R&R offers two ways to test routes

1. **R&R Automated Call Testing Facility with CLI verification**
2. **R&R Manual call testing Guide**

R&R Automated Call Testing Facility with CLI verification

Automated Call Testing facility is designed to simplify Carrier route testing, route quality monitoring, FAS detections and to verify correct CLI delivery.

Users can perform various tests using one or multiple suppliers, automate testing scenarios and be informed by email when quality drops or False Answer Supervision (FAS) is detected.

R&R is using advanced voice recognition algorithms to detect the following cases:

- Connect while ringing
- Silence after connect (Dead Air)
- Known IVR patterns
- Lack of audible ringback tone

Call Testing Facility records media, counts media packets exchanged in both directions and calculates Mean Opinion Score for the test call. There's a function to alert on low MOS. User is able to listen to each test call ta have own opinion about the quality.

Configure your Sonus partition

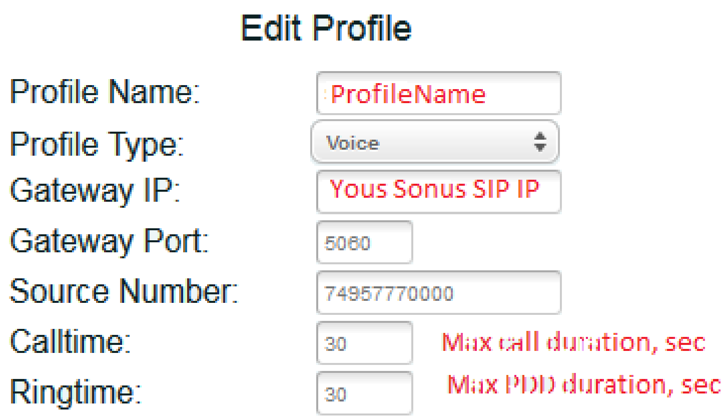
Create a Member and a Trunk Group for inbound SIP remote IP 38.125.23.203

Login to Test portal https://test.r-rtele.com

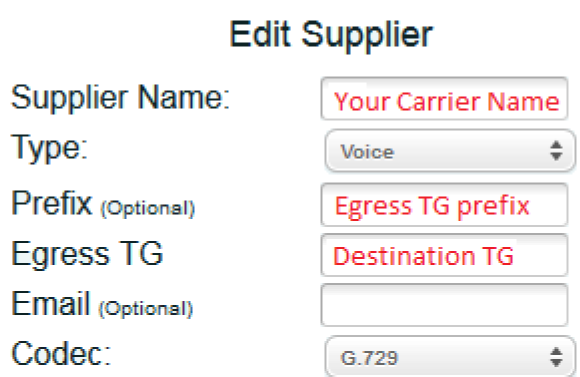
Login credentials are provided by R&R customer support.

Configure test Profile (one per account).

Destination IP must be our Sonus.



Configure Supplier

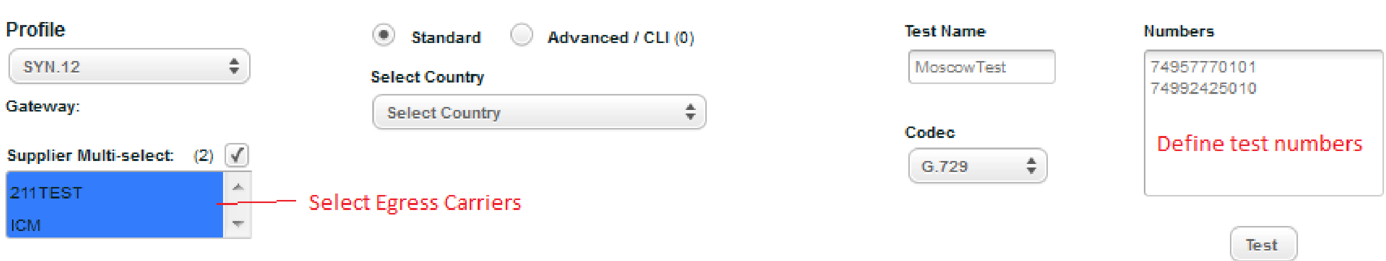


In Egress TG field user may enter full Outbound TG name; this will force routing out to that TG bypassing Routing Table.

When using EgressTG you must populate egress prefix (if any). Having prefix populated will distort Called number in reports but will send correct number to the Carrier.

If “Egress TG” field is omitted then call will be routed according to Routing Table for the Test Member.

