Application Note

ShoreTel™

ST-0224
December 14, 2006

Mediatrix 1204 Analog Gateway (SIP)

Analog gateways provide customers the ability to connect up to Central Offices and other Gateway solutions. There maybe times when a customer either doesn't have a ShoreGear switch or ports available on the switch. With the Mediatrix Gateway this will provide additional flexibility.

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Mediatrix

The Mediatrix 1204 is a high-quality and cost efficient VoIP gateway connecting IP networks to the PSTN.

It is the ideal solution to deploy private or small hosted toll bypass networks.

The Mediatrix 1204 bridges the gap between the IP and the PSTN. It connects up to four analog CO lines to the IP through a single 10/100 BaseT Ethernet port – providing convenient access to an IP telephony network for PSTN users around the world.

The Mediatrix 1204 provides PSTN access for Shoretel PBX System. It is an efficient solution to maintain local PSTN breakout in remote locations that are converted to IP.

By connecting CO lines from selected sites to a VoIP network, the Mediatrix 1204 enables enterprises to use a VoIP connection between pre-determined local networks. When used in conjunction with Shoretel PBX, routing schemes and calling rights can be programmed in order to optimize the use of resources and minimize long distance fees.

Features and Benefits

- IP connectivity for analog PSTN lines
- IP connectivity for legacy PBX Systems
- PSTN connectivity to IP-based telephone systems
- Provides PSTN users access to a VoIP Network
- PSTN-quality voice over IP networks
- Deployable in SIP or H.323 VoIP networks
- Auto configurable, remotely manageable and upgrade-able
- Interoperable with equipment from leading industry vendors
- Fax over IP support
- Multiple codec support
- Internal power supply
- Ideal for enterprise or carrier-based applications

Vendor Overview and Contact

Mediatrix Telecom, Inc. - Headquarters
4229 Garlock Street
Sherbrooke, Quebec
J1L 2C8
CANADA
Tel: 819-829-8749
Fax: 819-829-5100
http://www.mediatrix.com

Sales Department
Americas
Tel: 1-877-GET-VOIP / (877) 438-8647
Tel: (514) 285-0058
Fax: (514) 842-0125

Europe, Middle East, Africa
Tel: +39-02-84742280
Fax: +39-02-84742212

Resellers who want to start selling this solution should go to the following URL: http://www.mediatrix.com/partners_become.php

Vendor Product Information

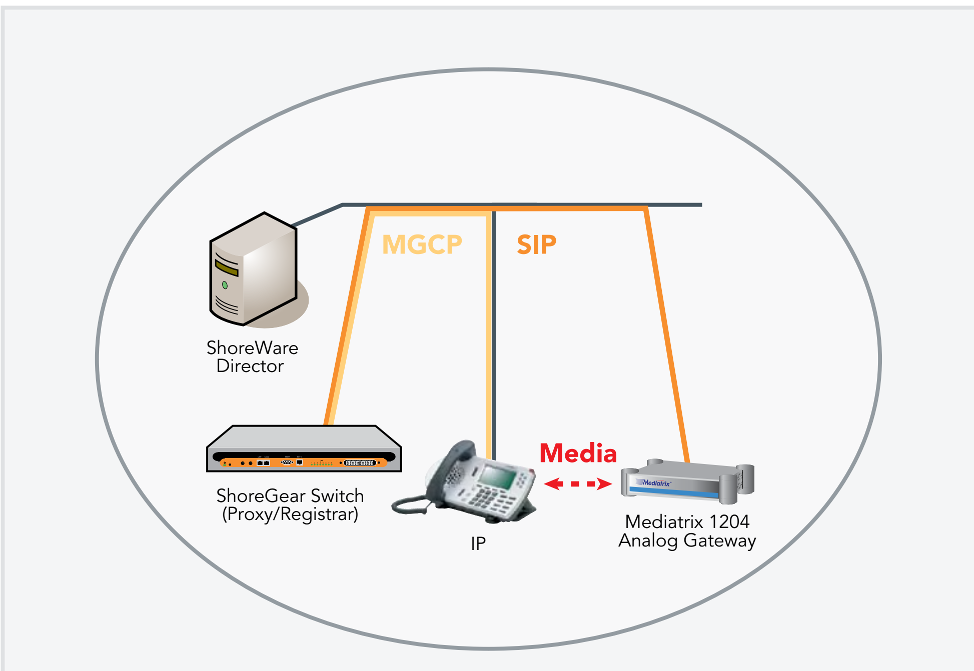
Model Number: Mediatrix 1204

Where to buy: http://www.mediatrix.com/buy.php

List Price: Please check your reseller

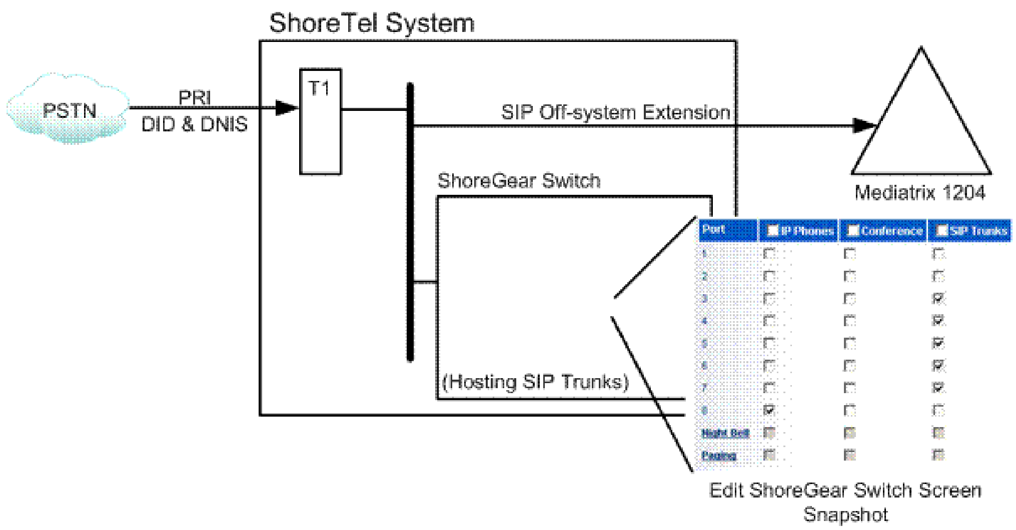
Architecture Overview

The following is a diagram of the solution architecture showing the integration between the Mediatrix 1204 Analog Gateway and the ShoreTel System:

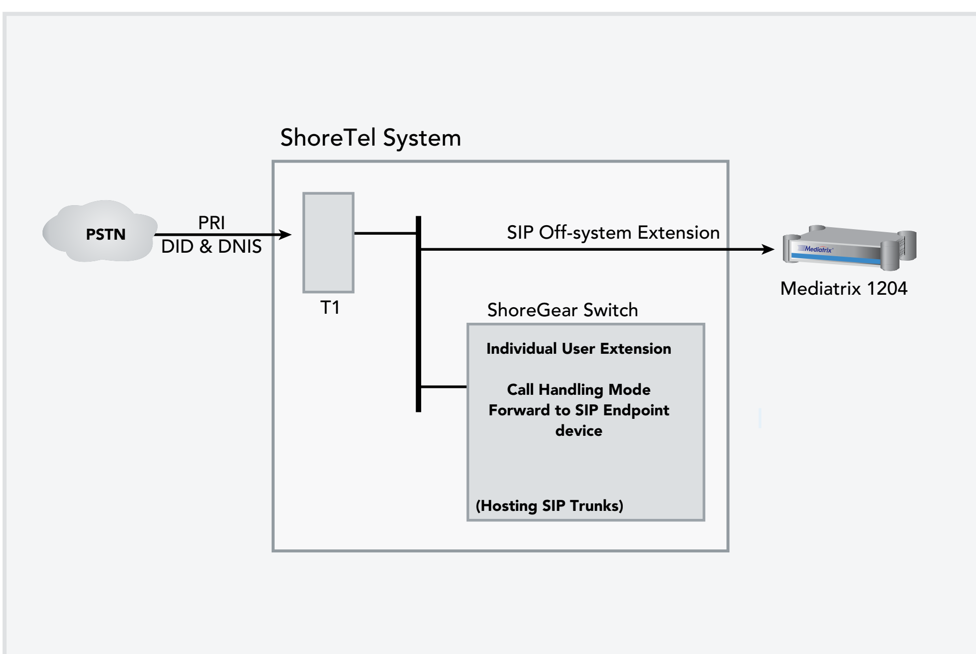


Sip devices are being implemented on the ShoreTel System with SIP trunks. The ShoreTel system will use outbound call routing for the SIP devices. There are various ways to handle the call routing for the SIP devices. The following drawings depict different call routing designs:

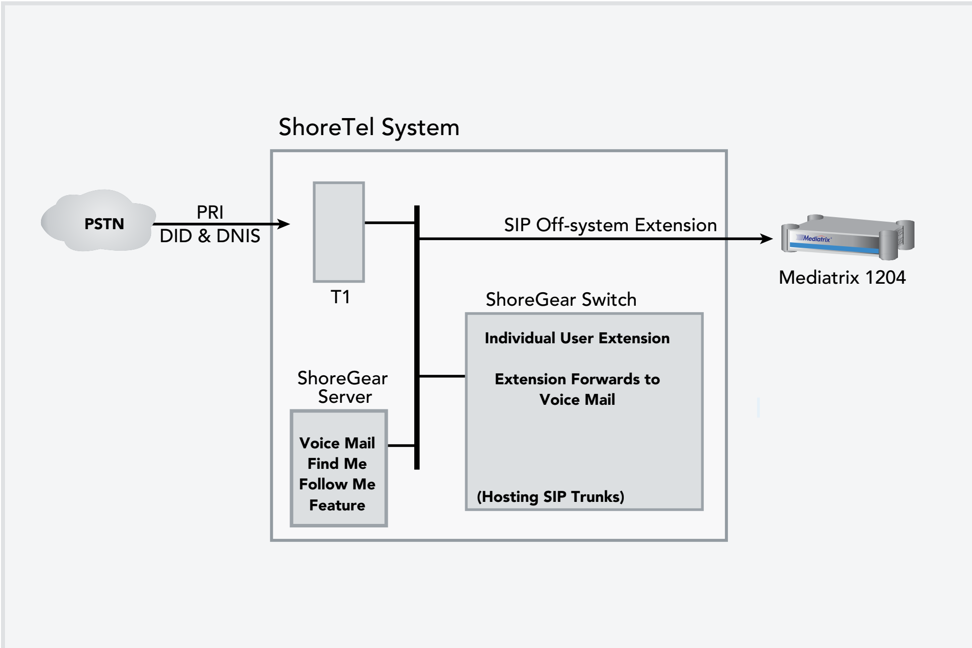
1. Route the call with DID (Direct Inward Dial) or DNIS (Dialed Number Identification Service) directly to the Mediatrix 1204 Analog Gateway extension. In this configuration tandem trunking is enabled and off system extensions are defined in the trunk group. Shown in the drawing is a snapshot of the "Edit ShoreGear Switch" screen indicating the selection of SIP trunks.



2. Routing through Individual User Extension with the Call Handling Mode set to forward (external destination) enter Mediatrix 1204 Analog Gateway extension.



3. Routing through Individual User Extension with extension forwarded to Voice Mail. The "Find Me Follow Me" feature of voice mail can be used to forward call (external) to Mediatrix 1204 Analog Gateway extension.



Requirements, Certification and Limitations

The following requirements are necessary to integrate a Mediatrix 1204 Analog Gateway to the ShoreTel IP Phone system as described in this Application Note.

ShoreTel Requirements

- ShoreWare Server Software, ShoreTel 6.1 (11.14.5501.0) or higher. Versions prior to this release will not support the Mediatrix 1204 product.
- ShoreTel SIP Trunk port licenses are required.

Mediatrix 1204 Analog Gateway Requirements

- Mediatrix 1204 Analog Gateway – this device should be running the latest firmware (see version support table below).

Certification

CTLABS TESTING SERVICES

CT LABS VoIP Endpoints INTEROP TEST 2006

Interoperability Test Program for VoIP Endpoints Test Results Overview

VOIP Endpoint Product Information:

Date:	December 11, 2006
Vendor Name:	Mediatrix
Product Name:	Mediatrix 1204 FXO Gateway
Product Modle No:	1204
Product Release:	5.0.15.92
Date Range of Tests Performed:	June 5 through December 6, 2006

Test Results Overview

This section presents an overview of the results of all the VoIP Endpoint test cases available in this test plan template.

