



Dialogic[®] 4000 Media Gateway Series Integration Note

Generic PBX

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1. Scope

This document is intended to detail a typical installation and configuration of a gateway from the Dialogic® 4000 Media Gateway Series if used to interface between a PBX or the PSTN and the Microsoft® Office Communications Server (OCS) 2007 application. Note that the term “gateway” is used herein to generically refer to a gateway from the Dialogic 4000 Media Gateway Series.

2. Configuration Details

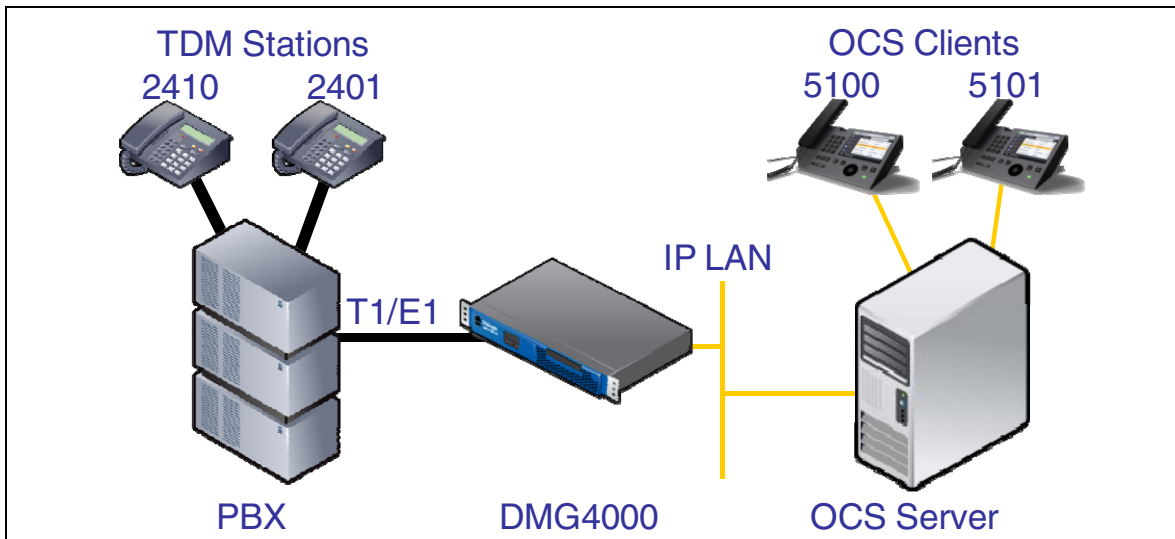
Listed below are the specific details of the gateway used during testing to construct the following documentation.

2.1 Gateway

Gateway Series	Dialogic® 4000 Media Gateway Series
Software Version(s)	Dialogic® Diva® System Release software version 8.3.2 build 459 (formerly called Diva® Server software) Dialogic® Diva® SIPcontrol™ Software version 1.6 build 46 (DSSIPControl.msi)
Protocol	T1 (NI2, QSIG, 5ESS) or E1 (QSIG, EuroISDN)

2.2 System Diagram

The diagram below details the setup used in the testing and creation of this technical document. In the diagram, the abbreviation “DMG4000” denotes a gateway from the Dialogic® 4000 Media Gateway Series. The term “OCS” stands for Microsoft® Office Communications Server (OCS) 2007 both in the diagram and throughout the document.



3. Prerequisites

3.1 PBX Cabling Requirements

Cabling for PRI connections must be CAT5e or better. A standard voice quality cable will not provide the appropriate signal quality and will cause the gateway to have issues establishing a connection on the D-channel. If connecting the gateway directly to the PBX, it may be necessary to use a T1 crossover cable (Pin 1 is connected to Pin 4 and Pin 2 is connected to Pin 5). The crossover cable is only necessary if the alarm light on the front of the gateway remains lit red after connecting the gateway to the PBX.

4. Summary of Limitations

No limitations noted.

5. Gateway Setup Notes

Steps for setting up the gateway:

1. Add the gateway to Microsoft® Office Communications Server (OCS) 2007.
2. Configure the Dialogic® Diva® Media Board drivers.
3. Configure the Dialogic® Diva® SIPcontrol™ Software.

5.1 Adding the Dialogic® 4000 Media Gateway Series to the Microsoft® Office Communications Server (OCS) 2007

It is necessary to add the gateway into the Active Directory containing the OCS environment:

1. Add the gateway to the OCS domain.
2. Add the Microsoft® Mediation Server on the gateway to OCS as follows:
 - Open a command prompt on the gateway.
 - Change the directory to: c:\program files\common files\microsoft communications server 2007 .
 - Execute:
 - `lscmd.exe /MedSrv /Action:Activate /Password:<Password-of-RTCComponentService-Account>`
 - It might be necessary to also add a username for the RTC component service account. To do this add a `/User:<user name>`.
3. Open the OCS MMC on the gateway:
 - `Start > Settings > Control Panel > Administrative Tools > Office Communications Server.`
 - Request, retrieve, and import a certificate from a CA (Certificate Authority) and add it to the Microsoft® Mediation Server. Refer to the OCS documentation for more information.

Note: This step needs to be done while you are directly logged onto the gateway. It cannot be done using the MMC of the OCS on another PC in the domain.

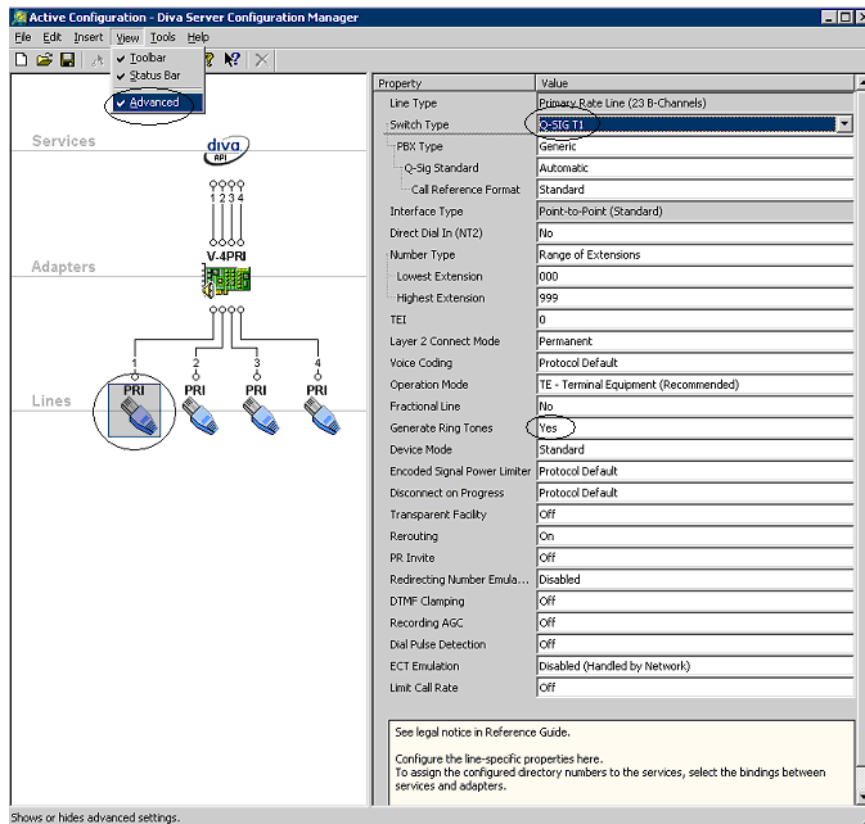
5.2 Dialogic® Diva® Media Board Configuration

The Diva Media Boards are configured in the Dialogic® Diva® Configuration Manager. To open the Configuration Manager, click:

Start > Programs > Dialogic Diva > Configuration Manager.

Note: In some instances within the software and documentation, Diva Media Boards are referred to as “Diva Server adapters”.

A screen similar to the one below will appear.



Note: The number of TDM circuits varies depending on the Dialogic® 4000 Media Gateway model that is used.

For this setup:

- Click the icon of one PRI line.
- Set the property `Switch Type` to the ISDN protocol that your PBX or telco supports. Common choices are:

NI2	USA/Canada, Nortel DMS 100 (National ISDN)
5ESS	USA/Canada, AT&T 5ESS (Custom ISDN)
Q.SIG	PBX, Q-SIG T1
EuroISDN	Europe/other countries, Euro-ISDN (ETSI-DSS 1)

Note: If you select PBX, Q-SIG T1 as Switch Type, you need to select the PBX Type from the dropdown menu below. If your PBX is not listed in this dropdown menu, select Generic, which will use a standard Q.SIG suitable for most PBXs.

