

Product: Ascom AA60 Voice Appliance, UNITE with Rauland Responder 5
 Purpose: Configuration of Ascom AA60, T1 VoIP Gateway and Unite with Rauland Responder 5
 Date: February 16, 2011

#### Introduction

This document outlines the necessary steps and guidelines to integrate the Ascom Unite Messaging System with Rauland Responder 5 System and the Ascom AA60, serving as the call management application for Rauland Responder 5, with the Ascom T1 VoIP Gateway. [NOTE: The AA60 is not required if Ascom handsets are to be treated simply as SIP endpoints in a PBX for which SIP interoperability has been tested and verified by Ascom]. There are two scenarios for this implementation; 1) The Responder 5 initiates calls to an Ascom portable when any selected patient event is triggered. 2) The Responder 5 sends a text message to the Ascom portable which then allows the user to call back into the patient room, if desired.

This guide is intended for someone knowledgeable on the configuration of the Ascom T1 VoIP Gateway and AA60 and is assumed that the user has already installed Unite products, according to their respective installation guides. (See Related Documents section). The steps, screenshots, and guidelines depicted throughout this document are based upon Ascom VoIP Gateway software version 7.00 hf3, AA60 software version C.3.1 and Unite Medamax Gateway software version 3.00.

The Rauland Responder 5 (R5) will register audio stations as SIP extensions on a Brekeke SIP Server, software version 2.3.8.2. . The AA60 establishes a SIP trunk with the T1 VoIP Gateway and another SIP trunk with the Brekeke SIP Server, in order to broker calls between the R5 and Ascom handsets.

Appendix A provides a description of the recommended settings and configuration for a Rauland Responder 5. For detailed information contact your Rauland technical support representative.

Manufacturer	Rauland Borg Corporation
Manufacturer products	Responder 5
Physical interface method	RS232 with 9600, 7, E, 1, no flow control
Protocol	TAP 1.8
UNITE Product	Medamax Gateway (SW version 3.00)

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### AA60 Initial Setup

The AA60 is initially configured to obtain an IP address using DHCP. Once the booted, a connected monitor will display the AA60's IP Address on the main screen. Using Firefox, or similar browser, enter the IP address of the AA60 into the address bar (http://XXX.XXX.XXX.XXX) on another computer connected to the same LAN.

#### Ascom AA60 Network Configuration

The AA60 shall be configured with a static IP Address. Follow the below steps to change the IP Address to a static address.

- 1. With a compatible web browser, navigate to <u>http://AA60-IPADDRESS:10000</u> (where AA60-IPADDRESS is the IP Address of the AA60).
- 2. Sign in with Username root and associated Password (i.e. password, changeme, etc.)
- 3. Click on the *Networking -> Network Configuration -> Network Interfaces* link and then choose the *Activated at Boot* tab.

Mod	lule	Ind	lex
11100	1010		~~~

## Network Interfaces

Active Now Activated at Boot					
Interfaces listed in this table will be activated when the system boots up, and will generally be active now too.					
Select all.   Invert sele	ction.   Add a new interfac	ce.   Add a new address ran	ge.		
Name	Туре	IP Address	Netmask	Activate at b	
🔲 ethO	Ethernet	172.20.96.200	255.255.255.0	Yes	
🔲 lo	Loopback	127.0.0.1	255.0.0.0	Yes	
рррО	PPP (PPP Dialup	Client) Automatic	Automatic	No	
Select all.   Invert selection.   Add a new interface.   Add a new address range.					
Delete Selected Int	terfaces Delete and /	Apply Selected Interfaces	Apply Selected Inte	rfaces	

🖕 Return to network configuration .

4. Click on the *eth0* link and after the *Edit Bootup Interface* appears, choose Static Configuration. Enter the desired IP Address, Netmask, and Broadcast values. Click SAVE.

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Name	eth0 Activate at boot?	💿 Yes 🔘 No
Address source	O From DHCP	
	O From BOOTP	
	Static configuration IP Address 172.20.96.200	
	Netmask 255.255.255.0	
	Broadcast 172.20.96.255	
мти	Default     O     Default	0 (Add virtual interface)

Edit Rootun Interface

 Return to the Network Configuration page and choose Routing and Gateways. For Default routes, enter the appropriate Gateway address for Interface eth0 and Save the settings

Module Index	Routing and Gateways
	0 ,

Boot time configuration Active configuration

Module Index

This section allows you to configure the routes that are activated when the system boots up, or when network settings are fully re-applied.

