Dialogic® 4000 Media Gateway Series Integration Note

Grandstream Handy Tone HT-502

August 2009 05-2690-001

www.dialogic.com

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1. Scope

This document is intended to demonstrate a typical installation and configuration of a Dialogic® 4000 Media Gateway Series with an Analog VoIP Phone adapter. In this case a Grandstream Handy Tone HT-502.

2. Configuration Details

Listed below are the specific details of the Gateway and Phone adapter used in the testing to construct the following documentation.

2.1 Gateway

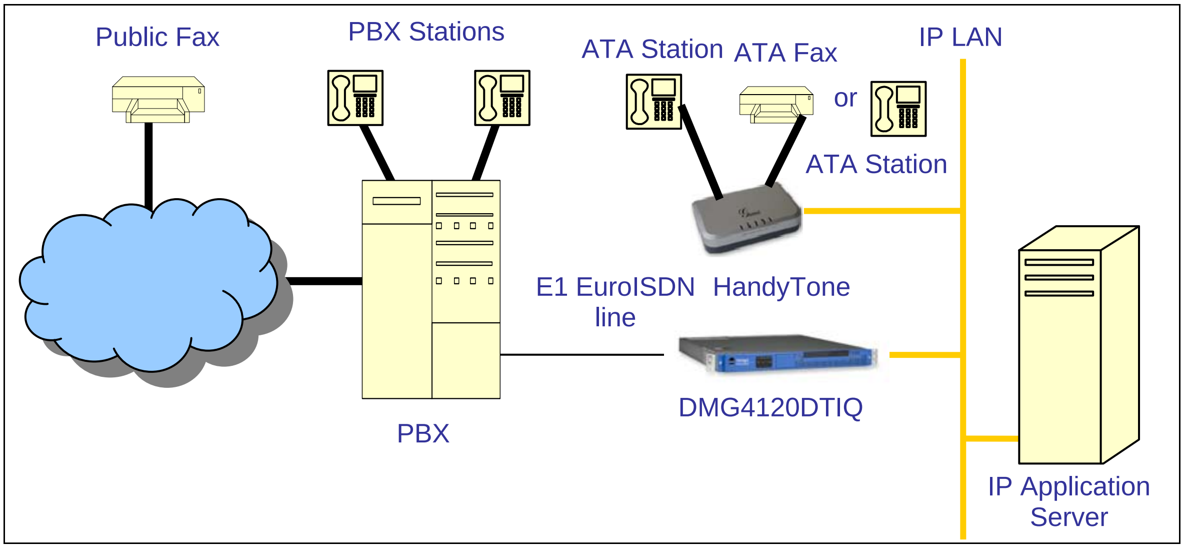
Gateway Model	Dialogic 4000 Media Gateway 4120DTIQ
Software Version(s)	Dialogic® Diva® System Release software version 8.5WIN SU3
Protocol	E1 EuroISDN
PBX/Integration	EuroISDN

2.2 VoIP Phone Adapter

Vendor	Grandstream
Model	Handy Tone HT-502 V1.1C
Software	Program-- 1.0.1.35 Bootloader-- 1.0.0.9 Core-- 1.0.0.34 Base-- 1.0.0.84
Information Link	http://www.grandstream.com/NEWSITE/ht502.html

2.3 System Diagram

The diagram below details the setup used in the testing and creation of the technical document.



3. Prerequisites

3.1 PBX Prerequisites

This document assumes that PBX programming or Direct PSTN connectivity has been established by using other published PBX specific configuration guides.

3.2 Gateway Prerequisites

The gateway used in this configuration guide is from DMG4120DTIQ series, but this configuration can apply to other Dialogic® 4000 Media Gateway Series.

4. Summary of Limitations

No limitations noted as of the last update to this document.

5. Network Configuration

This configuration guide assumes that the following IP addresses and subnet masks were assigned:

<u>ATA Ethernet:</u>
IP: 192.168.2.1 (Device Default) – Used for configuration
Subnet: 255.255.255.0

<u>ATA WAN:</u>
IP: 192.168.185.175
Subnet: 255.255.255.0

<u>Gateway:</u>
IP: 192.168.185.149
Subnet: 255.255.255.0

6. Grandstream Handy Tone Setup

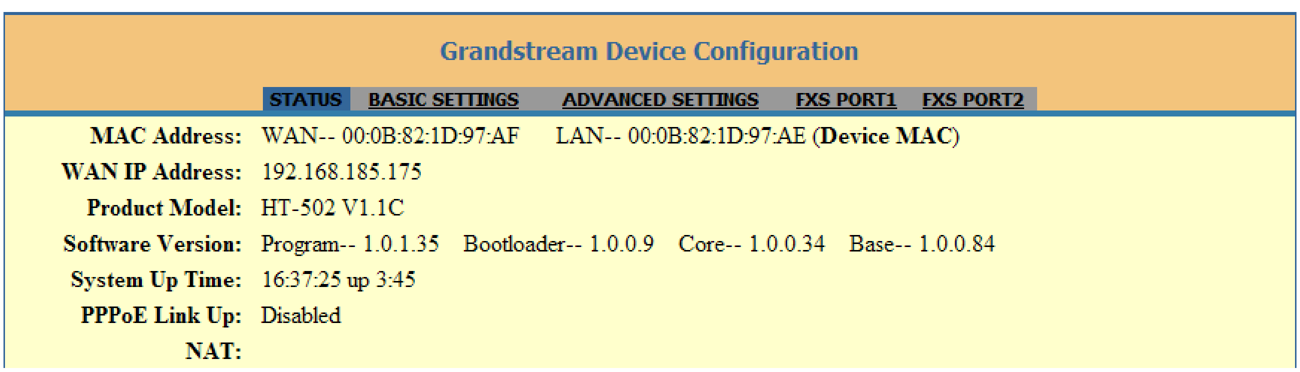
The Grandstream Handy Tone's Quick Install guide is posted at:
http://www.grandstream.com/NEWSITE/pdf/HandyToneInstallGuide.pdf