Passed	ID	Optional?	Name	Description
Passed	1.1	Mandatory	Device initialization with static IP address	Verify successful startup and initialization of the device up to a READY/IDLE state using a static IP address
Passed	1.2	Mandatory	Device reset – idle (for static configurations)	Verify successful re-initialization of device after power loss while device is idle
Passed	1.3	Mandatory	Device initialization with DHCP	Verify successful startup and initialization of the device up to a READY/IDLE state using DHCP
Passed	1.4	Mandatory	Device reset – idle (for dynamic configurations)	Verify successful re-initialization of device after power loss while device is idle
Passed	1.5	Mandatory	Verify Diffserv Code Point support	Verify the ability to set Diffserv Code Point from SIP DUT and verify via inspection of packet capture
Passed	1.6	Optional	Verify Date and Time Update support	Verify setting of Date and Time Update on SIP DUT. Time Zone can be updated by using the web interface
Passed	1.7	Mandatory	Place call	Verify successful call placement with normal dialing to a variety of terminating phones
Passed	1.8	Mandatory	Receive call	Verify successful reception of calls with normal dialing from a variety of calling phones
Passed	1.9	Optional	Place call – re-dial	Verify successful call placement using re-dial to SIP Reference
Passed	1.10	Optional	Place call – speed dial	Verify successful call placement using programmed speed dial
Passed	1.11	Mandatory for G.711, Optional for other CODECs	CODEC support – common (from DUT to ShoreTel Phone, REF-x)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)
Passed	1.12	Mandatory for G.711, Optional for other CODECs	CODEC support – common (from DUT to SIP Reference Phone, SIP-Ref)	Verify successful call connection and audio path using all supported CODECs (G.711-Ulaw and G.729)
Passed	1.13	Mandatory (only if more than 1 CODEC is supported)	CODEC support – negoti-ated	Verify successful negotiation between devices configured with different default CODECs (G.711-Ulaw and G.729)
Passed	1.14	Mandatory	Hold from DUT to SIP Reference	Verify successful hold and resume of connected call
Passed	1.15	Mandatory	Hold from DUT to ShoreTel Phone	Verify successful hold and resume of connected call
Passed	1.16	Mandatory	Forward	Verify successful forwarding of incoming calls
Passed	1.17	TBD	Forward from SIP DUT	Verify successful forwarding of incoming calls
Passed	1.18	Optional	Mute	Verify device's mute function
Passed	1.19	Mandatory	Out-of-band / In-band DTMF Transmission	Verify successful transmission of in-band and out-of-band digits (RFC2833) for calls placed to and from the DUT with a variety of other devices
Passed	1.20	Optional	Missed call notification	Verify that device notifies the user about missed calls
Passed	1.21	Optional	Volume	Verify the device's volume adjustment function

Table 1 1: Basic Feature Test Cases

Passed	ID	Optional?	Name	Description
Passed	2.1	Mandatory	Speech quality – Minimal impairment	Verify acceptable voice quality between two parties with minimal network impairment condition
Passed	2.2	Mandatory	Speech quality – Moderate Impairment	Verify acceptable voice quality between two parties with low-to-moderate artificial network impairment condition
Passed	2.3	Mandatory	Speech quality – High Impairment	Verify acceptable voice quality between two parties with moderate-to-high artificial network impairment condition

Table 1 2: Performance Test Cases

Passed	ID	Optional?	Name	Description
Passed	3.1	Mandatory	Call waiting	Verify appropriate notification and successful connection of incoming call while busy with another party
Passed	3.2	Mandatory	Park	Verify successful park and retrieval of connected call
Passed	3.3	Optional	Extended forward	Verify extended call forwarding options – busy forwarding, no-answer forwarding
Not Tested	3.4	Not Supported	Extended forward from SIP DUT	Verify extended call forwarding options – busy forwarding, no-answer forwarding
Passed	3.5	Mandatory	Transfer – blind	Verify successful blind transfer of connected call
Passed	3.6	Mandatory	Transfer – monitored	Verify successful monitored transfer of connected call
Failed	3.7	Mandatory	Conference – ad hoc	Verify successful ad hoc conference of three parties
Passed	3.8	Optional	Place call – secondary line	Verify successful call placement using secondary line
Passed	3.9	Optional	Receive call – secondary line	Verify successful connection of incoming call on secondary line
Passed	3.10	Optional	Callback	Verify successful connection of a call using the missed-call callback feature of the device
Passed	3.11	Optional	Headset	Verify the device's support for external headsets (using headsets supplied by the 3P phone vendor)
Passed	3.12	Optional	Ring selection	Verify the device's ability to change the ring type
Passed	3.13	Mandatory	Caller ID Name and Number	Verify that Caller ID name and number is sent and received from SIP endpoint device
Passed	3.14	Mandatory	SIP Device Generates Busy Tone.	Verify that SIP DUT generates busy tone when calling a busy extension
N/A	3.15	Optional	POTS Analog Gateway supports the transfer operation by "flashing"	Verify that the POTS Analog Gateway can support the transfer operation by "flashing"
Passed	3.16	TBD	Verify handling of "911"	Verify dialing "911" on DUT could connect with "911" services
N/A	3.17	N/A	Verify Fax Handling	Verify that fax can be sent and received through DUT

Table 1 3: Extended Feature Test Cases

Limitations

Office Anywhere is not supported with the ShoreTel Release 6.1. A future release will support Office Anywhere.

Currently these are known issues at the time of this release. These issues will be addressed in the future!

Interaction with "911" Support

With the release of ShoreTel 6.1 support for 911 calls is limited. If a 911 call is made whatever number is configured for that SIP device is what will be sent out the local trunk which services 911 calls based on the "Site" of the ShoreGear switch which is associated for the SIP Trunk.

Other items to consider:

- If the SIP device is ONLY configured for example with a four digit extension then that is number which will be sent! Depending how the trunk from the CO is configured it may default to the billing address or something else. Check with your phone company on how 911 calls are handled.
- If the SIP device is configured with a full 10 digit DID then once again that's what will be sent out the local trunk where the ShoreGear Switch is located supporting the SIP Trunk.
- Should the SIP device be in NY and the ShoreGear switch which supports the SIP Trunk for the SIP device in San Jose then the 911 call will go out through whatever 911 trunk is configured for the San Jose site!
- It is recommended 911 is fully tested for based on the design!
- Should it be desired to use 10 digit DID, using "Digit Translation" with "Off System" extensions can be used. Example: Dialing 3510 can translate to 408-331-3510. This is only needed when configuring the "Out Bound" portion of digit translation in the Trunk Group. See the Planning and Installation Guide for information on configuring Digit Translation.

Version Support

Product certified via the Technology Partner Certification Process for the ShoreTel system. Table below contains the matrix of Hitachi Wireless IP-5000 handset firmware releases certified on the identified ShoreTel software releases.

		Mediatrix 1204 Analog Gateway Firmware Version		
		5.0.15.92	N/A	N/A
ShoreTel Release	ShoreTel 6.1 11.14.5501.0 or newer	✔		
	N/A			
	N/A			
	N/A			

Configuration Overview

The following steps are required to configure the Mediatrix 1204 Analog Gateway to work with the ShoreTel system:

1. Configure General ShoreTel system settings
   a. Call Control Options, Site and Switch settings
2. Setup Trunk Groups
3. Setup Individual Trunks and User Groups
4. Configure Mediatrix 1204 Analog Gateway to function with the ShoreTel system
5. Configure Appropriate Call Routing options

ShoreTel Configuration

This section describes the ShoreTel system configuration to support the Mediatrix 1204 Analog Gateway. The section is divided into general system settings and trunk configurations (both group and individual) needed to support the Mediatrix 1204 Analog Gateway.

ShoreTel System Settings—General

The first settings to address within the ShoreTel system are the general system settings. These configurations include the call control, the site and the switch settings. If these items have already been configured on the system, skip this section and go on to the "ShoreTel System Settings – Trunk Groups" section below.

Call Control Settings

The first settings to configure within ShoreWare Director are the Call Control Options. To configure these settings for the ShoreTel system, log into ShoreWare Director and select "Administration" then "Call Control" followed by "Options" (Figure 1).

Figure 1 Administration Call Control Options

The "Call Control Options" screen will then appear (Figure 2).

