Application Note

Radio Relay Port to SIP

Software From
N/A

Revision
v1.6

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AN232 RADIO RELAY PORT TO SIP

1 Overview

This Application Note shows how to configure a BASICS Radio Relay port to SIP UA. The arrangement below is used to show how radio 104 can receive a SIP peer-to-peer call from SIP phone 102 and radio 105 can receive a SIP call from phone 103 through an Asterisk SIP proxy.

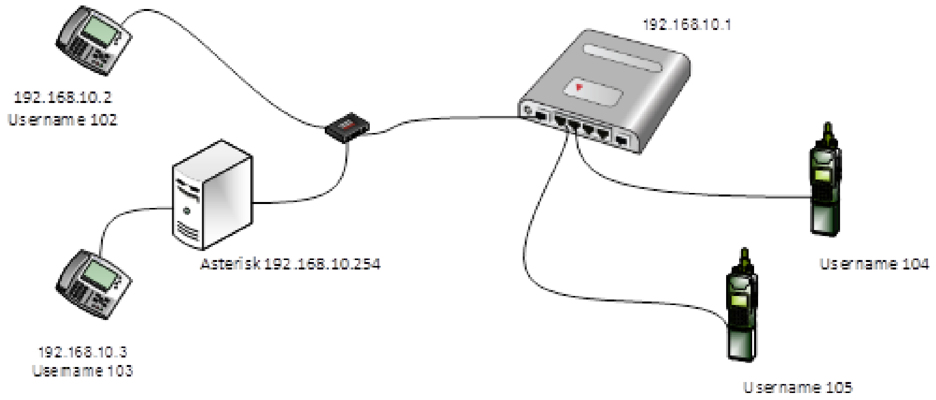


Figure 1 Example showing radios, phones and an Asterisk SIP proxy connected through BASICS Radio Relay

2 Pre-requisites

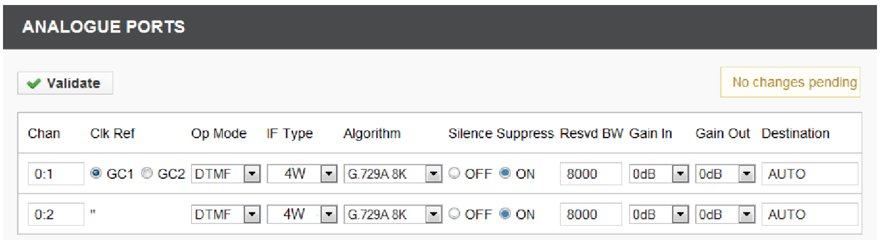
Application Note 230 – Radio Relay using BASICS Four Wire

3 Hardware configuration

None.

4 Software configuration

On the **Analogue Voice > Analogue Ports menu** change the Destination to 'AUTO'. Make sure that the Algorithm selected here matches the SIP Gateway channel algorithm.



ANALOGUE PORTS

Validate

No changes pending

Chan	Clk Ref	Op Mode	IF Type	Algorithm	Silence Suppress	Resvd BW	Gain In	Gain Out	Destination
0:1	GC1 GC2	DTMF	4W	G.729A 8K	OFF ON	8000	0dB	0dB	AUTO
0:2	"	DTMF	4W	G.729A 8K	OFF ON	8000	0dB	0dB	AUTO

Figure 2 Analogue Voice > Analogue Ports menu

On the **SIP Gateway > System menu** set the Mode to Gateway. The Re–Registration Interval should match what is specified in the Asterisk PBX.