Alternatively, you can follow the steps below:

6.1 Configuration Via Web Access

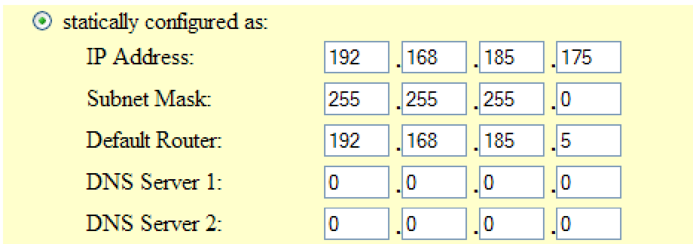
1. Connect your PC to the Handy Tone's LAN port.
2. Set your PC's IP address to 192.168.2.2.
3. Type 192.168.2.1 (The Handy Tone's default LAN port IP Address) in a web browser and hit Enter.

4. Login using the default password "admin".

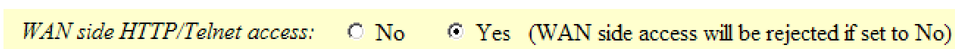


6.2 Handy Tone WAN Port Configuration

1. Click the **Basic Settings** Tab
2. Select the **statically configured as:** radio button
   - Set **IP Address** to: 192.168.185.175
   - Set **Subnet Mask** to: 255.255.255.0



3. Set **WAN side HTTP/Telnet access:** to Yes (If device WAN configuration access is desired.)



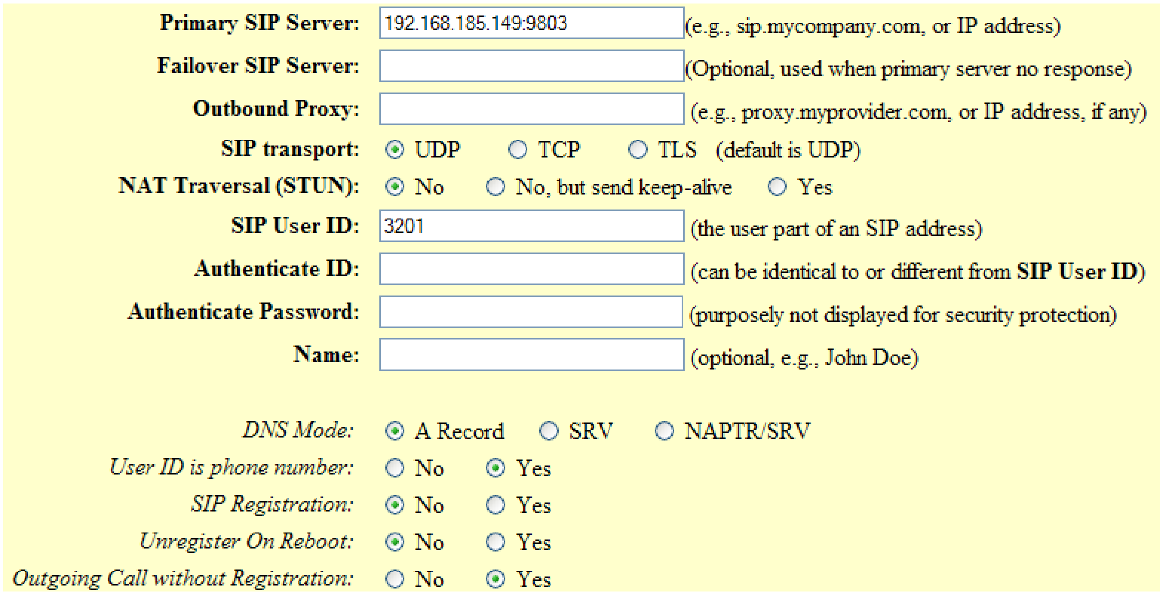
4. Click **Update**.
5. Click **Reboot** for the changes to take affect.

6.3 Telephone Line Configuration

After logging back in:

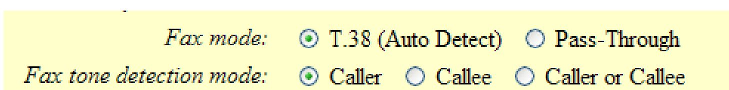
<u>FXS Port 1:</u>

1. Click the **FXS Port 1** tab.
2. Complete the following fields:
   - **Primary SIP Server:** 192.168.185.149:9803 (DMG IP Address and port)
   - **SIP transport:** UDP
   - **SIP User ID:** 3201
   - **User ID is phone number:** Yes
   - **SIP Registration:** No



Primary SIP Server:	192.168.185.149:9803	(e.g., sip.mycompany.com, or IP address)
Failover SIP Server:		(Optional, used when primary server no response)
Outbound Proxy:		(e.g., proxy.myprovider.com, or IP address, if any)
SIP transport:	◉ UDP ○ TCP ○ TLS	(default is UDP)
NAT Traversal (STUN):	◉ No ○ No, but send keep-alive ○ Yes	
SIP User ID:	3201	(the user part of an SIP address)
Authenticate ID:		(can be identical to or different from **SIP User ID**)
Authenticate Password:		(purposely not displayed for security protection)
Name:		(optional, e.g., John Doe)
DNS Mode:	◉ A Record ○ SRV ○ NAPTR/SRV	
User ID is phone number:	○ No ◉ Yes	
SIP Registration:	◉ No ○ Yes	
Unregister On Reboot:	◉ No ○ Yes	
Outgoing Call without Registration:	○ No ◉ Yes	