- Configure the switch type of the other PRI lines as described above.
- To activate the changes, click `File > Activate`.
- Restart the computer if you are asked to do so.
- To test the connection with the PBX or the PSTN, see the [Dialogic® Diva® Line Test tool](#) on page 30 for more information.

5.3 Dialogic® Diva® SIPcontrol™ Software Gateway Application

The Diva SIPcontrol software is configured via the web-based interface. To open the web interface, click `Start > Dialogic Diva > SIPcontrol Configuration`.

On the main page, click the `SIPcontrol` link to display the different configuration menus.

The `PSTN Interface Configuration` section should automatically include all ports detected in the system.

Note: If you do not see any detected ports, you may need to add <http://127.0.0.1> as a trusted site. From Microsoft® Internet Explorer®, click `Tools > Internet Options > Security > Trusted Sites`. Use <http://127.0.0.1:10005> to get to the configuration.

In order for the Diva SIPcontrol software to route calls, the proper routes must be created and configured. Each route consists of a source interface and a destination interface. PSTN controllers and SIP peers are considered either a source interface or a destination interface depending on the call direction.

The following components need to be configured:

- PSTN Interface
- Network Interface
- SIP Peer
- Routing
- Number Normalization (either an Address Map or a dialplan)

If you have a simple phone numbering scheme, all numbers are of the same number exchange (ex 973-967-1xxx). It may be possible to use a dialplan to configure the gateway rather than an Address Map. In North America it is more common to need to use Address Maps.

5.3.1 PSTN Interface

The **PSTN Interface Configuration** shows the available T1 or E1 interfaces and enables you to configure them separately for Address Maps and dialplans. It is also possible to disable interfaces that are not in use. To disable the interfaces that are not in use, click the check mark in the **Enabled** field to delete it.

PSTN Interface Configuration						
Name	Nr	Hardware Description	Channels	Dialplan	Enabled	
Controller1	1	Eicon Diva Server V-2PRI/E1/T1 - PORT 1 SN: 1016	23	none	<input checked="" type="checkbox"/>	Details
Controller2	2	Eicon Diva Server V-2PRI/E1/T1 - PORT 2 SN: 1016	24	none	<input checked="" type="checkbox"/>	Details

5.3.2 Network Interface

The **Network Interface Configuration** displays the network interface connections that are available for use by the Dialogic® Diva® SIPcontrol™ Software. You need to enable the IP address that the Diva SIPcontrol software should listen to. Since the Mediation Server is co-resident on the gateway enable the network interface connections of the main gateway.

Since the Microsoft® Mediation Server and the Diva SIPcontrol software share the same network interface connections, it is necessary to configure them to listen and broadcast on different TCP ports. For the main gateway network interface connection, configure the TCP listen port to be an unused port like 9803. Enter this number in **SIP Listen Port**.

Network Interface Configuration						
Name	Device	IP Address	Protocol	SIP Listen Port	Enabled	
Intel(R) 825xER PCI Adapter	Intel(R) 825xER PCI Adapter	192.168.1.9	all	5060	<input type="checkbox"/>	
Intel(R) PRO100 M Network Connec	Intel(R) PRO/100 M Network Connection	192.168.1.10	all	9803	<input checked="" type="checkbox"/>	
Local Loopback Interface	Local Loopback Interface	127.0.0.1	all	5060	<input type="checkbox"/>	
RTP Start Port:	30000					
RTP End Port:	40000					
Jitterbuffer Size Min [ms]:	0					

Note: Take note of the **SIP Listen Port** number because you need it for the configuration of the Microsoft® Mediation Server.

5.3.3 SIP Peer

Add a SIP peer to point to the Microsoft[®] Mediation Server. To do so, expand the SIP Peer Configuration menu and click Add. In the Edit SIP Peer Configuration window, configure the following:

Edit SIP Peer Configuration	
Name:	Mediation Server
Peer type:	MS OCS 2007 / Mediation Server
Host:	192.168.1.10
Port:	5060
IP protocol:	TCP
Domain:	

- Name: Enter a name for the SIP peer, e.g., Mediation Server
- Peer type: Select MS OCS 2007 / Mediation Server from the dropdown menu.
- Host: Set the IP address on which the Microsoft[®] Mediation Server is listening. On the gateway, it is the gateway's IP address.
- Port: Set the broadcast port that the Diva SIPcontrol software uses to communicate with the Microsoft[®] Mediation Server. It is the port on which the OCS Server listens.
- IP protocol: Select TCP for the Microsoft[®] Mediation Server.
- Click OK to save the settings.

5.3.4 Routing Configuration

In the Routing Configuration menu, you need to create two routes, one for the inbound direction (TDM to IP) and one for the outbound direction (IP to TDM). These routes together will connect a SIP peer and a PSTN interface. After the number normalization is configured you will need to return to these routes and add the Address Map or dialplan. To configure the routes, expand the Routing Configuration menu and click Add. In the Routing Configuration window, configure two routes; one for IP to TDM (outbound) and one for PSTN to IP (inbound).

For the IP to TDM (outbound) route, configure the following:

- Name: **Enter** OCS to PSTN (outbound).
- Direction: **From** the dropdown menu select SIP to PSTN.
- Select sources: **Enable** the Microsoft® Mediation Server as source.
- Select destinations: **Select** Controller1 as master destination.
- Click OK to save the settings.

Click Add to configure the PSTN to IP (inbound) route:

- Name: **Enter** PSTN to OCS (outbound).
- Direction: **From** the dropdown menu select PSTN to SIP.
- Select sources: **Select** Controller1 as source.

- Select destinations: Select Enable the Microsoft® Mediation Server as master destination.
- Click OK to save the settings.

After both routes are configured, the Routing Configuration display should look like the picture below.

Routing Configuration					
Name	Sources	Destinations	Address Map	Enabled	
OCS to PSTN (outbound)	Mediation Server	Controller1 (Master)	none	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
PSTN to OCS (inbound)	Controller1	Mediation Server (Master)	none	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
Add					

5.4 Number Normalization

There are two methods to do number normalization; via a dialplan or with the help of Address Maps. If your PBX dialplan is not very complex, or to just get started, a dialplan may be used. For most PBX using multiple DID ranges or fractured DID ranges, it is necessary to use Address Maps.

Note: Do not enable both a dialplan and an Address Map at the same time.

5.4.1 Dialplan

A dialplan allows you to enter details about your numbering scheme specific to the location of the gateway to enable the Dialogic® Diva® SIPcontrol™ Software to either add digits to a number it receives from a PBX before being sent to OCS or remove numbers it gets from Microsoft® OCS so it can be sent to the PBX in a dialable format. Microsoft® OCS will always send an E.164 number to the gateway. With this number, the gateway needs to determine if it is an internal extension that should be dialed as a 4 digit number or an external number that needs to have a trunk access number appended to it prior to sending it to the PBX. The dialplan defines what the basic number formats look like to enable this conversion. In setting the dialplan, you define the different parts of a number, national prefix, area code, base number, and number of extension digits.

