

Generating Traffic for VoIP Testing

Goal: Set up and run VoIP traffic.

In this example, LANforge-FIRE is used to set up two VoIP test calls that may be used as a basis for VoIP load testing or VoIP Gateway testing.

- **Test 1:** Directed VoIP call where a LANforge endpoint calls another LANforge endpoint.
- **Test 2:** Gateway VoIP call where two LANforge endpoints register with a VoIP Gateway so that the call from one endpoint to the other goes through the gateway. The VoIP Gateway used in this example is Asterisk.

 This cookbook does not cover FXS/FXO ports (analog RJ11 lines). It is possible that those setup require special commercial gateway features.

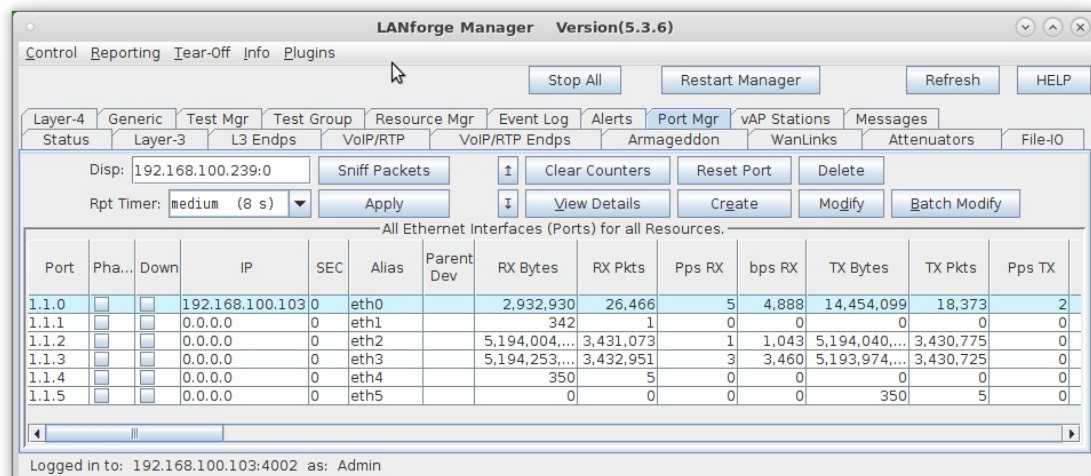
1. **Set up the LANforge physical connections**

The same two ports are used for both tests. Connect eth1 and eth2 from the LANforge-FIRE system to a network switch that is also connected to the VoIP Gateway. This example assumes that your VoIP Gateway is set up properly. If you need assistance, you can contact us at support@candelatech.com.

2. **Configure LANforge ports**

Ports require valid IP addresses and IP masks.