- **Fax Mode:** T.38 (Auto Detect)
- **Fax tone detection mode:** Caller

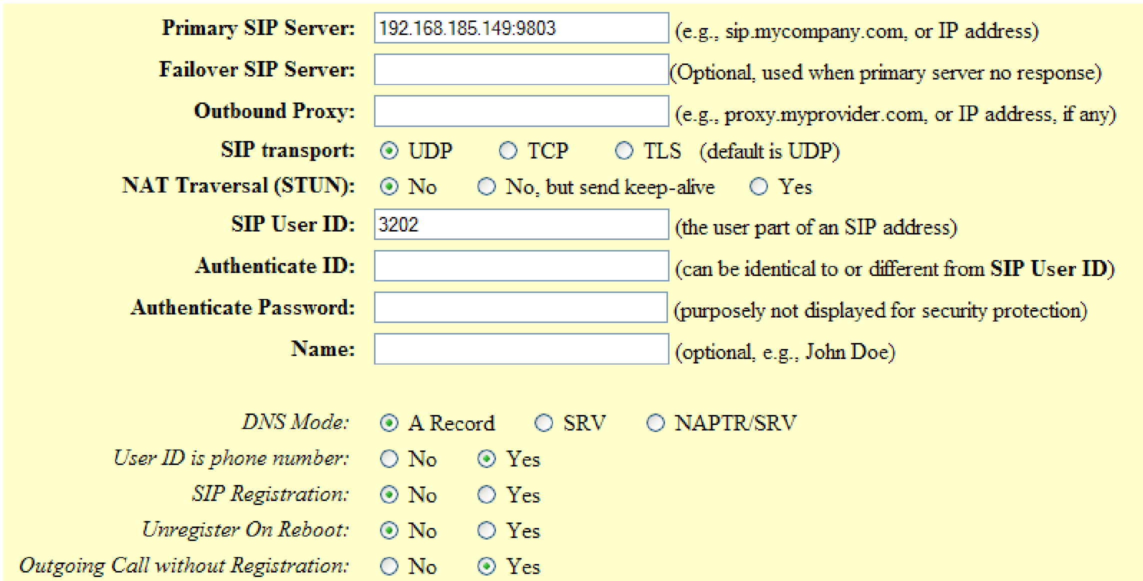


Fax mode:	◉ T.38 (Auto Detect) ○ Pass-Through
Fax tone detection mode:	◉ Caller ○ Callee ○ Caller or Callee

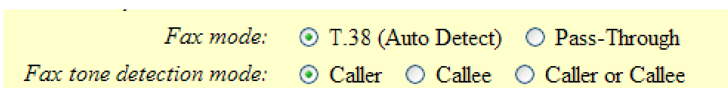
3. Click **Update**.

FXS Port 2:

1. Click the **FXS Port 2** tab.
2. Complete the following fields:
   - **Primary SIP Server:** 192.168.185.149:9803 (DMG IP Address and port)
   - **SIP transport:** UDP
   - **SIP User ID:** 3202
   - **User ID is phone number:** Yes
   - **SIP Registration:** No



- **Fax Mode:** T.38 (Auto Detect)
- **Fax tone detection mode:** Caller



3. Click **Update**.
4. Click **Reboot** for the changes to take affect.

7. Gateway Setup Notes

The following are the necessary steps to set up the gateway:

1. Connection
2. Parameter configuration
3. Dialogic® Diva® SIPcontrol™ Software configuration
   - Network Interface configuration
   - SIP Peer configuration
   - Routing configuration

7.1 Connection

Connect a mouse and keyboard to the gateway or connect using RemoteDesktop. Login (default username/password is Dialogic/Dialogic)

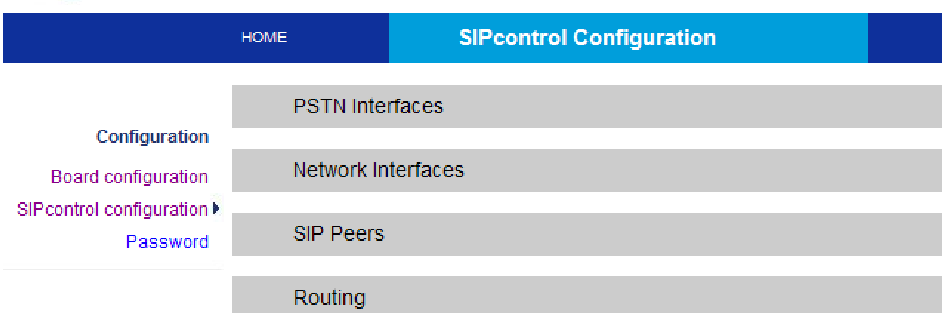
7.2 Parameter Configuration

Follow other published configuration guides depending on your IP application. For Microsoft® Office Communications Server (OCS) 2007, follow this link:
http://www.dialogic.com/microsoftuc/pbx_integration.htm
This document assumes that the connection to the PBX or PSTN has been setup and the Diva SIPcontrol software has been configured to route calls to and from your IP application server, e.g., Microsoft® OCS 2007.

7.3 Dialogic® Diva® SIPcontrol™ Software Configuration

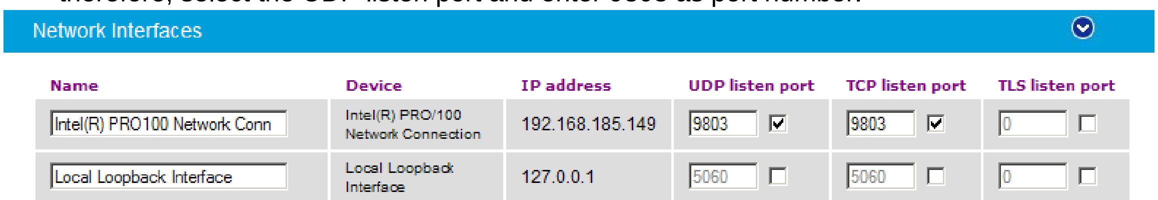
In this example, we assume that the two Grandstream® ATA extension numbers are 3201 and 3202. Their corresponding IP addresses on the ATA side are: 192.168.185.175:5060 and 192.168.185.175:5062.

1. Click **Start** > **Programs** > **Dialogic Diva** > **SIPcontrol Configuration**.
2. In the web interface, select **SIPcontrol configuration**.