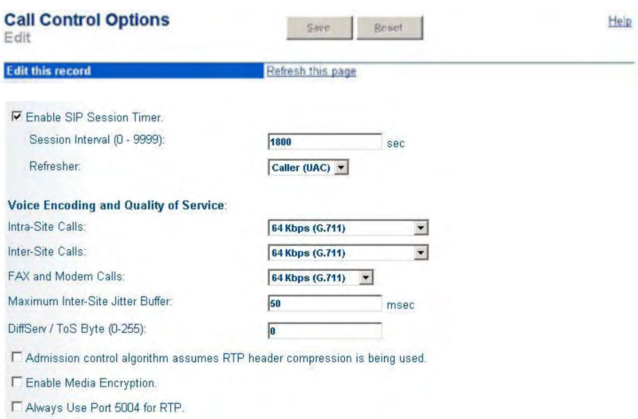


Figure 2 Call Control Options

Within the "Call Control Options" screen confirm the appropriate settings are made for the "Enable SIP Session Timer", "Intra-Site Calls", "Inter-Site Calls" and "Always Use Port 5004 for RTP" fields.
The first step is to ensure the box is checked for the "Enable SIP Session Timer". Next the Session Interval Timer needs to be set. The recommended setting for "Session Interval" is 1800 seconds. The last item to select is the appropriate refresher (from the pull down menu) for the SIP Session Timer. The "Refresher" field will be set either to "Caller (UAC)" [User Agent Client] or to "Callee (UAS)" [User Agent Server]. If the "Refresher" field is set to "Caller (UAC)" the Caller's device will be in control of the session timer refresh. If "Refresher" is set to "Callee (UAS)" the device of the person called will control the session timer refresh.
The next settings to verify are the "Intra-Site Calls" and the "Inter-Site Calls" settings under the" Voice Encoding and Quality of Service" prompt. For the Intra-Site Calls verify the desired audio bandwidth is selected for the CODEC for calls within the system. The settings should then be confirmed for the desired audio bandwidth for the CODEC for Inter-Site calls (calls between sites).

Note: SIP uses both G.711 and G.729 CODECs. The CODEC setting will be negotiated to the highest CODEC supported.

Un-checking the box for "Always Use Port 5004 for RTP" is required for implementing SIP on the ShoreTel system. For SIP configurations, Dynamic UDP must be used for RTP Traffic. Note: If the box is unchecked MGCP will no longer use UDP port 5004; MGCP and SIP traffic will use dynamic UDP ports.

Note: When unchecking port 5004 the system will then require a full reboot. This is a one time event! Make sure everything is rebooted such as "Servers, ShoreGear Switches, IP Phones, etc..."

Sites Settings:

The next settings to address are the administration of sites. These settings are modified under the Shore Ware Director by selecting "Administration" then "Sites" (Figure 3).

Figure 3 Administration Site

This selection brings up the "Sites" screen. Within the "Sites" screen select the name of the site to configure. The "Edit Site" screen will then appear. The only change required to the "Edit Site" screen is to the "Admission Control Bandwidth" field (Figure 4).

Figure 4 Admission Control Bandwidth

The Admission Control Bandwidth defines the bandwidth available to and from the site. This is important as SIP devices will be counted against the site bandwidth. See the *ShoreTel Planning and Installation Guide* for more information. When a call is "Intra-site" it will not count against the "Admission Control Bandwidth" unless "Teleworker" is checked in the "Trunk Group Page".

Switch Settings—Allocating Ports for SIP Trunks

The final general settings to input are the ShoreGear switch settings. These changes are modified by selecting "Administration" then "Switches" in ShoreWare Director (Figure 5).

Figure 5 Administration Switches

This action brings up the "Switches" screen. From the "Switches" screen simply select the name of the switch to configure. The "Edit ShoreGear …Switch" screen will be displayed. Within the "Edit ShoreGear …Switch" screen select the desired number of SIP trunks from the ports available (Figure 6).

Figure 6 ShoreGear Switch Setting

Each Check box designated as a SIP trunk enables the support for 5 individual trunks.

ShoreTel System Settings—Trunk Group

ShoreTel Trunk Groups support both Dynamic and Static SIP end point Individual Trunks.

Note: A ShoreGear switch can only support one Trunk Group with Dynamic IP addressing.

In trunk planning a couple of things need to be considered.

1. Are the SIP devices using DHCP or Static IP?
2. Are the SIP devices endpoints (like ATAs, Conference Phone or WiFi handset) or non-endpoint devices like an ISP?

If the SIP Trunk Groups have already been configured on the system, skip down to the "ShoreTel System Settings - Individual Trunks" section. The settings for Trunk Groups are changed by selecting "Administration" then "Trunks" followed by "Trunk Groups" within ShoreWare Director (Figure 7).

Figure 7 Administration Trunk Groups

This selection brings up the "Trunk Groups" screen (Figure 8).

Name	Type	Site	Trunks	DID	Destination	Access Code
Analog Loop Start	Analog Loop Start	Headquarters	0	No	1700	9
Digital Loop Start	Digital Loop Start	Headquarters	0	No	1700	9
Digital Wink Start	Digital Wink Start	Headquarters	0	No	1700	9

Figure 8 Trunk Groups Settings

From the pull down menus on the "Trunk Groups" screen select the "site" desired and then select "SIP" and click on the "Go" link. The "Edit SIP Trunk Group" screen" will appear (Figure 9).

Figure 9 SIP Trunk Group Settings

Determine whether the trunks need to be configured as inter-site trunks (trunks between sites) or intra-site trunks (trunks within a site). In the "Edit SIP Trunks Group" screen, input a name for the trunk group. When naming the trunk group it is recommended to include the trunk type and the extension or extension range of the trunk group being established. Having this detail in the name is a convenient way to recall the trunk type and extension range. In the example in Figure 9 the name "SIP x3000-3019 Intra-site" has been created. This name indicates that a SIP Intra-site trunk with twenty extensions in the range of 3000-3019 has been created. Or if this box is connected to a CO and no "Off System Extensions" are being used then a more appropriate name might be "SIP - CO XYZ City Name" - just needs to be something meaningful. The next step is to verify the setting of the "Teleworker" check box. The "Teleworker" check box needs to be checked if inter-site trunk groups have been configured. Checking this box will count against the site bandwidth.

The next item on this screen is the "Enable Digest Authentication" field. This is a SIP feature that allows a user ID and password to be established for security. If this feature is configured for the Trunk Group the same settings must be configured on the SIP endpoint devices.

The "Enable SIP Info for G.711 DTMF Signaling" box should not be checked. Enabling SIP info is currently only used with tie trunks between ShoreTel systems.

After these settings are made to the "Edit SIP Trunk Group" screen, press the "Save" button to input the changes.

The next item to change for the SIP trunk groups is the handling of the inbound trunk configuration (Figure 10).