- A. Go to the Port Manager

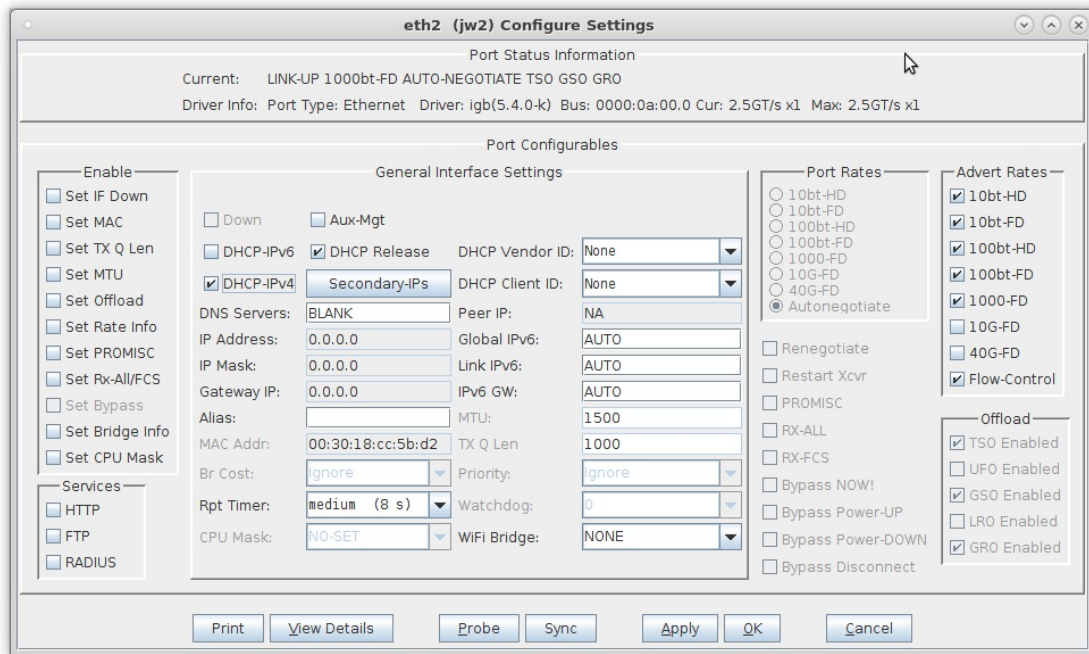


All Ethernet Interfaces (Ports) for all Resources.

| Port | Pha... | Down | IP | SEC | Alias | Parent Dev | RX Bytes | RX Pkts | Pps RX | bps RX | TX Bytes | TX Pkts | Pps TX |
|-------|--------------------------|--------------------------|-----------------|-----|-------|------------|---------------|-----------|--------|--------|---------------|-----------|--------|
| 1.1.0 | <input type="checkbox"/> | <input type="checkbox"/> | 192.168.100.103 | 0 | eth0 | | 2,932,930 | 26,466 | 5 | 4,888 | 14,454,099 | 18,373 | 2 |
| 1.1.1 | <input type="checkbox"/> | <input type="checkbox"/> | 0.0.0.0 | 0 | eth1 | | 342 | 1 | 0 | 0 | 0 | 0 | 0 |
| 1.1.2 | <input type="checkbox"/> | <input type="checkbox"/> | 0.0.0.0 | 0 | eth2 | | 5,194,004,... | 3,431,073 | 1 | 1,043 | 5,194,040,... | 3,430,775 | 0 |
| 1.1.3 | <input type="checkbox"/> | <input type="checkbox"/> | 0.0.0.0 | 0 | eth3 | | 5,194,253,... | 3,432,951 | 3 | 3,460 | 5,193,974,... | 3,430,725 | 0 |
| 1.1.4 | <input type="checkbox"/> | <input type="checkbox"/> | 0.0.0.0 | 0 | eth4 | | 350 | 5 | 0 | 0 | 0 | 0 | 0 |
| 1.1.5 | <input type="checkbox"/> | <input type="checkbox"/> | 0.0.0.0 | 0 | eth5 | | 0 | 0 | 0 | 0 | 350 | 5 | 0 |

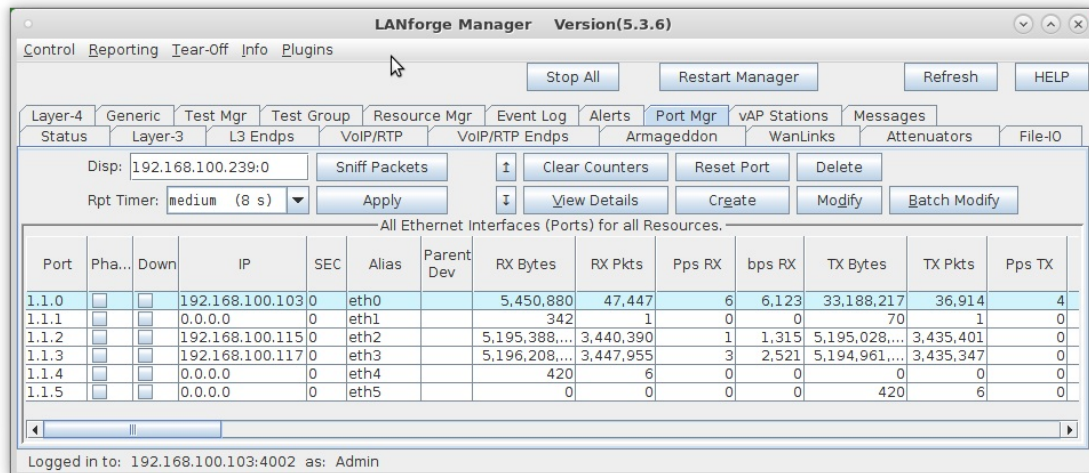
Logged in to: 192.168.100.103:4002 as: Admin