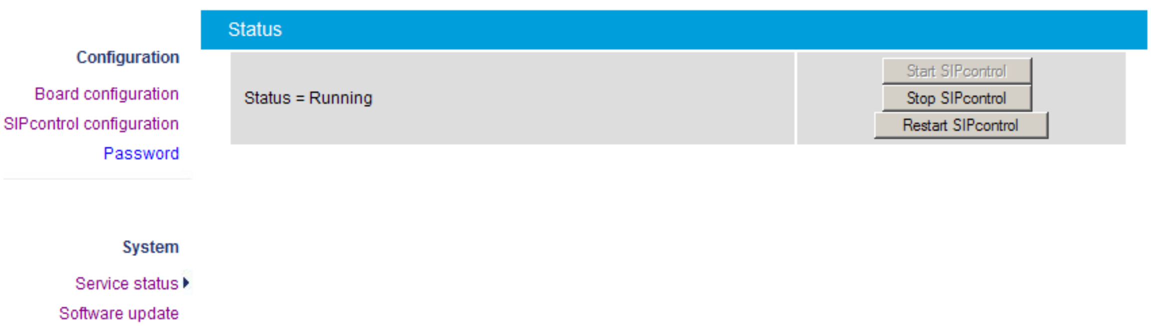
7.3.1 Network Interfaces

1. We will be using UDP for the connection from the ATA to the Diva SIPcontrol software; therefore, select the UDP listen port and enter 9803 as port number.



2. Click the **Save** button at the bottom of the Diva SIPcontrol software web interface to save the changes.

3. In the web interface under **System**, select **Service status** and click **Restart SIPcontrol** to implement the change to the Network interface. *Note that this will disconnect any active call.*



4. Click **SIPcontrol configuration** to configure the SIP peers.

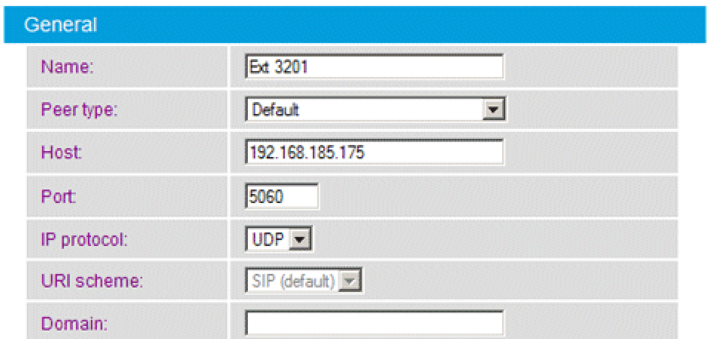
7.3.2 SIP Peer Configuration

When inbound TDM calls are targeted to ATA extensions 3201 or 3202, the gateway will use the routing engine rules in order to route these calls to the correct SIP peer destination.
This gateway has an existing SIP Peer defined for connecting Microsoft® OCS 2007:

Now add two new SIP Peers; one for each port on the ATA.

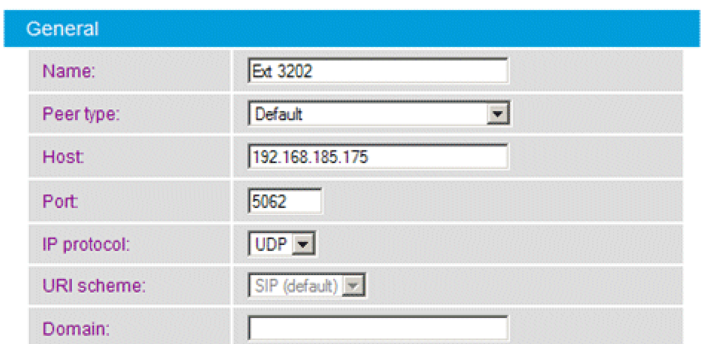
SIP Peer #1

Add a SIP Peer for calls to ATA Port 1:
Name: Enter a descriptive name, e.g., Ext 3201.
Peer type: Select Default.
Host: Enter 192.168.185.175 (IP address of ATA).
Port: Enter 5060 (Port 1 of ATA).
IP protocol: Select UDP.



SIP Peer #2

And similarly add a SIP Peer for calls to ATA Port 2. *Note that the port is 5062*.



Now the SIP Peers window should display three SIP Peers as shown below:

SIP Peers

Name	Default SIP to PSTN Peer	Host	Port	IP protocol	Display name to	Dialplan	Enabled
OCS local	☐	192.168.185.149	5060	TCP		Maidenhead	☑
Ext 3201	☐	192.168.185.175	5060	UDP		none	☑
Ext 3202	☐	192.168.185.175	5062	UDP		none	☑

Add

Click the **Save** button at the bottom of the Diva SIPcontrol software web interface to save the changes.

7.3.3 Routing table

You should already have two routes, one for PSTN to SIP calls that routes calls from the PSTN to your IP server/Microsoft® OCS 2007 and one for the reverse direction, SIP to PSTN that routes calls from your IP server/ Microsoft® OCS 207 to the PSTN connection.

Routing

Name	Sources	Destinations	Address map	Enabled
OCS to PSTN	OCS local	Controller1 (Master)	none	☑
PSTN to OCS	Controller1	OCS local (Master)	none	☑