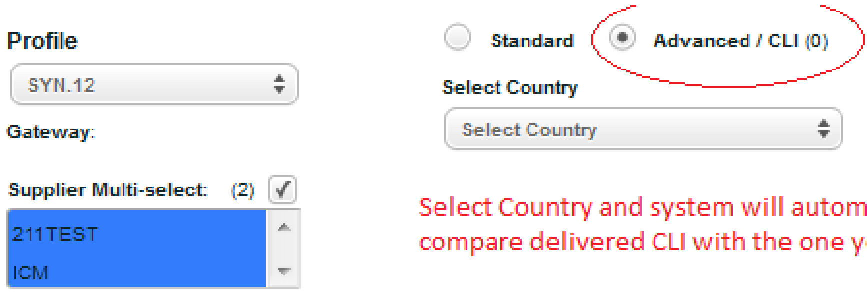
Perform manual testing



CLI testing

Each account is allocated free 50 CLI test credits per month.

More can be purchased through R&R.



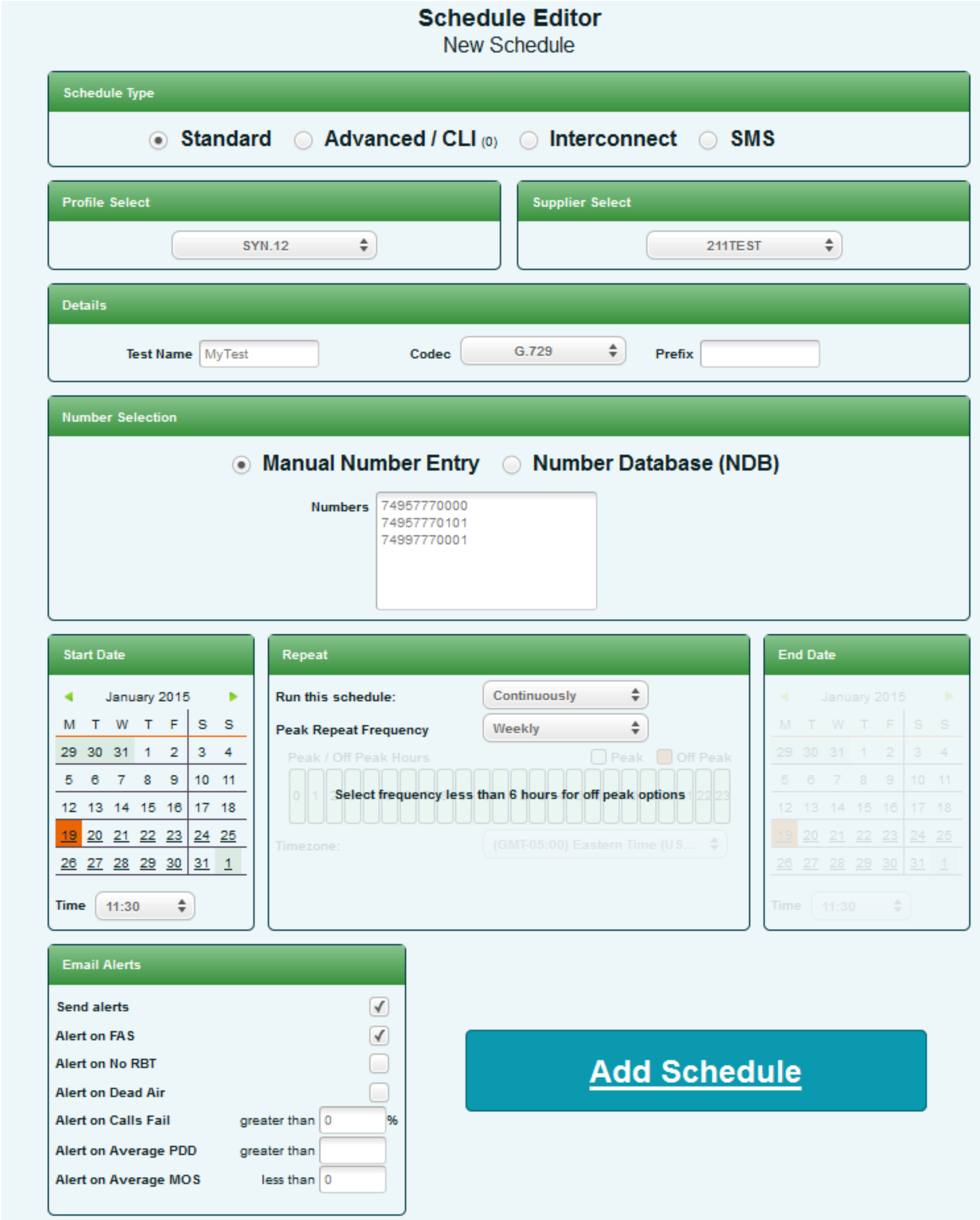
Select Country and system will automatically perform testing and compare delivered CLI with the one you used for test.

