



Enabling Live Communications at the Edge of IP Networks

Configuration Notes 292

PBX Remote Line Extension using Mediatrix 4104 and 1204

June 22, 2011

Proprietary

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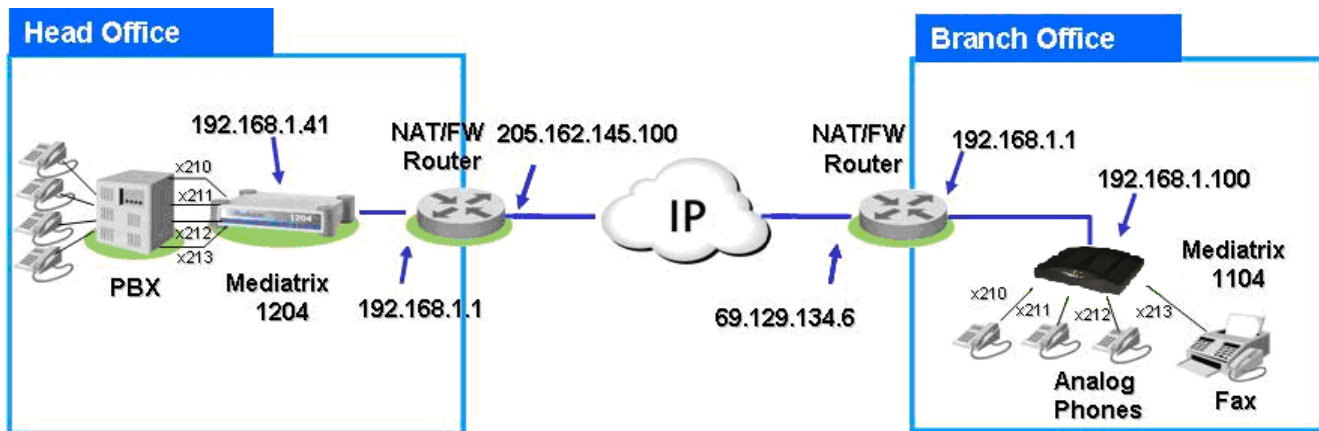
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Introduction

These Configuration Notes are distributed to partners and technical support personnel as an aid to configure units in a PBX Remote Line Extension scenario. They are specific to the SIP v5.0 firmware.

Application Scenario

- With VoIP, the distance limitation of extending copper wire from a PBX to a remote location is no longer an issue. Small remote branch offices (SOHOs) can now have existing analog phones installed at the remote site as analog extensions of the PBX system at the Main Office.
- Requirement for fax at remote branch office.
- Requirement for analog phone at remote branch office.
- Toll-Bypass for intra-office calls.



The following configuration notes are not a substitute for the Mediatix Administration Documentation for Mediatix analog products. Please have the following manuals available for reference:

- Mediatix 4104 SIP Reference Manual (SIP v5.0 version).
- Mediatix 1204 SIP Reference Manual (SIP v5.0 version).
- Mediatix Unit Manager Network Administration Manual and Quick Start guide (UMN v3.2).

Running the Unit Manager Network Software

The Unit Manager Network (UMN) software is a configuration and management tool for Mediatrix devices. The UMN can be downloaded from the Mediatrix Support Portal: <https://support.mediatrix.com/DownloadPlus/Download.asp>. It has a default 3-units limit upon installation. This will suffice for most configurations.

It is recommended that the administrator uses the UMN to configure the Mediatrix devices. Please refer to the UMN Quick Start guide for the installation of the software. Once the UMN software has been installed on your PC¹, proceed with the following steps.

Step 1: Start the UMN

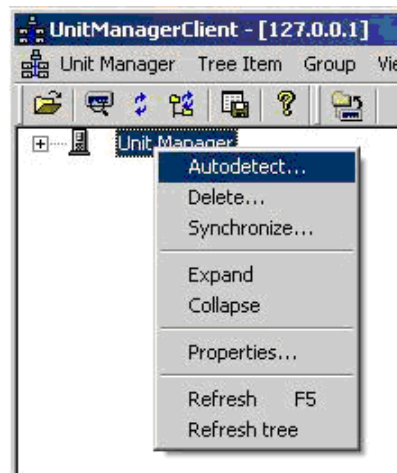
- Select from the *Start Menu > Programs > Unit Manager Network 3.2 > Unit Manager Network*.

Step 2: Login to the UMN

- In the *Administrator* login window (Connect to Unit Manager), a User Name and Password are not required. Click **OK** to proceed.

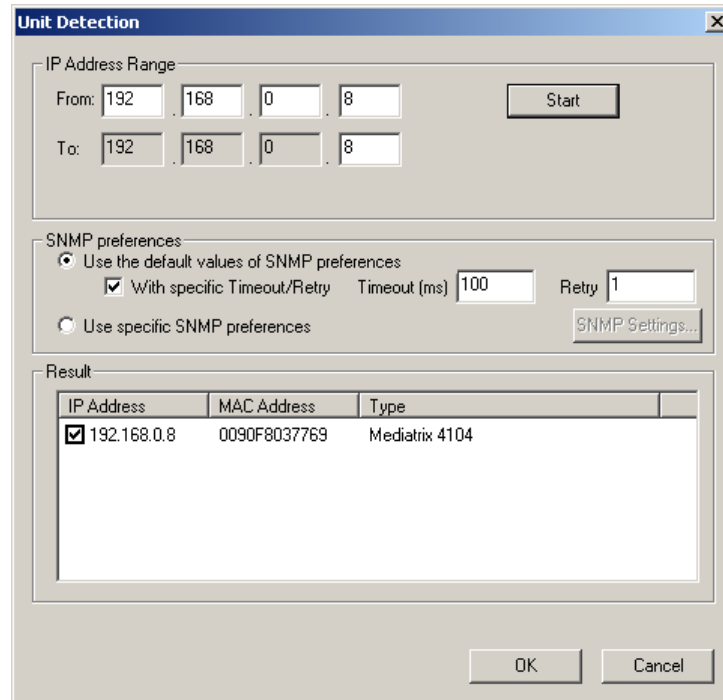
Step 3: Perform an Autodetect

- On the left pane, right-click the *Unit Manager* level, then select **Autodetect**.



