Dialogic® 4000 Media Gateway Series

Reference Guide

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Contractor/ manufacturer is:

Dialogic CORPORATION.

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This Agreement has been drafted in English at the express wish of the parties. Ce contrat a été rédigé en anglais à la demande expresse des parties.

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Dialogic® 4000 Media Gateway Series Reference Guide

About This Publication

The Dialogic® 4000 Media Gateway Series are pre-installed with the Windows® versions of the Dialogic® Diva® System Release software and the Dialogic® Diva® SIPcontrol™ software. This reference guide contains relevant information about both software versions, such as Diva SIPcontrol configuration parameters, the Dialogic® Diva® Diagnostics tool, the Dialogic® Diva® Line Test tool, and the SNMP configuration for Dialogic® Diva® Media Boards. Various configuration scenarios are also included, as well as information about the different support options.

The term "Dialogic® 4000 Media Gateways" is used herein to refer collectively to the Dialogic® 4000 Media Gateway Series and the term "Dialogic® 4000 Media Gateway" is used herein to refer a gateway in the Dialogic® 4000 Media Gateway Series.

Feature Overview

For the list of Dialogic® Diva® SIPcontrol™ software features, see below.

For the list of Dialogic® Diva® Media Board features, see page 12.

Dialogic® Diva® SIPcontrol™ software features

General features

- Support for Microsoft® Office Communications Server 2007 Release 2
- Noise suppression support
- Echo cancellation selectable via GUI
- Forward the display name from SIP to Q.SIG and vice versa
- Support for Windows Vista® and Windows Server® 2008
- Diva SIPcontrol software update via web interface
- Interoperability with Dialogic® Host Media Processing (HMP) software 3.0WIN and 3.1LIN
- Configuration via web interface
- Standard web browsers can be used for configuring. The Dialogic® Diva® SIPcontrol™ Software has been tested with the following browsers:
  - Microsoft® Internet Explorer® version 6 and 7
  - Mozilla Firefox version 2.0 and 3.0
- Remote configuration of the Diva SIPcontrol software from any computer in the network
- Cause codes: Configurable translation of ISDN cause code to SIP response code and vice versa; consequently, the Diva SIPcontrol software can adapt to the specific behavior of the PSTN, PBX, and/or SIP peer.
- Configuration changes during runtime: Modify most parameters of the Diva SIPcontrol software without the need to restart the service; active calls are not affected by configuration updates and continue undisturbed.
- Support for North American numbering plan: The configuration of multiple area codes is handled as local. Therefore, the Dialogic® Diva® SIPcontrol™ Software dialplan engine is able to automatically format dialed numbers according to local phone provider requirements without any additional regular expressions.
- Interoperability with the Dialogic® Brooktrout® Bfv API SDK: The Dialogic® Brooktrout® SR140 Fax Software version 5.2.1 has been confirmed via testing to be V.34/T.38 interoperable with the Diva SIPcontrol software. The Brooktrout SR140 Fax Software is high-performance, host-based T.38 fax software for IP networks.
- Codec configuration: Configuration options for supported audio and fax codecs. See Media processing on page 10 for supported codecs.
- Support for Proxy and Registrar authentication
- Support for registering the Diva SIPcontrol software as an e-phone gateway

- Support for Early Media: Early Media is supported for calls from SIP to the PSTN. For calls from the PSTN to SIP, it depends on the used line protocol.
- Configuration of Dialogic® Diva® Media Board parameters via the web interface
- Interoperability with the Dialogic® Host Media Processing (HMP) software
- Support for up to 64 ports per system for Dialogic® Diva® BRI and Analog Media Board installations
- Support for up to 240 ports per system for Dialogic® Diva® PRI Media Board installations

Call handling

- Support for TLS and SSL encryption and authentification
- Support for SRTP (secure Real-time Transport Protocol)
- Support for SIPS (Secure SIP)
- Support for RTCP
- SIP methods: ACK, BYE, INVITE, NOTIFY*, REFER, CANCEL, OPTIONS
- Configurable IP transport layer TCP, UDP, or TLS
- Basic call incl. numbering services:
  - Called Party Number
  - Calling Party Number
  - Redirecting Number
- Call Routing
- Call Hold/Retrieve (e.g., Re-Invite mapping towards ISDN)
- PSTN-side Call Transfer (i.e., Refer points to PSTN)
- PSTN-side incoming Call Diversion
- Message Waiting Activation / Deactivation
- Support of REDIRECT (Moved Temporarily)
- SIP Session Timer (RFC 4028)
- Simplified Number Normalization based on PSTN connection parameters
- Number Manipulation using Regular Expressions

* NOTIFY in combination with SUBSCRIBE are used to provide the feature Message Waiting Activation / Deactivation with regular SIP clients. However, in a gateway configuration applications are using the features without the need for the Dialogic® Diva® SIPcontrol™ software to use SUBSCRIBE.

