alert definition does not exist.	Notify your supervisor to configure the alert in the Symphony Configuration utility.
Auto SimulSelect not enabled for Page.	An alert was defined for this composite page without Auto-SimulSelect enabled.	Rebuild the composite page by enabling Auto-SimulSelect.
No destination found for picked Page.	One of the picked pages does not have a destination. It is possible that the destination entity no longer exists.	Notify your supervisor to validate that the page is defined correctly.
SimulSelect recall is not available when page is in progress.	An attempt was made to recall the last page SimulSelect while a page is in progress.	Wait for the page to complete or abort the page before recalling the last page SimulSelect.

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
There was a problem recalling the last SimulSelect.	There was a problem recalling the last SimulSelect. It is possible the entity does not exist anymore or one or more entities within the SimulSelect definition could not be SimulSelected.	Send logs to TAC.
There was an error sending Page.	There was an error while trying to send a composite page. It is possible that the sequences within the composite page were invalid or did not have destinations.	Rebuild the composite page.
This Page's destination cannot be added to the SimulSelect.	There was a problem trying to add a page to the composite page. It is possible that the page destination is not compatible with the other destinations in the list.	Send logs to TAC.
Page was terminated because it was preempted by higher priority call.	A higher priority call preempted the page.	Retry the page after the call ends.
Unsupported vocoder for Patch encryption.	The vocoder of this patch does not support encryption.	N/A
Unsupported vocoder for SimulSelect encryption.	The vocoder of this SimulSelect does not support encryption.	N/A
Not able to persist KMF data.	There was some problem trying to save KMF data.	Notify your administrator to make sure you have the right permissions.
Base Station report contained invalid values for the following fields <FIELD_NAME>	The UAC sent some bad values in its report to the console. This could be a result of improper configuration of the base stations.	Notify your supervisor or system administrator to make sure the base station is configured properly.
Invalid state for Base Station operation.	There was a problem trying to change the state of a control on the base station.	Notify your administrator to make sure that the control is configured correctly on the base station.
Requested Base Station operation for (<NAME>) involves unprovisioned channel <NAME>.	An attempt was made to change channel to an unprovisioned channel or lock an unprovisioned channel.	Notify your supervisor.
Group <NAME> for Base Station <NAME> delta provisioned.	A group that is part of a base station was just provisioned.	N/A
Group <NAME> for Base Station <NAME> delta unprovisioned.	A group that is part of a base station was just unprovisioned.	Notify your supervisor.
Cannot transmit as entity <NAME> does not exist.	This entity does not exist.	Notify your supervisor.
There is an active Patch preventing this Page from being picked.	This page cannot be picked because the destination of this page is not compatible with the destination(s) of the picked pages.	N/A
Base Station <NAME> is not available.	This base station is not available.	Notify your supervisor to make sure the base station is configured properly and running.
Base Station <NAME> is busy.	This base station is busy.	Try again.
Cannot declare emergency on a Base Station Channel.	Emergency cannot be declared on a base station channel.	N/A
Page destination not available. Base Station down.	This page's destination is a base station that is currently down.	Notify your supervisor to make sure the base station is configured properly and running.
Page <NAME> was terminated because it cannot preempt higher-priority call.	This page was terminated because a higher priority call is in progress.	Wait for the call to end before retrying page.