Review test results

Packet exchange counter

Click to open call log

Schedule automated tests


Schedule Editor
New Schedule
Schedule Type
Standard
Advanced / CLI (0)
Interconnect
SMS
Profile Select
SYN.12
Supplier Select
211TEST
Details
Test Name
MyTest
Codec
G.729
Prefix
Number Selection
Manual Number Entry
Number Database (NDB)
Numbers
74957770000
74957770101
74997770001
Start Date
January 2015
M T W T F S S
29 30 31 1 2 3 4
5 6 7 8 9 10 11
12 13 14 15 16 17 18
19 20 21 22 23 24 25
26 27 28 29 30 31 1
Time
11:30
Repeat
Run this schedule:
Continuously
Peak Repeat Frequency
Weekly
Select frequency less than 6 hours for off peak options
End Date
Email Alerts
Send alerts
Alert on FAS
Alert on No RBT
Alert on Dead Air
Alert on Calls Fail
greater than
0
%
Alert on Average PDD
greater than
Alert on Average MOS
less than
0
Add Schedule

R&R Manual call testing Guide

Clients are encouraged to create a test account for manual testing of outbound routes. This document outlines suggested procedure to establish a test account.

1. **Prepare VoIP testing device**

   Any VoIP origination device will work: softphones, smartphone SIP client applications, H.323 or SIP phone adapter, IP PBX.

   For a free PC softphone we recommend X-Lite that can be downloaded for free:

   http://www.counterpath.com/x-lite-download/

   X-lite supports Windows, Linux and MacOS.

2. **Identify your IP address**

   Most of DSL/Cable providers don't assign static IP but keep it same for long periods of time. To find out your public IP address check www.myipaddress.com

   If your IP changes just repeat steps 2 and 4.

3. **Configure your Sonus system**

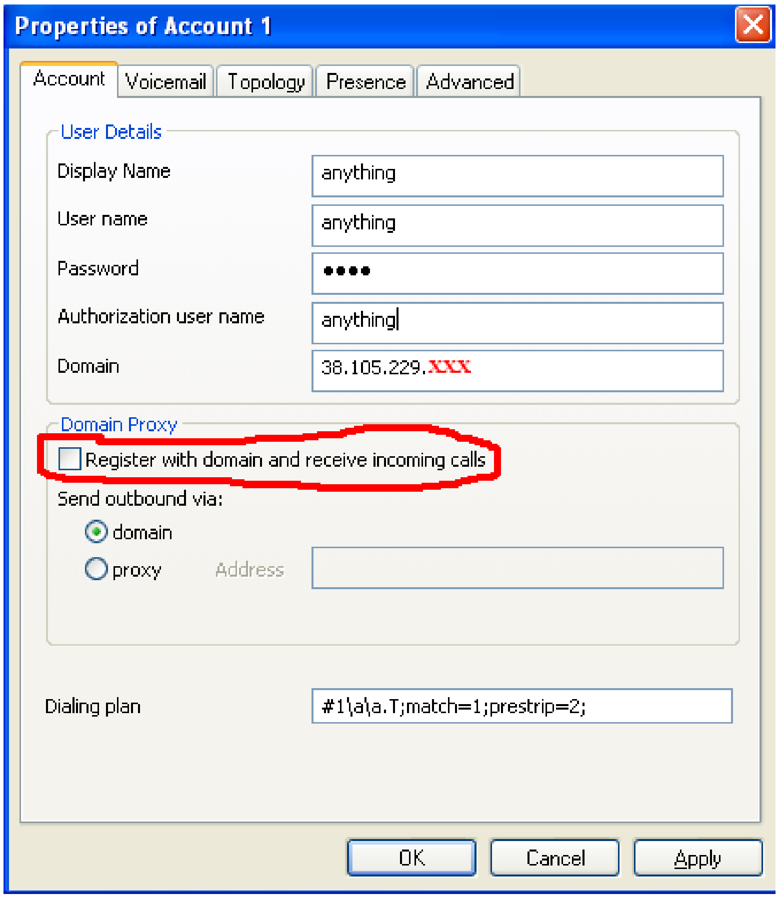
   Place provisioning order as usual. Test line will be treated as any other Member in your system.