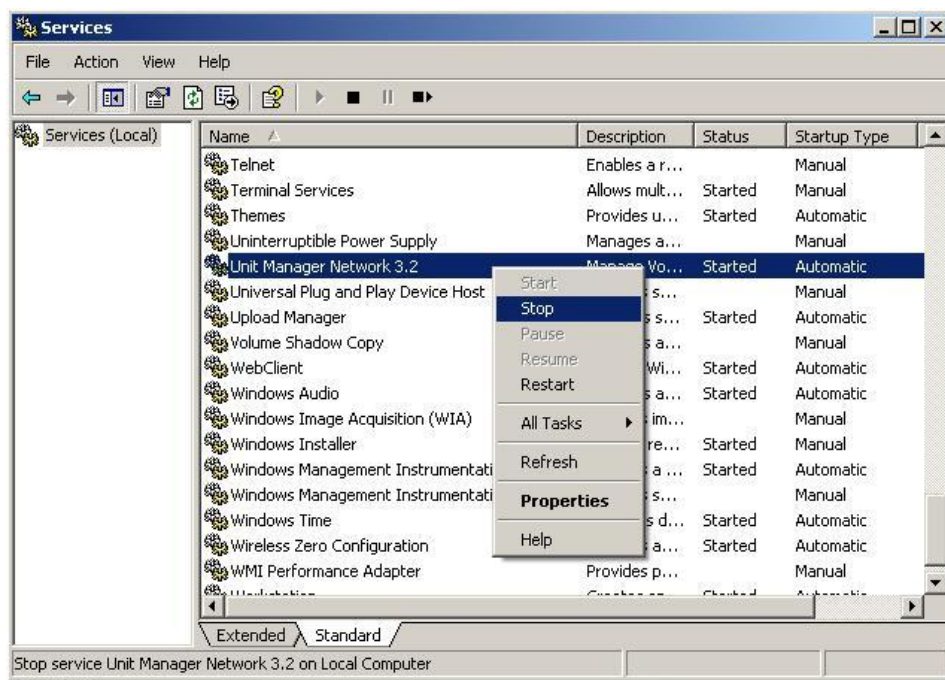
- Set the *IP Address Range* to minimize the time it takes to auto-detect the unit. Click **Start** to begin Mediatrix unit detection. When the unit is detected, the *Result* section lists the unit.
- Select the unit and click **OK**. If no DHCP server is used in your subnet, you must connect one unit 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 **Mediatrix 4104 and 1204 SIP Quick Start Guide** for more details on initial setup.

¹ If you are having trouble opening the UMN application, verify that the status of the "Unit Manager Network 3.2" service is started. See Page 5 to access Windows Services.



Note: The UMN runs in the Windows background. Media5 highly recommends that the administrator stops and restarts the UMN service whenever the IP address of the PC is changed. A restart may be needed 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.

Advanced TCP/IP Settings ? X

IP Settings | DNS | WINS | Options

IP addresses:

IP address	Subnet mask
192.168.0.98	255.255.255.0
192.168.1.98	255.255.255.0
192.168.0.98	255.255.255.0

Add... Edit... Remove

Default gateways:

Gateway	Metric
---------	--------

Add... Edit... Remove

☒ Automatic metric

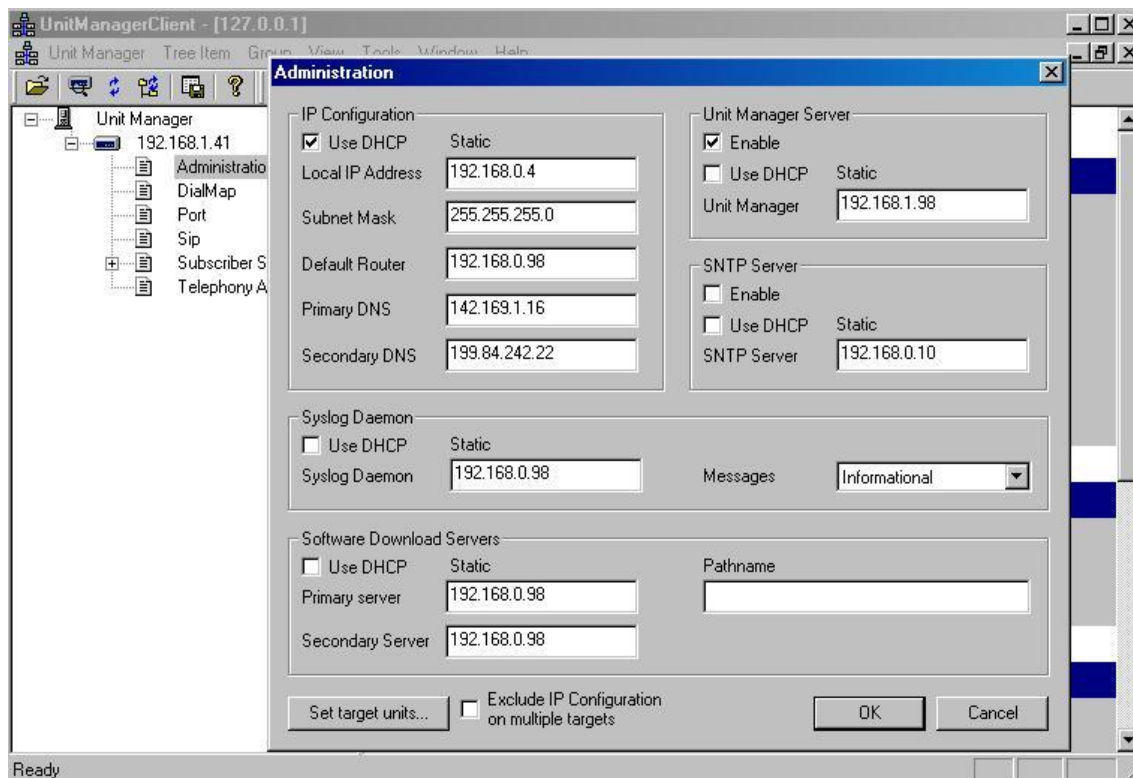
Interface metric:

OK Cancel

Configuring the Mediatrix 4104

Step 1: Setting Basic Network Parameters

- Once the Mediatrix 4104 has been auto-detected, it should appear under the *Unit Manager* level. Right-click the *Administration* level of the Mediatrix 4104 and select the *Edit* menu.
- Configure the IP addresses to reflect your network. Uncheck **Use DHCP** if you are using static IP addresses. Restart the unit after the configuration of IP addresses for the unit to accept the new addresses.
- When using DHCP: you can also reserve the IP address for the Mediatrix 4104 by assigning the MAC address of the unit to a specific IP address in your DHCP server – it will be easier to auto-detect the unit for provisioning. An example of a unit IP address fetched by DHCP is shown in the figure below.
- Another alternative to determine the IP address of the unit fetched by DHCP is to connect a telephone into port #2 of the Mediatrix 4104 and dial ***#*0**. An internal IVR will provide the IP address of the unit.



