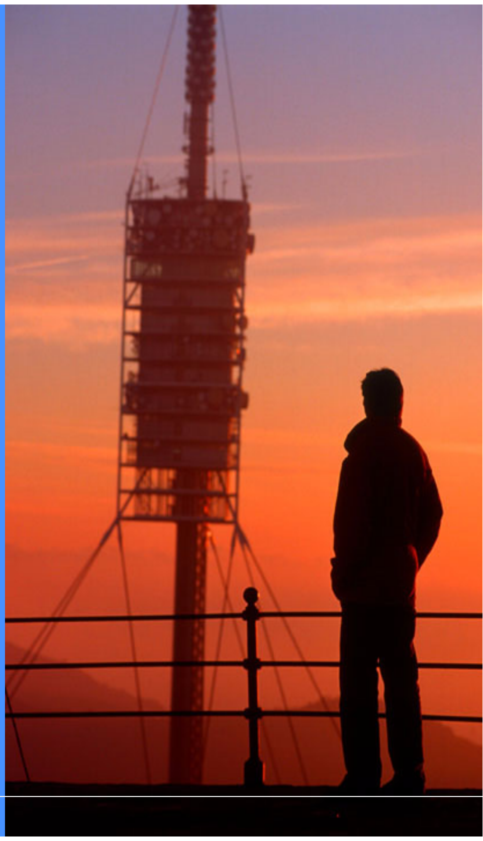
Verifying SIP Signalling

The Session Initiation Protocol (SIP) is an application-layer protocol used to establish, modify and tear down calls between users in IP networks.

by Francisco J. Hens



Testing the World's Networks

TrendCommunications

INTRODUCTION

The Session Initiation Protocol (SIP) is an application-layer protocol used to establish, modify and tear down calls between users in IP networks. SIP also locates and identifies Uniform Resource Identifier (URI) users. The protocol provides specific SIP URIs using a *sip:user@domain* structure similar to e-mail addresses. It can also handle traditional phone numbers.

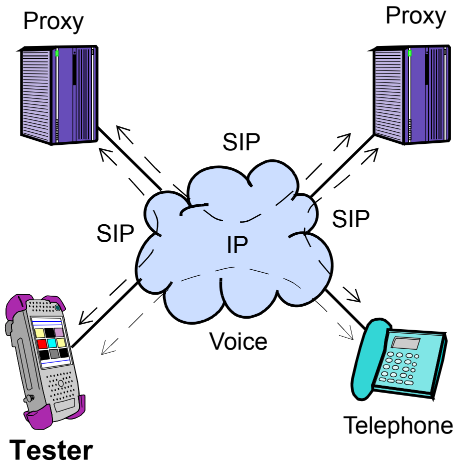


Figure 1 SIP test performed with Trend Multipro

There are various SIP service elements; for example, user location, supplementary service provision and connection to non-SIP devices.

- *User Agents* initiate requests and are usually the final destination of these requests. Some examples of user agents are Internet telephones and software phones (softphones).
- *Proxy Servers* are application-layer routers that forward SIP requests and responses. Their function can be compared to SMTP mail servers.
- *Redirect Servers* provide the location of alternative user agents or proxy servers when the original destination cannot be reached.
- *Registrars* keep track of their assigned network domain. They provide authentication of users and maintain information about the subscribers.

Proxies, redirect servers and registrars can be in the same physical device, or even in the same software.

VERIFYING SIP PROTOCOL OPERATION

1. Connect Trend Multipro to the network in one of the following ways:

   (a) from the subscriber LAN, by connecting Trend Multipro to an Ethernet interface located at the customer premises

   (b) from the local loop, by connecting the PIM's WAN connector to the provider network

   Note: If you are connected to the service provider network through a firewall, you can test the device by repeating the test from LAN and WAN interfaces. If the firewall is performing Network Address Translation (NAT), you will need a different configuration for the IP layer in each interface.

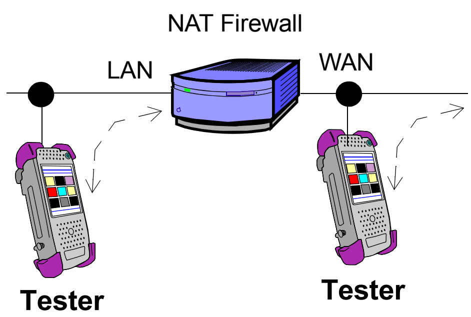


Figure 2 VoIP firewall traversal test performed with Trend Multipro

2. From the *Main* screen, choose  to display the *Modes* selection screen.
3. Select , and touch  to confirm your selection.

