SIP Module Installation Guide KI-2912

Responder® 5000 SIP Module Installation Guide

A Division of AMETEK, Inc.

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Table of Contents

1

1: General Information

Should you be new to configuring Nurse Call Systems or new to Responder 5000 Phone Integration, we **strongly recommend** that you

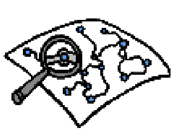
- familiarize yourself with the entire R5K product line (best explained in the *Responder 5000 Configuration Guide* and *Responder 5000 Component Installation Guide*) and
- take the time to review and understand this entire document before attempting to configure any system.

Revision History

This is the “A” version of KI-2912.

Revision	Summary
O	– Initial Release
A	– Updated Alerts, Cautions and Warnings – Updated SIP Server Section

Scope of this Document



Read this document if your duties include installing or maintaining the R5K system.

Customer Connection/Extranet

Rauland Customer Connection

You can use Rauland's secure Customer Connection site to find, view, and/or download many support documents—including manuals, drawings, and reports. To request an account, follow the online instructions at: http://customerconnection.rauland.com.

Alerts, Precautions, and Limitations

Observe the following alerts, precautions, and system limitations:

- ✓ The use of wireless phones is considered a secondary means of informing staff members of patient calls. All sources of patient calls must still be covered by one or more supervised, UL-Listed nurse call Consoles and/or Duty Stations. These devices are the primary means of annunciating calls and they should be located where staff members can see and hear them.
- ✓ SIP hardware and third party software should be installed by a trained installer ONLY.
- ✓ SIP component should be tested after installation
- ✓ Installer should ensure the third party SIP system is tested and verified with R5K system before facility go-live use.
- ✓ Installer should ensure the third party SIP system is designed to handle the call traffic R5K system will be sending to it.
- ✓ All system callpoints covered by SIP devices should also be covered by R5K Console(s).
- ✓ R5K system does not support placing calls on hold.
- ✓ It is recommended to avoid placing a SIP device as coverage on the last call stop unless that call stop is also covered by a console. SIP devices are not supervised. Therefore, the installer shall ensure any SIP device covering the last Call Stop of an Escalation Chain is also covered by a console. Without console coverage, a call could go un-answered due to a device that is turned off or out of range.
- ✓ Installer should assign SIP Devices to the MSC where majority of their covered devices reside.

Related Documents

Other, related information can be found in the following documents:

- ✓ *Responder 5000 Configuration Guide* (KI-2908)
- ✓ *Responder 5000 Component Installation Guide* (KI-2907)
- ✓ *Responder 5000 Telephony Solutions Application and Design Guide*

2

2: System Overview

Responder 5000 (R5K) is a flexible and adaptive platform that integrates with virtually all telephony systems in a clinical environment. R5K uses Voice over Internet Protocol (VoIP) technology and Session Initiation Protocol (SIP) for direct and cost-effective communication with a telephone system's devices/server. Alternatively, R5K can offer connectivity to legacy phone systems that do not use SIP.

SIP is used for all telephony integrations, both SIP and analog. The R5KMSIP Module on the R5KMSC is used to connect to a telephone system's SIP server where the phones/devices are registered. The telephone system's SIP Server creates a "SIP trunk" to the R5KMSC.

The benefits of establishing a SIP trunk between the phone system's SIP server and R5KMSIP are:

- ✓ Minimizes the nurse call footprint on telephone system (all audio nurse call components seen as a single trunk)
- ✓ Saves setup/programming time
- ✓ Reduces risk of incompatibility
- ✓ Provides a demark for system upgrades and changes

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General SIP Configuration Guidelines and Tips

Each wireless phone integration has a unique set of performance characteristics dependent upon many factors including:

- ✓ Capabilities of the handset models involved
- ✓ Versions of firmware and software within each model
- ✓ Facility network configuration
- ✓ Facility network and phone traffic
- ✓ Intended use of the integrated solution

Each R5KMSC can support up to 4 SIP audio sessions at a time. To avoid conflicting with the 4-audio-per-MSC limit, it is recommended the dial-plan will route dial-in audio to the MSC that “owns” the room being called. When this is the case, the SIP audio limit corresponds well to the K-Bus audio channel limit. Exceptions to this routing are permitted but dialing into an MSC to communicate to a room on another MSC will prevent the MSC from utilizing the 4 SIP audio channels for audio its own 4 K-Bus runs. Creating a dial-plan ensures SIP requests match system resources.

It is also recommended to configure SIP phones on the MSC that owns the bulk of the rooms that they annunciate. This way, answering calls use the local channels and the 4-audio-per-MSC limit is optimized.

Please also observe the following:

- ✓ Disable call-forwarding from the phones receiving nurse calls
- ✓ Voice mail should not be used for SIP handsets that receive patient calls (contact the phone system vendor for assistance configuring voice mail when receiving non-patient calls).
- ✓ Disable Re-Invite from the wireless phone system communicating with R5K
- ✓ Make sure to include SDP (Session Description Protocol) in the Invite sent from the phone system
- ✓ Use the proper codec for your region:
  - G.711 μ-law codec for North America
  - G.711 a-law for Europe and Asia
- ✓ Disable SIP Session Timer at both Responder SIP Server and wireless phone system

SIP System Integration

The diagram below is an example of a direct SIP integration to a wireless SIP-capable phone/device platform, without the use of middleware or third party integration vendors.