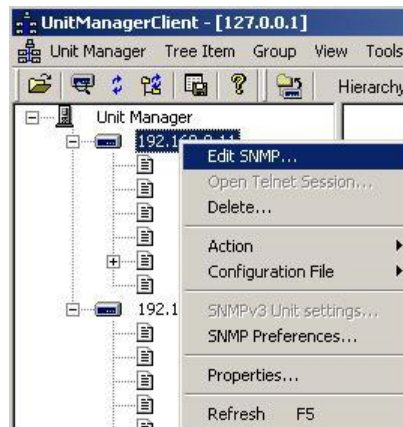
Step 2: Set the SIP Server Redirection

- Select the **Sip** page and set the following:
 - Uncheck **Use DHCP**.
 - Set the *SIP Registrar* field to empty since registration is not required.
 - Set the *SIP Proxy* field to be the IP address of the destination² device. In this scenario, the destination IP address is that of the Mediatrix 1204.
 - Click in the *User Name* field next to the port number and type the telephone, fax, or extension number of the Mediatrix 4104 port. Each port on the 4104 requires a User Name.

#	User Name	Friendly Name
1	210	Port 1
2	211	Port 2
3	212	Port 3
4	213	Port 4

Step 3: Access the Edit SNMP Window

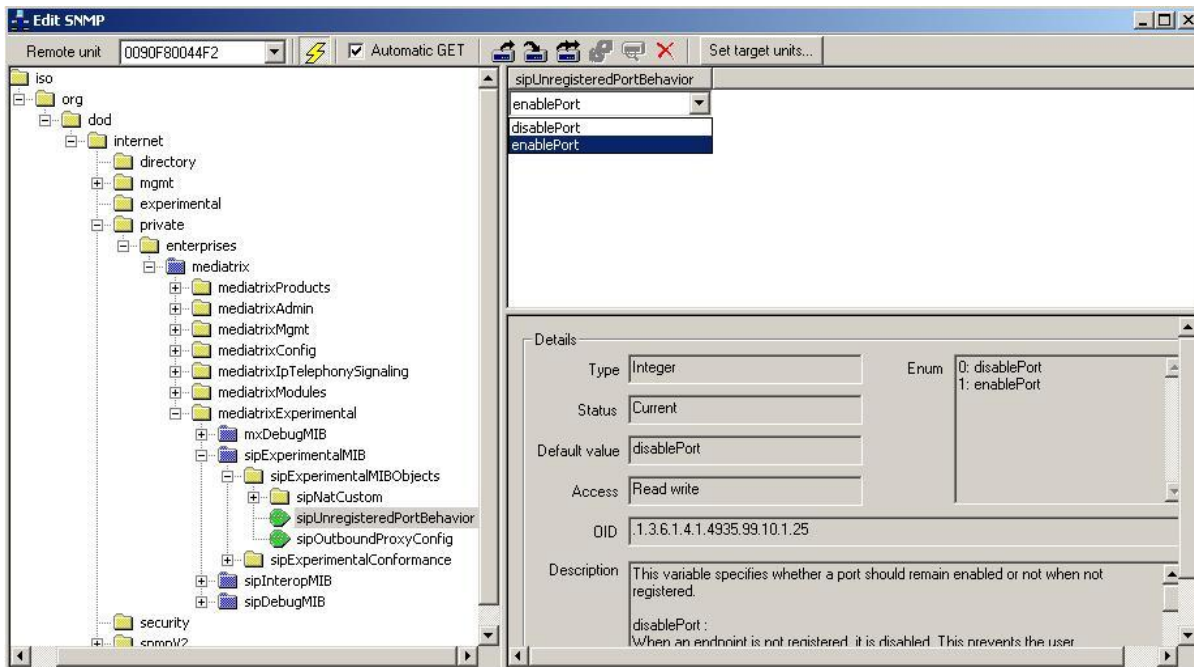
- Right-click the unit and select **Edit SNMP**.



² Using this setup will route all calls from the Mediatrix 4104 to the assigned IP address, that is the Mediatrix 1204. Note that you cannot use this configuration to route calls to more than one unit.

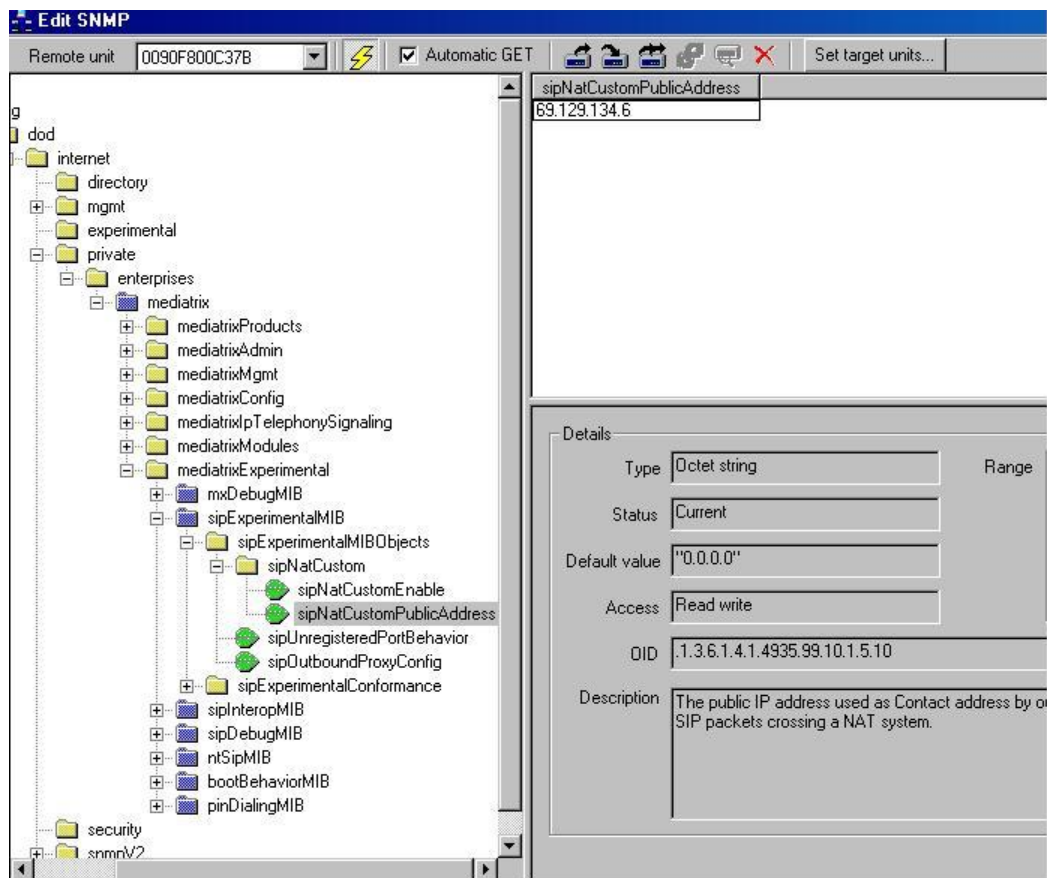
Step 4: Set the Port Behaviour

- On the top menu bar, check the option **Automatic GET**.
- Go to *iso>org>dod>internet>private>enterprises>mediatrix > mediatrixExperimental > sipExperimentalMIB > sipExperimentalMIBObjects > sipUnregisteredPortBehavior* and set this parameter to **enableport** in the drop-down menu that appears upon double-clicking the value located in the upper-right pane.