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
Page failed because the destination of Page <NAME> cannot be SimulSelected.	This page failed because its destinations cannot be SimulSelected. It is possible that one or more destinations are in another Patch or SimulSelect. Or the vocoders of the destinations are not compatible.	Notify your supervisor to configure the pages and/or destinations to allow the page to succeed.
A valid User ID to send pages in background is not available.	The console is configured for background paging but a valid User ID is not available.	Open Symphony Connection Manager and enter valid credentials for Background Paging User ID. You may have to log into UAS and enter new User IDs.
All pages have been revalidated.	A valid background paging ID is available. All pre-defined pages have been revalidated.	N/A
Page <NAME> was terminated because the destination Base Station channel changed during the page.	This page was terminated because the base station channel changed while the page was in progress.	Retry the page again.
Licensed number of setups has been exceeded. Truncating setups list.	More setups have been created than is licensed by the console.	Notify your supervisor if a higher license needs to be purchased.
Licensed number of workspace tabs has been exceeded. Truncating workspace tabs collection.	More workspace tabs have been created than is licensed by the console.	Notify your supervisor if a higher license needs to be purchased.
Licensed number of Patches has been exceeded. Truncating Patches collection.	More Patches have been created than is licensed by the console.	Notify your supervisor if a higher license needs to be purchased.
Licensed number of SimulSelects has been exceeded. Truncating SimulSelects collection.	More SimulSelects have been created than is licensed by the console.	Notify your supervisor if a higher license needs to be purchased.
Cannot select base station's channel! Select the Base Station instead.	A base station's channel cannot be directly selected.	Select the base station instead.
Cannot select a remote SimulSelect.	A remote SimulSelect cannot be selected.	N/A
Cannot select an inactive SimulSelect! Activate it before selecting it.	An inactive SimulSelect cannot be selected.	Activate the SimulSelect.
Cannot select an inactive Patch! Activate it before selecting it.	An inactive Patch cannot be selected.	Activate the Patch, then try again.
Invalid entity! Unable to select it.	This is an invalid entity and cannot be selected.	Notify your supervisor.
Cannot unmonitor <NAME> while it has an active local marker tone.	This entity cannot be unmonitored because it has an active local marker tone.	Disable the marker tone before trying to unmonitor the entity.
Cannot enable/disable marker tone for ICOM groups.	ICOM groups do not support marker tones.	N/A
Cannot enable marker tone. Exceeded maximum number of marker tones allowed.	The maximum number of active local marker tones has been reached. Cannot enable another marker tone.	Disable one or more marker tones to be able to enable a new marker tone.
Marker tone already active for entity <NAME>.	Marker tone is already active for this entity.	N/A
Marker tone already active for entity <NAME>. Enabled by remote console.	A marker tone has already been enabled on this entity by a remote console.	N/A
Marker tone already inactive for entity <NAME>.	Marker tone has already deactivated for this entity.	N/A

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
Marker tone activated for entity <NAME> by <NAME>.	Status message. No error.	N/A
Marker tone deactivated for entity <NAME> by <NAME>.	Status message. No error.	N/A
Marker tone enable/disable requested for unknown ID <NAME>.	An attempt was made to enable/disable marker tone on an unknown entity.	Notify your supervisor.
Marker tone disabled for ICOM group <NAME>.	Marker tone was disabled for this group because it has been updated to be an ICOM group.	N/A
Marker tone cannot be enabled on inactive patch <NAME>.	Marker tone cannot be enabled on an inactive patch.	Activate the patch, and then try again.
Marker tone cannot be enabled or disabled on a RX-only entity <NAME>.	Marker tone cannot be enabled or disabled on an entity that is Rx-only.	Open the directory and make the entity non-Rx-only before retrying.
Cannot enable marker tone. Marker tone feature not licensed.	Cannot enable marker tone because this feature is not licensed.	Notify your supervisor if a license needs to be purchased for marker tone.
A valid User ID to send marker tones is not available.	A valid Marker tone ID is not available.	Open Symphony Connection Manager and enter valid credentials for Marker Tone User ID. You may have to log into UAS and enter new User IDs.
Cannot enable RX-only on entity <NAME> with active marker tone.	This entity has an active local marker tone and cannot be made Rx-only.	Disable the marker tone before trying again.
Cannot change channel on Base Station <NAME> while there is an active marker tone.	When there is an active marker tone on a Base Station's current channel, the channel cannot be changed until the marker tone is disabled.	Disable the marker tone on the base station before attempting to change channels.
Base station <NAME> down! Disabling marker tone.	There was an active marker tone on the Base Station that is currently down and so the marker tone was disabled.	Notify your supervisor or network administrator and let them know that the Base Station is down. The marker tone can be re-enabled once the base station is back up. You can also press the Refresh button on the base station module to re-check the status.
Cannot enable marker tone on Base station <NAME> while it is down.	A marker tone cannot be enabled on a Base Station while it is down.	Notify your supervisor or network administrator and let them know that the Base Station is down. The marker tone can be enabled once the base station is back up. You can also press the Refresh button on the base station module to re-check the status.
Tone (<NAME>) is missing or cannot be loaded properly.	A problem occurred attempting to send a tone because the tone file is either missing or is corrupted.	Notify your supervisor or network administrator and let them know that the tone file cannot be read. You may have to do a repair operation using the Symphony installer to get the tone file installed.
Duration of tone (<NAME>) is too short.	A problem occurred attempting to send a tone because the tone is too short. This typically happens if a pre-installed tone file was replaced by a custom tone file.	Notify your supervisor or network administrator and let them know that the tone is too short. To get the previous tone file installed, do the repair operation using the Symphony installer. Otherwise, replace the current tone file with a longer tone.