B. Modify eth2 and eth3 to set a valid network IP address and mask



A. If your network has DHCP service, you can select the 'DHCP-IPv4' checkbox so that each port is a DHCP client and will acquire its IP address from your DHCP server

C. Verify the port configuration

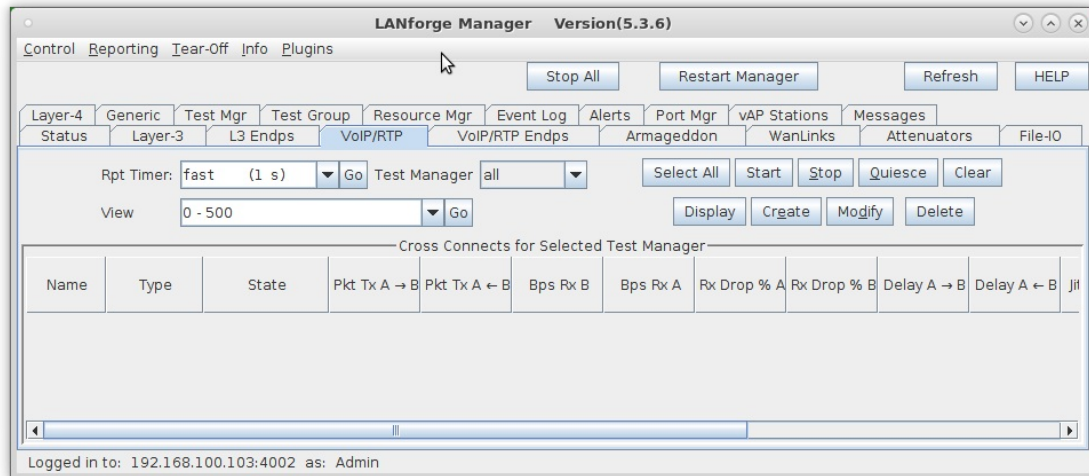


For more information see [LANforge User's Guide: Ports \(Interfaces\)](#)