Routing configuration activated at boot time				
Default routes	Interface		Gateway	
	eth0 💌		172.20.96.2	
	~			
Act as router?	🔿 Yes 💿	No		
Static routes	Interface	Network	Netmask	Gateway
Local routes	Interface	Network		Netmask
Save				



6. Navigate to the *System->Bootup and Shutdown* page and select the *Reboot System* button. The system will take a couple of minutes to reboot.

Login: root Webmin Sustem	Change to runlevel: 3 💌
Bootup and Shutdown Change Passwords	Reboot System
Disk Quotas Disk and Network Filesystems	Shutdown System
Filesystem Backup	

### Upload Default Configuration for R5

The AA60 default configuration backup for R5 includes necessary system settings as well as predefined configuration examples to assist in initially configuring the unit. To upload the configuration file, you will need to connect to the AA60 via a SFTP application, such as WinSCP.

- 1. Connect to the AA60 as 'root' with an SFTP application.
- 2. Navigate to the following directory: "/var/lib/asterisk/gui\_backups"
- 3. Copy the default configuration backup for R5 to this directory.
- 4. Connect to the AA60 configuration GUI with a compatible web browser
- 5. Navigate to the *Backup* tab and restore the default configuration backup for R5.

If unable to connect to the AA60 with an SFTP application, please ensure the following settings are enabled.

#### SSH allows root login

- 1. Connect to the AA60 as 'root' with an SSH client, such as putty.
- 2. Edit the sshd\_config file (vim /etc/ssh/sshd\_config)
- 3. Uncomment out the line "PermitRootLogin yes".
- 4. Save the file (":wq").

#### Root password established for SSH

- 1. Connect to the AA60 as 'root' with an SSH client, such as putty.
- 2. At the root prompt, enter command "passwd root" and follow instructions on setting the password.



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## Scenario I – R5 Calls to Ascom Portable Handsets

#### **SIP Trunk to IGWP**

If the R5 will be originating calls to Ascom handsets (no UNITE included in solution), then a SIP trunk to the IGWP must be configured. [NOTE: Scenario 1 requires and additional Gatekeeper License in the VoIP Gateway].

#### AA60 setup

- Connect to the AA60 configuration GUI with a compatible web browser, using the following address: <u>http://AA60-IPADDRESS:8088</u> (where AA60-IPADDRESS is the IP Address of the AA60).
- 2. Navigate to the *Trunks* tab and click on *VoIP trunks*.
- 3. A predefined VoIP trunk for the IGWP is included in the default configuration for R5, called "IGWP".
- 4. Edit this trunk and set the IP Address to the T1 VoIP Gateway IP Address into the hostname field.

#### ascom **Application Note** Product: Ascom AA60 Voice Appliance, UNITE with Rauland Responder 5 Configuration of Ascom AA60, T1 VoIP Gateway and Unite with Rauland Purpose: Responder 5 Date: February 16, 2011 Analog Trunks Service Providers VOIP Trunks 11/E1/BREEPunks 88 mISDN Config + New StellAX Trunk Trunks are outbound lines used to allow the system to make calls IGWP. SIP 172.20.96.120 IGWP. to the real world. Trunks can be BREKEKE SIP 172.20.96.193 BREKEKE VolP lines or traditional telephony lines. Edit SIP trunk trunk 1 X Provider Name 🛈 IGWP Hostname () Se Outgoing Calling Rules 172.20.96.120 22 Dial Plans Username ① ## Users IGWP Password : ## Music On Hold Codecs: Second : None Mind : None Y First : u-law ## Voice Menus Fourth : None Y Fifth : None V 88 Time Intervals CallerID (1) ## Incoming Calling Rules 88 Voicemail FromDomain: 22 Paging/Intercom FromUser: 22 Conferencing 88 Follow Me AuthUser: 22 Call Features insecure: по 💙 88 VoiceMail Groups Outbound Proxy: ## Voice Menu Prompts ## System Info Enable Remote MWI: î٢ 88 Backup 🛇 Cancel 🛛 🗹 Save 99 Ontinns

## SIP Trunk to Brekeke SIP Server

If the Rauland 5 will be registering audio stations through a Brekeke SIP Server, then a SIP trunk from the Brekeke must be configured.