Add a dialplan from Dialplan Configuration. Complete the settings and click OK. For the dialplan below, the following phone number characteristics are used:

National Prefix: 1
 Area code: 716
 Base number: 639
 Extensions: 4 digits
 Access code: 9

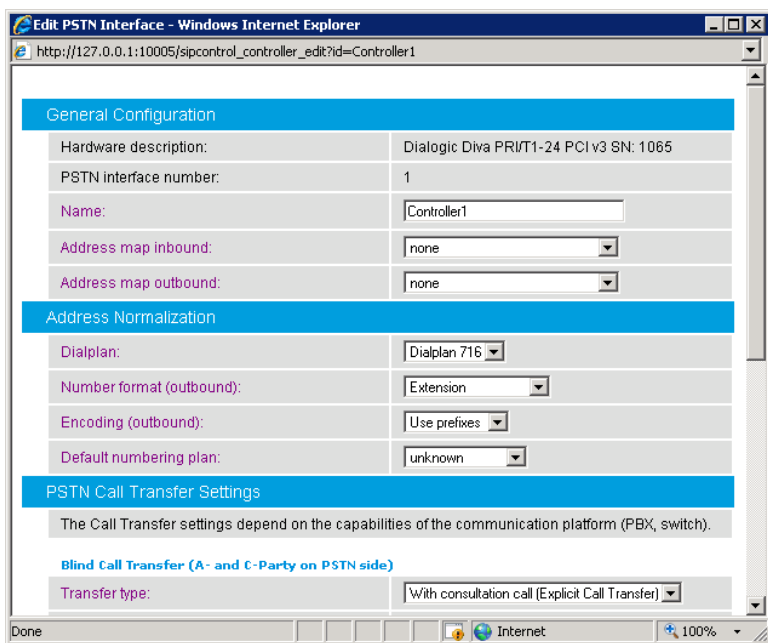
- Enter the local **Country code** (1 for North America).
- Enter the local **Area code**.
- Enter the **Base number** of your DID phone numbers. This is the non-variable portion of your DID numbers.
- Enter the number of extension digits.
- Enter the prefix necessary to dial an international call.
- Enter the prefix to dial a national call.
- Enter the trunk access number to get an external line on the PBX if necessary. (9 for North America).
- Click **OK** to save the settings.

For a dialplan to be applied to outbound PSTN calls, click **Details** on the PSTN controller.

PSTN Interface Configuration

Name	Nr	Hardware Description	Channels	Dialplan	Enabled	
Controller1	1	Dialogic Diva PRI/T1-24 PCI v3 SN: 1065	23	none	<input checked="" type="checkbox"/>	Details

The following window opens:



Configure the Address Normalization settings as follows:

- Select the Dialplan you have just created.
- Set the numbering format to Extension.
- Set the encoding to Use Prefixes.
- Set the Default numbering plan to unknown.
- Click OK to save the settings.

This configuration converts the dialed numbers from OCS into the format based on the dialplan so the PBX will be able to dial the number.

For a dialplan to be applied to inbound SIP calls, click Details on the SIP Peer.

SIP Peer Configuration							
Name	Default SIP to PSTN Peer	Host	Port	IP Protocol	Display Name to	Dialplan	Enabled
Mediation Server	<input checked="" type="checkbox"/>	192.168.1.10	5060	TCP		none	<input checked="" type="checkbox"/>

The following window opens:

Address Normalization	
Dialplan:	Dialplan 716
Number format (outbound):	International number
Encoding (outbound):	Use type flag
OK Cancel	

Configure the Address Normalization as follows:

- Dialplan: Select the dialplan, you configured in [Dialplan](#) on page 10.
- Number format (outbound): From the dropdown menu, select International number.
- Encoding (outbound): From the dropdown menu, select Use type flag.
- Click OK to save the settings and to close the window.
- Click Save on the main page for the changes to take effect.

This configuration converts the phone numbers into the E.164 format needed by OCS.

5.4.2 Address Map

The Address Map entry uses regular expression (as does OCS) for converting the call address format for inbound or (and) outbound direction.

Note: Call address for outbound calls (IP to TDM) includes “@hostname” part. For example, [+17166391234@DMG4000.bufocs.local](#) is the call address, not just +17166391234. For inbound calls (TDM to IP), call address is the called or calling number with the possible addition of the prefix +, N or S. For example, an inbound call has called number 1234 with ISDN type of numbering flag set to Subscriber, and calling number 49715233334444 with ISDN type of numbering flag set to International. The called address will be S1234 and calling address will be +49715233334444. If ISDN type of numbering flag is set to National, prefix N will be used with the call number. If type is Unknown, no prefix is used. This is important to know before applying RegEx rules in Address Map.

5.4.2.1 Outbound calls (VoIP to TDM):

OCS sends the E.164 dial number format to the gateway. Both called and calling numbers need to be converted to a format that the PBX can accept. Using the same PBX dialplan as in the previous section, the following conversions are needed:

- In the “non-DID internal” rule, the gateway will automatically extract the number string following the “ext” field of the Invite and pass it to the routing table. Because of this mechanism it is only necessary to pass the extension in the CPID manipulation.
- The “Internal” rule will extract the last 4 numbers from the right side to create a 4-digit extension to be dialed on the local PBX.
- The “National” rule adds the trunk access number to the 11 digits following the “+”. Depending on your configuration, this can be changed or deleted.

- The “International” rule adds the trunk access number (in our case 9) and the international access number (e.g., 011). If necessary, you can change or delete the trunk access number.

Calling number	From OCS	To PBX
Internal	+1716639xxxx	716639xxxx

Called number	From OCS	To PBX
4-digit extension from non-DID number	+17166391000;ext=xxxx	xxxx
To Internal	+1716639xxxx	xxxx
To National	+1xxxxxxxxxxx	91xxxxxxxxxxx
To International	+xxx...xxx	9011xxx...xxx

Below are the regular expression for the conversion tables above:

Sub rule name	Expression	Format	Stop on match
Calling number	^\+1(716639\d{4})	\$1	Not checked
Called - non-DID	^\+17166391000\;ext=(\d{4})	\$1	Check
Called - Internal	^\+1716639(\d{4})	\$1	Check
Called - National	^\+1	91	Check
Called - International	^\+	9011	Check

The graphic below shows the Address Map entries for outbound calls.

Address Map Configuration ▼

Name	Rule Name	Stop on Match	Enabled	
Outbound (VoIP to TDM)	Calling Number	<input type="checkbox"/>	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
	Called - non-DID	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
	Called - Internal	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
	Called - National	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
	Called - International	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
Add Rule				

The order of the above 5 sub rules and the Stop on Match check mark are important.

Below is the first sub rule, **Calling Number**, that converts a E.164 calling number into 10-digit national number.

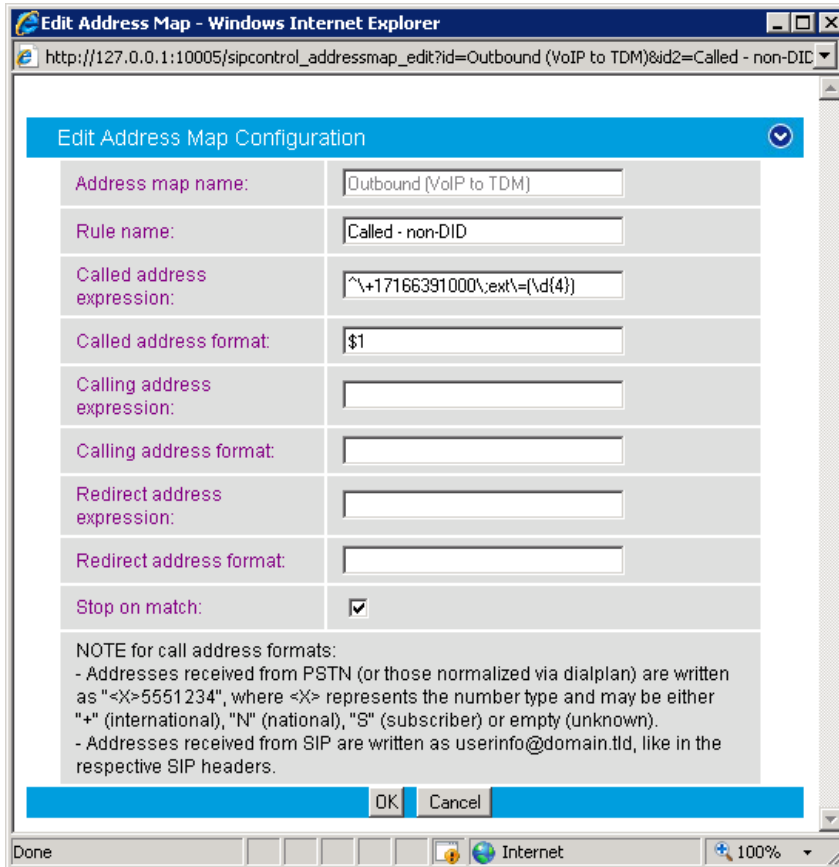
Edit Address Map Configuration

Address map name:	Outbound (VoIP to TDM)
Rule name:	Calling Number
Called address expression:	
Called address format:	
Calling address expression:	^\\+1(716639\\d{4})
Calling address format:	\$1
Redirect address expression:	
Redirect address format:	
Stop on match:	<input type="checkbox"/>

NOTE for call address formats:
- Addresses received from PSTN (or those normalized via dialplan) are written as "<X>5551234", where <X> represents the number type and may be either "+" (international), "N" (national), "S" (subscriber) or empty (unknown).
- Addresses received from SIP are written as userinfo@domain.tld, like in the respective SIP headers.