Media processing

- Support for the following codecs:
  - G.711 A-law and u-law
  - G.726 (16, 24, 32, and 40 kbps)
  - G.729
  - GSM-FR
  - iLBC

  Note: For G.729, you need to purchase and activate a license before you can use it. See the Dialogic® 4000 Media Gateway Quickstart Guide for more information.

  Note: iLBC is only available on Dialogic® Diva® V-2PRI and V-4PRI Media Boards.
- RTP dynamic payload audio/telephony event
- RTP profile RTP/AVP
- DTMF via RTP payload/telephony event (RFC 2833)

- PSTN-side fax tone detection via RTP event (RFC 2833)
- Echo Canceller with up to 256 ms
- T.38 Fax up to V.34 (SuperG3 Fax)

Reliability

- Load balancing and failover on PSTN side
- Load balancing and failover on SIP side (optionally uses OPTIONS for keep-alive check)
- Alive check for active calls on SIP side via SIP session timer

Supported RFCs

- RFC2617 - HTTP Digest Authentication
- RFC2833 - RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC3261 - Session Initiation Protocol
- RFC3262 - Reliability of Provisional Responses in Session Initiation Protocol (SIP)
- RFC3264 - An Offer/Answer Model with Session Description Protocol
- RFC3265 - SIP-specific Event Notification
- RFC3326 - The Reason Header Field for the Session Initiation Protocol (SIP)
- RFC3389 - RTP Payload for Comfort Noise
- RFC3398 - ISDN to SIP mapping
- RFC3420 - Internet Media Type message/sipfrag
- RFC3515 - REFER method
- RFC3550 - Realtime Transport Protocol (RTP)
- RFC3551 - RTP/AVP profile
- RFC3711 - The Secure Real-time Transport Protocol (SRTP)
- RFC3842 - Message Waiting Indication for SIP
- RFC3891 - SIP "Replaces" header
- RFC3892 - SIP Referred - By Mechanism
- RFC3951 - Internet Low Bit Rate Codec (iLBC)
- RFC3952 - Real-time Transport Protocol (RTP) Payload Format for internet Low Bit Rate Codec (iLBC) Speech
- RFC3960 - Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP)
- RFC4028 - Session Timers in SIP
- RFC4497 - Interworking between SIP and QSIG
- RFC4566 - Session Description Protocol (SDP)
- RFC4568 - SDP Security for Media Streams
- Draft: Diversion Indication in SIP (draft-levy-sip-diversion-08)

Enhanced routing

- Defines which CAPI controller is used for which calls from SIP
- Increased flexibility of load balancing and failover functionality; load balancing and failover can be used together and are available for calls to the PSTN as well
- Number-based routing also available for calls to the PSTN
- Matching rules for number-based routing can contain regular expressions
- Routing based on calling or redirected number, the redirected number is only available for calls from the PSTN

Enhanced address manipulation

- Define the number manipulation on three different stages of the call routing (inbound, route selection, outbound)
- Unlimited number of regular expressions for number manipulation at each stage of call routing
- Different dialplans can be entered for each controller and each SIP peer, which can ease the deployment in an environment with multiple locations

Dialogic® Diva® Media Board features

This list includes only features relevant for the Dialogic® 4000 Media Gateways. If you are interested in the complete list of features, consult the Dialogic® Diva® System Release Reference Guide available at: www.dialogic.com/manuals.