AA60 setup

- Connect to the AA60 configuration GUI with a compatible web browser, using the following address: <u>http://AA60-IPADDRESS:8088</u> (where AA60-IPADDRESS is the IP Address of the AA60).
- 2. Navigate to the *Trunks* tab and click on *VoIP trunks*.
- 3. A predefined VoIP trunk for the Brekeke is included in the default configuration for R5, called "Brekeke".

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4. Edit this trunk and set the IP Address to the the Brekeke SIP Server IP Address in the hostname field.

ee Configure naroware		Analog Trunks	Service Providers	VOIP Trunks	T1/E1/BRI Trunks
88 mISDN Config	🛉 New SIP/IAX Trunk				
Trunks are outbound lines used	Provider Name	Туре	Hostname/IP	Username	
to allow the system to make calls	IGWP	SIP	172.20.96.120	IGWP	E
to the real world. Trunks can be	BREKEKE	SIP	172.20.96.193	BREKEKE	E
VolP lines or traditional telephony lines		N950			
	Edit SIP trunk trunk	4			х
	-		Provider Name		
61			BREKEKE		
			Hostname ①		
器 Outgoing Calling Rules		172	20 96 193		
## Dial Plans			llearnama ()		
## Users		в			
## Ring Groups			Password :		
器 Music On Hold					
## Call Queues			Codecs:		
## Voice Menus		First : U-law	Second : None M Thin	d : None 💙	
## Time Intervals		Fourth	Fifth : None		
St Incoming Calling Rules			CalleriD U:		
P Voicemail			EcomDomain:		
			FromUser:		
as conterencing					
aa Follow Me			AuthUser:		
III Directory					
## Call Features			insecure:		
## VoiceMail Groups			Outbound Proxy:		
88 Voice Menu Prompts					
## System Info			Enable Remote MWI;		
## Backup			F		
RR Options			Cancel Save		

#### T1 VoIP Gateway setup

- 1. Connect to the T1 VoIP Gateway configuration GUI with a compatible web browser.
- 2. Navigate to *PBX* → *Objects* and add a new Gateway object for "IGWP" with the following parameters:
  - a. Long Name = "IGWP"
  - b. Name = "IGWP"

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	lowp	
Long Name	IGWP	Display Name
Name	IGWP	Number Critical
Password		retype Password
Hardware ID		
Node	root 💌	Hide from LDAP
PBX	. 💌	Local
Config Template		
Filter	~	Diversion Filter Reject ext. Calls
Response Tin	neout	Busy On Calls
No Inband Dis	sconnect	
Gateway		
Enblock Cour	nt 🗌	
Enblock as D	iverting No 📃	
Prefix		
International N	Match	
National Matc	:h	
Subscriber M	atch	
ОК	Apply Delete C	ancel

- 3. Navigate to the *Gateway* tab and click on *GK* tab.
- 4. Define a GW interface with the following parameters:
  - a. Name = "From\_AA60"
  - b. Protocol = "SIP"
  - c. Mode = "Gateway without Registration"
  - d. Domain = AA60-IPADDRESS
    - (where AA60-IPADDRESS is the IP Address of the AA60)
  - e. General Coder Preference = "G711u"
  - f. Local Coder Preference = "G711u"

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Name	From_AA60			
Disable				
Protocol	SIP 🔽			
Mode	Gateway without Regist	ration 🔽		
Domain	172.20.96.200			
Proxv		(ontional)		
STUN Server				
STUN Server		(optional)		
Local Port				
Authorizatio	on			
Name				
Password		Retype		
Alias List -				
Name	Number			
Media Prop	erties			
General Cod	er Preference G711u	🖌 Framesize [ms] 30 Silence Compression 🔲 Exclusive 📃		
Local Netwo	rk Coder G711u	Framesize [ms] 30 Silence Compression 📃		
Enable T.38	Enable SRTP	lo DTMF Detection 📃 Enable PCM 📃		
SIP Interop	Tweaks			
Accept INVI	TE's from Anywhere	(affects registered interfaces only)		
Enforce Sending Complete (affects outgoing SIP calls only)				
No Inband In	formation on Error	(affects incoming SIP calls only)		
From Heade	r when Sending INVITE	Fixed AOR (affects registered interfaces only)		
Identity Head	der when Sending INVITE	CGPN in user part of URI 💌 (affects registered interfaces only)		
Reliability of	Provisional Responses	Supported 🛩 (affects outgoing SIP calls only)		

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- 5. Navigate to Gateway  $\rightarrow$  GK
- 6. Make sure the From\_AA60 registers as 0.0.0.0, if you see a nonzero IP address, delete the From\_AA60 Interface and rebuild it.