OK Cancel

The following sub rule, **Called – Non-DID**, extracts the 4-digit extension following the E.164 “ext=” field.



Following is the third sub rule, **Called – Internal**, that converts E.164 for internal called number into 4-digit extension.

Edit Address Map Configuration

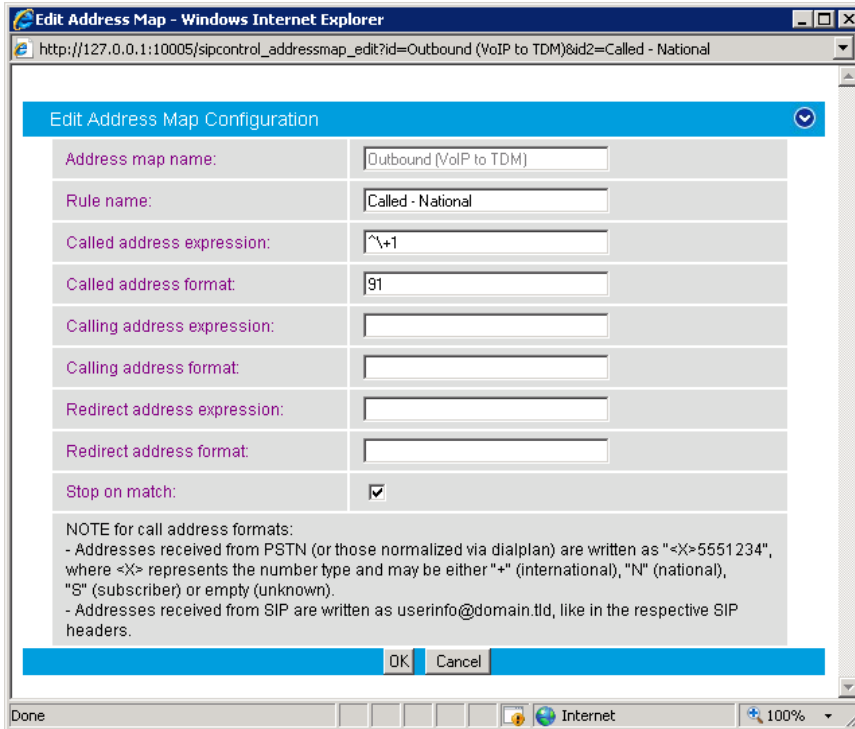
Address map name:	Outbound (VoIP to TDM)
Rule name:	Called - Internal
Called address expression:	^\\+1716639(\\d{4})
Called address format:	\$1
Calling address expression:	
Calling address format:	
Redirect address expression:	
Redirect address format:	
Stop on match:	<input checked="" type="checkbox"/>

NOTE for call address formats:
- Addresses received from PSTN (or those normalized via dialplan) are written as "<X>5551234", where <X> represents the number type and may be either "+" (international), "N" (national), "S" (subscriber) or empty (unknown).
- Addresses received from SIP are written as userinfo@domain.tld, like in the respective SIP headers.

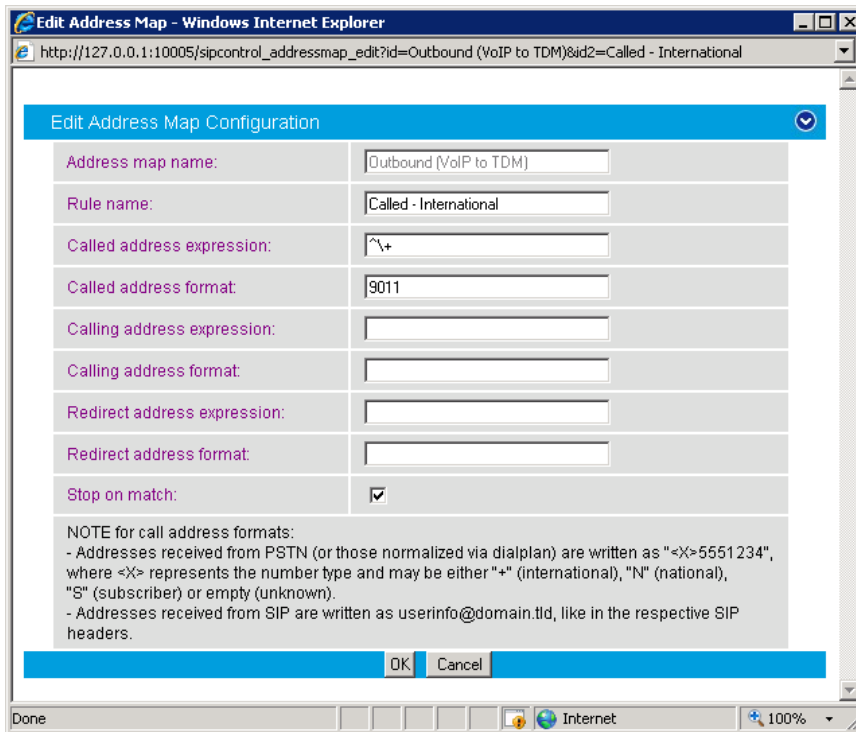
OK Cancel

Done Internet 100%

The following sub rule, **Called – National**, is for converting an E.164 national number into 10-digits national number with prefix 91.



The below rule converts an international called number by prefixing 9011.



5.4.2.2 Inbound Calls (PSTN to VoIP):

These examples assume that the PBX sends the inbound calls using 4-digit extensions. The rules will need to be modified if your PBX is sending different number of extensions.

- Converting the called party number to E.164 format.**
 It is not necessary to convert the called party number to E.164 before it is sent to OCS, as the existing normalization rules in OCS will take the number the gateway is receiving from the PBX, and normalize it to the correct format.
- Converting the calling party number to E.164 format.**
 If you want OCS to be able to do a reverse look up in the global address list (GAL) or in the users contact list, it is necessary to have the gateway convert the calling party number to E.164 to match the format in the GAL. This manipulation of the calling party information depends on the information that the gateway receives from the PSTN or PBX.
- Removing the ISDN number type indicator.**
 Dialogic® Diva® SIPcontrol™ Software appends the ISDN call type (N = National, + = International and S = Station) at the beginning of calling, called, and redirect numbers. Therefore, it is necessary to have a rule to remove these, as OCS does not make use of them. Here are a few examples of necessary conversions and the rules to achieve the needed conversions.

Called number	From PBX	To OCS
All types	[N,S,+]xxxx	xxxx

Calling number	From PBX	To OCS
Calling from internal	xxxx	+1716639xxxx
Calling from national	xxxxxxxxxxx	+1xxxxxxxxxxx
Calling from international	xxx...xxx	+xxx...xxx

Sub rule name	Expression	Format	Stop on match
Remove ISDN type Called Calling Redirect	^[NS\+]?		Not checked
Calling - internal	^\d{4}\$	+1716639\$1	Checked
Calling - national	^\d{10}\$	+1\$1	Checked
Calling - international	(.*)	+\$1	Checked

Create an Address Map named **Inbound (TDM to VoIP)** and its 4 sub rules as shown below.

Address Map Configuration ▼

Name	Rule Name	Stop on Match	Enabled	
Outbound (VoIP to TDM)	Calling Number	<input type="checkbox"/>	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
	Called - non-DID	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
	Called - Internal	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
	Called - National	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
	Called - International	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
Add Rule				
Inbound (TDM to VoIP)	Remove ISDN type	<input type="checkbox"/>	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
	Calling - internal	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
	Calling - national	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
	Calling - international	<input checked="" type="checkbox"/>	<input checked="" type="checkbox"/>	↑ ↓ Details Delete
Add Rule				
Add				

The rule below removes Dialogic® Diva® SIPcontrol™ Software's marking for the ISDN number type for the called number, calling number, and the redirect number if there is one. If OCS does not make use of it, it is removed.