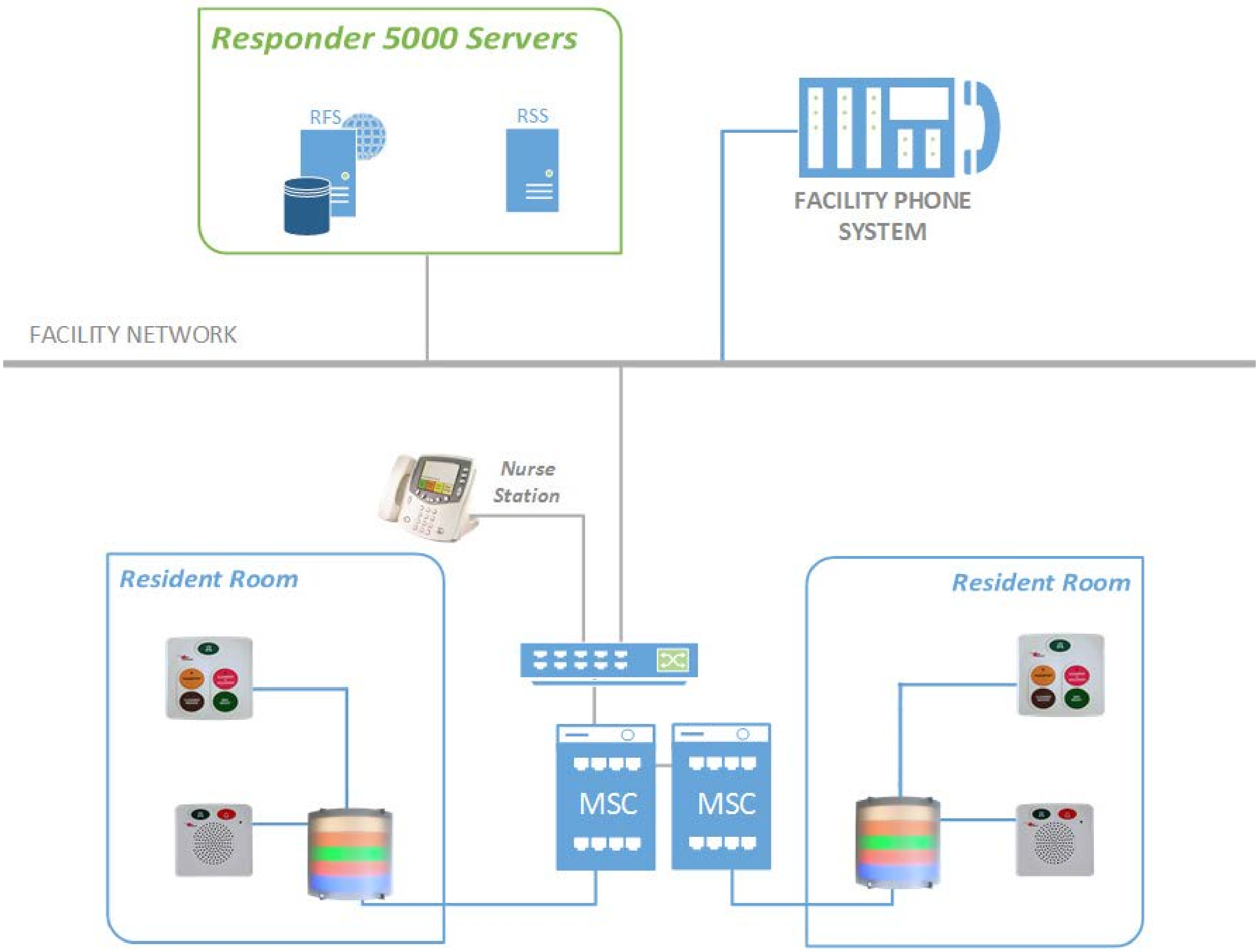


Figure 1: SIP Phone System Integration

The MSC is directed to the facility phone system with a SIP trunk. The IP address for the phone system SIP Server (or the optional Responder SIP Server shown in above diagram) is set in R5KWare under System Options>Servers & Email>SIP Server Settings:

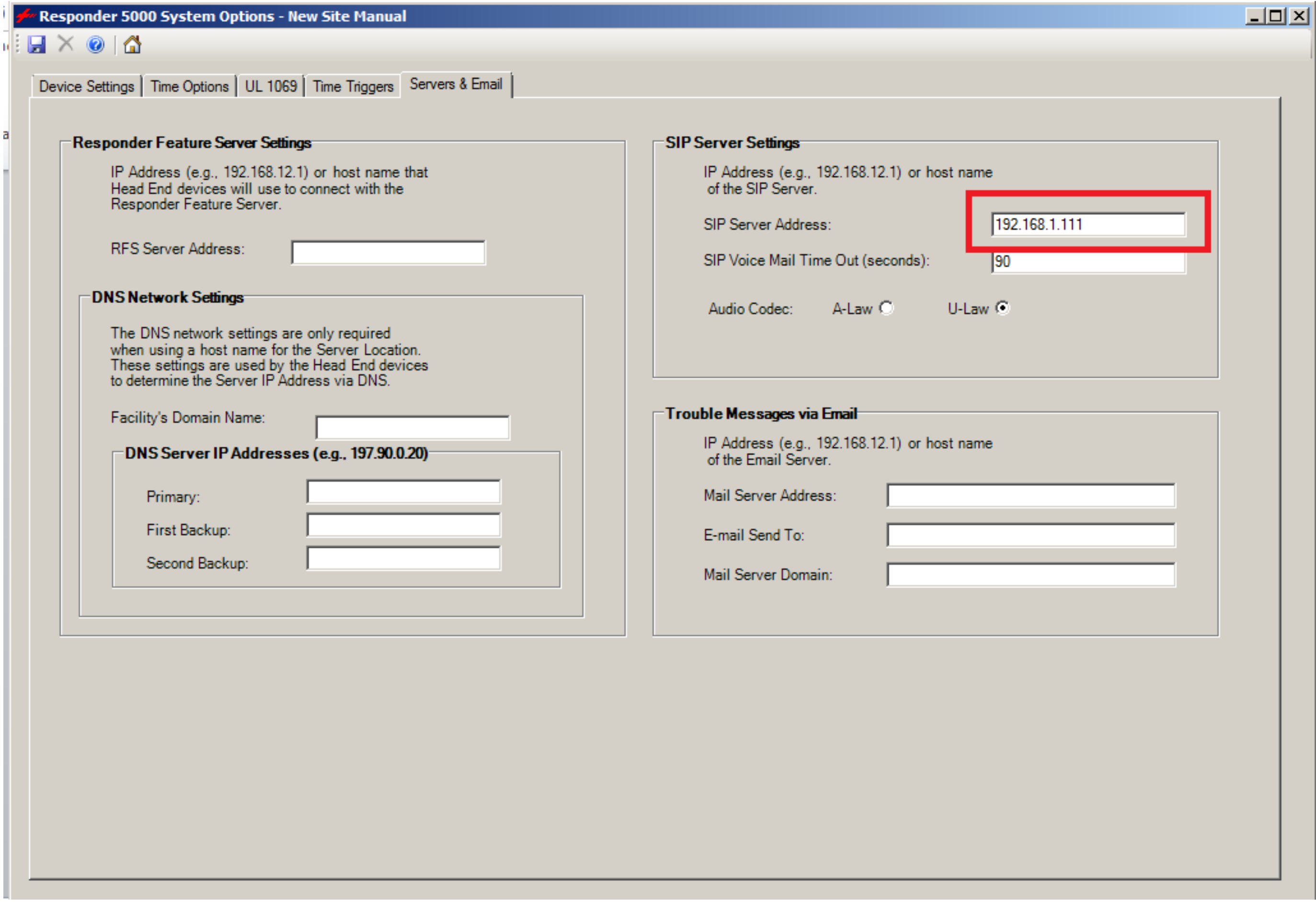


Figure 2: SIP Server IP Address

Analog Phone System Integration

If the phone system is not SIP-capable (analog PBX/FXO port), you can use a VoIP gateway router to convert the SIP call as shown below:

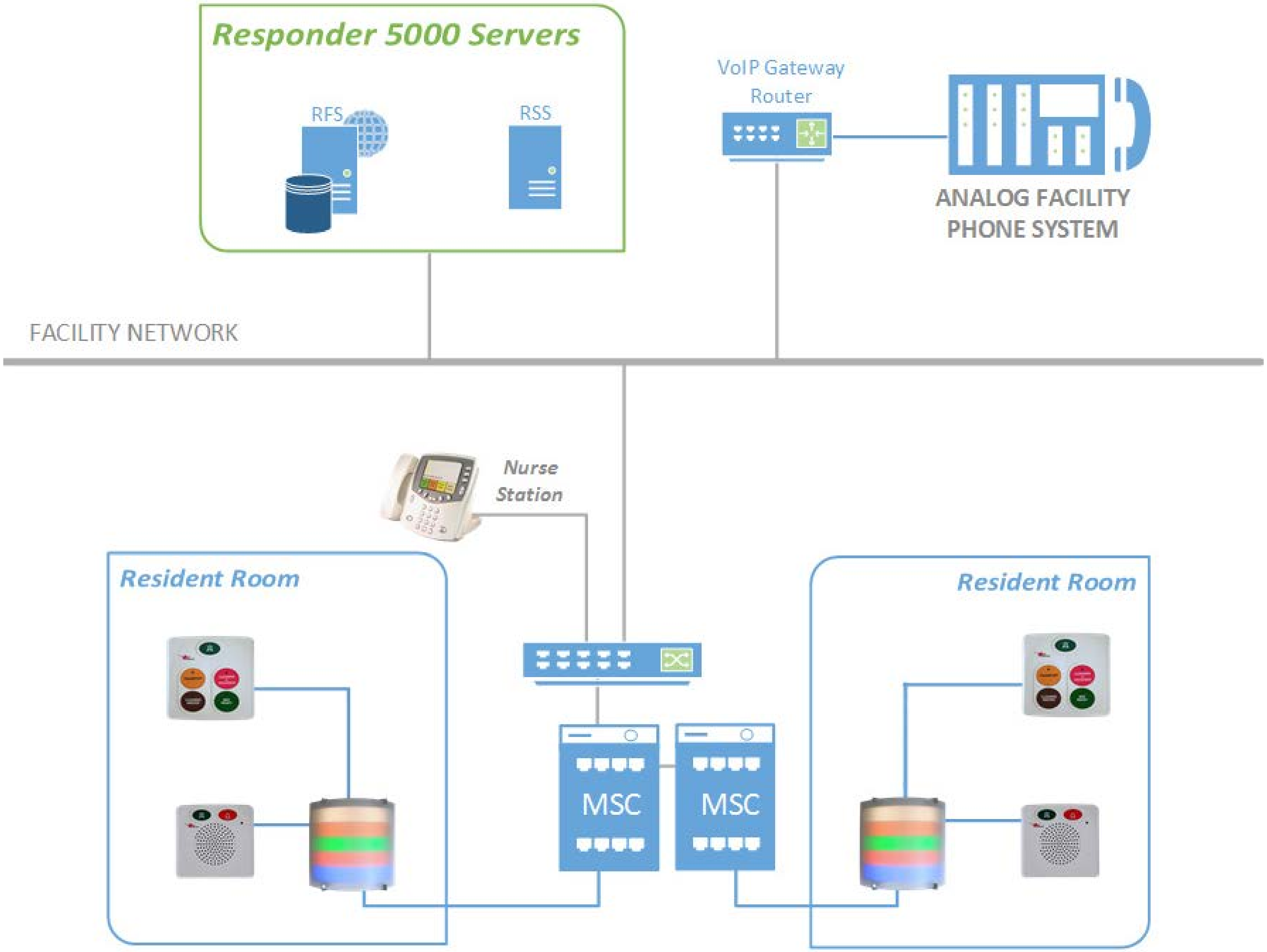


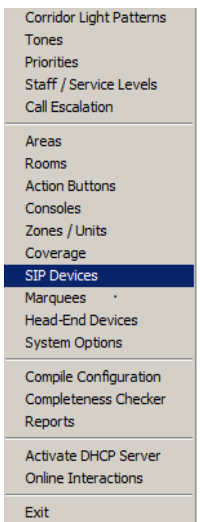
Figure 3: Analog Phone System Integration

3

3: R5KWare SIP Editor

Finding the Editor

Open Responder 5000 Configuration software (R5KWare) and select SIP Devices from the menu:



- The use of wireless phones is considered a secondary means of informing staff members of patient calls. All sources of patient calls must still be covered by one or more supervised, UL-Listed nurse call Consoles and/or Duty Stations. These devices are the primary means of annunciating calls and they should be located where staff members can see and hear them.
- SIP hardware and third party software should be installed by a trained installer ONLY.
- SIP component should be tested after installation.
- Installer should ensure the third party SIP system is tested and verified with R5K system before facility go-live use.
- Installer should ensure the third party SIP system is designed to handle the call traffic R5K system will be sending to it.
- All system callpoints covered by SIP devices should also be covered by R5K Console(s).
- R5K system does not support placing calls on hold.
- It is recommended to avoid placing a SIP device as coverage on the last call stop unless that call stop is also covered by a console. SIP devices are not supervised. Therefore, the installer shall ensure any SIP device covering the last Call Stop of an Escalation Chain is also covered by a console. Without console coverage, a call could go un-answered due to a device that is turned off or out of range. Installer should assign SIP Devices to the MSC where majority of their covered devices reside.