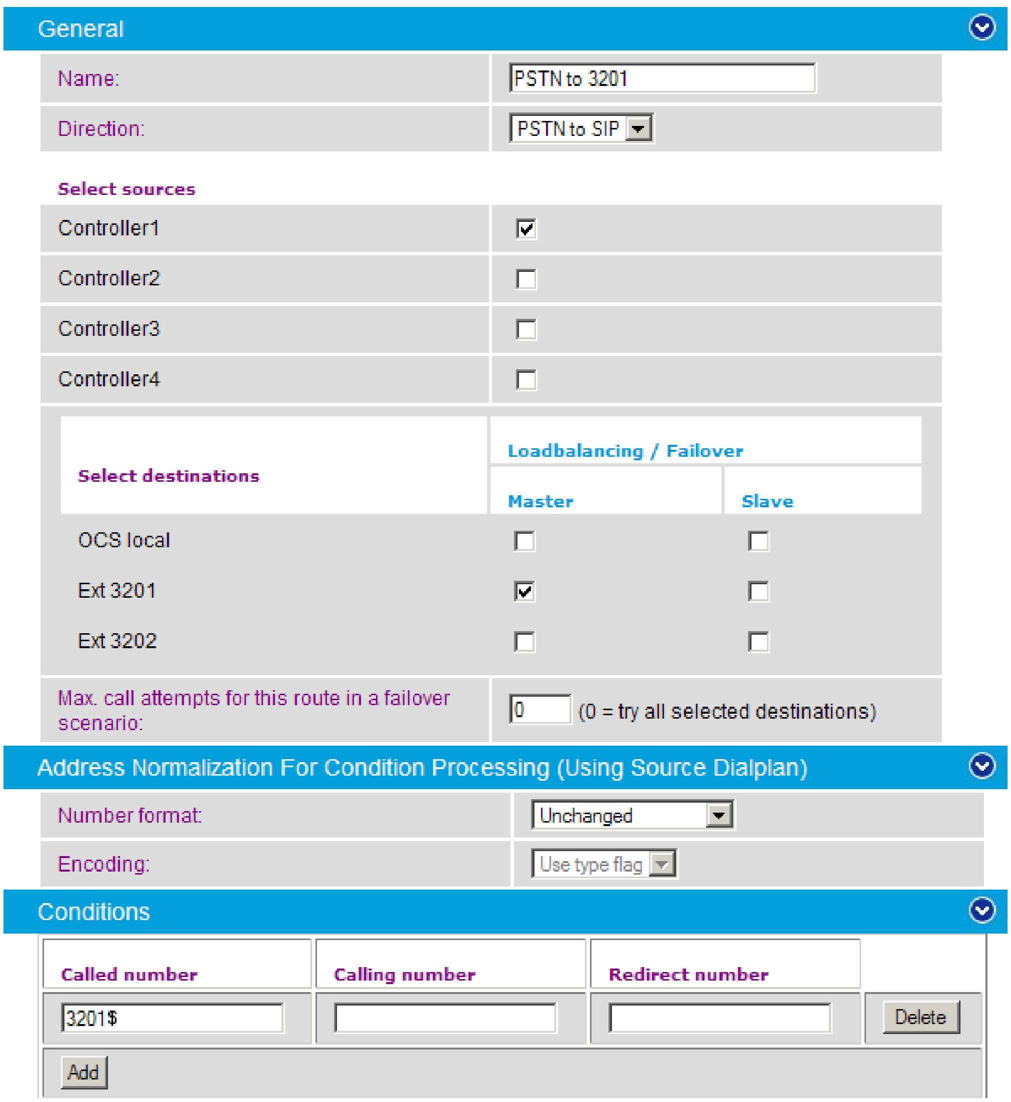
Add

Now add two new routes in order to route TDM calls targeted for the two ATA extensions (3201 & 3202) to their respective SIP peers and also modify the existing Microsoft® OCS 2007 to PSTN route to support calls from the ATA to the PSTN.

Routing Rule #1

Add a new route from PSTN to SIP that that matches a called number of 3201 and routes it to the SIP Peer for ATA Port 1 created in the previous step.

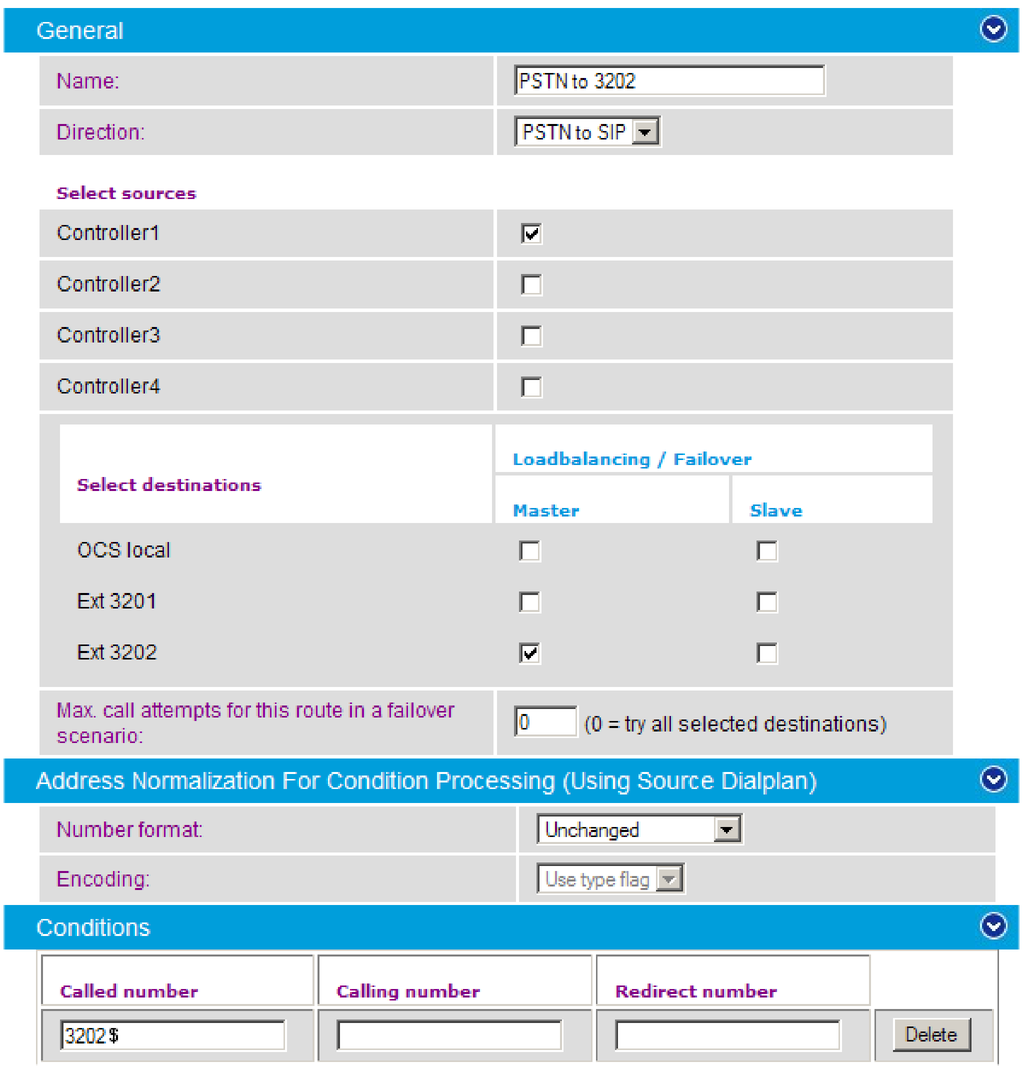
Name: Enter a descriptive name.
Direction: PSTN to SIP
Select sources: Select the controllers (gateway PSTN ports) that will be used.
Select destinations: Select the SIP peer destination for ATA port 1 that was created above (Ext 3201).
Conditions: Enter 3201$ in the Called number, this will ensure this route is used for calls to 3201 only.