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
Duration of tone (<NAME>) is too long.	A problem occurred attempting to send a tone because the tone is too long. This typically happens if a pre-installed tone file was replaced by a custom tone file.	Notify your supervisor or network administrator and let them know that the tone is too long. To get the previous tone file installed, do the repair operation using the Symphony installer. Otherwise, replace the current tone file with a shorter tone.
Marker tones turned off due to VNIC being down.	The connection to the VNIC has been lost and therefore all active Marker Tones have been turned off.	Notify your supervisor or network administrator and let them know that your connection to the VNIC has been lost.
Marker tone disabled for <NAME>! It is in a patch now.	An active Marker Tone for this entity has been disabled because the entity is now in an active patch.	Either enable marker tone on the talkgroup/patch or wait until the patch is deactivated.
Marker tone request for <NAME> denied by another console.	This happens when an attempt was made to enable a Marker Tone on an entity that already has a Marker Tone enabled by another console. The console has not yet been updated to show the enabled status. Depending on the marker tone interval, it can be a few seconds or several minutes before the marker tone status gets updated.	Wait for a few minutes and verify that the Marker Tone button on the talkgroup module shows the enabled status. If it does not update, then try to re-enable marker tone on that entity. If you still get the same message, notify your supervisor to address the matter.
Max active marker tones exceeded. Request denied for <NAME>.	The maximum of 1 in-progress local Marker Tones has already been reached.	N/A
Max active background pages exceeded. Request denied for <NAME>.	There can only be one active background page at a time.	Either wait for the current active page to complete or abort it and then try again.
Cannot set RX-only on entity <NAME> while it is in a Patch or SimulSelect.	Entities that are part of an active Patch or SimulSelect cannot be made RX-only.	Wait for the Patch or SimulSelect to become deactivated then try again.
U2U Endpoint <NAME> already in active U2U Session.	An attempt was made to place an I-Call to a user that is already in an active I-Call.	Wait for the I-Call session to complete then try again.
Cannot unmonitor Local Auxiliary Outputs.	Local Auxiliary outputs cannot be unmonitored.	N/A
Invalid attempt to TX to <NAME> while console in Backup Radio Mode.	An attempt was made to transmit on the console while Backup Radio mode was enabled. Console can be used for transmit operations only when not in Backup Radio mode.	N/A
Cannot clear emergency on group with active inbound/outbound call.	An attempt was made to clear emergency on an entity while there was an active inbound or outbound call on it.	Wait until the active call on the entity is complete then try again.
Console is not licensed for AES Encryption.	An attempt was made to enable encryption on a talkgroup or user module when there is no license for AES encryption.	Notify your supervisor if a license needs to be purchased for AES encryption.
Invalid System Key.	An attempt was made to encrypt a user when there was no valid System Key.	Notify your supervisor about this issue. The encryption keys may not be properly loaded.
Cannot change encryption state of user that is in an active I-Call session.	An attempt was made to change the encryption state of a unit while it was in an active I-Call session.	Wait for the session to end before trying again. It may be necessary to end the call.
I-Call with <NAME> failed to be setup before timing out.	An I-Call setup failed because the setup timer expired at the recipient side.	Try again.

