

Making a Call Using SIP – Quick Start

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Scope

This document covers the configuration of the Tenor AS/AX/AF/BX/DX/CMS Gateway products. This quick-start guide enables you to configure a single phone call on a Tenor using SIP. As a test, you can send the SIP call through a publicly available Proxy Server maintained by Quintum Technologies.

Introduction

SIP (Session Initiation Protocol) offers a new level of mobility for IP calls, as well as integration of call-related services. Routing in SIP is accomplished by using two servers: Proxy and Registrar. Because in most cases you will point to the same box and configure the same IP address for both servers, the Quintum Tenor combines the two.

The Proxy Server maintains a table of IP addresses and their corresponding Directory Numbers. This table is used by the Proxy to find the right place to send your call. When you dial a number, the Proxy looks up and finds out where the call must be sent. That information is used to route the call for you.

Prerequisites

- The Tenor's system software must be version P103-08-00 or later (which includes SIP). To find out the version number, Telnet into the unit and type **show -v**.
- Ensure you have your Tenor ready for connection, located on a LAN that is connected to the Internet.
- For the purposes of this quick start, your connection to the Internet is assumed to be through a DSL or cable connection. Other gateway connections are possible; use the principles in this document as a guideline for configuration.

Connect Hardware

Your Tenor needs to be connected to your LAN. If you have already set it up, you can jump to the next section, *Configure the Tenor*. Otherwise, you will need to achieve a few basic steps.

1. **Assign the Tenor an IP address.** Connect the Tenor directly to the PC with the supplied serial cable. Power up the Tenor and open a HyperTerminal session (or use another terminal emulation program). Assign the Tenor an available IP address on your LAN.
2. **Make the Tenor IP address publicly available.** When you connect the Tenor to your LAN (using the supplied Ethernet cable), you should list the Tenor in the DMZ of your gateway router. This allows your Tenor to have a publicly accessible IP.

Caution: The gateway router serves as a firewall to protect your LAN from malicious intrusion. Place the IP address of the Tenor in the DMZ so that callers may access your Tenor, even though it is “behind” a firewall. Please change the Tenor’s administration password as soon as possible to prevent malicious intrusion.

3. **Determine the WAN IP.** When you are configuring the gateway router, use its software to find out the WAN IP that the router has received from your ISP.

Once the Tenor is connected to your LAN, you can configure the unit using *Tenor Configuration Manager*, and make a SIP call.

Configure the Tenor

Note: The instructions below include information for making a test SIP call using a Proxy Server maintained by Quintum Technologies. You can hear a Quintum recording as a test of the first call.

1. Launch *Tenor Configuration Manager*, and when prompted, enter the IP address that you assigned to your Tenor.

Systemwide Configuration

2. From the main menu tree, select *System-Wide Configuration > Dial Plan > General tab*.
3. In the Country Code field, enter the dial code for the country where the Tenor is located, and in the **Area Code** field, enter the area code for the DN from which you are calling.
4. Adjust the Minimum and Maximum Dial Digit Length, as required.
5. Change Long Distance and International Prefix for local dialing rules, as required.
6. Click **Confirm/OK**.
7. From the top menu bar, select *File > Submit* to submit the changes.

VoIP Configuration

8. From the main menu tree, select *VoIP Configuration > Gateway*. In the **Outgoing IP Routing** field, click on the **SIP only** radio button. Click **Confirm/OK**.

9. From the main menu tree, select *VoIP Configuration > SIP Signaling Group-1*. On the **General** tab, change the **Register Expiry Time** to 30. In the **Primary SIP Server** field, enter the IP address **12.176.187.191**. Click **Confirm/OK**.
10. On the **User Agent** tab, click **Add** to display the **Add User Agent** dialog. Enter your full international format phone number in the **Primary User** field (such as 17324609000), and enter an alternative number in the **Contacts[1]** field. Click **OK** on the dialog, then click **Confirm/OK** at the bottom of the window. From the top menu bar, select *File > Submit* to submit the changes.

Circuit Configuration

11. From the main menu tree, select *Circuit Configuration > Line Routing Configuration > Hunt LDN Directories > Hunt LDN Directory-pub1*. Click **Add** to display the **Add Hunt LDN Number** dialog. Enter your phone number without area code (for example, 5551000) in the **Number Pattern** field. Click **OK**, then click **Confirm/OK** at the bottom of the **Hunt LDN Directory-pub1** window. From the top menu bar, select *File > Submit* to submit the changes.
12. Select *Circuit Configuration > Line Routing Configuration > Line Circuit Routing Groups > Line Circuit Routing Group-phone*. On the **General** tab, in the **SIP User Agent** field, select the User Agent that you defined previously from the drop-down box. Click **Confirm/OK** at the bottom of the window. From the top menu bar, select *File > Submit* to submit the changes.

Phone (FXS)/Line (FXO) Configuration

13. From the main menu tree, select *Phone (FXS)/Line (FXO) Configuration*. Under Analog Online Setting for Phone-Line/FXS-FXO Pair, enable Phone-Line 1 and enable Phone-Line 2, if required, by checking the checkboxes. Click **Confirm/OK** at the bottom of the window.
14. From the top menu bar, select *File > Submit* to submit the changes.

Give the Proxy's Registrar a minute or two to store the registration information.

Make First SIP Call

Dial the first call using Quintum's DN for test purposes: 17324609000. The call is then sent to our test Tenor. After a few rings, an answering machine picks up the call, your call is completed, and you should hear the greeting played back to you.