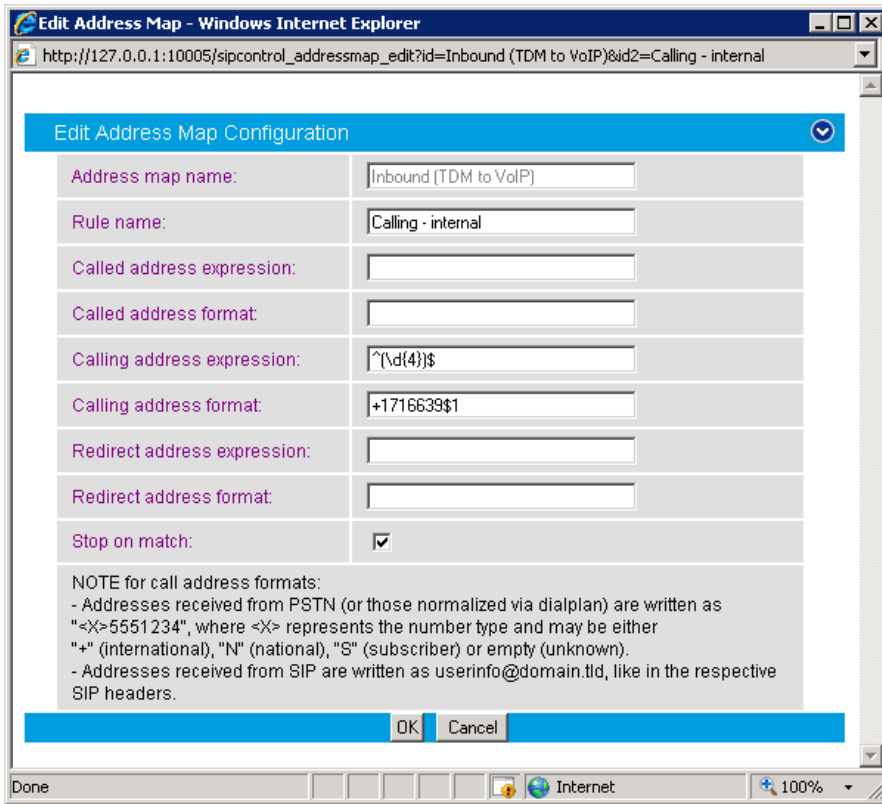
Edit Address Map Configuration

Address map name:	Inbound (TDM to VoIP)
Rule name:	Remove ISDN type
Called address expression:	^[NS\+]?
Called address format:	
Calling address expression:	^[NS\+]?
Calling address format:	
Redirect address expression:	^[NS\+]?
Redirect address format:	
Stop on match:	<input type="checkbox"/>

NOTE for call address formats:
- Addresses received from PSTN (or those normalized via dialplan) are written as "<X>5551234", where <X> represents the number type and may be either "+" (international), "N" (national), "S" (subscriber) or empty (unknown).
- Addresses received from SIP are written as userinfo@domain.tld, like in the respective SIP headers.

OK Cancel

The second rule converts a 4-digit calling number into an E.164 format to enable OCS to perform a reverse look-up on the calling party.



Like the previous rule, this rule also converts the calling party to E.164 format; but this is looking for a 10-digit external number.

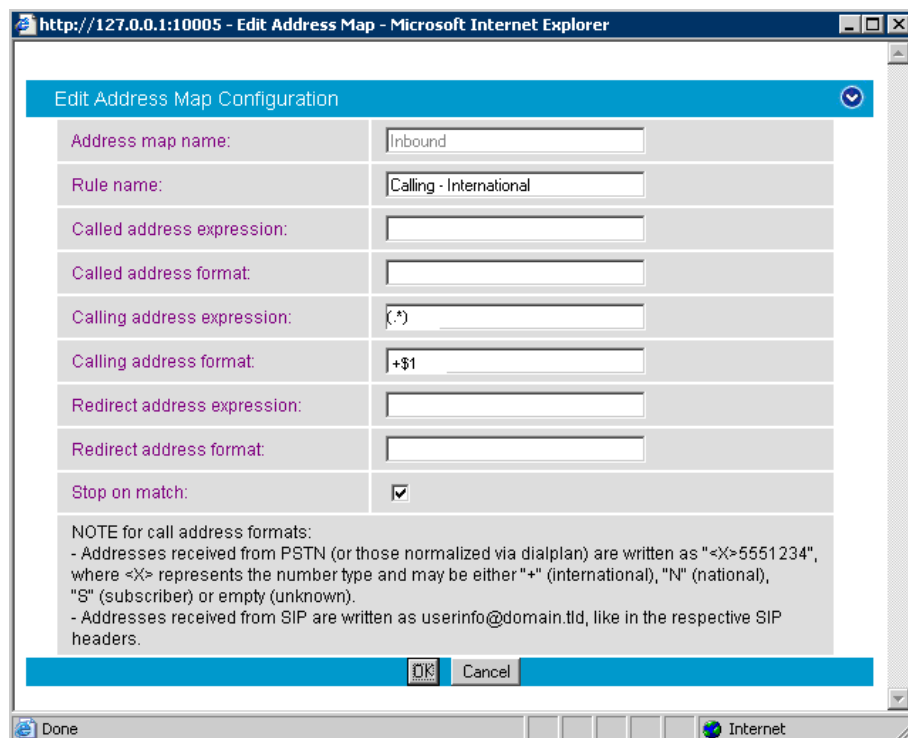
Edit Address Map Configuration

Address map name:	Inbound (TDM to VoIP)
Rule name:	Calling - national
Called address expression:	
Called address format:	
Calling address expression:	<code>^(\d{10})\$</code>
Calling address format:	<code>+1\$1</code>
Redirect address expression:	
Redirect address format:	
Stop on match:	<input type="checkbox"/>

NOTE for call address formats:
- Addresses received from PSTN (or those normalized via dialplan) are written as "<X>5551234", where <X> represents the number type and may be either "+" (international), "N" (national), "S" (subscriber) or empty (unknown).
- Addresses received from SIP are written as userinfo@domain.tld, like in the respective SIP headers.

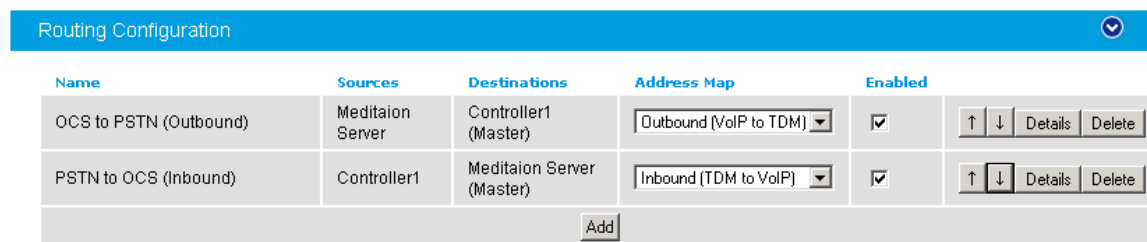
OK Cancel

This final inbound rule converts the calling ID for an international caller into E.164 format.



5.5 Applying an Address Map to a Route

After you created the inbound and outbound Address Maps, it is necessary to link them to the inbound and outbound routes. To do so, expand the **Routing Configuration** dialog and in the dropdown menu under **Address Map** select **Outbound (VoIP to TDM)** for the outbound route and **Inbound (TDM to VoIP)** for the inbound route as shown in the graphic below.



5.6 Saving the Settings

After you made the changes, you need to save the configuration. To save the configuration, scroll down to the bottom of the main page and click *Save*.

The screenshot shows the Dialogic SIPcontrol Configuration page. The navigation bar includes 'HOME', 'SIPcontrol Configuration', 'REFERENCE GUIDE', and 'LOGOUT'. The left sidebar has 'Configuration' selected, with sub-items 'SIPcontrol' and 'Password'. The main content area lists various configuration options: PSTN Interface Configuration, Network Interface Configuration, SIP Peer Configuration, Routing Configuration, Dialplan Configuration, Address Map Configuration, Cause Code Mapping Configuration, and System Settings. At the bottom, there are three buttons: 'Save', 'Show Configuration', and 'Revert to Saved Configuration'.