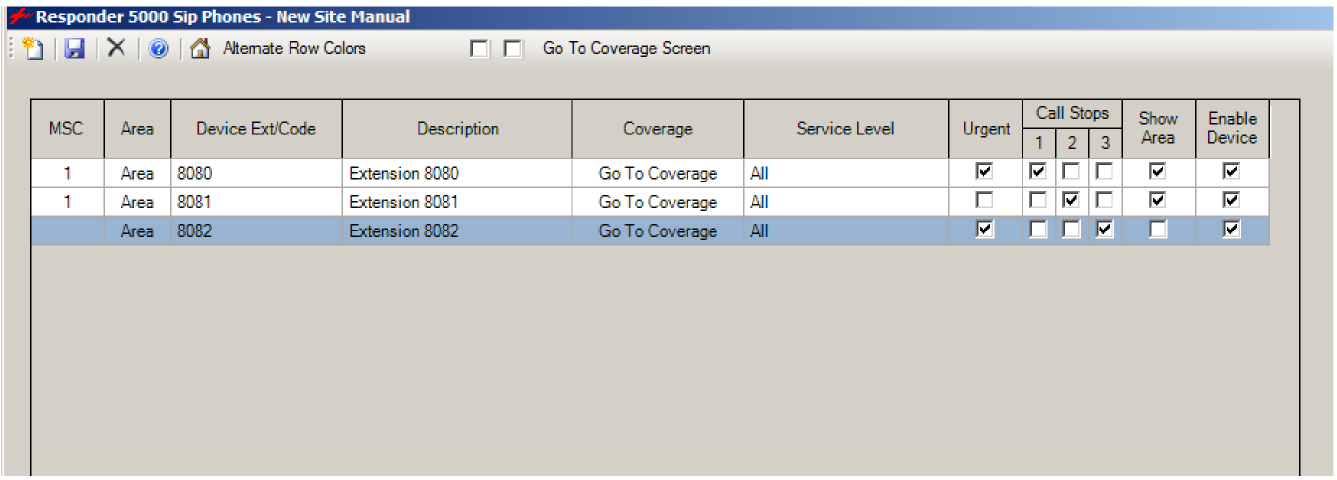
SIP Basics

Here's what you should keep in mind as you use the SIP phone editor:

- ✓ Calls annunciated to SIP phones are based on coverage for that phone.
- ✓ Coverages need to be assigned to each SIP phone.
- ✓ The R5K supports a maximum of ten (10) Phones per MSC.

First Look: SIP Configuration

When the configurator first appears, it will feature the following configuration fields:



MSC	Area	Device Ext/Code	Description	Coverage	Service Level	Urgent	Call Stops 1	Call Stops 2	Call Stops 3	Show Area	Enable Device
1	Area	8080	Extension 8080	Go To Coverage	All	☑	☑	☐	☐	☑	☑
1	Area	8081	Extension 8081	Go To Coverage	All	☐	☐	☑	☐	☑	☑
	Area	8082	Extension 8082	Go To Coverage	All	☑	☐	☐	☑	☐	☑

Figure 4: SIP Configurator

SIP Phone Information

The SIP Phone list displays already created SIP Phone:

MSC

Use this field to associate the SIP phone with a R5KMSC using the drop-down list of MSCs already configured. If you do not see the MSC in this list check the Headend Devices screen to see if the MSC has previously been configured. The MSC selected will be responsible for sending the call annunciation session invites to the SIP device.

Area

The Area is a dropdown field that allows you to select the area in which phone is associated. This field is used primarily to organize the device in the coverage screen explorer tree.

Device extension/Code

Use this ten (10) digit field (numbers only) to specify the extension of the SIP device on the SIP system.

Description

Description is a text field that describes the SIP phone. This is used as a reference in configuration software only.

Coverage

Selecting the coverage field will allow the user to configure the selected SIP phone. The SIP phone coverage can be customized on the Coverage screen. **Note**: when adding a new SIP phone, it has no coverage; you must assign it on the Coverage screen.

Service Level

It is a drop-down field that allows the user to set the service request this phone will receive based on covered by the SIP phone. User must select from one of the service levels listed, default is ‘None’.

Urgent

Selecting this box will have all Escalation Chains marked Urgent, within the defined coverage annunciate on the SIP phone.

Call Stops

Call Stops allows a SIP phone to annunciate only on the defined call stops checked. If the following criteria are met:

1. The call is within the SIP phone’s coverage
2. The Escalation Chain is not marked Urgent

3. The Escalation Chain is marked "Evaluate Staff Coverage"

The Device will annunciate only on the specified call stop. For example, when a phone has call stops 2 and 3 checked, it will annunciate calls only when they occur on the second and third call stops. This could be explained as only notify the SIP phone when a call has been escalated.

Show Area

It is a checkbox field that allows users to show the Area on the SIP phone when a call is initiated. If the box is unchecked, SIP phone display will only show the Room Name and Bed.

Enable Device

It is a checkbox field that will allow the users to enable the SIP phone. If the box is unchecked, the SIP phone is disabled in the system and does not cover any calls.

How It's Done

The SIP Configurator allows you to:

- ✓ Create a New SIP phone
- ✓ Delete Existing SIP phone
- ✓ Configure a SIP server

To Add a new SIP phone:

1. Access the SIP configurator (Screens | SIP Devices).
2. Click on new device to add a new SIP phone:

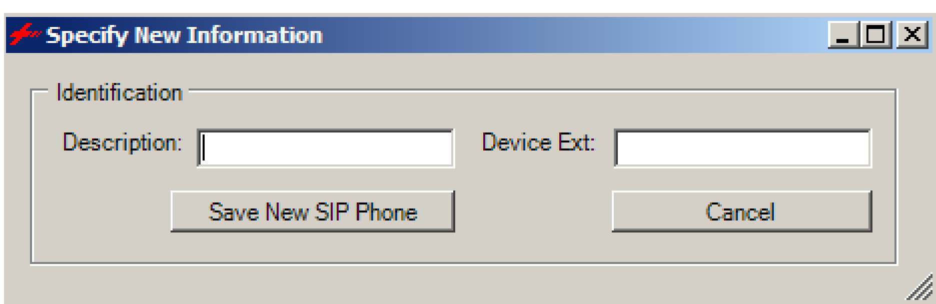


Figure 5: SIP Device - New

3. Enter the Description and the Device Extension and save the new device information. (Figure 5)
4. Enter the MSC that you want the phone to be configured on. (Figure 4)
5. Enter the Coverage for the phone by selecting coverage for a room in that area you want the phone to cover. (Figures 6 and 7)