Routing Rule #2

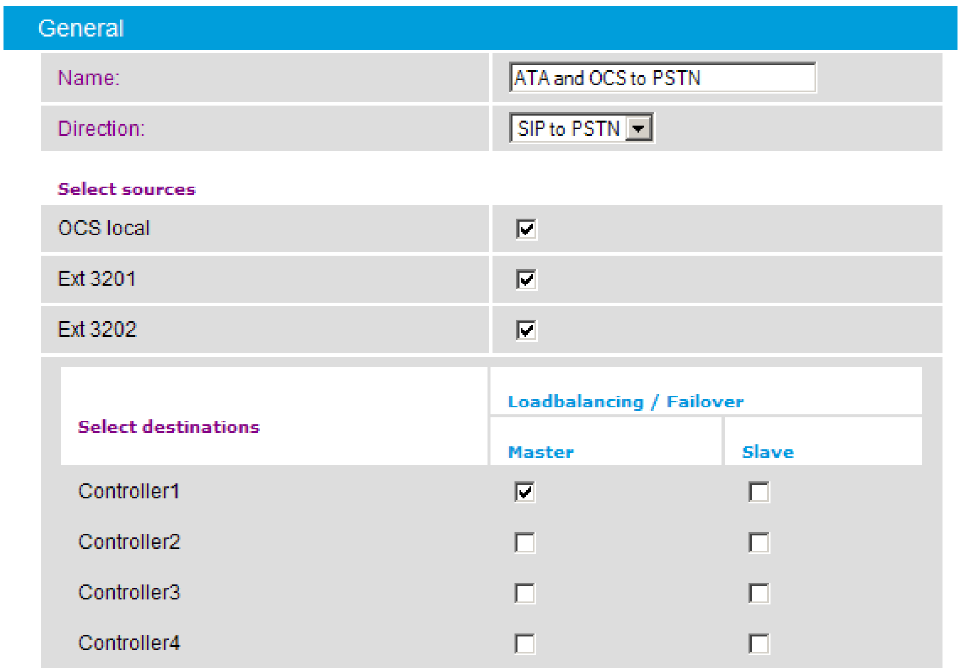
Add a new route from the PSTN to SIP that matches a called number of 3202 and routes it to the SIP peer for ATA Port 2 created in the previous step.

Name: Enter a descriptive name.
Direction: PSTN to SIP
Select sources: Select the controllers (gateway PSTN ports) that will be used.
Select destinations: Select the SIP peer destination for ATA port 2 that was created above (Ext 3202).
Conditions: Enter 3202$ in the Called number, this will ensure that this route is used for calls to 3202 only.

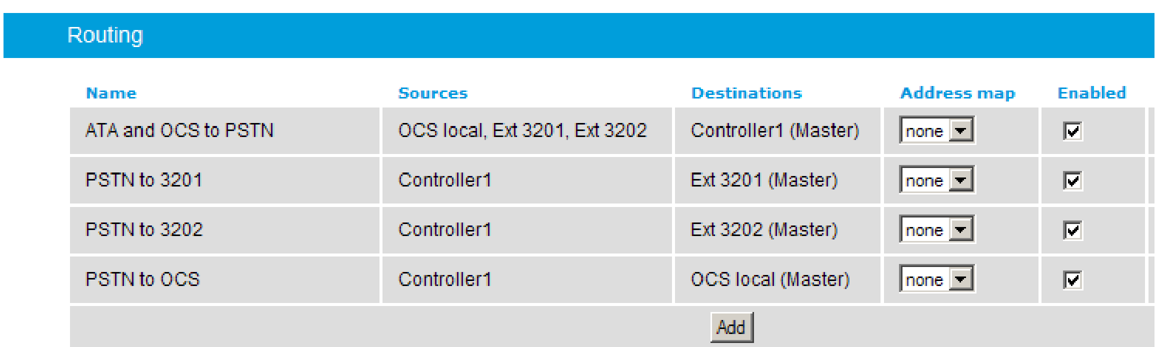


Routing Rule #3

Finally, we need to modify the existing Microsoft® OCS 2007 to PSTN rule to enable calls from the ATA to the PSTN.
In **Select sources** select the two SIP Peers you added for the ATA in the Select sources fields and change the Route name to reflect the new role.



Now the Routing rules should look like this.
Note: The existing PSTN to Microsoft® OCS 2007 rule MUST be moved using the up and down arrows to be below the PSTN to 3201 and PSTN to 3202 rules to ensure that the routes to 3201 and 3202 are processed first.



Click the **Save** button at the bottom of the Diva SIPcontrol software web interface to save the changes.