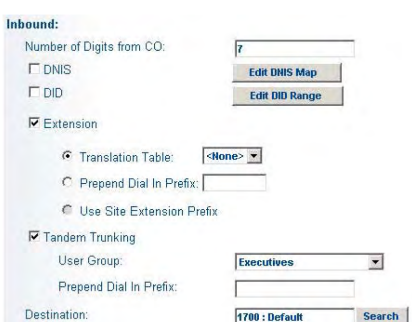


Figure 10 Inbound Trunk Configuration

For the Inbound Trunk Settings ensure the "Number of Digits from CO" is correct. Select the "DNIS" (Dialed Number Identification Service) box to create a DNIS to extension mapping. Select the "DID" (Direct Inward Dial) box to input a DID range. The next step is to make sure the "Extension" box is checked along with the "Translation Table" button along with the translation table from the pull down menu. The last step for the Inbound Trunk configuration is to select the "Tandem Trunking" box if needed. Selecting this box will allow trunking between trunk groups. The "Destination:" contains the number for the Auto Attendant. Trunk calls not routed by previous entries under the Inbound Trunk routing will be routed to the Auto Attendant.

The next item to change for the SIP trunk groups is the handling outbound trunk configuration (Figure 11).

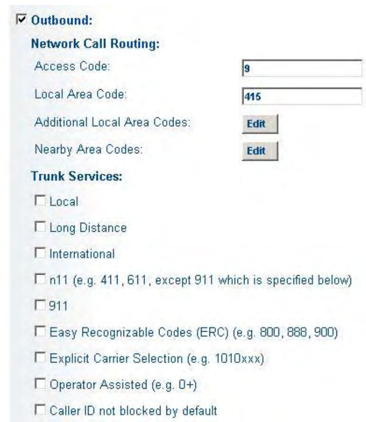


Figure 11 Outbound Trunk Configuration

For the Outbound Trunk Configuration set the "Access Code" and "Local Area Code" fields for the system. For the "Trunk Services" boxes, typically none of these will be selected.

Note: If connecting to a CO via the analog gateway use of "Off System Extensions" may not be needed.

The next item to change for the SIP Trunk Groups is the Off System Extensions (Figure 12).

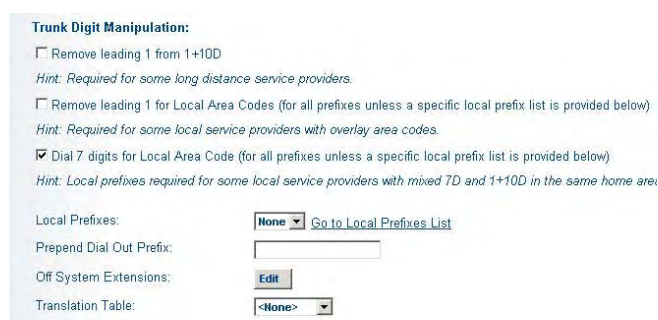


Figure 12 Off System Extension

Selecting the "Edit" button for the Off System Extensions will bring up the"Off System Extension Ranges" dialog box (Figure 13).

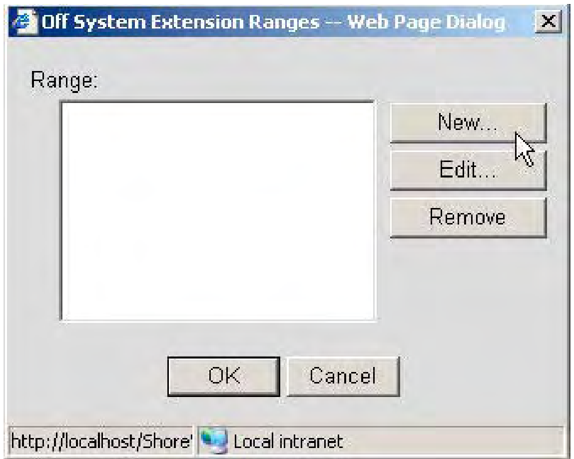


Figure 13 Off System Extension Ranges

Selecting the "New" button will bring up the "New Range" dialog box (Figure 14).

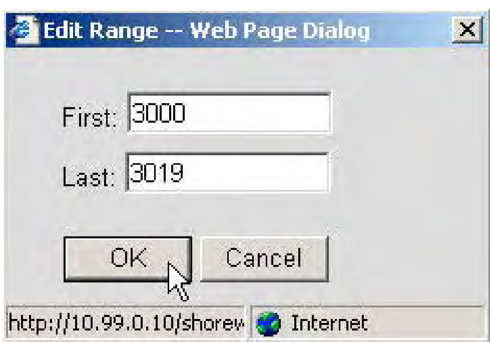


Figure 14 New Range Settings

Input a new range of off system extensions for the SIP devices and click "OK". For this example the extension range 3000-3019 has been input, which corresponds to the naming convention used in the previous example. This completes the settings needed to set up the SIP Trunk Groups on the ShoreTel system.

Note: It is important to note that only one dynamic trunk group can be set up per ShoreGear switch. If another dynamic trunk group is desired another ShoreGear switch will be needed.

ShoreTel System Settings—Individual Trunks:

This section covers the configuration of the individual trunks. Select "Administration" then "Trunks" followed by "Individual Trunks" to configure the individual trunks (Figure 15).

Figure 15 Individual Trunks

The "Trunks by Group" screen, used to change the individual trunks settings, then appears (Figure 16).

Figure 16 Trunks by Group

Select the site for the new individual trunk(s) to be added and select the appropriate trunk group from the pull down menu. In this example the site is "Headquarters" and the trunk group is SIP x3020 Intra-site".

Note: If this box is connected to a CO and no "Off System Extensions" are being used then a more appropriate name might be "SIP - CO XYZ City Name" - just needs to be something meaningful.

Click on the "Go" button to bring up the "Edit Trunk" screen (Figure 17).

Figure 17 Edit Trunks Screen for Individual Trunks

From the individual trunks "Edit Trunk" screen, input a name for the individual trunks, select the appropriate switch, select the SIP trunk type and input the number of trunks. When selecting a name, the recommendation is to name the individual trunks the same as the name of the trunk group so that the trunk type and extension range can easily be tracked. Select the switch upon which the individual trunk will be created. For the SIP Trunk Type decide whether the trunks are to be configured as dynamic or static. Dynamic trunks are typically configured for endpoint devices like wireless handsets, conference phone or Integrated Access Devices (IADs). For the "SIP Trunk Type" field select either "Dynamic" or for a static configuration select "Use IP Address" button and input an IP address. For best results with the Mediatrix 1204 the administrator should select "Use IP Address" and put the IP address of the Mediatrix (Note: Configure the Mediatrix device with a static IP address, using dynamic could break the configuration if the IP address was to change). Next set the "Number of Trunks". Because the Mediatrix has 4 analog ports the most individual trunks that will need to be set it "4". Though should only 2 ports be used then configure with "2" for example. Once these changes are complete, select the "Save" button to create the list of individual trunks (Figure 18).