	VoIP Gateway							
Configuration	General	Interfaces	SIP GK	Routes	CDR0	CDR1	Calls	
General								
IP	Interface	CGP	N-In CDPN-In	CGPN-Out	CDPN-Ou	t Alias	Registra	ation Product
ETH0	GW1 From	_AA60 +					0.0.0.0	
ETH1	GW2	+						
LDAP	GW3	+						
PRI1	GW4	+						
PRI2	GW5	+						
DDI2	GW6	+						
PRIJ	GW7	+						
PRI4	GW8	+						
TEL	GW9	+						
Administration	GW10	+						
PBX	GW11	+						
Gateway	GW12	· · · +						
Download	L							

- 7. Define a GW interface for the local IGWP objects with the following parameters:
  - a. Name = "Local-IGWP"
  - b. Protocol = "H323"
  - c. Mode = "Register as Gateway"
  - d. Gatekeeper Address = "127.0.0.1"
  - e. Alias Name = "IGWP"
  - f. General Coder Preference = "G711u"
  - g. Local Coder Preference = "G711u"

Applicati	on Note ascom
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Name Disable Protocol Mode Gatekeeper Addr Gatekeeper Addr Mask Gatekeeper Ident Local Port Authorization Password Alias List Name	Local-IGWP H323 Register as Gateway ess 127.0.0.1 (primary) ess (secondary) iffer Retype Number
Media Propertie General Coder Local Network Enable T.38 H.323 Interop T No Faststart Suppress HLC OK Ca	s Preference G711u  Framesize [ms] 30 Silence Compression Exclusive  Coder G711u  Framesize [ms] 30 Silence Compression  I Enable SRTP No DTMF Detection Enable PCM  weaks No H.245 Tunneling  Suppress FTY Suppress Subaddr  ncel Apply Delete

- 8. Navigate to Gateway  $\rightarrow$  Routes.
- 9. Add a route that allows calls from the AA60 to the IGWP objects.
  - a. Description = "from AA60"
  - b. Check GW interface defined as "From\_AA60" as the origin interface
  - c. Enable Interworking (QSIG,SIP)
  - d. Select GW interface defined as "Local-IGWP" as the destination interface

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Description from AA60	Disable 🗌	
PRI1 PBX_1       ✓ GW1 From_AA60         PRI2       GW2         PRI3 PBX_2       GW3         TEST       GW4         TONE       GW5         HTTP       GW6         ECHO       GW7         CONF       GW8         SIP1 To_AA60       GW9         RS1 To_AA60       GW10         SIP2       GW12 Local-IGWP	Add UUI         Final Route         Final Map         No Reroute on wrong No         Verify CGPN         Interworking(QSIG,SIP)         Rerouting as Deflection         Routing on Diverting No         Force enblock         Add #         Disable Echo Canceler	GW12 Local-IGWP



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# Scenario II – Text Messaging with Callback

## **Ascom Unite Configuration**

#### **Medamax Gateway**

The Medamax Gateway is a Unite module based on the ELISE hardware. It receives input from other Unite modules or from external equipment. The main functionality of the Medamax Gateway is to handle different types of protocol. It will convert events to actions in our systems, and also to provide an assignment interface to offer the ability for users to dynamically assign recipients to events.

#### **Configuration of Medamax Gateway**

1. Navigate to the Medamax Gateway web administration page (http://xxx.xxx.xxx). This will take you to the "Basic Setup" screen Click on the "Advanced" button. A login pop-up window will appear. Log into the Medamax Gateway using the appropriate Username and Password.

	XGate
Duty Assignment	Administration of duty assignments
Administration	
Access Rights	Administration of access rights in the XGate module
Action Configuration	Configuration of available events and which actions to take
Event Assignment	Conditions to be fulfilled indicating that an event has occurred
Advanced	Configuration of the XGate module

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2. On the "Basic Setup" screen, click on the "Backup/Restore" button on the left-hand side of the page

		XGate		
	Basic Advanced			
Basic Setup Set Language	Basic Setup			
Backup/Restore User Administration				
	Access Rights	Set up access rights for the Unite User Teams		
	Action Handling			
	Action Configuration	Set up of actions for the event that occurred. Define what to transmit and success and failure conditions		
	Assignments			
	Event Assignment	Set up which Event Elements that correspond to a certain Event		
	Duty Assignment	Set up addressees for the actions		
	Input Data Conversion	s		
	Translation Tables			

- 3. The next screen that is displayed is the "Backup/Restore" screen. Click on the "Browse..." button. When the Choose file pop-up window appears, navigate to where you have stored a Medamax/Rauland 5 basic configuration template. If one does not exist contact your support team. Select the file and click the open button.
- 4. Click the "Restore" button. The "Browse" and "Restore" buttons will disappear momentarily while the file is being restored. Wait until the restoration has completed and click on the "Advanced" tab.