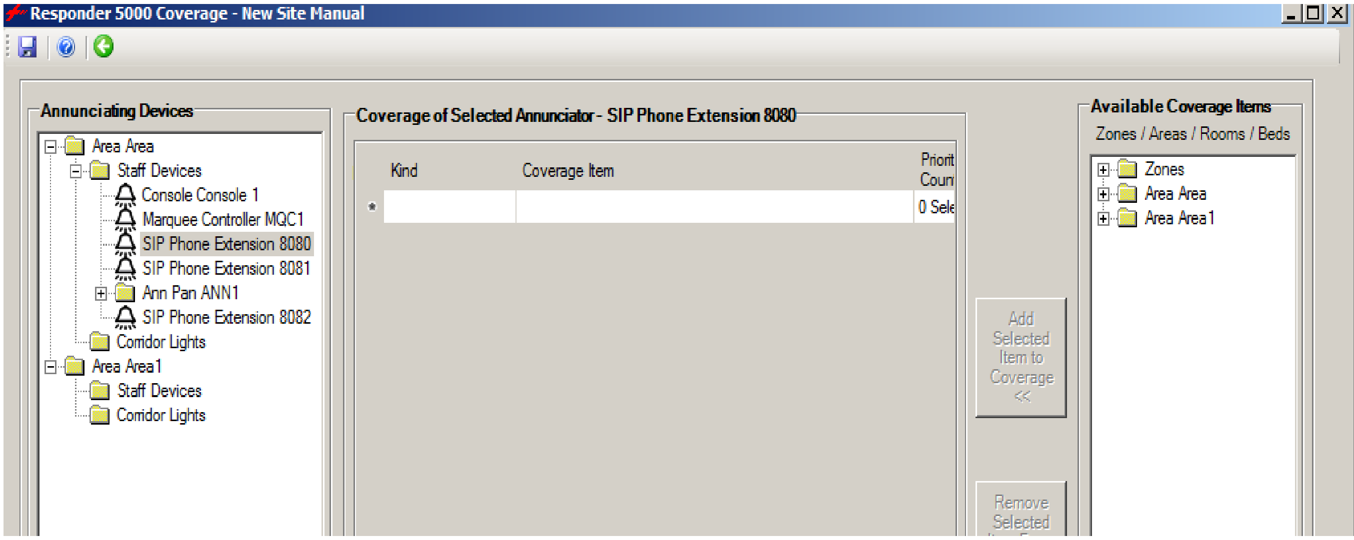


Figure 6: Coverage Screen

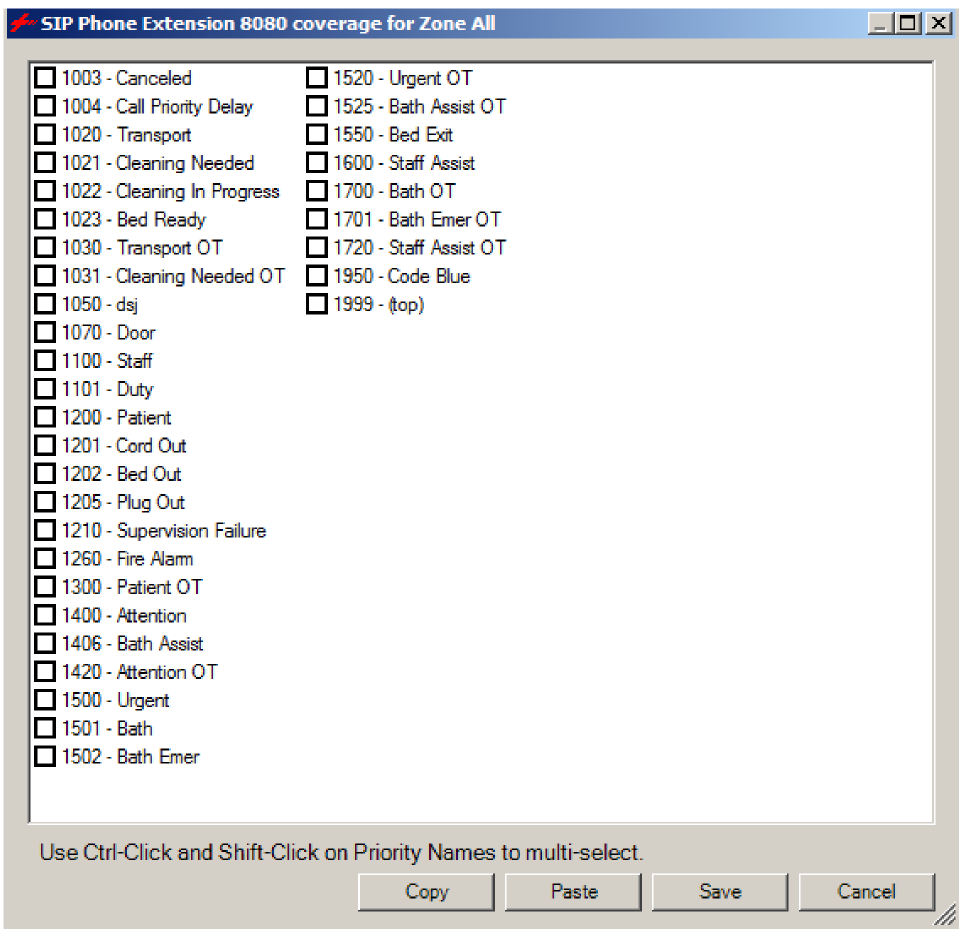


Figure 7: Coverage for SIP Phone

6. Select the service level that you want the SIP phone to cover using the drop-down box. (Figure 4)
7. Select whether the SIP phone received Urgent call escalation chains. (Figure 4)
8. Set the Call Stops that you want the phone to attend. (Figure 4)
9. Check the 'Show Area' box as needed. (Figure 4)
10. Check the 'Enable Device' as needed. (Figure 4)

To Remove a SIP phone from the configuration

1 Access the SIP configurator (Screens | SIP Devices).
2 Click to select any SIP phone in the configuration.
3 Click on 'Delete Selected Device' from menu.
- The SIP phone will disappear from the configuration.

To Configure a SIP server in the R5K configuration

1 Access the SIP Server settings (Screens | System Options | Servers & Email).

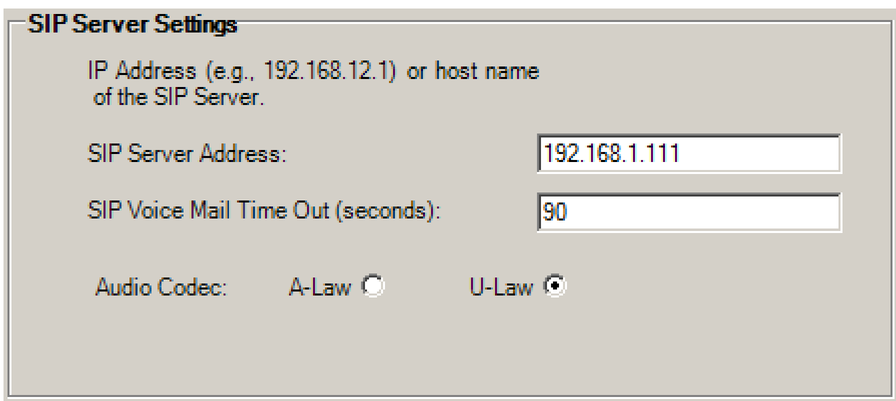


Figure 8: System Options SIP Server Settings

2 Enter a valid IP address for the SIP server.
3 Set the Voicemail timeout in seconds.
- This is the time that the SIP phone will ring without timing out and allowing the call to go to voicemail.
- Set the Audio Codec to either A-Law or U-Law.
  - G.711 a-law – check with SIP provider, typically Europe and Asia
  - G.711 µ-law – check with SIP provider, typically North America

4

4: Responder SIP Server Integration

A Responder SIP Server (RSS, SKU 355005) can optionally be used for all telephony integrations. This SIP Server performs as a registrar and proxy for SIP-enabled wireless phones that may not be part of an in-house phone system (phones only used for Responder 5000 calls). The SIP Server is also used to connect to a telephone system's SIP server where the phones/devices are registered. The telephone system's SIP Server creates a "SIP trunk" to the Responder SIP Server. In this application the R5KMSC creates a SIP trunk to the RSS and the RSS creates a SIP trunk to the phone system SIP server.

Responder SIP Server Specifications