8. Testing Validation Matrix

The table below shows various test scenarios that were executed in this configuration and their results:
- ATA Port 1 is connected to an analog telephone
- ATA Port 2 is connected to an analog telephone

Note: The current version of the Dialogic® 4000 Media Gateway does not support SIP to SIP routing; therefore, all calls from one ATA port to another ATA port would be routed to the PBX and back to the gateway resulting in two PSTN channels being used.

Test Number	Call Scenario Description	Notes
Inbound TDM calls		
1	Inbound call to PBX destination 3201	Pass
2	Inbound call to PBX destination 3202	Pass
Outbound to VOIP (From one ATA port to the other)		
1	Outbound from 3201 to 3202	Pass (Uses two gateway channels)
2	Outbound from 3202 to 3201	Pass (Uses two gateway channels)
Outbound to TDM		
1	Outbound from 3201 to PBX Extension	Pass
2	Outbound from 3202 to PBX Extension	Pass
3	Outbound from 3201 to External Number	Pass
4	Outbound from 3202 to External Number	Pass
Blind Call Transfer		
1	Inbound to 3201, 3201 transfer to 3202	Pass (Uses three gateway channels)
2	Inbound to 3201. 3201 transfers to	Pass (Uses two gateway channels)

	PBX Extension	
3	Inbound to 3201. 3201 transfers to External Number	Pass (Uses two gateway channels)
Attended Call Transfer		
1	Inbound to 3201, 3201 transfers to 3202	Pass (Uses three gateway channels)
2	Inbound to 3201. 3201 transfers to PBX Extension	Pass (Uses two gateway channels)
3	Inbound to 3201. 3201 transfers to External Number	Pass (Uses two gateway channels)
Bellcore Style 3-Way Conference		
1	Inbound to 3201, 3201 initiates a 3-way conference to 3202, 3202 answers, 3201 completes the conference.	Pass (Uses four gateway channels)
2	Inbound to 3201, 3201 initiates a 3-way conference to a PBX #, PBX # answers, 3201 completes the conference.	Pass (Uses two gateway channels)
3	Inbound to 3201, 3201 initiates a 3-way conference to an external #, External # answers, 3201 completes the conference.	Pass (Uses two gateway channels)
Fax		
1	Inbound to 3201 and 3202 Ricoh 5800	Pass
2	Outbound from 3201 and 3202 Ricoh 5800	Pass

Notes:

Blind Transfer
Assume that call caller A and B are in conversation. Caller A wants to *Blind Transfer caller* B to caller C:
1. Caller A presses **FLASH** on the analog phone to hear the dial tone.
2. Caller A dials ***87**, then dials caller C's number, and then # (or wait for 4 seconds)
3. Caller A will hear the confirm tone. Then, A can hang up.

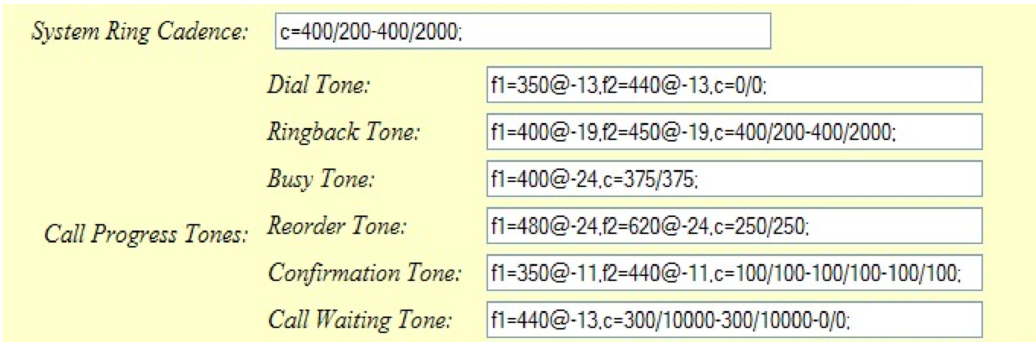
Attended Transfer
Assume that caller A and B are in conversation. Caller A wants to *Attend Transfer* B to C:
1. Caller A presses **FLASH** on the analog phone for dial tone.
2. Caller A then dials caller C's number followed by # (or wait for 4 seconds).
3. If caller C answers the call, caller A and caller C are in conversation. Then A can hang up to complete transfer.
4. If caller C does not answer the call, caller A can press **FLASH** to resume call with Caller B.

Bellcore Style 3-way Conferencing
Assume that call party A and B are in conversation. Caller A (HT502) wants to bring third caller C into the conference:
1. Caller A presses **FLASH** (on the analog phone, or Hook Flash for old model phones) to get a dial tone.
2. Caller A dials C's number then # (or wait for 4 seconds).
3. If caller C answers the call, then caller A presses **FLASH** to bring callers B and C in the conference.
4. If caller C does not answer the call, caller A can press **FLASH** to talk to caller B.
5. If caller A presses **FLASH** during the conference, caller C will be dropped out.
6. If caller A hangs up, the conference will be terminated for all three parties when configuration "Transfer on Conference Hangup" is set to "No". If the configuration is set to "Yes", caller A will transfer caller B to caller C so that B and C can continue the conversation.