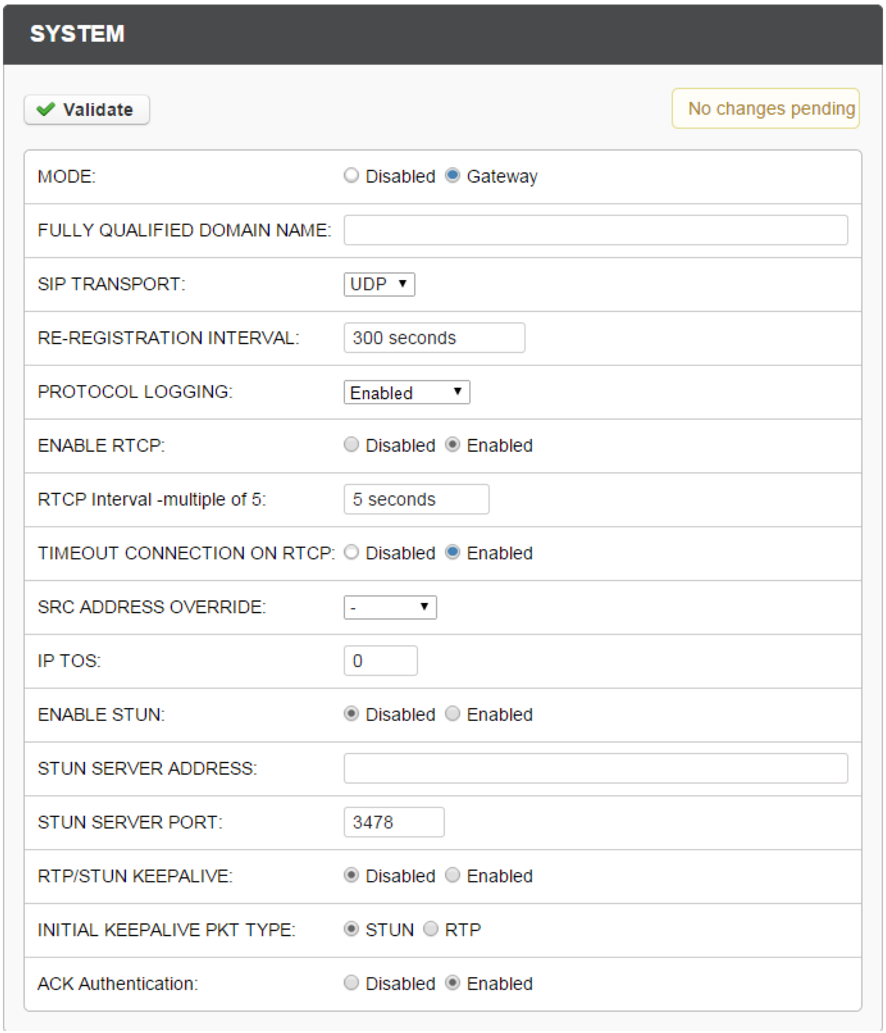


Figure 3 SIP Gateway > System menu

In the following configuration SIP Gateway channel 1 is configured for peer-to-peer calls from SIP phone 102. Only the User ID, Destination, Algorithm and Silence Suppression need to be specified on the **SIP Gateway > Channel Details menu**. The Algorithm needs to match the analogue port channel algorithm. The Silence Suppression setting for the SIP Channel also needs to match that for the analogue port. When the gateway receives a SIP INVITE for User ID 104 it will map this to the radio connected to Node2:Slot 0:Channel 1.

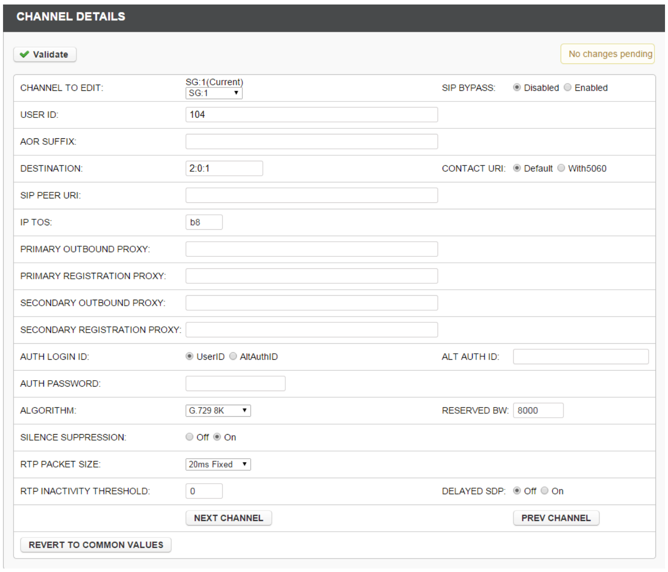


Figure 4 SIP Gateway > Channel Summary > Channel Details menu

A SNOM 360 phone will be used to place the peer-to-peer call. You will need to define a function key in the SNOM phone as a speed dial to 'sip:104@192.168.10.1' as seen below:

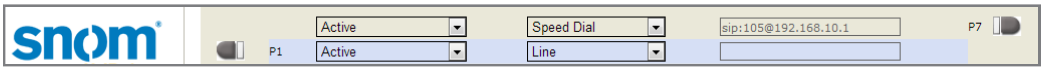


Figure 5 SNOM menu set to speed dial sip:104@192.168.10.1

SIP Gateway channel 2 is configured to receive calls from SIP phone 103, which is registered with the Asterisk PBX. On the **SIP Gateway > Channel Details menu** User ID 105 will associate the radio connected to Node 2: Slot 0: Channel 2 with this SIP Gateway channel. This Gateway channel also needs to be registered with Asterisk to complete the call. To do this you need to specify an Outbound Proxy and Registration Proxy, using the IP address of the Asterisk PBX. In addition, you will need to enter the Auth Password for the 105 extension, as set in Asterisk. The Algorithm needs to match the analogue port channel algorithm.