Responder SIP Server should always be installed on a standalone server (can be virtualized), with specifications similar to the Responder Feature Server. Java SE 7 or 8 (32bit/64bit provided by Oracle Corporation) is also required. The software is purchased from Rauland. First tier support is provided by Rauland Technical Services, and as with other Rauland software, included as part of the distributor's SMA. Once you have downloaded the software from Customer Connection you will need to activate it with a product ID (provided by Customer Service when item "ships"). For ease of installation, please make sure you have an internet connection to the machine you will be activating the software on. You can activate the software without a connection, but it is best to have the connection available.

Step 1: Install Java
Install Java SE before installing the Responder SIP Server software:

1. Go to: http://www.oracle.com/technetwork/java/javase/downloads/index.html
2. Download and install the appropriate version of JRE or JDK for the type of Windows OS you are running.

   If you already have Java in your computer, please make sure that the Java version is 1.6 or later. We recommend using Java provided by Oracle.

Step 2: Install Responder SIP Server

1. Obtain the executable installer and a Product ID for Responder SIP Server.
2. Start the installer.
3. Continue the installation by following the installer's instructions. The SIP Server will be installed automatically.

   If you check the [Run Responder SIP Server] box at the last stage of the installation and push the [Finish] button, the SIP Server's HTTP service will start automatically.

Step 3: Start Responder SIP Server's HTTP service

If you did not check [Run Responder SIP Server] at the last stage of the installation, please start the SIP Server's HTTP Service by the following:

1. Open [Control Panel] > [Administrative Tools] > [Service].
2. Select [Responder SIP Server] and start the service.
3. Set server "Startup Type" as "Automatic"
4. Restart your computer. The SIP Server's HTTP service will automatically start.