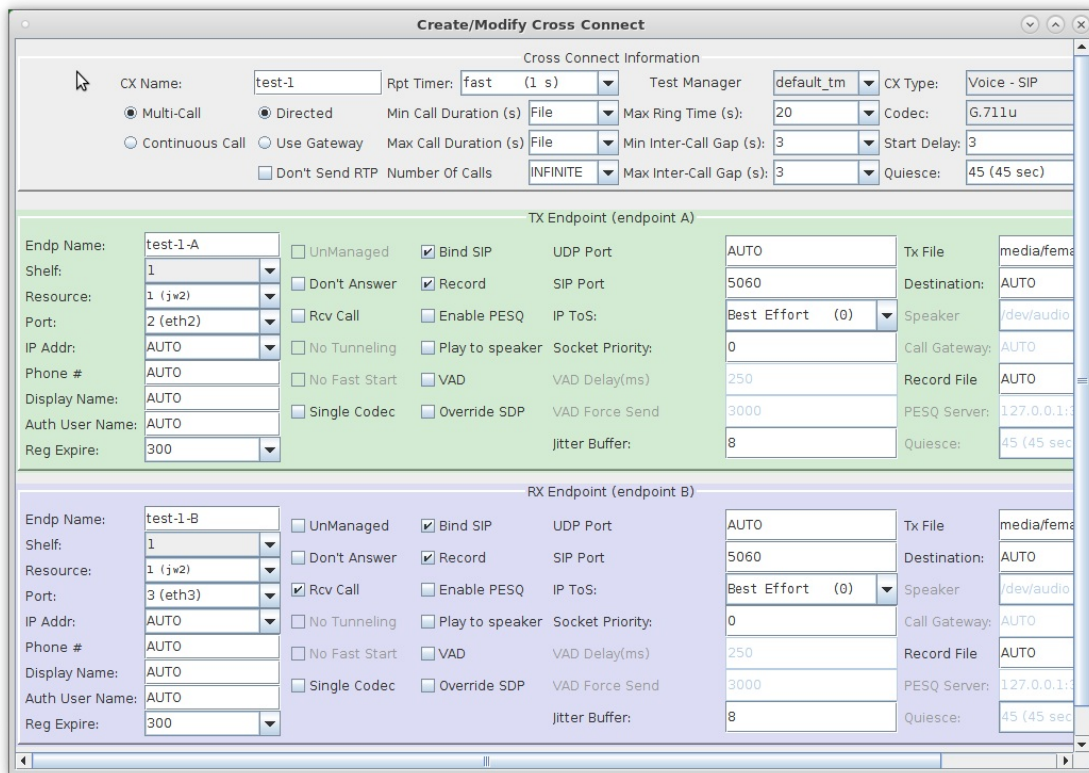
3.

Set up Test 1, a Directed VoIP call.

A. Go to the **VoIP/RTP** tab



B. Click the **Create** button:



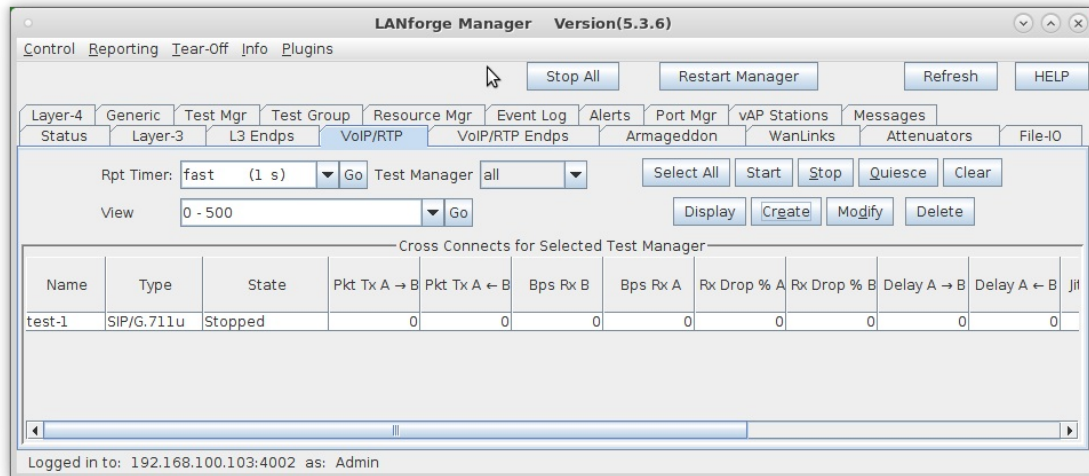
- Enter test-1 in the **CX Name** field
- Select the **Multi-Call** and **Directed** buttons
- Endpoint A is on port eth2 in this example. If you have a PESQ licensed server available, you can select Record and Enable PESQ.
- Endpoint B is on port eth3. If you are using PESQ, be sure to enter a Record File and the IP address and port of your PESQ licensed server. Be sure to select the **Rcv Call** checkbox for this endpoint to receive the call.
- Be careful of the VoIP phone number: you might have to format the number as extension@IP-address, E.G.:
`5678@192.168.1.10`
- Click **OK** to create the VoIP Directed call