Step 5: NAT Traversal Configuration

- You must configure the following parameters for the Mediatrix unit to work behind a NAT/Firewall router:
 - Go to *mediatrix > mediatrixExperimental > sipExperimentalMIB > sipExperimentalMIBObjects > sipNatCustom > sipNatCustomEnable* and set this parameter to **enable**.
 - Go to *mediatrix > mediatrixExperimental > sipExperimentalMIB > sipExperimentalMIBObjects > sipNatCustom > sipNatCustomPublicAddress* and set the parameter to be the public IP address of the NAT/Firewall router where the Mediatrix 4104 is located. In this scenario, the IP address would be **69.129.134.6** as shown in the following figure.



Step 6: Configure Traffic Routing to the Mediatrix Unit

- To allow traffic from the NAT/Firewall router to be routed to the Mediatrix unit, there are two alternatives:
 - Option 1 – Configure the NAT/Firewall router's DMZ (demilitarized zone) to include the Mediatrix unit. Please refer to the documentation provided with your NAT/Firewall router for configuration details.
 - Option 2 – Enable the port forwarding on the NAT/Firewall router. The following ports should be forwarded to the IP address of the Mediatrix unit behind the NAT/Firewall:
 - UDP 5060 for SIP Signaling
 - UDP 5004 to 5010 for Voice RTP streams
 - UDP 161 (optional for management of unit via UMN-SNMP)

Configuring the Mediatrix 1204

Step 1: Autodetect the Mediatrix 1204

- Autodetect the Mediatrix 1204 after a recovery reset (in a recovery reset, the unit restarts with the address 192.168.0.1) and configure the IP addresses to reflect your network parameters.
- Uncheck **Use DHCP** if you are using static IP addresses. An example is shown in the following figure.
- Restart the unit after the configuration of IP addresses for the unit to accept the new addresses.
- When using DHCP, you can reserve the IP address for the Mediatrix 1204 by assigning the MAC address of the unit to a specific IP address in your DHCP server – it will be easier to auto-detect the unit for provisioning.

The screenshot shows the 'Administration' window with the following configuration details:

Section	Option	Value
IP Configuration	Use DHCP	☐ Static
	Local IP Address	192.168.1.100
	Subnet Mask	255.255.255.0
	Default Router	192.168.1.1
	Primary DNS	142.169.1.16
	Secondary DNS	199.84.242.22
Unit Manager Server	Enable	☒
	Unit Manager	192.168.1.98
SNTP Server	Enable	☐
	SNTP Server	192.168.1.98
Syslog Daemon	Use DHCP	☐ Static
	Syslog Daemon	192.168.1.98
Software Download Servers	Use DHCP	☐ Static
	Primary server	192.168.1.98
	Secondary Server	192.168.0.10
	Pathname	
Messages		Informational
Set target units...		☐ Exclude IP Configuration on multiple targets

Buttons: OK, Cancel

Step 2: Configure SIP Parameters

- Select the **Sip** page of the 1204 and set the following:
 - Uncheck **Use DHCP**.
 - Set the *SIP Proxy* field to be the IP address of the destination devices, the Mediatrix 4104 IP address in this case.
 - Use default values for the other entries.

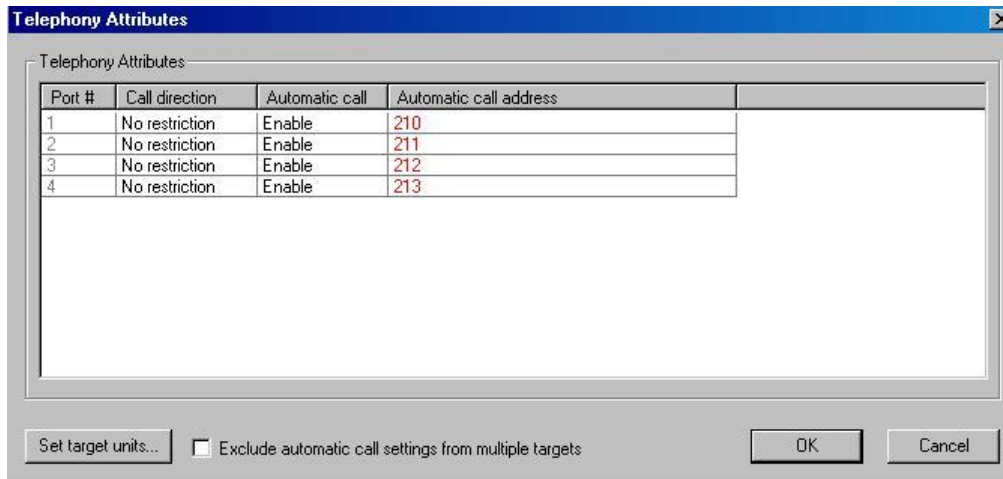
#	User Name	Friendly Name	
1	3330001		
2	3330002		
3	3330003		
4	3330004		

Step 3: Configure Incoming Calls from PSTN Behaviour

- The Mediatrix 1204 exhibits two possible behaviours when an incoming call from the PSTN is received:
 - The Mediatrix 1204 goes off-hook and produces a second dial tone from which the PSTN user can dial a number in IP.
 - The Mediatrix 1204 redirects the call to a specific SIP endpoint upon off-hook.

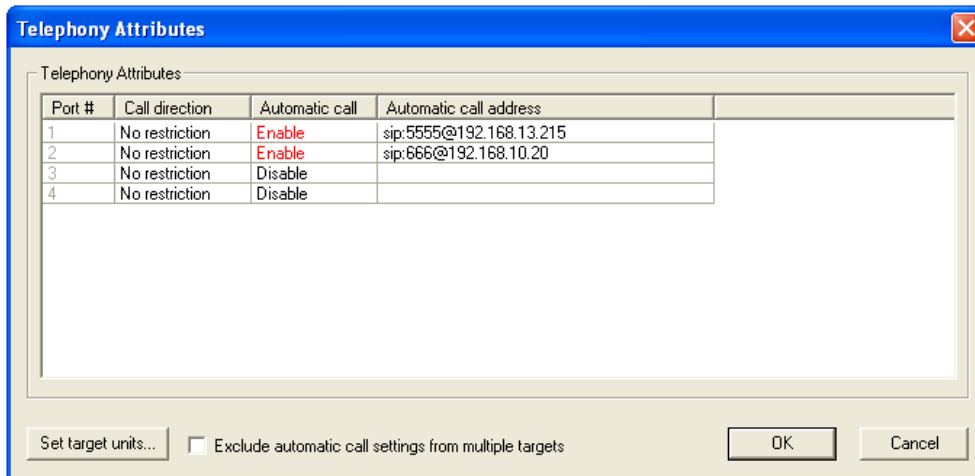
For the purpose of this application, the second behaviour is required. The goal is to tie the PBX analog extension to a specific FXS point located remotely in the IP network.