Step 4: Start Responder SIP Server Administration Tool (Admintool)

1. Select [Start] > [All Programs] > [Responder SIP Server] > [Responder SIP Server Admintool]. A web browser will open and you will see the License Agreement page. Copy and paste the Product ID you have to product ID field.
   Follow the instruction to activate product. (Entering the same product ID on multiple machines is not allowed.)
2. At the Admintool Login page, enter User ID and Password and push [Login] button. The default administrator's User ID is "sa" and its Password is "sa".
3. After the login, push the [Start] button at [Status] -> [Start/Shutdown] page. If the Status is Active, the Responder SIP Server has started successfully. If the Status is Inactive, the server has not started successfully, the error should be shown.

Note: When the Responder SIP Server's port number (default port 5060) is already in use by another application, the server status will be shown as Inactive. For example, if you attempt to start the server while another SIP UA is running on the same computer, the server may fail to start. In this case, please stop the other SIP UA, and click the [start] button on the Admintool's [Start/Shutdown] page.

Responder SIP Server Programing

Most direct SIP programming for Responder SIP Server should follow the defaults for all items except as noted below. The two step setup process involves:

- ✓ Set up an Interface Address of the Responder SIP Server host.
- ✓ Setting one or more Dial Plans to direct specific ranges of phone numbers to the phone system's SIP server/gateway so that calls from R5K to those phones are routed correctly

Using the Responder SIP Server Admin interface, select Dial Plan.

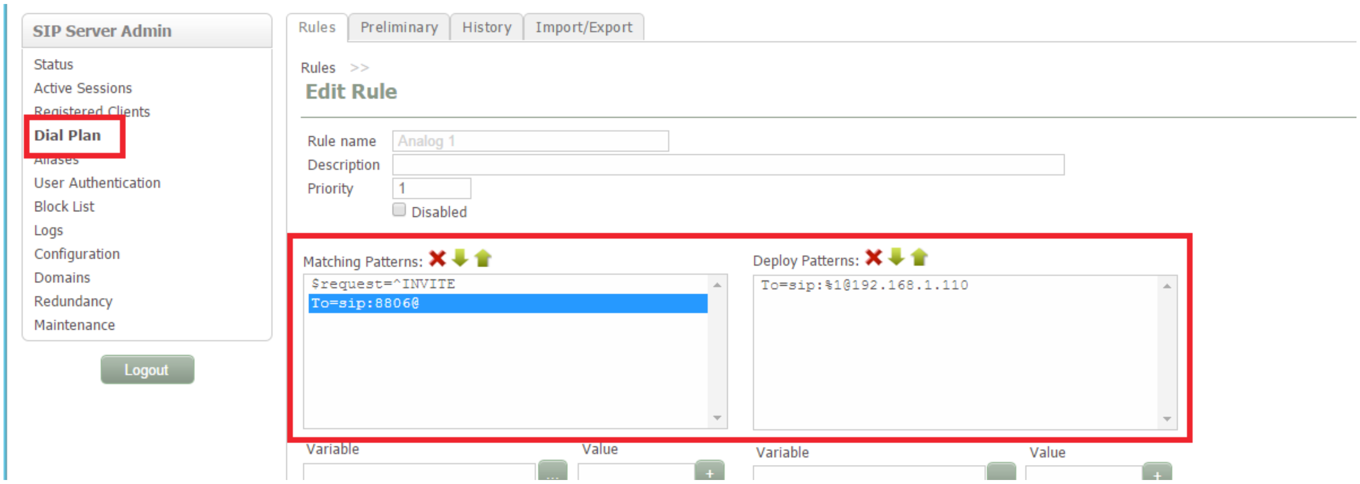


Figure 9: Responder SIP Server Admin Interface

1. Add a new rule in “Matching Patterns”:
   - ➢ $request=^INVITE
   - ➢ To=sip:[extension of the SIP phone]@
2. Add a new rule in “Deploy Patterns”:
   - ➢ To=sip:%1@[IP Address of the SIP PBX]
3. Repeat steps 1 and 2 to create Dial Plans for all SIP phones.

Note: If all extensions start with the same number (ex. 7XXX) you can use To=sip:(7.+)@ for creating a single “Matching Pattern”.

After a SIP session has been established from a patient call to a wireless phone, the actual audio conversation or RTP (Real-Time Transport Protocol) is conducted between the R5K MSC and the wireless phone.

Some integrations such as Avaya and certain SIP client phone apps require having the RTP routed through the RSS so that it appears to these particular phone systems that all voice packets are originating from a single IP address. By enabling RTP Relay in the RSS, all of the RTP conversations flow through the RSS. Since all of the communication must now flow through the SIP Server instead of a direct MSC to handset approach, it is strongly recommended to use a separate RSS when enabling RTP Relay.

The additional programming steps involve:

- ➢ Ensuring Network Address Translation and Authentication are turned off. Enabling RTP Relay in the Configuration > RTP section.
- ➢ Within NAT Traversal, set Keep address/port mapping to off and within Authentication set REGISTER to off and INVITE to off.
- ➢ Set RTP Relay to on, RTP Relay (UA on this machine) to auto, Port Mapping to source port and click Save and restart to complete.