C. Verify that the test call is created

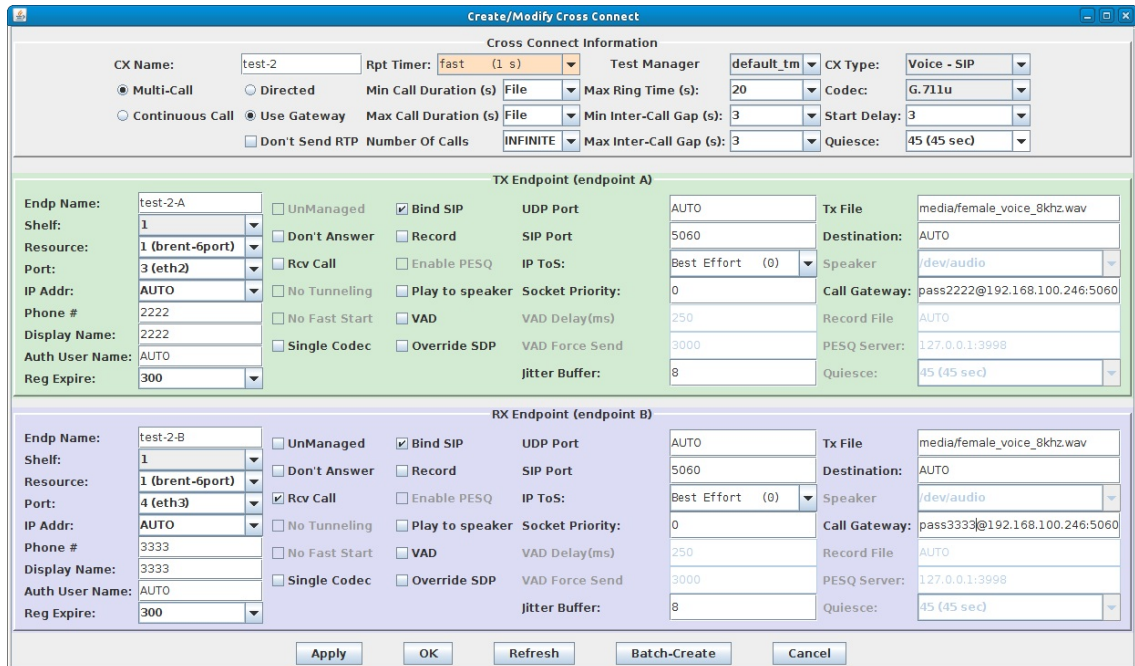
4.

Set up Test 2, a Gateway VoIP call.