- Select the **Telephony Attributes** page and set the following:
 - Enable **Automatic call** on all ports.
 - Set an **Automatic call address** for each port.



Port #	Call direction	Automatic call	Automatic call address
1	No restriction	Enable	210
2	No restriction	Enable	211
3	No restriction	Enable	212
4	No restriction	Enable	213

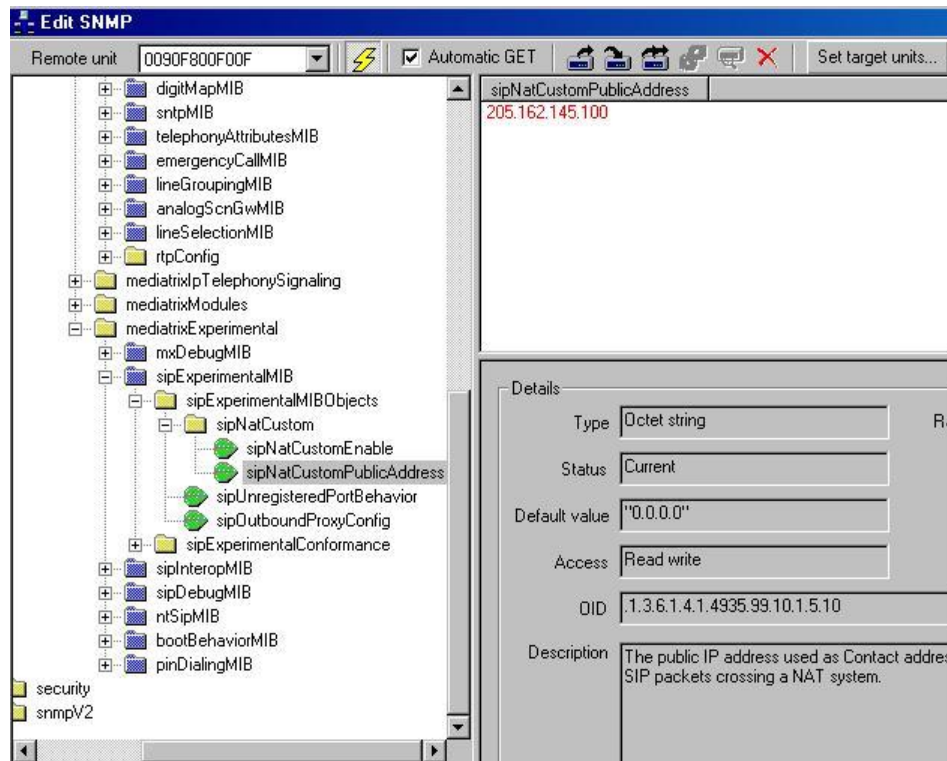
- If you want to map ports of the 1204 to multiple units
 - Set the **Automatic call address** including the IP address of the destination units.



Port #	Call direction	Automatic call	Automatic call address
1	No restriction	Enable	sip:5555@192.168.13.215
2	No restriction	Enable	sip:666@192.168.10.20
3	No restriction	Disable	
4	No restriction	Disable	

Step 4: NAT Traversal Configuration

- You must configure the following parameters for the Mediatrix unit to work behind a NAT/Firewall router:
 - Go to *mediatrix > mediatrixExperimental > sipExperimentalMIB > sipExperimentalMIBObjects > sipNatCustom > sipNatCustomEnable* and set this parameter to **enable**.
 - Go to *mediatrix > mediatrixExperimental > sipExperimentalMIB > sipExperimentalMIBObjects > sipNatCustom > sipNatCustomPublicAddress* and set the parameter to be the public IP address of the NAT/Firewall router where the 1204 is located.



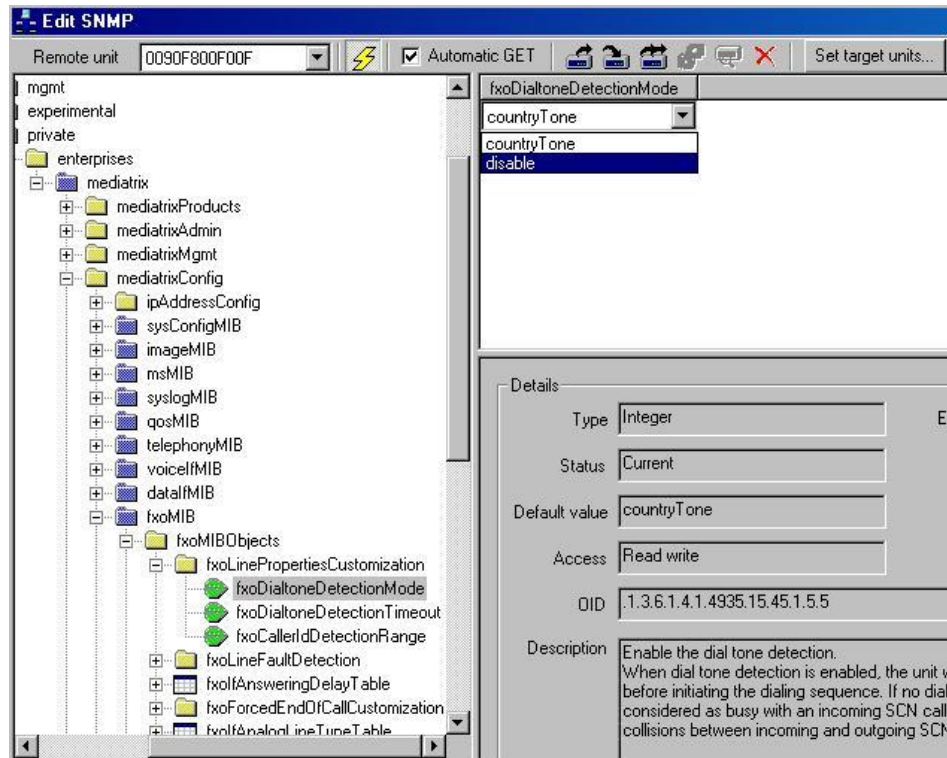
Step 5: Configure Traffic Routing to the Mediatrix Unit

- To allow traffic from the NAT/Firewall router to be routed to the Mediatrix unit, there are two alternatives:
 - Option 1 – Configure the NAT/Firewall router's DMZ (demilitarized zone) to include the Mediatrix unit. Please refer to the documentation provided with your NAT/Firewall router for configuration details.
 - Option 2 – Enable the port forwarding on the NAT/Firewall router. The following ports should be forwarded to the IP address of the Mediatrix unit behind the NAT/Firewall:
 - UDP 5060 for SIP Signaling
 - UDP 5004 to 5010 for Voice RTP streams
 - UDP 161 (optional for management of unit via UMN-SNMP)

Step 6: Configure Dial Tone Detection

The FXO dial tone detection mechanism allows the Mediatrix 1204 to wait for a dial tone on the PSTN side before initiating the dialing sequence. If no dial tone is detected, the line is considered busy with an incoming PSTN call. This mechanism helps to avoid collisions between incoming and outgoing calls.