General features

- Support for the ISDN basic rate interface (BRI), the ISDN primary rate interface (PRI), the channelized E1 interface, and the channelized T1 interface
- Support for fractional PRI, E1, and T1 lines
- Support for multiple PRI, E1, and T1 lines
- Automatic Diva Media Board detection
- Support for ISDN lines with a transfer rate of 64 kbps or 56 kbps (for example some regions in the USA)
- Support for channelized T1 lines with a transfer rate of 56 kbps (see Channelized T1 (robbed bit signaling) on page 14)
- Support for unchannelized lines with a transfer rate of 64 kbps or 56 kbps
- Support for R2 signaling E1 lines with a transfer rate of 64 kbps
- Support for up to 120 B-channels
- Support for all known switch types (ISDN protocols)
- Support for Q.SIG protocol (see Q.SIG features on page 14)
- Additional security through support of RSA
- Dialogic® Diva® V-PRI Multiport Media Boards: Creation of a trace message in the trace file if maximum operation temperature is exceeded.
- Support for collecting phone number ranges or a specific number on incoming calls by the software.
- Support for a wide range of Windows® event logs. Driver and connection errors and informative messages are listed in the MOM (Microsoft® Operation Manager). For a detailed description of the errors and messages see the Dialogic® Diva® Configuration Manager Online Help file (DSMain.chm).
- Support for call deflection or call rerouting
- Support for redirecting number emulation (for incoming calls). In this case, the called party number is delivered as redirecting number to the application.
- ECT Link Balance: To avoid confusion with call transfer and multiple incoming calls, each incoming call is delivered to a separate TEI. This feature is only valid for Dialogic® Diva® BRI Media Boards and Point-to-Multipoint interfaces.

- Call Rate Limiter: Limitation of the amount of outgoing calls per second. Some switches may require limitation of calls in order to grant stability of the PSTN network.
- Support for DTMF transmission, DTMF detection, DTMF clamping

DSP-based features

- Real time protocol (RTP)
- Dynamic anti-jitter buffering
- Comfort noise generation (CNG)
- Voice activity detection (VAD)
- Support for 256 ms echo cancellation on all channels in parallel on Diva V-PRI Multiport Media Boards

Fax and modem features

Note: These features are available only after activating the corresponding license. See the Dialogic® 4000 Media Gateway Quickstart Guide for more information.

- TDM fax support, up to V.34 (33.600 bps and lower bit rates)
  - Support for Fax G3, T.30, V.34 HDX, V.17, V.29, V.27ter, V.21, V.34
  - Fax Compression MH, MR, MMR
  - Error Correction Mode ECM
  - Fax Polling
  - Reversal Fax Direction
  - Fax Password, Sub Addressing, 'new header line'
  - Page Formats A4, B4, A3
  - Resolutions fine, super fine, ultra fine
  - Color Fax JPEG format
  - T.38 FoIP (PSTN - IP Gateway mode)
  - Support for color fax via CAPI (JPEG format; sending and receiving single or multi-page documents; fallback to gray scale if remote side does not support color fax)
- Data modem support, up to V.90
  - All modem modulations POS up to V.90 (client and server side)
  - V.21, V.23, V.22, V.22bis, Bell 103, Bell 212A, V.32, V.32bis, V.34, V.90, including error correction MNP, V.42, SDLC and compressions V.42bis, MNP 5
  - POS modulations V.22 FC, V.22bis FC, V.29 FC
  - Text telephone modem: V.18, V.21, Bell 103, V.23, EDT, Baudot 45, Baudot 47, Baudot 50, DTMF
  - Extended modulations V.23 half duplex, V.23 on hook (SMSC mode), V.23 off hook, Bell 202 (POS), Telenot
- RAS (Remote Access Service) support
  - Connection to ISDN routers, enabling access to a remote LAN or the Internet
  - Network access for PPP-compatible clients
  - Connection to a Windows® server from digital, analog, and mobile networks with only one telephone number
  - Automatic detection of ISDN service, synchronous/asynchronous framing, and B-channel protocol of incoming calls
  - Synchronous/asynchronous conversion
  - Support for LAN protocols: TCP/IP, IPX/SPX, NetBIOS, NetBEUI, LAN Manager API