A. Go to the VoIP/RTP tab

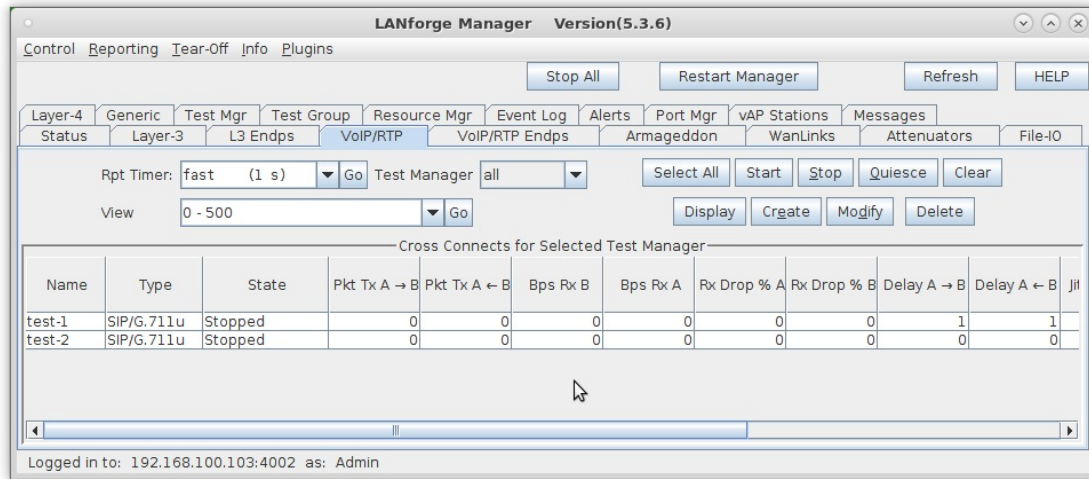


B. Click the Create button:



- A. Enter test-2 in the **CX Name** field
- B. Select the **Multi-Call** and **Use Gateway** buttons
- C. Endpoint A is on port eth2 in this example. Be sure to enter the proper username and password for the endpoint so that it can authenticate with the VoIP Gateway if necessary.
- D. Configure gateway authentication:
 - i. To register with the gateway, often the **Auth User Name** is the phone extension (like 3333).
 - ii. The **call gateway** begins with the extension password:
`pass333@192.168.100.245:5060`
- E. Endpoint B is on port eth3. Be sure to select the **Rcv Call** checkbox for this endpoint to receive the call.
- F. Click **OK** to create the VoIP Gateway call

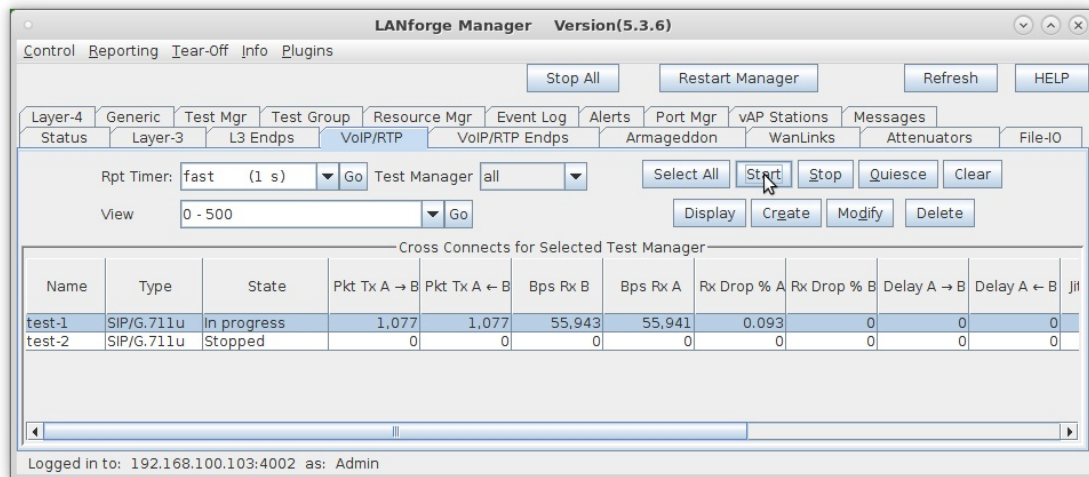
C. Verify that the test call is created



For more information see [LANforge User's Guide: VoIP Call Generator](#)

5. Run test-1 and test-2 individually.

A. Select test-1 and click the **Start** button



B. Go to the **VoIP/RTP Endps** tab to see detailed results:

LANforge Manager Version(5.0.9)

Control Reporting Tear-Off Help

Stop All Restart Manager Refresh HELP

Collision-Domains File-IO Layer-4 Generic Test Mgr Resource Mgr Serial Spans PPP-Links Port Mgr Messages

Status Layer-3 L3 Endps VoIP/RTP VoIP/RTP Endps Audio/Visual AV Endps Armageddon WanLinks