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	XGate	

	Basic Advanced	
Basic Setup	Backup/Restore	
Set Language Backup/Restore	Backup parameters	
	Backup	
	Restore parameters	
		Browse Restore

5. Click on the "Configuration" link. You will be taken to the "Event Handler Configuration" screen.

		XGate
	Basic Advanced	
Advanced Setup Translate GUI	Advanced Setup	
I/O Setup	Event Handler	
Data Monitor		
Basic	Configuration	Assignment of Event Elements
Administration	Overview	Display an overview of the Event Handler programming
	Log	Display the Event Handler log
	Database administration	Administration of the Event Handler configuration and block databases

6. On the Event Hander Configuration screen, click on the "+" sign next to the Event Triggers" and make sure you see Rauland 5 in the expanded list. If there is no Rauland 5 folder under Event Triggers, repeat steps 2 through 4.

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7. Modify the Event Elements and Action Configuration to the specific customer. The default configuration details are outlined in Appendix B.

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8. To complete the restoration, we now need to activate it. Click on the Activate configuration button at the top center of the "Event Handler Configuration screen. A new pop-up window will appear. Click on the "Update Persistent elements" radio button and click "Continue". Another pop-up window will appear saying the "Configuration successfully activated", click "OK"

🦻 Persistent element values Webpage Dialog	×
http://172.20.96.65/StartConfiguration/include/persistentalert.html	-
Update currently active Persistent elements with values from the new Configuration	
C Keep the values of currently active Persistent elements	
Continue	
http://172.20.96.65/StartConfiguration/include/persistentalert.html	

## Medamax TAP Port Settings

If the R5 Nurse Call system will be sending messages to the Ascom handsets, then a TAP port must be configured on the Medamax. See XGate – Installation and Operation Manual TD 92338GB for TAP Setup information. Set the TAP values to match those of the Rauland R5.

## SIP Trunk from IGWP

If the Ascom handsets (UNITE included in solution) will be originating calls to the R5 Nurse Call, then a SIP trunk from the IGWP must be configured.

#### AA60 setup

- Connect to the AA60 configuration GUI with a compatible web browser using the following address: <u>http://AA60-IPADDRESS:8088</u> (where AA60-IPADDRESS is the IP Address of the AA60).
- 2. Navigate to the Users tab.
- 3. A predefined user for the IGWP is included in the default configuration for R5, extension "6001". Edit this extension if it conflicts with the overall dial plan.



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#### SIP Trunk from Brekeke

If the Rauland 5 will be registering audio stations through a Brekeke SIP Server, then a SIP trunk from the Brekeke must be configured.

AA60 setup

- Connect to the AA60 configuration GUI with a compatible web browser, using the following address: <u>http://AA60-IPADDRESS:8088</u> (where AA60-IPADDRESS is the IP Address of the AA60).
- 2. Navigate to the Users tab.
- 3. A predefined user for the Brekeke is included in the default configuration for R5, extension "6000". Edit this extension if it conflicts with the overall dial plan.

88 Configure Hardware	+	Create New User	Modify Selected Users	X Dele	e Seled	ed Users			
88 mISDN Config									
## Trunks				List of	user	Extens	sions		
## Outgoing Calling Rules		Extension	Full Name	Port	SIP	IAX	DialPlan	OutBound CID	
## Dial Plans	Г	6000	FROM IGWP		Yes	12	TO BREKEKE	none	Edit
## Users		C004					TO 10110		
Users is a shortcut for quickly adding and removing all the necessary configuration components for any new phone.		0001	FROM_BREKEKE		res		TO_IGWP	none	Eoit

#### T1 VoIP Gateway setup

- 1. Navigate to *PBX* → *Objects* and add a new Gateway object for "To\_AA60" with the following parameters:
  - a. Long Name = "To\_AA60"
  - b. Name = "To\_AA60"
  - c. Number = Defined R5 extension number plan prefix (i.e. 6 when all extensions start with a 6)

Note: a universal number plan must exist between the R5 audio station extensions, Ascom handsets, and any extensions on the PBX that shall be dialed from Ascom handsets.