Trunks by Group Help

Add new trunk at site: Headquarters in trunk group: SIPx3000-3019 Intra-site Go

Name/Number	Group	Site	Switch	Port/Channel	IsDigital	TrunkTypeID	DeviceIPAddress
SIPx3000-3019 Intra-site	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (1)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (10)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (11)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (12)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (13)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (14)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (15)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (16)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (17)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (18)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (19)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (2)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (3)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (4)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (5)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (6)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (7)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (8)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0
SIPx3000-3019 Intra-site (9)	SIPx3000-3019 Intra-site	Headquarters	SG-40/8	0	1	6	0.0.0.0

Figure 18 Trunks by Group

Note: Individual SIP trunks cannot span networks. SIP trunks can only terminate on the switch selected. There is no failover to another switch.

After setting up the trunk groups and individual trunks refer to the ShoreTel Product Installation Guide to make the appropriate changes for the User Group settings. This completes the settings for the ShoreTel system side.

Mediatrix 1204 Analog Gateway Configuration

To begin the configuration of the Mediatrix 1204 FXO gateway, you must first install a configuration and management software tool provided by Mediatrix called the Unit Manager Network (UMN). The UMN is provided on the CD included with the unit. It has a default 3-units limit upon installation. This will suffice for most configurations.

Please refer to the following documentation for the installation and administration of the software:

- UMN Quick Start guide
- Unit Manager Network Administration Manual

Once the UMN software has been installed on your PC , proceed with the following steps.

Step 1: Start the UMN

- Select from the Start Menu > Programs > Unit Manager Network 3.2 > Administrator.

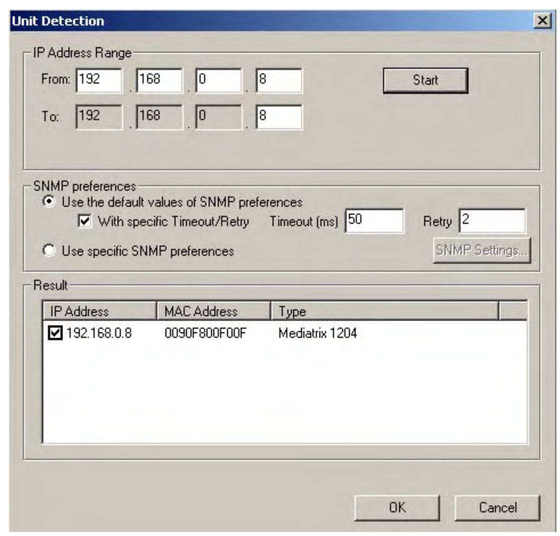
Step 2: Login to the UMN

- In the Administrator login window (Connect to Unit Manager), the IP address 127.0.0.1 is a loopback address that refers to the local PC to which the UMN is connecting. A User Name and Password are not required. Click OK to proceed.

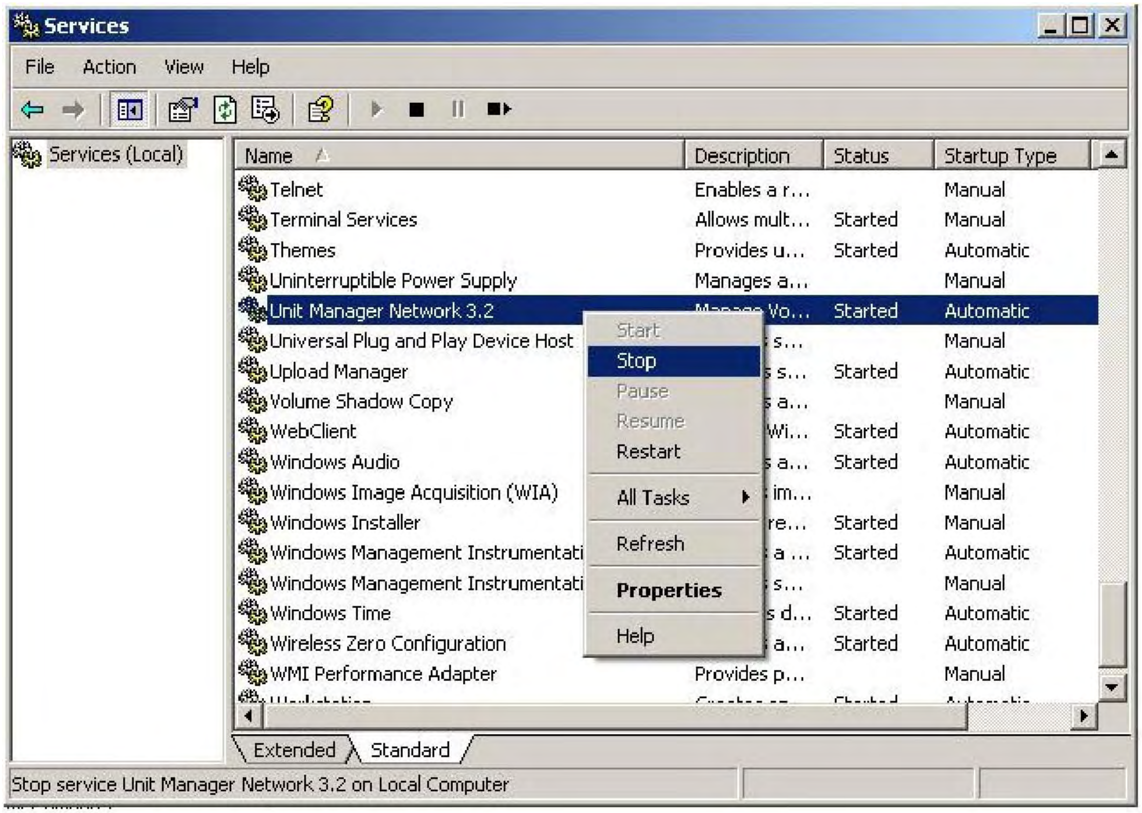
Step 3: Perform an Autodetect

- On the left panel, right-click the Unit Manager level, then select Autodetect.
- Set the IP Address Range to minimize the time it takes to auto-detect the unit. Click Start to begin the Mediatrix unit detection. When the unit is detected, the Result section will show the unit.
- Select the unit and click OK. If no DHCP server is used in your subnet, you must connect one unit at a time since they will start by using the default IP address 192.168.0.1 after a recovery reset. You will have to set a different static IP address for every unit. Please see the product SIP Quick Start Guide for more details on the initial setup.