4. **Configure VoIP test line**

   Configure destination IP address (your system SIP address)
   Disable phone Registration
   Use any username/password as authentication is based on IP

Note: User name & password call authentication is disabled as such method assumes that firewall allows any IP address to access your system, which is dangerous.

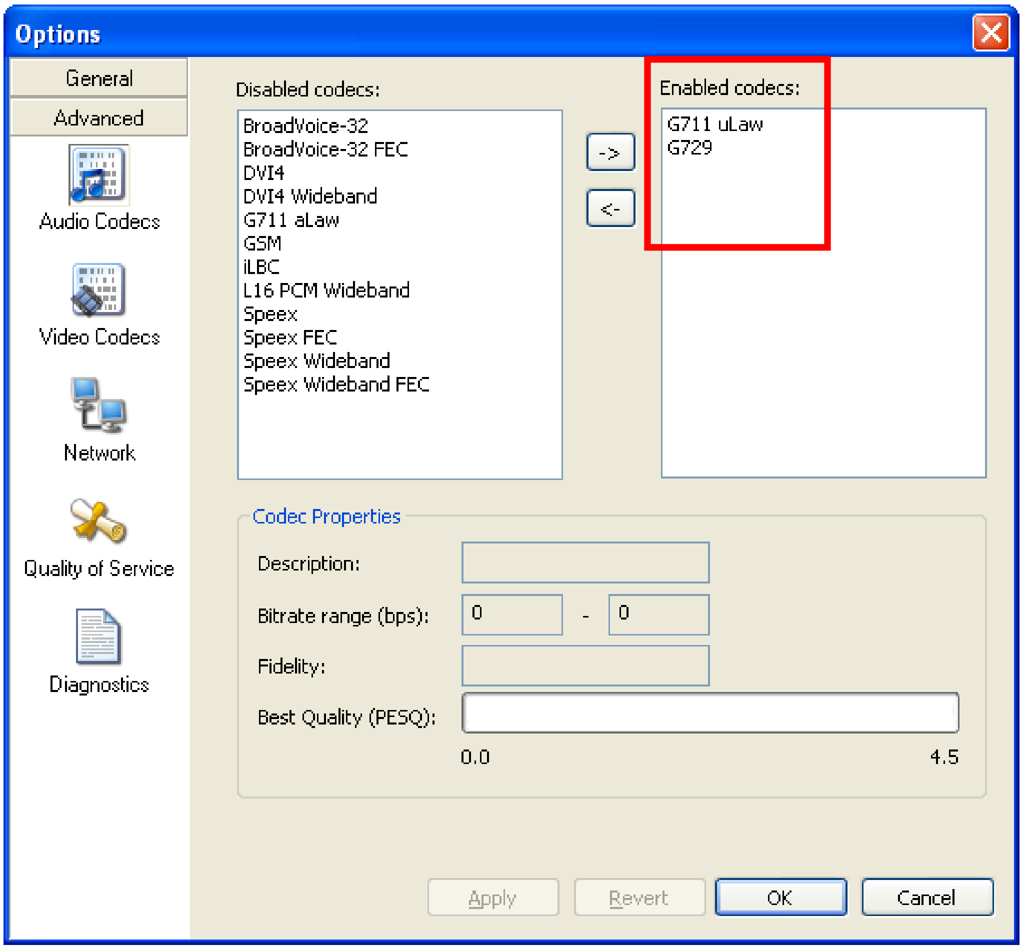
Example : X-Lite configuration



EyeBeam (paid version of X-lite) SIP protocol settings:

Make sure to use UDP for SIP phones
Select G711, G729 or G723 codecs


Properties of Account 5
Account
Voicemail
Topology
Presence
Storage
Security
Advanced
Signalling Transport
UDP
Encrypted or TLS must be selected to enable media encryption.
Media Encryption
Make and accept only encrypted calls
Prefer to make and accept encrypted calls
Make unencrypted calls, accept all calls
Do not allow encrypted calls


Options
General
Advanced
Audio Codecs
Video Codecs
Network
Quality of Service
Diagnostics
Disabled codecs:
BroadVoice-32
BroadVoice-32 FEC
DVI4
DVI4 Wideband
G711 aLaw
GSM
iLBC
L16 PCM Wideband
Speex
Speex FEC
Speex Wideband
Speex Wideband FEC
->
<-
Enabled codecs:
G711 uLaw
G729
Codec Properties
Description:
Bitrate range (bps):
0
-
0
Fidelity:
Best Quality (PESQ):
0.0
4.5
Apply
Revert
OK
Cancel