

Verifying VoIP Capacity

SIP IETF RFC 3261 Version 1.0

Objective

This test will determine the maximum number of simultaneous calls a VoIP network can handle on a local or wide area network. The test determines the capability of the network elements to receive and hold VoIP calls while managing the setup and tear down of those calls.

Overview

How can you find out if your VoIP system handles the number of calls you need? When the system is first installed or after an increase in users, it's important to know if the current system has to be completely replaced or modified.

Equipment suppliers can make recommendations from their experience based on other installations that use their equipment. They also can provide a list of equipment proven interoperable. But other networks are not identical to yours, and your system may require hardware and software changes.

The only way to know the capacity of your network is to connect a call generator that makes real calls over your network and provides information that describes your network. You need to know how many calls the network can handle along with the quality of the voice. This includes what kind of delays or distortions are present when performing various functions such as conferencing, announcements, and voice mail retrieval.

This test will provide the information you need.

VoIP network elements

- Softswitch (also known as a proxy server)
- Media gateway
- Signalling gateway
- Edge router
- Switches
- Application server
- Registration server
- DNS server
- Voice mail server
- Media server
- Conferencing Server
- IP PBX
- Management, provisioning, and billing interfaces

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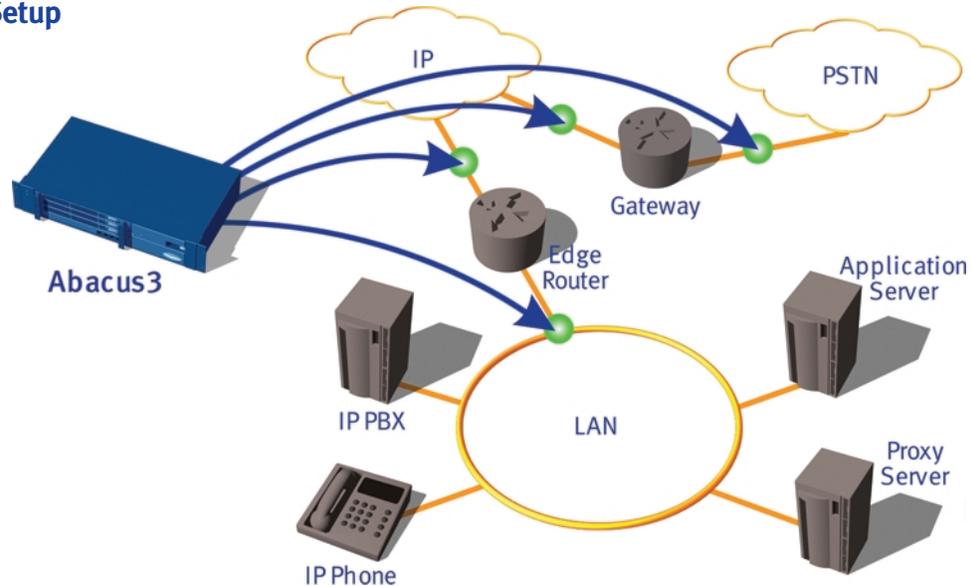
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This test uses the Abacus3 acting as many SIP telephones that register with the network, receive information from a DNS server, and make calls over the network. Abacus3 uses the SIP protocol to send call setup messages and real voice files through the network under test to verify proper operation.

The built-in protocol analyzer verifies proper signalling while the test set makes a variety of measurements that characterize the network. The Abacus3 uses “Packet Path Confirmation” to provide a pass or fail test for voice transmissions.

The user sets the threshold for packet loss and delay. When the measurements exceed any of these values, an event is displayed detailing the issue. When no event is created, the call has completed with minimal packet loss and without significant delays.

Setup



Test Steps

1. Connect the Ethernet cabling from the rear of the Abacus3 to the device or network under test.
2. Configure the test set by:
 - a. Clicking on “Phones” and entering the originating and terminating phone numbers.
 - b. Clicking on “Ptl Selection” and selecting SIP protocol – the Channels tab allows you to configure the voice encoding types, and the ICG tab allows you to configure the IP addresses of the test ports.
 - c. Clicking on “Partition” to create the script and timing for each set of ports. This configures the test set to either act as two end points, one generating call setup signalling and the other receiving the call setup information, or all ports calling another device or phone.
3. Click on Configure, Channels, and the Path Confirm tab. Under voice type select “packet only.” This provides you with a window to the right titled Packet. In the window, set the packet loss percentage to the amount you consider as a threshold for a poor connection. If packet loss exceeds this amount, you will receive an event notification for that call. Additionally, set the Max. Network Delay to the delay that is least acceptable for the network.
4. Start by setting the call load at only a few calls to verify the signalling is correct and the media path is established.
5. Start the test and check all measurements in the Variances window. Open the VoIP monitor windows and verify that the protocol signalling between all devices is correct. Check the Events window to identify any problems.

6. When a few calls are completed properly, take a base line measurement of all parameters. These are used to compare to the measurements with a higher load.
7. Once you are certain the signalling can transcend the network and complete a call, increase the call load to what you expect the network needs to handle.
8. Check for delays and other test parameters listed below. If there are no malfunctions or significant delays, increase the call setup rate and retest.
9. When you see errors, delays, or malfunctions, you may have reached the capacity of that network. The type of problem or delay can identify the component that limits the capacity. Sometimes decreasing a buffer size or increasing the processor speed in a router, switch, or gateway can increase the capacity of the network.
10. Repeat the tests with calls being made to application servers, conference servers, and other devices on the network.

Test Parameters

Observable results

- Call attempts.
- Call completions.
- Percentage of call completions.
- Call attempts per hour.

Measurable results

- Packet loss.
- Latency.
- Jitter.
- Packets out of order.
- Connection speed.
- Throughput.
- Call setup time.
- Message response time.

Test Outcome

When there is a significant change in any of these measured results, record the Call attempts, Call completions, and Call attempts per hour numbers. These results describe your network's capacity.

Expert Tips

Changes to the configuration of the network elements may change the results significantly. Test with multiple configurations of the devices in the network.