    - Support for ISDN B-channel protocols: HDLC, X.75, V.120, V.110, PIAFS 1.0 and 2.1, modem V.34+ and V.90, fax connections, V.42/LAPM (error correction), and V.42bis compression
    - Encryption, data compression, number checking, shorthold mode, callback function
- Modem emulation support
    - COM port for 16-bit Windows® applications
    - TAPI-compliant pre-initialized Dialogic® Diva® modems (Diva V.120 Modem (64K), Diva Fax Modem (Fax Class 1/ Fax Class 2), Diva Analog Modem)
    - Diva V.120 Modem (56K)
    - Diva V.110 Modem
    - Diva X.75 Modem (64K)
    - Diva X.75 Modem (56K)
    - Diva PPP-Modem (64K)
    - Diva PPP-Modem (56K)
    - Diva X.25 Modem
    - Diva Generic Modem (network access for PPP-compatible clients, automatic detection of ISDN service, synchronous/asynchronous framing and B-channel protocol, synchronous/asynchronous conversion, encryption, data compression, number checking, shorthold mode, callback function)

Q.SIG features

- Support for generic Q.SIG according to ECMA and ISO (For more information, see "Supplementary services" in the Dialogic® Diva® System Release Reference Guide available at: www.dialogic.com/manuals.)
- Tests have been conducted for the various switch types (for a complete list of all supported types see "Supplementary services" in the Dialogic® Diva® System Release Reference Guide available at: www.dialogic.com/manuals.)

Channelized T1 (robbed bit signaling)

- Trunk modes (loop, ground, and wink start)
- Tone dialing (DTMF and MF)
- Pulse dialing
- Ringer and busy tone detection
- 56 kbps transfer rate
- Call transfer

PBX Interoperability

Dialogic® 4000 Media Gateways are designed and tested for PBX interoperability with the installed base of enterprise communications systems. They are also tested and approved for use with Microsoft® Unified Communications. The general use configuration guides and PBX interoperability matrix at http://www.dialogic.com/microsoftuc/pbx_integration.htm provide guidance and configuration information for many different PBX vendors and models.

Dialogic® Diva® SIPcontrol™ Software Configuration

This section provides a detailed description of the Diva SIPcontrol software configuration parameters. Dialogic® 4000 Media Gateway-specific configuration examples are provided in Configuration Scenarios on page 62.

The Diva SIPcontrol software can be configured via the Diva SIPcontrol software web interface.

To open the Diva SIPcontrol software web interface:

1. Click **Start** > **Programs** > **Dialogic Diva** > **SIPcontrol Configuration**. By default, the access to the web interface is only allowed from localhost (127.0.0.1) and by default, the port number to which the server is listening is set to 10005.
2. If you need to access the configuration via remote access, you must set a password. To do so, open the main configuration web interface locally and click **Password** on the left hand side under **Configuration**. Enter a minimum 7 digit long password and confirm it. Click **Save** to make the new password active.
3. If necessary, open the port in the local firewall settings.

   To do so:

   - Click **Start** > **Settings** > **Control Panel** > **Windows Firewall**.
   - In the **Windows Firewall** dialog box, click the **Exceptions** tab and click **Add Port...**.
   - In the **Add a Port** dialog box, enter a name, e.g., Diva SIPcontrol, and enter the port number 10005. Select **TCP** as protocol and click **OK** to close the dialog box.

Now you may access the Diva SIPcontrol software web interface on any of the IP addresses of the computer where SIPcontrol is installed, and configure the settings according to your needs. The Diva SIPcontrol software configuration is divided into the following sections:

Mandatory configurations are:

1. Choose and enable one network interface.
2. Create and enable one SIP peer.
3. Create and configure one route for PSTN to SIP calls and another route for SIP to PSTN calls.

Before you start configuring, you might want to take a look at the Configuration Tips and Hints on page 16 that include useful information for the configuration.

Configuration Tips and Hints

- Changes to the configuration will only take effect after you click **Save** at the bottom of each configuration page.
- The settings will be lost if you close the Dialogic® Diva® SIPcontrol™ software web interface without having saved the configuration at the bottom of each configuration page.
- A restart of the Diva SIPcontrol software is recommended if you change the IP address or the port on which SIPcontrol is listening. If you do not restart, the Diva SIPcontrol software will continue listening on the previously configured port and IP address.