5.7 Restarting the Dialogic® Diva® SIPcontrol™ Software

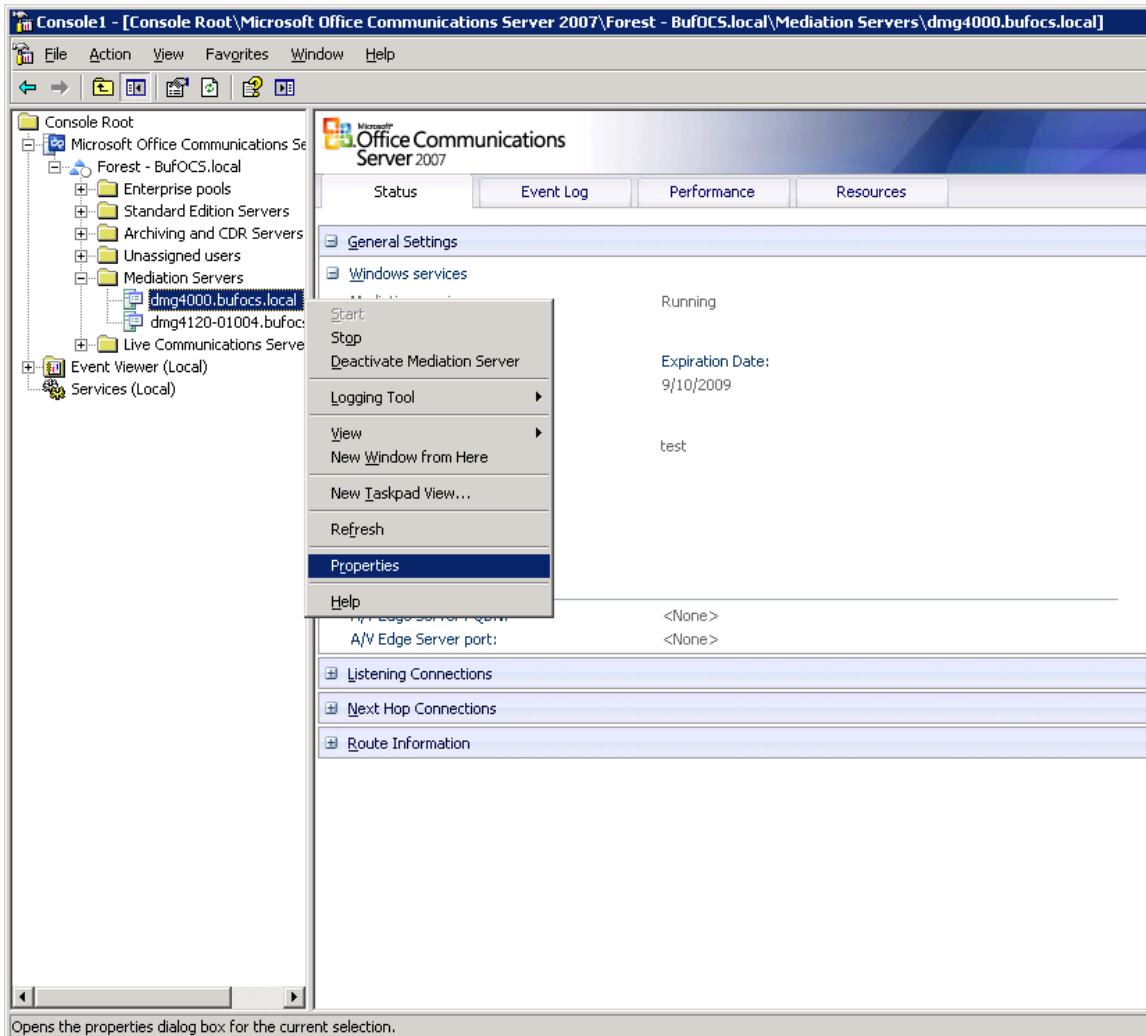
Note: A restart of the Diva SIPcontrol software service is needed only if the settings under Network Interface were changed.

Save the configuration and restart the Diva SIPcontrol software service for the changes to take effect. To do so, click *Service Status* on the left hand side of the main configuration page, and then click *Restart SIPcontrol*. The *Last operation log* will show that the service has been stopped and started again.

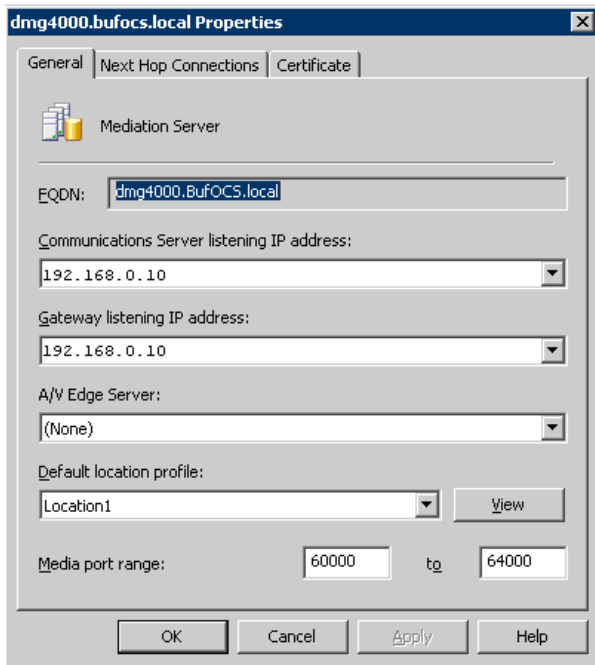
The screenshot shows the Dialogic Service Status page. The navigation bar includes 'HOME', 'Service Status', 'REFERENCE GUIDE', and 'LOGOUT'. The left sidebar has 'Service Status' selected under the 'System' category. The main content area shows 'Status = Running' and three buttons: 'Start SIPcontrol', 'Stop SIPcontrol', and 'Restart SIPcontrol'.

6 Microsoft® Mediation Server Configuration

From OCS MMC snap-in, right click on the detected Microsoft® Mediation Server and select Properties.



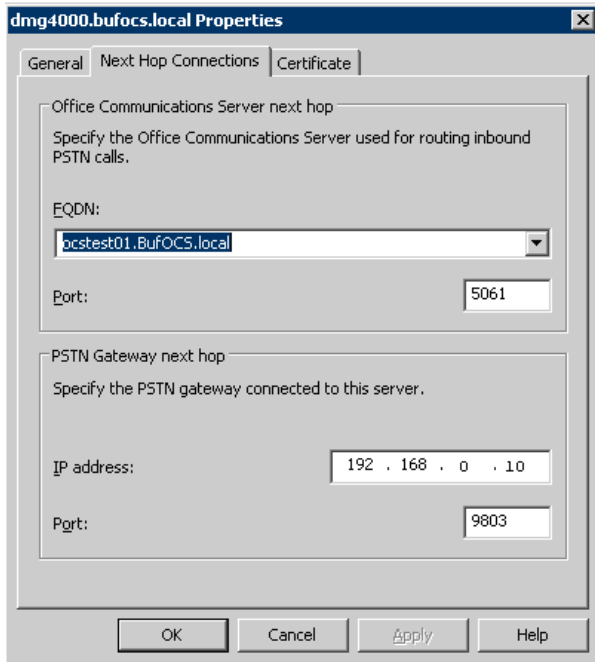
The following window opens:



On the General tab, configure the following settings:

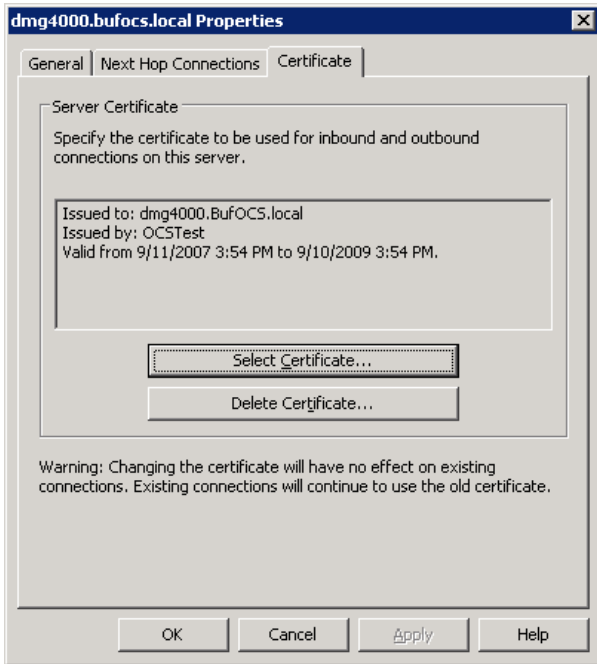
- Under Communications Server listening IP address and Gateway listening IP address, enter the IP address of the gateway. Since the gateway and Microsoft® Mediation Server are running on the same PC, the one address may be used for both gateway listening and OCS listening address.
- Under Default location profile, enter the location profile that corresponds to the deployment location of the gateway. This is used for defining the normalization rules for the called party numbers.