For more details, consult the Grandstream HT-502 manual found here:
http://www.grandstream.com/NEWSITE/ht502.html

9. Configuring Analog Tones (Optional)

In some regions, it is useful to modify the default tones generated by the HT-502 so that users will hear their regional tones such as busy, ringback, etc.
The following graphic shows the settings for the UK, the system ring cadence, ringback, and busy tone were all modified as shown on the HT-502 Advanced settings screen:



System Ring Cadence:	c=400/200-400/2000;	
Call Progress Tones:	*Dial Tone:*	f1=350@-13,f2=440@-13,c=0/0;
	Ringback Tone:	f1=400@-19,f2=450@-19,c=400/200-400/2000;
	Busy Tone:	f1=400@-24,c=375/375;
	Reorder Tone:	f1=480@-24,f2=620@-24,c=250/250;
	Confirmation Tone:	f1=350@-11,f2=440@-11,c=100/100-100/100-100/100;
	Call Waiting Tone:	f1=440@-13,c=300/10000-300/10000-0/0;

10. Troubleshooting

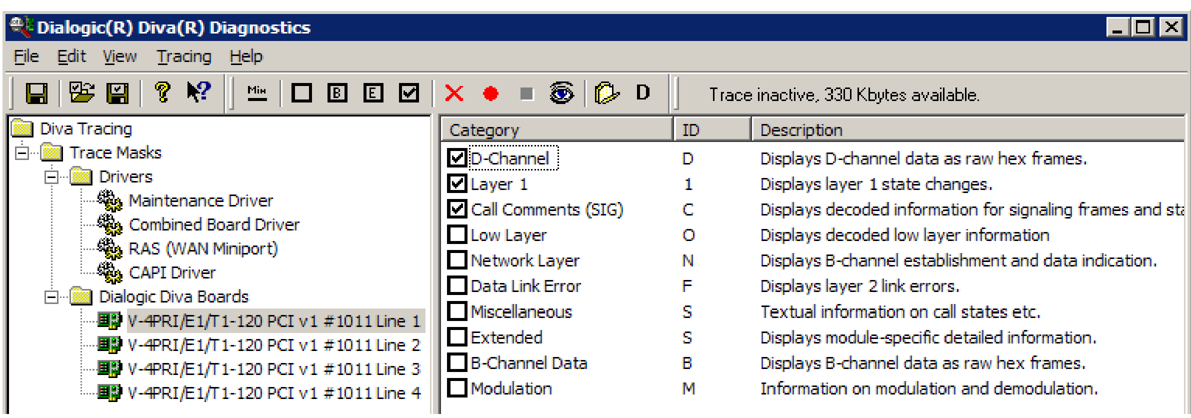
10.1 Important Debugging Tools

Ethereal/Wireshark – Used to capture, view, and analyze the network traffic from HT-502 to Dialogic® 4000 Media Gateway.
Dialogic® Diva® Diagnostics tool – Captures and views the ISDN and SIP traffic from the Dialogic® 4000 Media Gateway to the PSTN and SIP.

10.2 Using the Dialogic® Diva® Diagnostics Tool

1. Before using the Diva Diagnostics tool, you need to enable the tracing of SIPcontrol messages into the Diva Diagnostic tool.
   - Open the Diva SIPcontrol software web interface.
   - Under **SIPcontrol configuration** > **System settings** set the **Debug level** to **Extended**.
   - Click **Save** to save the changes.

2. Start the Diva Diagnostics tool by clicking on **Start** > **Programs** > **Dialogic Diva** > **Diagnostics**.



3. Click one line of the Dialogic® Diva® Media Board on the left hand pane, and click on the toolbar to activate Basic ISDN tracing. Repeat this for all of the lines of your Diva Media Boards.

4. Click the CAPI driver in the left hand pane, and click on the toolbar to activate the Basic CAPI tracing.

5. Click on the toolbar to start tracing. Click on the toolbar to zero the trace. Now reproduce the problem or scenario you are trying to capture.

6. To stop tracing, click on the toolbar. Click on the toolbar to open the trace into Notepad.

Basic notations for reading the trace

- SIG-R: Received ISDN message
- SIG-X: Transmitted ISDN message
- SIPR: Received SIP message
- SIPX: Transmitted SIP message

Sample transmitted ISDN Call Setup:

```
SIG-X(032) 08 02 00 BF 05 04 03 80 90 A3 18 03 A1 83 9F 1E 02 80 83 6C 06 00 80 33
32 30 32 70 03 80 34 31
                     Q.931  CR00bf SETUP
                            Bearer Capability 80 90 a3
                            Channel Id a1 83 9f
                            Progress Indicator 80 83
                            Calling Party Number 00 80 '3202'
                            Called Party Number 80 '41'
```

Sample received SIP INVITE:

```
SIPR (1153 byte) Remote 192.168.185.175:5062/UDP Local 192.168.185.149:9803/UDP
>INVITE sip:41@192.168.185.149:9803;user=phone SIP/2.0
>Via: SIP/2.0/UDP 192.168.185.175:5062;branch=z9hG4bK140096430;rport
>From: <sip:3202@192.168.185.149:9803;user=phone>;tag=1846592258
>To: <sip:41@192.168.185.149:9803;user=phone>
>Call-ID: 1537862311-5062-2@192.168.185.175
>CSeq: 20 INVITE
>Contact: <sip:3202@192.168.185.175:5062;user=phone>
>Max-Forwards: 70
>User-Agent: Grandstream HT-502  V1.1C 1.0.1.35
>Privacy: none
>P-Asserted-Identity: <sip:3202@192.168.185.149:9803;user=phone>
>Supported: replaces, path, timer
>Allow: INVITE, ACK, OPTIONS, CANCEL, BYE, SUBSCRIBE, NOTIFY, INFO, REFER, UPDATE
>Content-Type: application/sdp
>Accept: application/sdp, application/dtmf-relay
>Content-Length:   437
>
>v=0
>o=3202 8002 8000 IN IP4 192.168.185.175
>s=SIP Call
>c=IN IP4 192.168.185.175
>t=0 0
>m=audio 5012 RTP/AVP 8 0 4 18 2 97 103 102 101
>a=sendrecv
```

```
>a=rtpmap:8 PCMA/8000
>a=ptime:20
>a=rtpmap:0 PCMU/8000
>a=rtpmap:4 G723/8000
>a=rtpmap:18 G729/8000
>a=rtpmap:2 G726-32/8000
>a=rtpmap:97 iLBC/8000
>a=fmtp:97 mode=20
>a=rtpmap:103 AAL2-G726-40/8000
>a=rtpmap:102 G729E/8000
>a=rtpmap:101 telephone-event/8000
>a=fmtp:101 0-16,32-36,54
```