View 0 - 400 Go Delete

All Endpoints

| Name | State | Reg State | PESQ | Tx Pkts | Rx Pkts | Tx Bytes | Rx Bytes | Dropped | OOO Pkts | Dup Pkts | jB Silence | jB Under |
|----------|-------------|-----------|---------|---------|---------|------------|------------|---------|----------|----------|------------|----------|
| test-1-A | In progress | Unreg | 7: 4.21 | 13,551 | 13,558 | 2,168,1... | 2,169,2... | 0 | 0 | 0 | 0 | 0 |
| test-1-B | In progress | Unreg | 7: 4.21 | 13,561 | 13,551 | 2,169,7... | 2,168,1... | 0 | 0 | 0 | 0 | 0 |
| test-2-A | Stopped | Unreg | 0: 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |
| test-2-B | Stopped | Unreg | 0: 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 |

Logged in to: localhost:4002 as: Admin

- A. The PESQ score will be reported after the first successful call is completed and updated after each subsequent call
- B. **NOTE:** Endpoints are unregistered while the call is in progress because they are not calling through the VoIP gateway

C. Stop test-1, select test-2 and click **Start**

LANforge Manager Version(5.3.3)

Control Reporting Tear-Off Info Plugins

Stop All Restart Manager Refresh HELP

File-IO Layer-4 Generic Test Mgr Test Group Resource Mgr Event Log Alerts Port Mgr Messages

Status Layer-3 L3 Endps VoIP/RTP VoIP/RTP Endps Armageddon WanLinks Attenuators Collision-Domains

Rpt Timer: default (5 s) Go Test Manager all Select All Start Stop Quiesce Clear

View 0 - 200 Go Display Create Modify Delete

Cross Connects for Selected Test Manager

| Name | Type | State | Pkt Tx A → B | Pkt Tx A ← B | Bps Rx B | Bps Rx A | Rx Drop % A | Rx Drop % B | Delay A → B | Delay A ← B | Jit |
|--------|------------|-------------|--------------|--------------|----------|----------|-------------|-------------|-------------|-------------|-----|
| test-1 | SIP/G.711u | Stopped | 15,752 | 15,762 | 57,886 | 57,923 | 0.07 | 0 | 0 | 0 | 0 |
| test-2 | SIP/G.711u | In progress | 1,185 | 1,185 | 56,531 | 56,531 | 0.084 | 0 | 0 | 0 | 0 |

Logged in to: brent-6port:4002 as: Admin

D. Go to the **VoIP/RTP Endps** tab to see detailed results:

| Calls Attempted | Calls Completed | Calls Failed | CF 404 | CF 408 | CF Busy | CF Cancel... | Calls Ans... | Destination Addr | Source Addr | Elapsed |
|-----------------|-----------------|--------------|--------|--------|---------|--------------|--------------|------------------|-------------|---------|
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 02161386 | 4826976 | 348 |
| 0 | 0 | 0 | 0 | 0 | 0 | 0 | 0 | 04826976 | 2161386 | 348 |
| 4 | 3 | 0 | 0 | 0 | 0 | 0 | 0 | 09201601 | 9502721 | 114 |
| 0 | 3 | 0 | 0 | 0 | 0 | 0 | 0 | 49502721 | 9201601 | 114 |

Logged in to: brent-6port:4002 as: Admin

- A. PESQ remains 0:0 when it is disabled for the call in progress
- B. **NOTE:** Endpoints are registered with the VoIP gateway while the call is in progress
- C. Calls Attempted, Calls Completed and Calls Failed can be viewed by scrolling to the right on the **VoIP/RTP Endps** tab

For more information see [LANforge User's Guide: VoIP Call Generator](#)

6.

Diagnosing Problems

- A. If your VoIP endpoint is not going on-hook, check your VoIP gateway to see if the extension is failing to register.
- B. Extensions failing to register might be missing name hostname or IP of the voice gateway they are calling.
- C. Some gateways want **Auth User Name** to include the IP or hostname, E.G. `3333@grandstream` or `3333@192.168.100.245`. Make sure any hostnames are resolvable using `nslookup $name` or `host -v $name`.
- D. Your VoIP gateway should log when extensions go on-hook, please check the gateway logs
- E. Some gateways will not accept direct extension dialing but require `extension@gateway` style dialing, E.G. the phone number that extension 2222 would call wants to look like `3333@192.168.100.245`