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Long Name	To_AA60	Display Name		
Name	To_AA60	Number 6	Critical 🔲	
Password		retype Password		
Node	root 🛩	Hide from LDAP		
PBX	. 💌	Local		
Config Template	•			
Config		Diversion Filter Income I R Deiset		
Response Tin	normai	Busy On Calls		
No Inband Dis	sconnect			
Gateway				
Enblock Cour	nt 🗌			
Enblock as D	iverting No 🔲			
Prefix				
International N	Match			
National Mate	h			
Subscriber M	atch			
ОК	Apply Delete	Cancel		

- 2. Connect to the T1 VoIP Gateway configuration GUI with a compatible web browser.
  - a. Navigate to the Gateway tab and click on SIP tab.
- 3. Define a SIP interface with the following parameters:
  - a. Name = "To\_AA60"
  - b. ID = "6000" (or whatever extension defined above in the AA60)
  - c. @ = IP Address of AA60
  - d. Proxy = IP Address of AA60
  - e. Username = "6000" (or whatever extension defined above in the AA60)
  - f. General Coder Preference = "G711u"
  - g. Local Coder Preference = "G711u"
  - h. Internal Registration Protocol = "SIP"
  - i. Server Address = "127.0.0.1"
  - j. ID = "To\_AA60"
  - k. @ = "127.0.0.1"
  - I. Username = "To\_AA60"

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Name To_AA60				
Disable				
ID 6001 @ 172.20.106.106				
Proxy 172.20.106.106				
STUN Server				
Authorization				
Username 6001				
Password Retype				
- Media Properties				
General Coder Preference G711u 💌 Framesize [ms] 20 Silence Compression 🔲 Exclusive 🗹				
Local Network Coder G711u 💌 Framesize [ms] 20 Silence Compression 🗌				
Enable T.38 🗌 Enable SRTP 🔲 Media-Relay 🗹 No DTMF Detection 🗌 Enable PCM 🔲				
SIP Interop Tweaks				
Proposed Registration Interval [s]				
Accept INVITE's from Anywhere				
Enforce Sending Complete   (affects outgoing SIP calls only)				
From Header when Sending INVITE Fixed AOR				
Identity reader when generating inverte CGPN in user part of URI				
Reliability of Provisional Responses Supported 💌 (affects outgoing SIP calls only)				
Internal Registration				
Protocol SIP				
Server Address 127.0.0.1 (primary)				
Server Address (secondary)				
ID To_AA60 @ 127.0.0.1				
Username To_AA60				
Password Retype				

4. Navigate to the *Gateway* tab and select *Routes*. Delete the route from the AA60 to the T1 VoIP Gateway (SIP1 to RS1). The only SIP1 route shall be RS1 to SIP1.



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### CABLING REQUIREMENTS

Please see XGate – Installation and Operation Manual TD 92338GB for cabling instructions

#### RELATED DOCUMENTS

TD 92232GB	Installation Guide ELLISE2
TD 92338GB	Installation and Operation Manual XGate
TD 92364GB	User Manual Administration, XGate
TD 92329GB	Programming Guide, Event Handler

#### Additional Information

If you have any questions or need additional information, please contact Ascom Technical Assistance Center at 1-877-71-ASCOM, option 3.

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Appendix A

## **Rauland Pager Configuration Setup**

Enter all settings as shown in the screenshot below.

## Pager configuration:

Com Port: 1 NOTE: Enter 0 to	o close the Com Port
Baud: 9600 💽 Stop Bits:	1 • Parity: Even • Data Bits: 7 •
Mode: PET1 💌 RTS Use Hold to	ue 💌 CTS Use Ignore 💌 Flow Control FLOW OFF 💌
Inactive limit: 0	Access Limit: 0
Internet port: 5051	NOTE: you must reboot if you update the Internet port for the change to take effect
Trace: 🔽 Recognize ID=: 🔽	
Configuration Update	

The AA60's IP address needs to be set in the SIP Proxy Address and SIP Registration Server fields in the RGS server. This field can be found in: *C:\Program Files\Rauland-Borg\R5RGS\RGSregedit.exe* 