- Under Edit SNMP, go to *mediatrix > mediatrixConfig > fxoMIB > fxoMIBObjects > fxoLinePropertiesCustomization > fxoDialtoneDetectionMode* and set this parameter to **countryTone**.



Note: Certain PBX manufacturers do not produce the country dial tone on the extension line. If this is the case, make sure that the dial tone detection is disabled, otherwise the Mediatrix 1204 will not output dialed digits.

Step 7: Configure the Answering Delay

- Under Edit SNMP, go to *mediatrix* > *mediatrixConfig* > *fxoMIB* > *fxoMIBObjects* > *fxoIfAnsweringDelayTable*.

The screenshot shows the 'Edit SNMP' window for remote unit 0090F8004BD3. The left pane displays a tree view of the MIB structure, with 'fxoIfAnsweringDelayTable' selected under 'fxoMIBObjects'. The right pane shows a table with three columns: 'fxoPreAnswerDelay', 'fxoAnswerOnCallerIdDetectionEnable', and 'fxoWaitForCalleeToAnswerEr'. The table contains four rows, all with a value of 8000 for the first column and 'enable' for the second column, and 'disable' for the third column. Below the table, a 'Details' pane shows the following information:

Type	Other	Range	
Status	Current		
Default value			
Access	Not accessible		
OID	1.3.6.1.4.1.4935.15.45.1.15		
Description	This table holds the answering delay configuration. Each row in this table configures the answering delay of a line.		

- The parameter *fxoAnswerOnCallerIdDetection* allows the Mediatrix 1204 to go off-hook upon caller ID detection during the ringing of the PBX extension.

Note: If the PBX does not pass the caller ID to the Mediatrix 1204, you can reduce the `<fxoPreAnswerDelay = 1000>` to reduce the time before the Mediatrix 1204 goes off-hook upon ring detection.

Step 8: Configure the Line Selection Parameters (Port-to-Port Mapping)

- Under Edit SNMP, go to *mediatrix > mediatrixConfig > lineSelectionMIB > lineSelectionMIBObjects > lineSelectionIfCustomizationTable*:
 - Set the **lineSelectionEnable** variable to **enable**.
 - Set the **lineSelectionDigitMap** variable to the number corresponding to the Mediatrix 4104 FXS user name.

This feature enables the port-to-port mapping between the Mediatrix 4104 and Mediatrix 1204. Each port of the Mediatrix 1204 corresponds to a specific port of the Mediatrix 4104.

The screenshot shows the 'Edit SNMP' window with the 'Remote unit' set to '0090F800F00F'. The left pane shows a tree view of the SNMP MIB structure, with 'lineSelectionMIBObjects' selected. The right pane displays a table with two columns: 'lineSelectionEnable' and 'lineSelectionDigitMap'. The table contains four rows, all with 'enable' in the first column and digit map values (210, 211, 212, 213) in the second column. Below the table, the 'Details' section shows the following configuration:

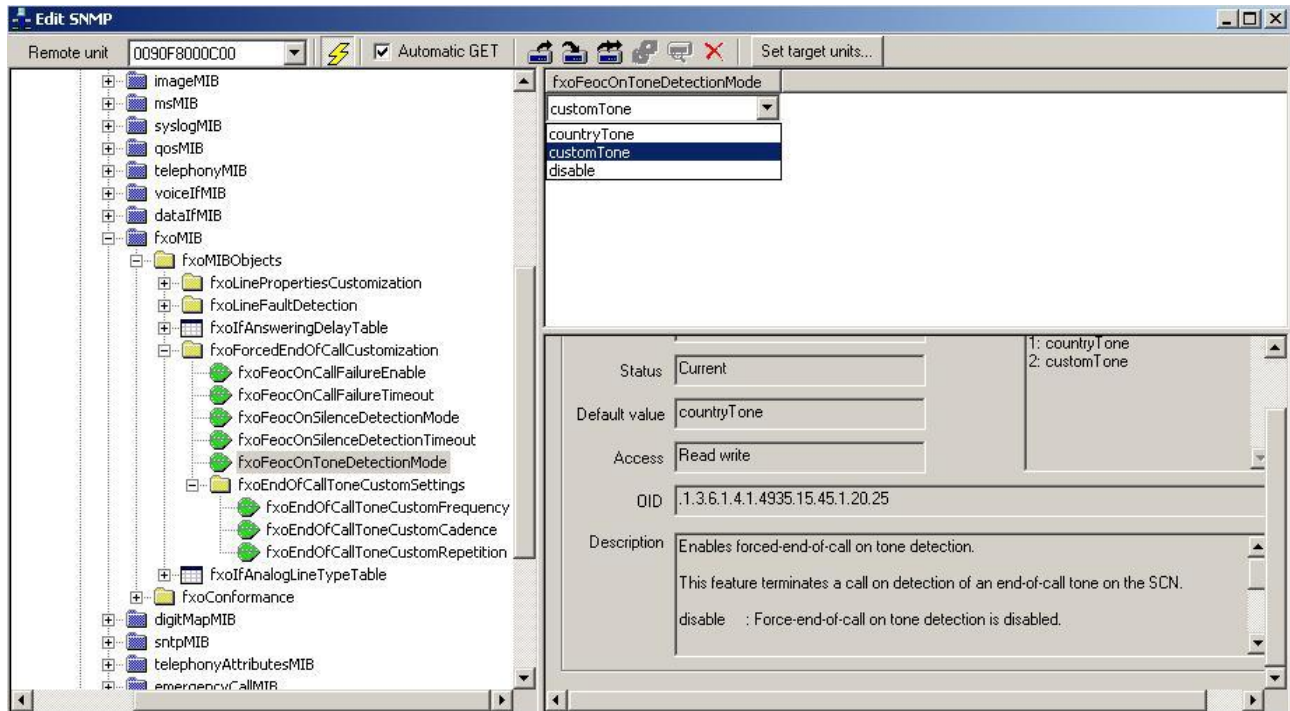
Type	Octet string	Range
Status	Current	
Default value	""	
Access	Read write	
OID	.1.3.6.1.4.1.4935.15.90.1.10.5.10	
Description	Call sources that match this digit map will be permitted. See digitMapMIB for syntax. Note that this digit map will not have any effect unless the feature's status is 'enabled'.	