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
Page to user <NAME> terminated because it failed to setup the I-Call.	I-Call setup to a unit failed the current page. The unit may be turned off or busy.	Try again.
Policy violation! Unable to change unselect speaker.	The console is probably configured not to allow speaker changes.	Notify your supervisor to check permissions.
Policy violation! Unable to change volume.	The console is probably configured not to allow volume changes, or the volume is being changed below the min allowed volume.	Notify your supervisor to check permissions.
Policy violation! Unable to change mute state.	The console is probably configured not to allow mute.	Notify your supervisor to check permissions.
Policy violation! Unable to change monitor state.	The console is probably configured to not allow changes to the monitored state of the entities.	Notify your supervisor to check permissions.
Policy violation! Unable to change encryption state.	The console is probably configured not to allow encryption state changes.	Notify your supervisor to check permissions.
Policy violation! Unable to change RX-only state.	The console is probably configured not to allow Rx-only to be set on entities.	Notify your supervisor to check permissions.
Policy violation! Unable to change Default Select Entity.	The console is probably configured not to allow changes to the Default Select Entity.	Notify your supervisor to check permissions.
Unavailable! Replay file not found.	An attempt to replay a call failed because the replay file could not be found.	Send logs to TAC.
Unavailable! Console not licensed for setup management.	This console is not licensed for managing or creating setups.	Notify your supervisor if a license for setups needs to be purchased.
Unavailable! VIDA I-Call TX to <NAME> not a licensed feature.	An outbound I-Call to a unit was rejected because this console is not licensed for I-Calls.	Notify your supervisor if a license for I-Calls needs to be purchased.
Unavailable! Unlicensed VIDA I-Call RX from <NAME> denied.	An inbound I-Call from a unit was rejected because this console is not licensed for I-Calls.	Notify your supervisor if a license for I-Calls needs to be purchased.
Call Alert failed. Unit <NAME> unregistered in system.	Call alert to this unit failed because this unit is not registered. It is possible that this unit is turned off.	Notify your supervisor.
Unsupported User Action! Setup request would exceed max licensed setups of <COUNT>.	A new setup cannot be added because the licensed number of setups has been reached.	Notify your supervisor if a higher license is required to be purchased.
Unsupported User Action! Talkgroups in Patch <NAME> have mismatched vocoders.	This entity cannot be added to the Patch because its vocoder is not compatible with the other entities in the Patch. It is possible that the console is configured not to allow entities with mixed vocoders to be added to a Patch.	Notify your supervisor if that setting needs to be changed. Otherwise, create the Patch without that entity.
Unsupported User Action! Entities in SimulSelect <NAME> have mismatched vocoders.	This entity cannot be added to the SimulSelect because its vocoder is not compatible with the other entities in the SimulSelect. It is possible that the console is configured not to allow entities with mixed vocoders to be added to a SimulSelect.	Notify your supervisor if that setting needs to be changed. Otherwise, create the SimulSelect without that entity.
Unsupported User Action! SimulSelects cannot contain Users in Motorola Mode.	The console is currently configured to be in Motorola mode. In this mode, SimulSelects cannot contain units.	Notify your supervisor if this is in error.
Unsupported User Action! Maximum Sequence limit exceeded for Page.	More sequences have been picked than is allowed in a page.	N/A

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
Unsupported User Action! The attempted selection would exceed the destination limit.	This page cannot be picked because the page destination limit will be exceeded.	If SimulSelect is not needed, then uncheck the Auto Simul box and re-pick this page.
Unsupported User Action! The attempted selection has a destination with a mismatched vocoder.	This page cannot be picked because its destination has a vocoder that is incompatible with the remaining destination entities in the page list.	If SimulSelect is not needed, then uncheck the Auto Simul box and re-pick this page.
Unsupported User Action! Page contains Base Station which would exceed SimulSelect limit.	This page cannot be picked because it has a base station in its destination list and this would exceed the number of destinations allowed in a SimulSelect.	If SimulSelect is not needed, then uncheck the Auto Simul box and re-pick this page.
Unsupported User Action! Page contains multiple channels from same Base Station.	This page cannot be picked because it has multiple channels from the same base station in its destination list.	If SimulSelect is not needed, then uncheck the Auto Simul box and re-pick this page.
Invalid User Action! Already transmitting to <NAME>.	There is already a transmit in progress. Cannot initiate another transmit until it ends.	N/A
Invalid User Action! State of User alert on <NAME> invalid for ACK.	The user alert is in an invalid state and cannot be acked.	Send logs to TAC.
Invalid User Action! State of User alert on <NAME> invalid for CLEAR.	The user alert cannot be cleared without being acked.	Ack the user alert before trying to clear it. If that does not work, send logs to TAC.
Invalid User Action! ACK required before Clear on emergency <NAME>.	The emergency cannot be cleared without being acked.	Ack the emergency before trying to clear it. If that does not work, send logs to TAC.
Invalid User Action! Setup <NAME> already exists.	This setup cannot be added or renamed because a setup with the same name already exists.	N/A
Invalid User Action! Requested setup name too long: <NAME>.	This setup name is too long. It needs to be within 14 characters.	N/A
Invalid User Action! Setup name contains invalid character(s): <NAME>.	This setup name contains invalid characters.	Try again without invalid characters in the name.
Invalid User Action! Setup operation requires setup name.	The setup name cannot be empty.	Try again with a valid setup name.
Invalid User Action! Password for setup <NAME> too long.	This setup password is too long. It needs to be within 14 characters.	N/A
Invalid User Action! Patch <NAME> does not contain enough groups.	This patch does not contain enough entities.	Update the patch definition to contain at least two entities.
Invalid User Action! SimulSelect does not contain enough entities.	This SimulSelect does not contain enough entities.	Update the SimulSelect definition to contain at least two entities.
Invalid User Action! Call Director is on hook.	Cannot talk on the Call Director since it is on hook.	Call Director should be off hook before being used.
Invalid User Action! The attempted selection has a destination that is already active.	This page cannot be added to the composite page because its destination is part of another Patch or SimulSelect, or is an All-Call group.	Wait for the Patch or SimulSelect to be deactivated before trying again. Or send the composite page without that page.
Warning! One or more Sequences may not have reached Page destination. Base Station down.	The base station that is part of the current page's destination is down. The page may or may not have successfully reached the station.	Try again if needed.
Incoming Unit Call from {0} failed encryption negotiation - non-P25 vocoder.	Could not accept incoming encrypted I-Call because console's vocoder does not support encryption.	Changing the console's vocoder to one that supports encryption can avoid this issue. Notify your administrator or supervisor.