On the Next Hop Connections tab, configure the following settings:



- Next to IP address, enter the IP address of the gateway.
- Next to Port, enter the same port number as in the Dialogic® Diva® SIPcontrol™ Software configuration. See section 5.3.2 [Network Interface](#) for more information.

On the Certificate tab, configure the following:



Select the certificate that will be used to communicate with OCS. You need to restart Microsoft® Mediation Server for the changes to take effect.

7 Troubleshooting

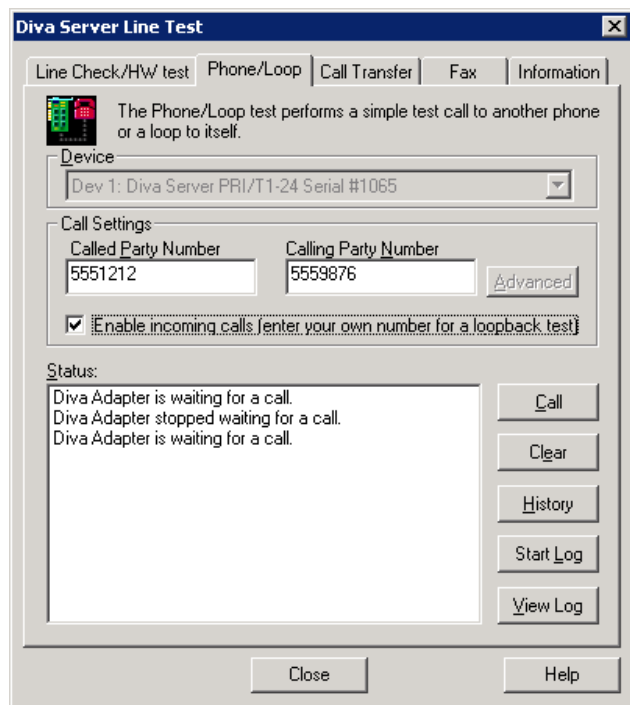
7.1 Important Debugging Tools

- Dialogic® Diva® Line Test tool
- Dialogic® Diva® Diagnostics tool: Used to review and analyze all SIP and ISDN traffic that relates to calls going into and leaving the gateway

7.2 Dialogic® Diva® Line Test tool

Diva Line Test tool enables you to test the PSTN configuration and connection. With this tool, you may place calls to the PBX or PSTN as well as receive calls for the PSTN or PBX without having other components set up on the IP side of the gateway.

To start Diva Line Test tool, click: Start > Programs > Dialogic Diva > Line Test.
Select the Phone/Loop tab to test the T1 or E1 connection.



To test an outgoing call, do the following:

- In Called Party Number, enter a PBX extension. If you want to place an external call, enter an external phone number.
- Click Call to initiate the outbound call.

To test an incoming call, do the following:

- Select `Enable incoming calls [...]` box so that the Diva Line Test tool application answers a call from the PBX or PSTN.
- Click `Call` to initiate the outbound call.

Note: For incoming calls, the called, calling, and redirecting number information is presented if available.

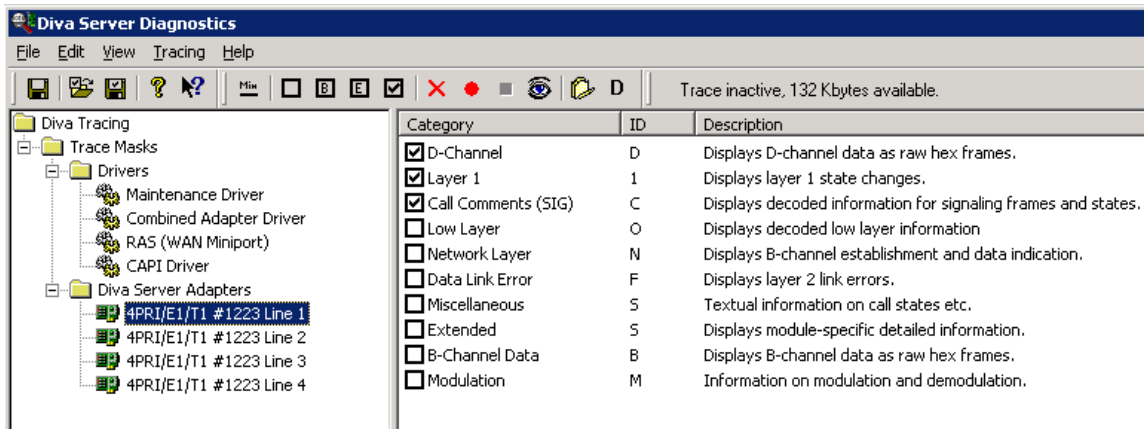
Note: For the Diva Line Test tool to answer an incoming call it is necessary to stop the Dialogic® Diva® SIPcontrol™ Software process. See [Restarting the Dialogic® Diva® SIPcontrol™ Software](#) on page 25 for how to stop the service.

7.3 Dialogic® Diva® Diagnostics tool



Before using the Dialogic® Diva® Diagnostics tool, you would need to enable it by setting the Dialogic® Diva® SIPcontrol™ Software debug. To do so, open the Diva SIPcontrol software web interface, click the link `System Settings`, and set `Debug Level` to `Extended`. Click `Save` for the changes to take effect.

Now, you can start the Diva Diagnostics tool. To do so, click:

`Start > Programs > Dialogic Diva > Diagnostics.`



1. Click one line of your Dialogic® Diva® Media Board in the left pane and click **B** on the toolbar to activate the Basic tracing level. This level captures Q.931 ISDN messages.
2. Click CAPI driver in the left pane and activate the Basic tracing level as explained in step 1.
3. Start tracing. To do so, click the start icon **●** or select the `Start Tracing` option from the `Tracing` menu.
4. Reproduce the issue.

5. To stop tracing, click the stop icon  on the tool bar or select the Stop Tracing option from the Tracing menu.
6. To view your collected trace, click the view icon  on the toolbar or select the View Recorded Trace option from the View menu. A notepad window will open with the recorded log.

Examples of Dialogic® Diva® Diagnostics traces for an inbound (TDM to IP) call to OCS

Basic notations for reading the trace:

- SIG-R: RX Q.931 ISDN message
- SIG-X: TX Q.931 ISDN message
- SIPR: RX SIP message
- SIPX: TX SIP message

< Below is RX Q.931 ISDN message for an inbound call >

```
...
9:16:28.431 C 3 21:2389:383 - SIG-R(030) 08 02 00 17 05 04 03 80 90 A2 18 03 A9 83 81 6C 06 01 A0 33
30 30 32 70 05 C1 35 31 30 31
      Q.931 CR0017 SETUP
      Bearer Capability 80 90 a2
      Channel Id a9 83 81
      Calling Party Number 01 a0 '2401'
      Called Party Number c1 '5101'
```