Step 9: Configure the Far End Disconnect Parameters

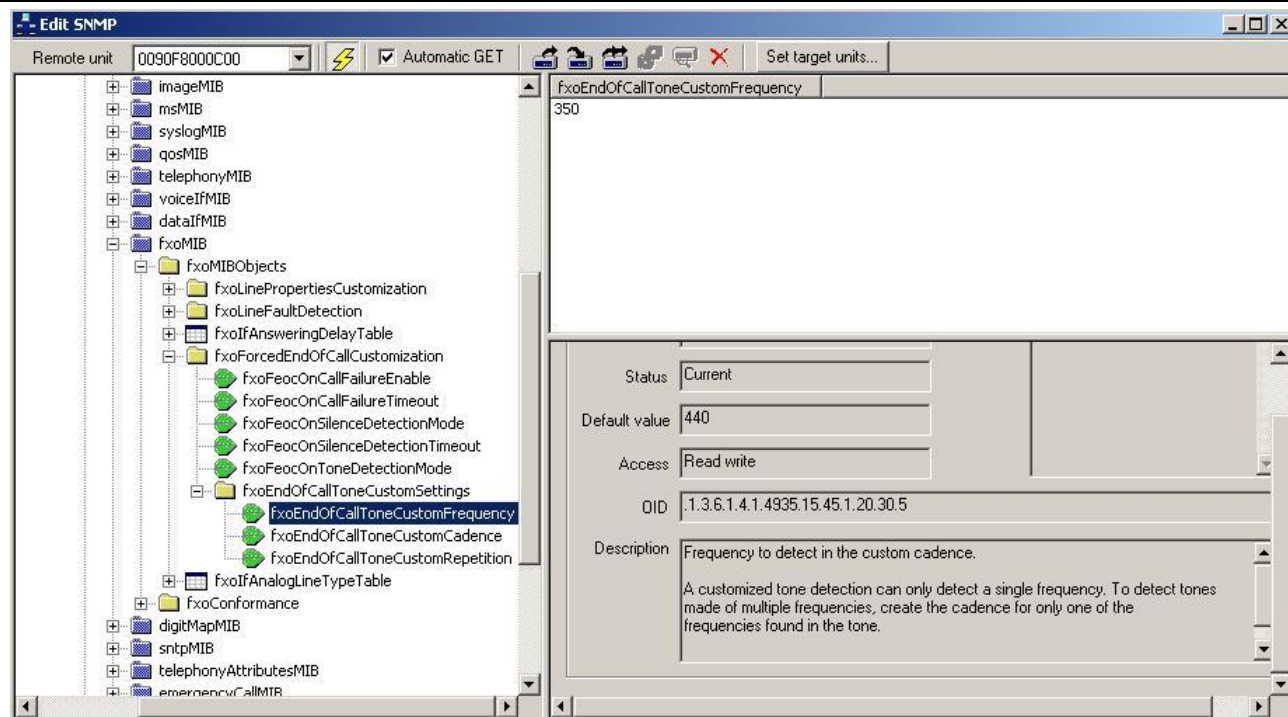
Far end disconnect refers to methods for detecting that a remote party has hung up. The far end disconnect signal is not mandatory in Loop Start and this could create problems. The lack of a far end disconnect signal in Loop Start lines is one of the main reasons why the Ground Start lines were created.

If the far end disconnect signal is not sent to or properly detected by the Mediatrix 1204, the connection will not be released by the unit, thus freezing the FXO port in the off hook state.

- Under Edit SNMP, go to *mediatrix > mediatrixConfig > fxoMIB > fxoMIBObjects > fxoForcedEndOfCallCustomization*.
- Set the **fxoFeocOnToneDetectionMode** variable to **customTone**.



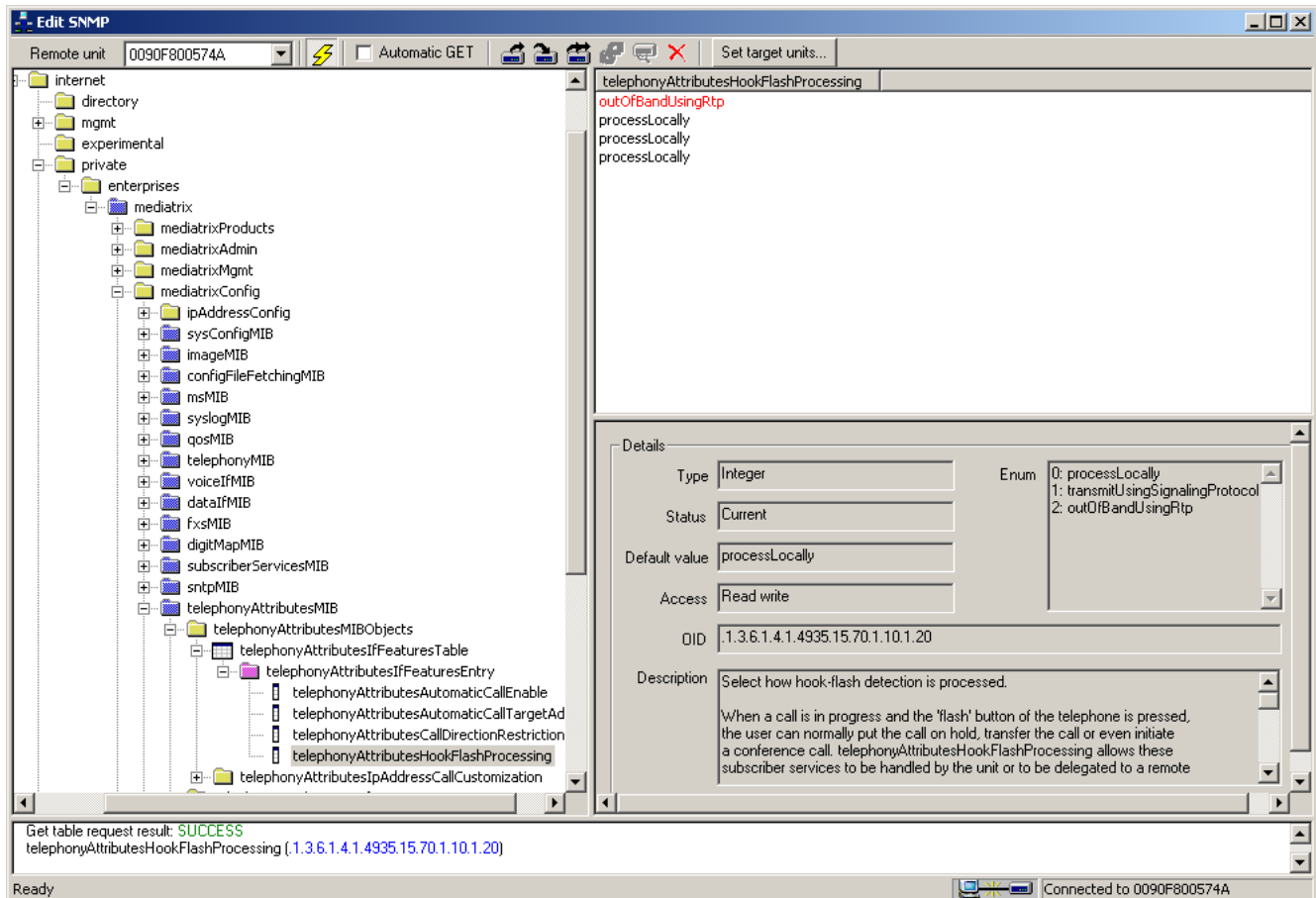
- Set the frequency that the Mediatrix 1204 needs to detect for far end disconnect. This frequency is typically the dial tone frequency (350 Hz for North America) but will vary from one PBX to another. **Please verify with your PBX vendor to determine what tone (exact frequency and cadence) the PBX produces on the extension when the far end is disconnected.**
- Set the **fxoEndOfCallToneCustomFrequency** variable to **350** Hz.
- Set the cadence and repetition required with the *fxoEndOfCallToneCustomCadence* and *fxoEndOfCallToneCustomRepetition* variables.