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
Outgoing Unit Call to {0} failed encryption negotiation - non-P25 vocoder.	Attempted encrypted I-Call with a unit whose vocoder does not support encryption.	Attempt an unencrypted I-Call. Or changing the unit's vocoder to one that supports encryption can avoid this issue. Notify your administrator or supervisor.
Page terminated! The current selected entity is in an active TX.	Console terminated the current page, as the current selected entity is in an active transmission.	Try paging again after the TX is finished on the selected entity.
Cannot ack, Aux I/O Point <NAME> is not configured as Acknowledged.	Console could not acknowledge the Aux I/O point as it is not configured as acknowledged.	If this is not the desired behavior, then notify your supervisor to configure the Aux I/O point to acknowledged.
Aux I/O alarm <NAME> acked remotely by User <NAME> is not configured as Acknowledged.	The Aux I/O alarm could not be acknowledged remotely by User, as the Aux I/O point is not configured as Acknowledged.	If this is not the desired behavior, then notify your supervisor to configure the Aux I/O point to acknowledged.
Corrupt Provisioning data received from the VNIC.	Console received a corrupted provisioning data from VNIC.	Restart the console. If the issue persists, notify your supervisor.
Page with a SimulSelect failed because it exceeded the destination limit.	Console failed a page with SimulSelect since the destination limit for the page is exceeded.	Try paging to fewer destinations (check the destination limit with your supervisor).
Replay file no longer available due to configured limit.	Replay file was made unavailable by console due to the configured time limit.	Notify your supervisor and check the configured limit for replay file.
The ALC has been disabled because an Audio parameter is out of range.	Console disabled ALC, due to an out-of-range audio parameter.	Notify your supervisor to check the ALC settings.
Aux I/O network device <NAME> is configured with an invalid community string!	Aux I/O device configuration error occurred due to the invalid community string used.	Notify your supervisor to configure the Aux I/O device with the valid community string.
Aux I/O network device <NAME> has a problem!	Problem occurred in Aux I/O network device.	Notify your supervisor.
Load keys operation included invalid FIPS bindings.	Load keys operation was failed since the FIPS mode is enabled and DES keys are loaded.	Notify your supervisor to disable the FIPS mode to load DES keys.
Patch <NAME> was deactivated with Reason Code: <REASON CODE>!	Console deactivated the patch due to the mentioned reason code.	Notify your supervisor.
Group <NAME> is active in call, try to unmonitor later!	Console failed to unmonitor the group as it is in an active call.	Try unmonitoring the group after the call is done.
Unable to TelePatch <NAME>. Reason - <REASON CODE>.	Attempt to TelePatch to an entity failed.	Address the issue raised by the <REASON CODE> and try again.
A valid User ID to do Call Director TelePatch is not available.	A configured User ID needs to be entered in Connection Manager before attempting to do TelePatch.	Run Connection Manager to enter an ID for Call Director TelePatch. If an ID is not available, contact your administrator.
Cannot TelePatch! Call Director should be off hook.	Attempt to TelePatch failed because the Call Director is not off hook.	Make a call to or from the Call Director to be able to TelePatch.
Cannot change setup! There is an active TelePatch.	Attempt to change setup failed because there is an active TelePatch.	End the TelePatch before attempting to change setup.
Cannot send page to destination in an active TelePatch.	Attempt to page failed because the destination is in a TelePatch	Try paging after ending the TelePatch.
Cannot set AutoSimulSelect because the selected entity is in an active TelePatch.	Attempt to set AutoSimulSelect for page failed because the selected entity is in a TelePatch.	Try paging after ending the TelePatch or page without AutoSimulSelect.