Outsid         Diagnostic         Advanced           Nurse Call Address - From the network administrator         Nurse Call Subnet - usually 295 295 295 0           10.196.12.11         [295.295.295.0           Data Address - if static Stat + End. if DHCP writer stating and ending range of DHCP range           Stat         End           [10.196.17.65         [10.196.17.65           SIP Proxy Address - If therwall + murse call friewall adt, else use Rieg six address below           [10.196.12.103           SIP Registration Server - SIP Registration server, blank if no phones           [10.196.12.103           SIP Contact - usually the same as Nurse Call Address if phones in use, blank otherwise	Config Trouble Pagers Save	
Quest Disgnostic         Advanced           Nurse Call Address - From the metwork administrator         Nurse Call Submet - usually 295 295 295 0           [10.196.12.11         [295.295.295 0           Data Address - if static Stat + End. #DHCP write stating and ending range of DHCP range           Stat         End           [10.196.17.65         [10.196.17.65           SIP Proxy Address - If fremval + murse call freewall adt. else use Rieg six address below           [10.196.12.103         SIP Registration server, blank if no phones           [10.196.12.103         SIP Contact - usually the same as Nurse Call Address if phones in use, black otherwise		
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### Appendix B

The default Medamax Gateway configuration has the following events:

## Event Assignment

Event	Description
Nursecall - Normal	Normal Priority Message with Callback to room is an option.
High Message Trigger	High Priority Message with no Callback to room as an option
Low Message Trigger	Low Priority Message with Callback to room is an option.
Medium Message Trigger	Medium Priority Message with Callback to room is an option.
Tagged Text Interactive Message	A tagged plain text message with Callback to room as an opti
Plain Text Message	A non tagged plain text message (non interactive)

Normal Priority – Message event is triggered by a message which includes any of the following words:

• Staff, Duty, Patient, Cord Out, Bed Out, Water, Patient OT, Attention, Go To Toilet, Bath Assist, In Pain, Attention OT 1, Urgent, Bath Assist O, Urgent OT 1, Bath Assist OT

High Priority – Message event is triggered by a message which includes any of the following words:

• Vent Alarm, Staff Assist, Staff Assist OT, Rapid Response, Code Blue

Medium Priority – Message event is triggered by a message which includes any of the following words:

• NA Rnd OT, Rn Rnd OT, Plug Out, Supervision Failure, Bath, Bath Emerg, Bath OT 1, Bath Emerg OT, Aux Alarm, Bed Alarm

Low Priority – Message event is triggered by a message which includes any of the following words:

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 NA Rounds, RN Rounds, Transport, Cleaning Needed, Cleaning In Progress, Bed Ready, Patient Transport OT, Cleaning Needed OT

Tagged Text – Message event is triggered by a message which includes a Tag (Area) (Room) (Bed) and is not a Normal, High, Medium, or Low priority message.

Plain Text – Message event is triggered by any non tagged message

The default configuration has the following actions:

Type	Name	Add
Interactive Message	Normal Priority - Interactive M	Auu
Interactive Message Interactive Message	High Priority - Interactive Mes Medium Priority - Interactive	Edit
Interactive Message	Low Priority - Interactive Mes	
Message	Non-Tagged Plain Text Messa	Delete
		Close

Normal Priority – Interactive Message will result in an Interactive message of priority normal, being displayed on the screen of the destination handset, with two options "Talk" and "Close". Selecting the "Talk" key will result in a call to <Area>\*<Room Number>\*<Bed>

High Priority – Interactive Message event will result in an Interactive message of priority alarm, being displayed on the screen of the destination handset, with one option "Close".



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Medium Priority – Interactive Message will result in an Interactive message of priority high, being displayed on the screen of the destination handset, with two options "Talk" and "Close". Selecting the "Talk" key will result in a call to <Area>\*<Room Number>\*<Bed>

Low Priority - Message event will result in an Interactive message of priority low, being displayed on the screen of the destination handset, with two options "Talk" and "Close". Selecting the "Talk" key will result in a call to <Area>\*<Room Number>\*<Bed>

Non-Tagged Plain Text Message event will result in a Message of priority normal, being displayed on the screen of the destination handset

Tagged Plain Text Interactive Message event will result in a Message of priority normal, being displayed on the screen of the destination handset, with two options "Talk" and "Close". Selecting the "Talk" key will result in a call to <Area>\*<Room Number>\*<Bed>