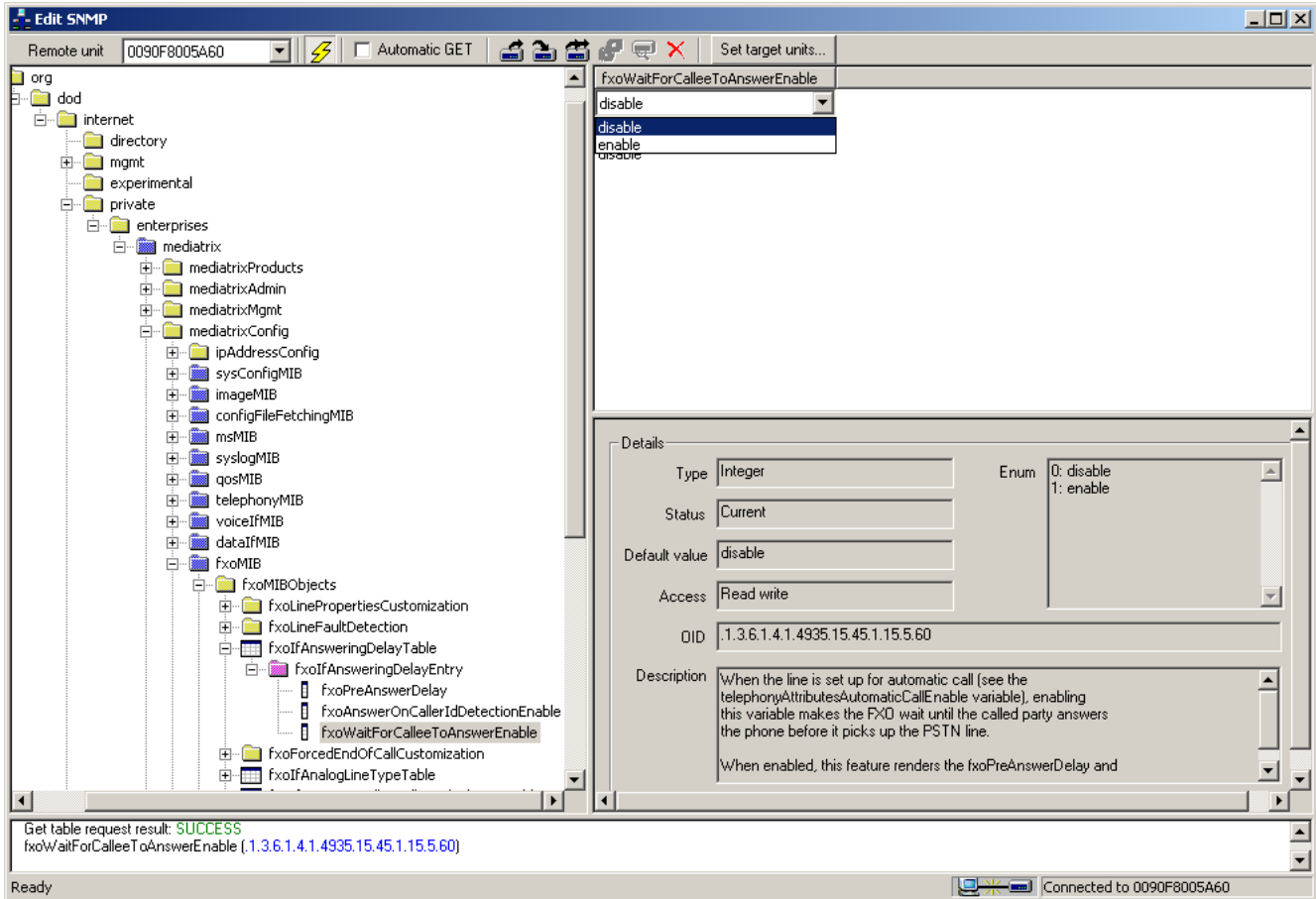
Step 10: Configure the Hook Flash Relay

If your PBX uses the Hook Flash feature, you must configure both the Mediatrix 4104 and Mediatrix 1204 so that they can properly exchange the Hook Flash events. See also the Mediatrix 4104 configuration.

- Go to *mediatrix > mediatrixConfig > telephonyAttributesMIB > telephonyAttributesMIBObjects > telephonyAttributesIfFeaturesTable > telephonyAttributesIfFeaturesEntry > telephonyAttributesHookFlashProcessing* and set this parameter to **outOfBandUsingRtp**.
- The hook-flash events that are received via an RFC 2833 RTP packet is executed.



- Go to *mediatrix > mediatrixConfig > fxoMIB > fxoMIBObjects > fxoIfAnsweringDelayTable > fxoIfAnsweringDelayEntry > fxoWaitForCalleeToAnswerEnable* and set it to **enable**.
- This instructs the Mediatrix 1204 to wait until the called party answers the phone before it picks up the PSTN line.



Step 11: Make a Call

- Connect an analog phone to a port of your Mediatrix 4104 device.
- Pick up the phone, you should hear a dial tone.
- Dial any valid extension number of your PBX. The dialed extension should ring.

Appendix A – Troubleshooting Tips

1. It is strongly recommended to disable the Call Waiting on the extension connected to the Mediatix 1204. Call Waiting tone provided by the PBX to the Mediatix 1204 may trigger the far end disconnect of your Mediatix 1204 and drop the call.
2. Enabling Current Drop on your PBX extensions connected to the Mediatix 1204 will improve the release line.
3. *No dial tone on the Mediatix 4104 when picking up the phone.* Verify the configuration of your Mediatix 4104 or your hardware connection on the Mediatix 4104. The problem would reside on the Mediatix 4104's since it provides dial tone to its ports.
4. *Cannot establish calls.* Can you ping the Mediatix 1204 from the Mediatix 4104 and vice-versa?
5. *Some ports of the Mediatix 1204 cannot seize the PBX line.*
 - a. PBX calibration may vary from one port to another.
 - b. Disable the Stutter dial tone played by the PBX to announce a voicemail message to the port connected on the Mediatix 1204.