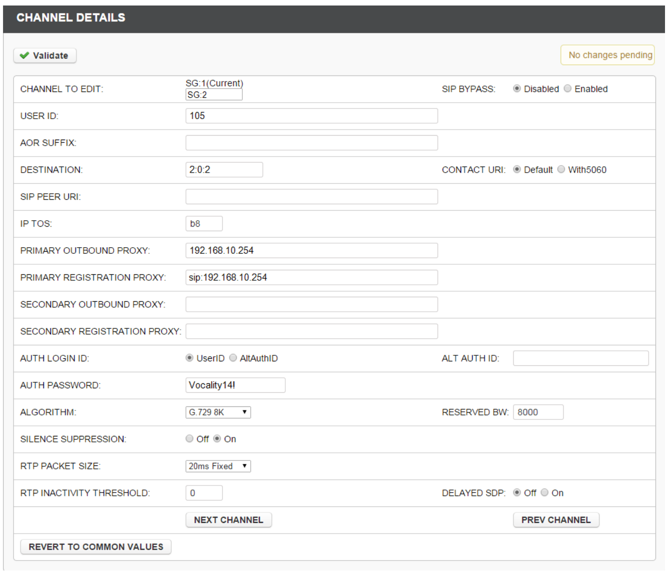


Figure 6 SIP Gateway > Channel Summary > Channel Details menu with extra details for Asterisk

5 Testing

From SIP phone 102 dial extension 104 and verify that radio 104 relays audio to the radio under test. From SIP phone 103 dial extension 105 and verify that radio 105 relays audio to the radio under test. For each test, the SIP phone user will need to mute their phone to allow the radio to transmit. If muting does not work consider other options described in the next section.

6 How different VoIP phones affect VoIP phone - radio interaction

There are several ways to control how a full-duplex VoIP phone will communicate with a half-duplex radio. If a simple mute is not enough, you may need to use another method to either prevent the VoIP phone from transmitting packets or have your unit discard packets, so that the radio user can PTT and talk to the VoIP phone user. Possible scenarios suitable for three different VoIP phone behaviors are described below:

1. **VoIP phones which stop transmitting RTP packets when set to MUTE -**

   When your unit is not receiving packets from the VoIP phone, the radio user can then PTT and talk to the VoIP phone user.

2. **VoIP phones which still send RTP packets when set to MUTE, but will mark those transmitted packets as 'MUTE packets' -**

   Some VoIP phones will still send RTP packets when set to MUTE, but will mark the transmitted packets as 'MUTE packets'. In this situation, your unit can check whether the incoming voice packets from the VoIP phone are marked as MUTE packets. If they are, they will be discarded by the Vocality allowing the radio user to PTT and talk to the VoIP phone user. To do this you need to set the RTP FILTER parameter to InboundSilence on the **SIP Gateway > Channel Details menu**.

   Note: You can only choose this method when the SIP Bypass Channel, Analogue Port and VoIP Phone are all set to use either G.711a or G.711u codecs.

3. **VoIP phones which keep transmitting RTP packets when set to MUTE, or do not have a MUTE option -**

   a. Your first option here is to use DTMF tones, where assigned keys from the VoIP phone control when your unit discards RTP packets from the VoIP phone.

      In the **Analogue Voice > Radio Relay menu** select the Mode DTMF-PTT to enable this function on a particular channel. Once enabled, the VoIP phone user can choose whether they are talking to the radio user or listening to the radio user, using the # and * buttons by default. When the VoIP phone user is listening, the radio user can PTT and talk to the VoIP phone user. The talking/listening (or Talk/Clear) buttons can be remapped to different keys by setting the DTMF Talk, DTMF Clear parameters (before VOS08_15_01 these parameters could only be set on the separate **Analogue Voice > Radio Relay Advanced menu**).

b. As an alternative, your unit can check whether the incoming voice packets from the VoIP phone are below an audio level threshold, which you set as the 'silence' level. If they are, they will be discarded by your unit, allowing the radio user to PTT and talk to the VoIP phone user.

   The RTP FILTER parameter needs to be set to InboundLevel on the **SIP Gateway > Channel Details menu**. The audio level threshold can be configured in the **SIP Gateway > RTP Payload Options**. If you need more help, please contact Vocality for assistance with changing these advanced options.

   Note: You can only choose this method when the SIP Bypass Channel, Analogue Port and VoIP Phone are all set to use either G.711a or G.711u codecs.

7 Diagnostics

If the above tests do not work then check the Vocality **Diagnostics > Logs > All Logs** to identify the cause. For further assistance contact Vocality Technical Support.

8 About Application Notes

Application Notes are intended as a supplement to, rather than a substitute for, your User Manual. Should you have queries which are not answered by our current documentation, your local Vocality support team would be happy to hear from you.
E-mail **support@vocality.com**.