   This is an intrusive operation mode that enables Trend Multipro to send data to the network and analyse the responses. If you are performing a WAN test, you will need to enable the PIM when configuring the operation mode.

4. Select  from the *Applications* screen to display the VoIP test selection screen.
5. Select  to display the *VoIP Call* test screen.

   This enables you to call remote telephones or receive SIP calls. You can perform several tests on an active call.

6. Select one of the following:

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 – to load a predefined configuration

•the *Setup* tab – to enter your own configuration.

Setting Up the Call Manager Test

1. From the *Call Manager* test screen, select the *Layers* tab.
2. Check to see if you need to configure the protocol stack before performing the Call-Manager Test.
3. Select the *VoIP Call Setup* tab to display the *VoIP Call Setup 1* screen.
4. From the *VoIP Call Setup 1* screen, select *Display Name* and enter the DNS name to display on the destination phone.
5. Select *User Name* and enter a valid name.

   Note: The user name must be recognized by the network, and the network must be configured to accept and deliver SIP calls to this user.

6. Enter the domain name or the IP address of a SIP Proxy that will route the call to the destination.

   If the SIP port is not available, for example due to a firewall, enter an alternative port number in the SIP port field to redirect the call to an open TCP/UDP port.

7. Enter the *Authorization User* and *Password* if any of the following conditions are met:

   •Your SIP proxy has to authenticate any outgoing calls that you route through the SIP proxy.

   •Your SIP proxy behaves as a registrar. It keeps track of your presence in the network, and it is only able to route calls to you if you are on line and registered.

8. Select *Automatic*, if you want Trend Multipro to automatically register with a remote proxy/registrar, or select *Manual* and enter your own remote proxy/registrar.
9. Enter the Registration *Timeout*. This is the time that Trend Multipro will wait before timing out when waiting to register with a remote proxy/registrar.

Setting Up the Codec Types

You do not have to set up the codec types to perform VoIP signalling, but it may be necessary if you want to carry out performance measurements over the voice signal. We recommend that you enable all the codec types.

Making and Receiving VoIP Calls

Trend Multipro enables you to use two lines. A green screen LED indicates an active call. If both lines are active, select either or to activate the call.

A grey LED indicates a non-active line. Select to make it active.

During an active call, you can make a second call by selecting the other grey LED .

1. Select the *VoIP Call Status* tab.
2. To make a call to a remote SIP phone (or another type of telephone connected to the network by means of a SIP gateway), you will normally need to register with the service provider VoIP network.

   If you are not registered, select Register.

   If the *Registered* screen LED is green, Trend Multipro has already successfully registered with the network.

   Note: The Register button Register is only available if you have set up the *Registration* to *Manual*. When registration has been successful, the registered screen LED is green.
3. Choose or to activate the line.
4. Select the <T> field, or select and enter the destination URL.

   The sender <From> and <Codec> type are displayed.
5. Select to make a call.

   The <Duration> of the call is displayed.
6. To end the call, select .

Receiving and Answering Incoming Calls

To receive and answer incoming calls, do the following:

1. Select the *VoIP Call Status* tab.
2. If you are not registered, select .

   If the *Registered* screen LED is green, Trend Multipro has already successfully registered with the network.

   Note: The Register button is only available if you have set up the *Registration* to *Manual*. When registration has been successful, the Registered screen LED is green.

   Note: You are not usually authorised to receive calls if you are not registered.
3. When you hear the ringing tone, the status LCD display area flashes. This indicates an incoming call.
4. Select to answer the incoming call.

   If you want to reject the incoming call, select .
5. To end the call, select .

Viewing the VoIP Call History

To view the VoIP call history, do the following:

1. If the *VoIP Call Status* tab is not selected, select it.
2. Select to display the *VoIP Call History* screen.

   The number of *outgoing*, *answered*, *rejected* and *missed* calls are displayed.

   To restart the counters, select .

If you are able to call and receive calls to and from destinations defined by the testing protocol, the test has been successful.

Failures can be due to SIP telephones or proxies being incorrectly configured. Firewalls may also block calls. Failure of communications behind a gateway can be due to the operation of the gateway.

Finally, call establishment failures can be caused by problems in the lower transmission layers. In this case you can test the network connections using a ping test. If the phone is directly connected to the network, it should answer ping requests. If the phone is behind a firewall, it may not answer ping requests, but the firewall should still answer ping requests. If the phone is behind a gateway, it should still be possible to ping the gateway.

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