<Below is TX SIP message with SDP>

```
9:16:28.431 1 L 12 00010000-SIPX begin to IP:192.168.0.106 port:5060 socket:3 Proto:TCP
9:16:28.431 1 L 12 00010000- >INVITE sip:+17166395101@dmg4000.bufocs.local:5060 SIP/2.0
9:16:28.431 1 L 12 00010000- >Via: SIP/2.0/TCP 192.168.0.106:9803;branch=z9hG4bK2617534104-7603272
9:16:28.431 1 L 12 00010000- >Max-Forwards: 70
9:16:28.431 1 L 12 00010000- >Allow: INVITE,ACK,CANCEL,BYE,OPTIONS,NOTIFY,REFER
9:16:28.431 1 L 12 00010000- >Accept: application/sdp,application/simple-message-summary
9:16:28.431 1 L 12 00010000- >Supported: timer,replaces
9:16:28.431 1 L 12 00010000- >From: "Dialogic Diva SIPcontrol"
<sip:+17166392401@192.168.0.106;user=phone>;tag=sipcontrol_2617534104-7668808
9:16:28.431 1 L 12 00010000- >To: "Default" <sip:+17166395101@bufocs.local;user=phone>
9:16:28.431 1 L 12 00010000- >Call-ID: 9c046698-730448-17@dmg4000
9:16:28.431 1 L 12 00010000- >CSeq: 1 INVITE
9:16:28.431 1 L 12 00010000- >Min-SE: 90
9:16:28.431 1 L 12 00010000- >Session-Expires: 600;refresher=uac
9:16:28.431 1 L 12 00010000- >Contact: <sip:+17166392401@192.168.0.106:9803>
9:16:28.431 1 L 12 00010000- >Content-Type: application/sdp
9:16:28.431 1 L 12 00010000- >Content-Length: 253
9:16:28.431 1 L 12 00010000- >
9:16:28.431 1 L 12 00010000- >v=0
9:16:28.431 1 L 12 00010000- >o=SIPcontrol 7472200 7472200 IN IP4 192.168.0.106
9:16:28.431 1 L 12 00010000- >s=-
9:16:28.431 1 L 12 00010000- >c=IN IP4 192.168.0.106
9:16:28.431 1 L 12 00010000- >t=0 0
9:16:28.431 1 L 12 00010000- >m=audio 30060 RTP/AVP 8 0 101 13
```



```
9:16:28.431 1 L 12 00010000- >a=rtpmap:8 PCMA/8000
9:16:28.431 1 L 12 00010000- >a=rtpmap:0 PCMU/8000
9:16:28.431 1 L 12 00010000- >a=rtpmap:101 telephone-event/8000
9:16:28.431 1 L 12 00010000- >a=fmtp:101 0-15
9:16:28.431 1 L 12 00010000- >a=rtpmap:13 CN/8000
9:16:28.431 1 L 12 00010000- >a=sendrecv
9:16:28.431 1 L 12 00010000-SIPX end
```

...

<Below is RX SIP message>

```
9:16:28.431 1 L 12 00010000-SIPR begin (331 byte) from IP:192.168.0.106 PORT:5060 on socket 3 port 5060
TCP
9:16:28.431 1 L 12 00010000- >SIP/2.0 100 Trying
9:16:28.431 1 L 12 00010000- >FROM: "Dialogic Diva
SIPcontrol"<sip:+17166392401@192.168.0.106;user=phone>;tag=sipcontrol_2617534104-7668808
9:16:28.431 1 L 12 00010000- >TO: "Default"<sip:+17166395101@bufocs.local;user=phone>
9:16:28.431 1 L 12 00010000- >CSEQ: 1 INVITE
9:16:28.431 1 L 12 00010000- >CALL-ID: 9c046698-730448-17@dmg4000
9:16:28.431 1 L 12 00010000- >VIA: SIP/2.0/TCP 192.168.0.106:9803;branch=z9hG4bK2617534104-7603272
9:16:28.431 1 L 12 00010000- >CONTENT-LENGTH: 0
9:16:28.431 1 L 12 00010000- >
9:16:28.431 1 L 12 00010000-SIPR end
```

...

```
9:16:28.665 0 L 12 00010000-SIPR begin (408 byte) from IP:192.168.0.106 PORT:5060 on socket 3 port 5060
TCP
9:16:28.665 0 L 12 00010000- >SIP/2.0 183 Session Progress
9:16:28.665 0 L 12 00010000- >FROM: "Dialogic Diva
SIPcontrol"<sip:+17166392401@192.168.0.106;user=phone>;tag=sipcontrol_2617534104-7668808
9:16:28.665 0 L 12 00010000- >TO:
Default<sip:+17166395101@bufocs.local;user=phone>;epid=CE4C602FA5;tag=3f5ea65423
9:16:28.665 0 L 12 00010000- >CSEQ: 1 INVITE
9:16:28.665 0 L 12 00010000- >CALL-ID: 9c046698-730448-17@dmg4000
9:16:28.665 0 L 12 00010000- >VIA: SIP/2.0/TCP 192.168.0.106:9803;branch=z9hG4bK2617534104-7603272
9:16:28.665 0 L 12 00010000- >CONTENT-LENGTH: 0
9:16:28.665 0 L 12 00010000- >SERVER: RTCC/3.0.0.0 MediationServer
9:16:28.665 0 L 12 00010000- >
9:16:28.665 0 L 12 00010000-SIPR end
```

...

```
9:16:28.869 1 L 12 00010000-SIPR begin (399 byte) from IP:192.168.0.106 PORT:5060 on socket 3 port 5060
TCP
9:16:28.869 1 L 12 00010000- >SIP/2.0 180 Ringing
9:16:28.869 1 L 12 00010000- >FROM: "Dialogic Diva
SIPcontrol"<sip:+17166392401@192.168.0.106;user=phone>;tag=sipcontrol_2617534104-7668808
9:16:28.869 1 L 12 00010000- >TO:
Default<sip:+17166395101@bufocs.local;user=phone>;epid=CE4C602FA5;tag=3f5ea65423
9:16:28.869 1 L 12 00010000- >CSEQ: 1 INVITE
9:16:28.869 1 L 12 00010000- >CALL-ID: 9c046698-730448-17@dmg4000
9:16:28.869 1 L 12 00010000- >VIA: SIP/2.0/TCP 192.168.0.106:9803;branch=z9hG4bK2617534104-7603272
9:16:28.869 1 L 12 00010000- >CONTENT-LENGTH: 0
9:16:28.869 1 L 12 00010000- >SERVER: RTCC/3.0.0.0 MediationServer
9:16:28.869 1 L 12 00010000- >
9:16:28.869 1 L 12 00010000-SIPR end
```

...

```
9:16:30.197 1 L 12 00010000-SIPR begin (836 byte) from IP:192.168.0.106 PORT:5060 on socket 3 port 5060
TCP
9:16:30.197 1 L 12 00010000- >SIP/2.0 200 OK
9:16:30.197 1 L 12 00010000- >FROM: "Dialogic Diva
SIPcontrol"<sip:+17166392401@192.168.0.106;user=phone>;tag=sipcontrol_2617534104-7668808
9:16:30.197 1 L 12 00010000- >TO:
Default<sip:+17166395101@bufocs.local;user=phone>;epid=CE4C602FA5;tag=3f5ea65423
```

```
9:16:30.197 1 L 12 00010000- >CSEQ: 1 INVITE
9:16:30.197 1 L 12 00010000- >CALL-ID: 9c046698-730448-17@dmg4000
9:16:30.197 1 L 12 00010000- >VIA: SIP/2.0/TCP 192.168.0.106:9803;branch=z9hG4bK2617534104-7603272
9:16:30.197 1 L 12 00010000- >CONTACT:
<sip:dmg4000.BufOCS.local:5060;transport=Tcp;maddr=192.168.0.106>
9:16:30.197 1 L 12 00010000- >CONTENT-LENGTH: 253
9:16:30.197 1 L 12 00010000- >SUPPORTED: 100rel
9:16:30.197 1 L 12 00010000- >CONTENT-TYPE: application/sdp; charset=utf-8
9:16:30.197 1 L 12 00010000- >ALLOW: UPDATE
9:16:30.197 1 L 12 00010000- >SERVER: RTCC/3.0.0.0 MediationServer
9:16:30.197 1 L 12 00010000- >ALLOW: Ack, Cancel, Bye, Invite
9:16:30.197 1 L 12 00010000- >
9:16:30.197 1 L 12 00010000- >v=0
9:16:30.197 1 L 12 00010000- >o=- 0 0 IN IP4 192.168.0.106
9:16:30.197 1 L 12 00010000- >s=session
9:16:30.197 1 L 12 00010000- >c=IN IP4 192.168.0.106
9:16:30.197 1 L 12 00010000- >b=CT:1000
9:16:30.197 1 L 12 00010000- >t=0 0
9:16:30.197 1 L 12 00010000- >m=audio 62438 RTP/AVP 8 101
9:16:30.197 1 L 12 00010000- >c=IN IP4 192.168.0.106
9:16:30.197 1 L 12 00010000- >a=rtcp:62439
9:16:30.197 1 L 12 00010000- >a=label:Audio
9:16:30.197 1 L 12 00010000- >a=rtpmap:8 PCMA/8000
9:16:30.197 1 L 12 00010000- >a=rtpmap:101 telephone-event/8000
9:16:30.197 1 L 12 00010000- >a=fmtp:101 0-16
9:16:30.197 1 L 12 00010000- >a=ptime:20
9:16:30.197 1 L 12 00010000-SIPR end
```

...

<Below is TX Q.931 ISDN message, after SIP session is established>

```
9:16:30.212 C 3 21:2391:136 - SIG-X(005) 08 02 80 17 07
Q.931 CR8017 CONN
```