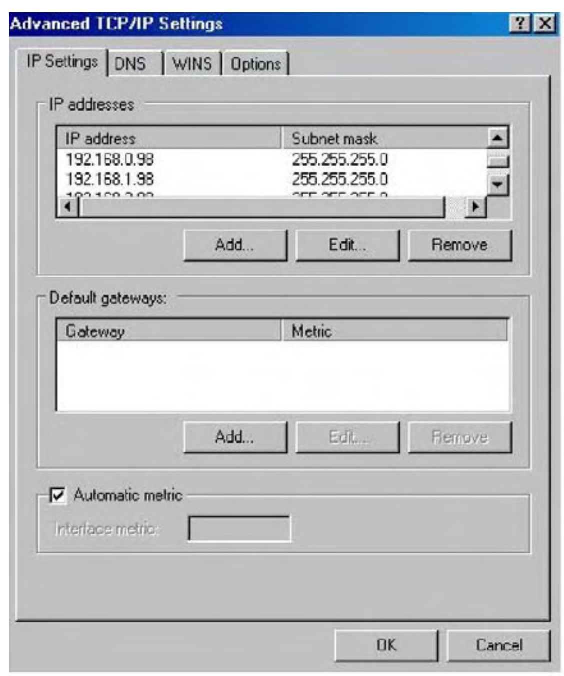
Note: The UMN runs in the Windows background. Mediatrix recommends that the administrator stops the UMN service before changing the IP address of the PC network interface and restart the service when completed. A reboot of the PC may be required depending on your Windows O/S.



- Go to Start Menu > Settings > Control Panel > Administrative Tools > Services.
- Stop and restart the Unit Manager Network 3.2 service.

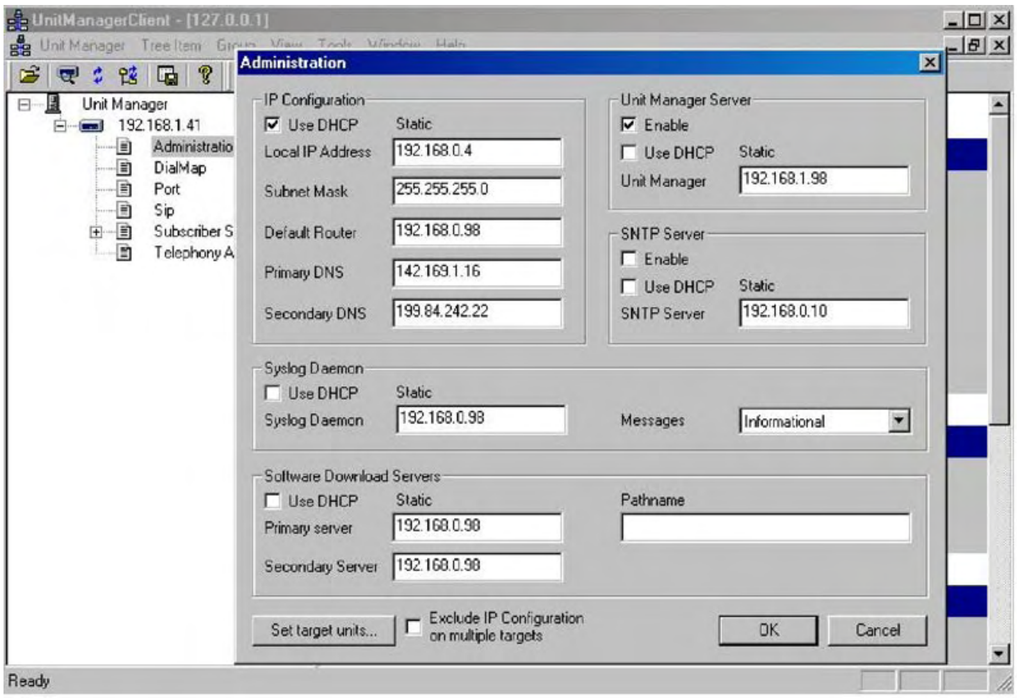


Alternatively, you can configure multiple IP addresses on the same PC network interface card (only in static IP mode – not DHCP) to avoid restarting the computer or stopping and restarting the UMN service whenever you change the IP address and subnet of the Mediatrix unit. This is recommended for advanced IP network users, the current configuration notes do not cover the step by step setting for this alternative.



Step 1: Setting Basic Network Parameters
You can also reserve the IP address in DHCP for the Mediatrix 1204 FXO gateway by assigning the MAC address of the unit to a specific IP address – in this fashion, it will be easier to auto-detect the unit for provisioning.

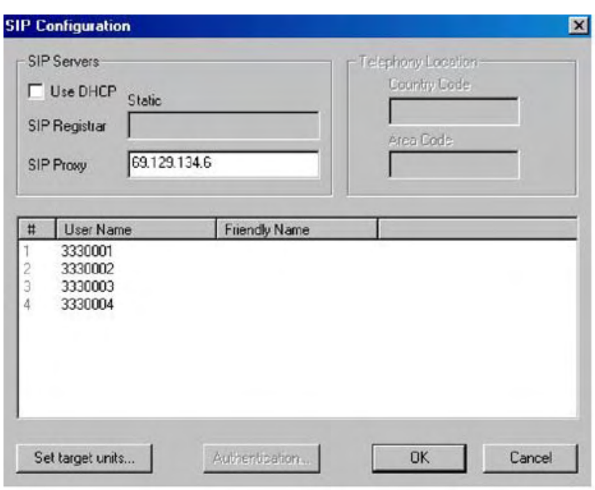
Alternatively, you can auto-detect the Mediatrix 1204 after a recovery reset (in a recovery reset, the unit boot up with address 192.168.0.1) and reconfigure the IP addresses to reflect the new network. An example is shown in the diagram below. Remember to restart the unit after the configuration of IP addresses to accept the changes.



Step 2: Set the SIP Server Registrar and Proxy

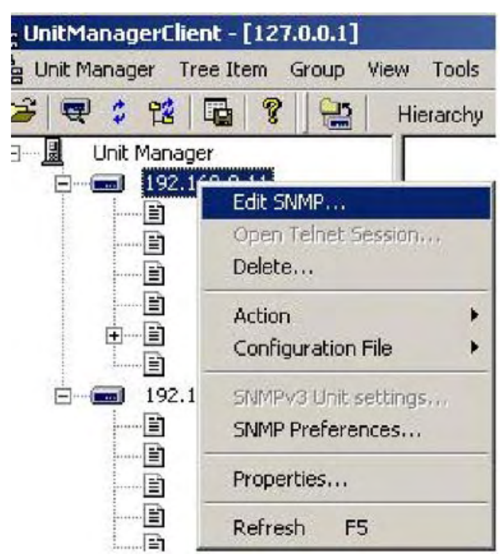
Select the Sip page and set the following:

- Uncheck Use DHCP.
- Set the SIP Proxy field to be the IP address of the Shoretel device.
- Use default values for the other entries.