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
Cannot send page with AutoSimulSelect because the selected entity is in an active TelePatch.	Attempt to send page with AutoSimulSelect failed because the selected entity is in a TelePatch.	Try paging after ending the TelePatch or page without AutoSimulSelect.
Cannot activate the Patch because one of the member entities is in an active TelePatch.	Attempt to activate a patch failed because one of the member entities is in a TelePatch.	Try again after ending the TelePatch.
Cannot activate the SimulSelect because one of the member entities is in an active TelePatch.	Attempt to activate a SimulSelect failed because one of the member entities is in a TelePatch.	Try again after ending the TelePatch.
Cannot deactivate Patch because it is in an active TelePatch.	Attempt to deactivate a patch failed because the patch is in a TelePatch.	Try again after ending the TelePatch.
Remote Patch deactivation is forcing the current TelePatch to end.	Console ended the TelePatch to remote patch because it has been deactivated.	N/A
Remote Patch activation is forcing the current TelePatch to end.	Console ended the TelePatch to entity because it is now in a remote patch.	N/A
Cannot declare an emergency on the Patch because it is in an active TelePatch.	Attempt to declare emergency on a patch failed because it is in a TelePatch.	Try again after ending the TelePatch.
A remotely declared emergency is forcing the current active TelePatch to end.	Console ended the TelePatch because the entity is now in emergency.	N/A
Max TelePatch calls exceeded. Request denied for <NAME>.	Only one TelePatch call can be established at a time.	N/A
A Base Station channel change is forcing the current TelePatch to end.	Console ended the TelePatch to base station because its current channel has changed.	N/A
Cannot pick entity because it is in an active TelePatch.	Attempt to pick entity for Patch/SimulSelect failed because the entity is in a TelePatch.	Try again after ending the TelePatch.
Talkgroup <NAME> cannot be unmonitored while in an active TelePatch.	Attempt to unmonitor a talkgroup failed because it is in a TelePatch.	Try again after ending the TelePatch.
TelePatch denied or torn down because VNIC not connected.	Attempt to establish or maintain TelePatch failed because VNIC is disconnected.	Notify your network administrator.
TelePatch denied or torn down because Audio Box not connected.	Attempt to establish or maintain TelePatch failed because Audio Box is disconnected.	Notify your administrator.
Modify request on <NAME> failed for invalid state or mode: <REASON CODE>.	Attempt to modify Patch or SimulSelect failed.	Address the issue raised by the <REASON CODE> and try again.
VNIC NAKed modify request: <REASON CODE>.	VNIC denied request to modify Patch or SimulSelect.	Address the issue raised by the <REASON CODE> and try again.
SIP Server is disconnected.	The connection to the SIP server has failed.	Check for other SIP related error messages to determine root cause of the failure.
SIP account <account> registration error. Server error <error>	There was an error registering an account with the SIP server. The issue was related to a PBX error	Ensure that the PBX server is correctly configured before attempting reconnection.
ABR call is queued.	Another call is already in progress.	N/A
The ABR is still in transmit state.	The ABR is currently in a call	Wait for the current call to end
The ABR is still in the receive state	The ABR is currently in a call	Wait for the current call to end.
There is no ABR audio stream. Cannot start transmit.	The ABR connection has failed.	Restart Symphony.

DISPLAYED MESSAGE	EXPLANATION/CAUSES	RECOMMENDED SOLUTION OR ACTION
There is no ABR receive audio stream.	The ABR connection has failed.	Restart Symphony.
There is no ABR audio stream.	The ABR connection has failed.	Restart Symphony.
ABR audio connection is lost.	The ABR connection has failed.	Restart Symphony.
The ABR is connected.	The ABR is connected	N/A
The ABR is disconnected.	The ABR connection has failed.	Check that the ABR is powered on and connected to the Symphony console. Reconnect if necessary.
Trying to connect to the ABR. Please wait.	Symphony is attempting to reconnect to the ABR	N/A
Cannot clear emergency on entity with active ABR inbound/outbound call.	The ABR is currently in a call	Wait for the current call to end.
Cannot clear emergency using ABR when VNIC is connected.	Emergencies can only be cleared from the ABR when the VNIC is disconnected.	Clear the emergency from the Emergency Side Panel or the corresponding talkgroup.
Cannot transmit from the ABR when another call is in progress.	Another call is already in progress.	Wait for the current call to end.

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About L3Harris Technologies

L3Harris Technologies is an agile global aerospace and defense technology innovator, delivering end-to-end solutions that meet customers' mission-critical needs. The company provides advanced defense and commercial technologies across air, land, sea, space and cyber domains.