Step 3: Access the Edit SNMP Window

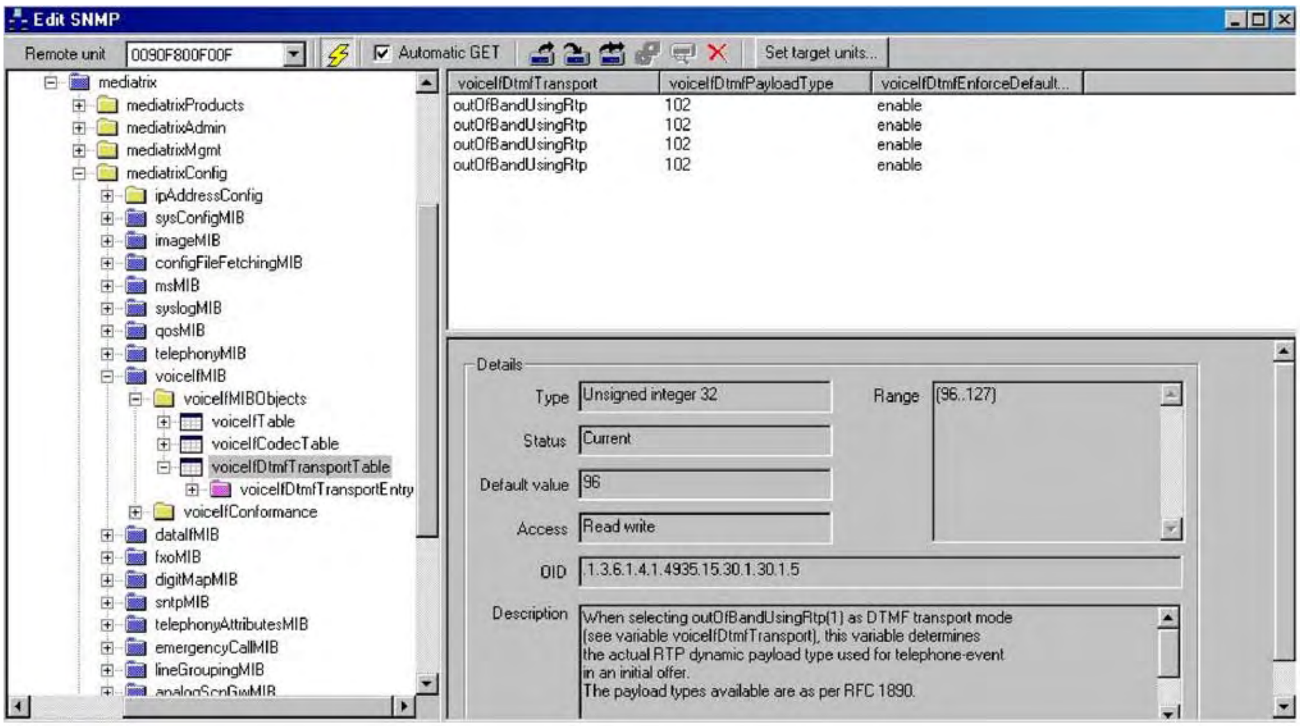
- Right-click the unit and select Edit SNMP.



Step 4: Set the DTMF transport method and payload type

Set the DTMF transport method to out-of-band using RFC2833 and payload type 102 (or any payload type matching the Shoretel system.)

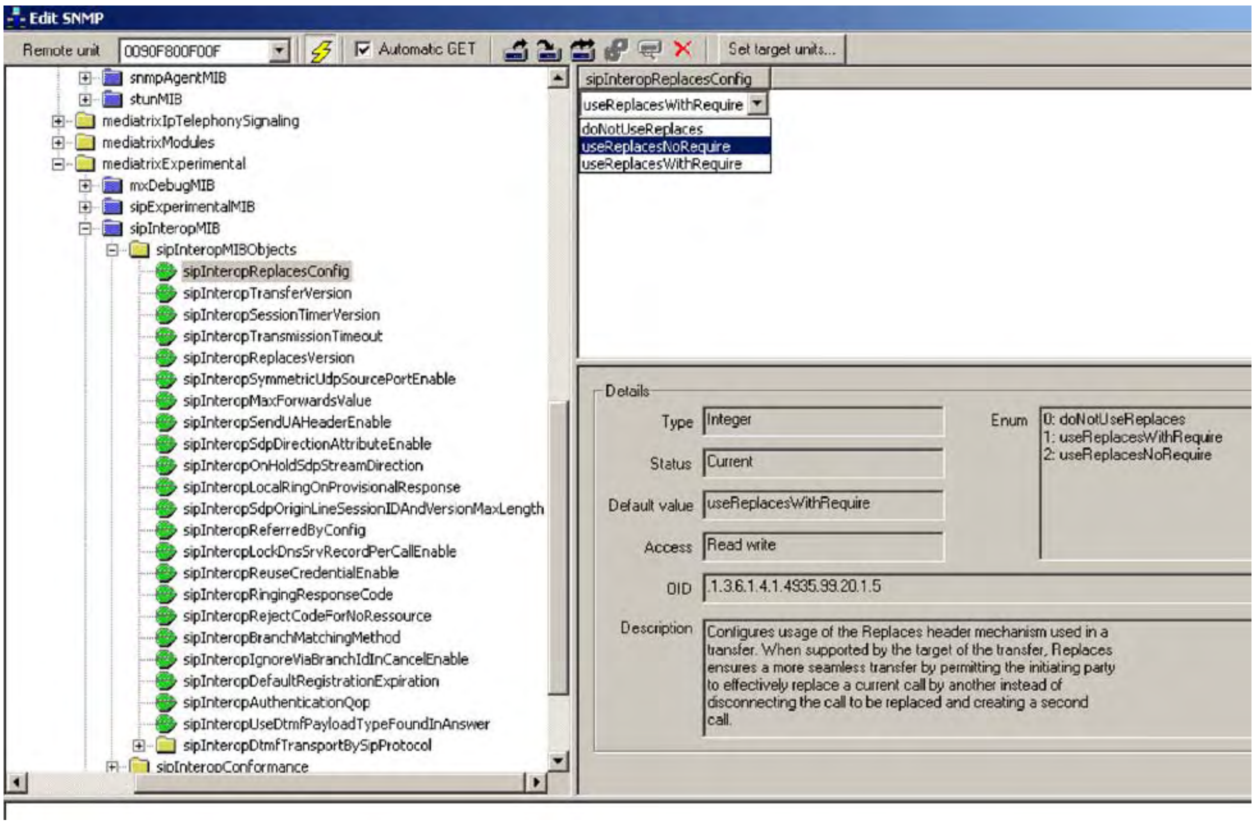
- On the top menu bar, check the option Automatic GET.
- Go to iso>org>dod>internet>private>enterprises>mediatrix > mediatrixConfig > voiceIfMIB > voiceIfMIBObjects > voiceIfDtmfTransportTable
- Select the MIB voiceIfDtmfUsingRtp and set FXO ports to outOfBandUsingRtp. Select the MIB voiceIfDtmfPayload-Type and set FXO ports to 102 as shown below.



Step 5: Set SIP Refer Interop Parameter (for Call Transfer)

Set the Interop MIB as follows:

- On the top menu bar, check the option Automatic GET.
- Go to iso>org>dod>internet>private>enterprises>mediatrix > mediatrixExperimental > sipInteropMIB > sipInteropMIBObjects > sipInteropReplacesConfig
- Set it MIB to useReplacesNoRequire. And hit "enter" to make the change effective.



Record of Change

This application note is subject to change. Updates and corrections are always welcome. Please submit any updates or corrections to info@shoretel.com.

Issue	Author	Reason for Change	Date
1.0	J. Casselman	Initial Release	December 14, 2006

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