  Note: The restart will terminate active connections.
- The names for specific configuration elements are limited to 32 alphanumeric characters and must not be repeated, i.e., you cannot assign the same name for two SIP peers.
- The configuration session times out after 30 minutes of inactivity and a new login is required to access the session again. If the new login screen appears when you try to save the configuration, login again and click the "Back" button of the browser. The configuration session opens with the settings before the time out and you can save the configuration.
- To remove the password login page, logout from the web interface and restart the Dialogic® Diva® WebConfig service as described below. Then open Windows® Explorer, go to C:\Program Files\Diva Server\httpd\login, and delete the login file.
- To restart the Dialogic® Diva® WebConfig service, click **Start** > **Settings** > **Control Panel** > **Administrative Tools**. In the **Administrative Tools** window, select **Services**. In the **Services** window, right-click the **Dialogic Diva WebConfig** service and select **Restart**.
- The Diva SIPcontrol software provides a secure configuration via the web interface (HTTPS). The default port for the HTTPS is 10006. The Diva SIPcontrol software provides a default certificate, but for security reasons you should install your own webserver certificates. The location depends on the operating system:
  - Windows Server® 2003 (32-bit):
    Program Files\Diva Server\divawebconfig\cert
  - Windows Server® 2008 (64-bit):
    ProgramData\Dialogic Diva\divawebconfig\cert
- To use TLS for SIP calls, you need to upload the certificates as described under Security Profiles on page 30 and enable the TLS port as described under Network Interfaces on page 21.
- To open the online help for a specific parameter, click the parameter and a window with the help text will pop up.

PSTN Interfaces

This section describes the Diva SIPcontrol software's PSTN interface related settings, e.g., which lines are used by the Diva SIPcontrol software or how Call Transfer is performed on this line. Line Parameters such as the signaling protocols (Q.Sig, ETSI) can be configured on the **Board Configuration** page. For more information, see Dialogic® Diva® SIPcontrol™ Software Reference Guide available via the web interface under **Documentation**.

At least one PSTN interface must be enabled for the Diva SIPcontrol software to be able to work. Disabled PSTN interfaces are ignored for both inbound and outbound calls. For each line, you may select a dialplan that you can configure as described in Dialplans on page 32.

To change the settings for the enabled interface, click the **Details** button on the right hand side. To open the online help for a specific parameter, click the parameter and a window with the help text will pop up.

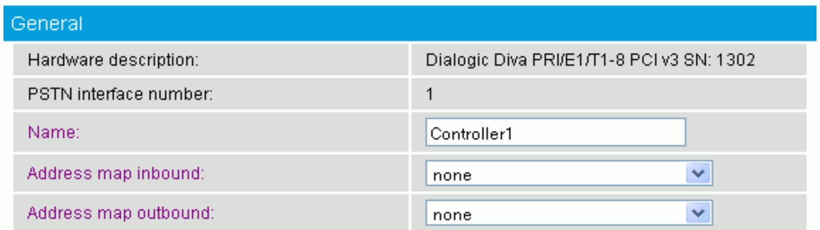
Note: PSTN interfaces without a binding to the CAPI service in the Dialogic® Diva® Configuration Manager are disabled in the Diva SIPcontrol software web interface and cannot be configured.

The following configuration menus are available for each Diva Media Board:

- General on page 17
- Enhanced on page 17
- Address Normalization on page 18
- PSTN Call Transfer Settings on page 19
- Message Waiting Indication (MWI) on page 20

General

You may configure the parameters shown in the graphic and explained in the table below:



Hardware description: Displays the installed Dialogic® Diva® Media Board. This entry is predefined by the system and cannot be changed.

PSTN interface number: Displays the number of the CAPI controller. The number is set automatically by the system.

Name: Displays the name of the installed Dialogic® Diva® Media Board. The name may be modified in order to display the purpose of the interface or the name of the PBX it is connected to.

Address map inbound: Select the name of a regular expression list to be applied on calls received on this interface. See Address Maps on page 35 for more information about setting up a regular expression list. If you upgraded from Dialogic® Diva® SIPcontrol™ Software version 1.5 or 1.5.1, an address map is automatically generated here to provide the same number processing behavior in the current Diva SIPcontrol software version as in former Diva SIPcontrol software versions. If you used regular expressions in Diva SIPcontrol software version 1.5.1, they will be included in this address map as well, unless they cannot be converted to the new scheme. In this case, the entry **<Use Windows Registry values>** is available. The Diva SIPcontrol software will then use the regular expressions defined in the registry keys that were used by Diva SIPcontrol software 1.5.1.

Regular expressions may be used to add or remove dial prefixes required by a PBX or to rewrite public phone numbers of different number ranges into a common format. See the Examples on page 65 for more information.

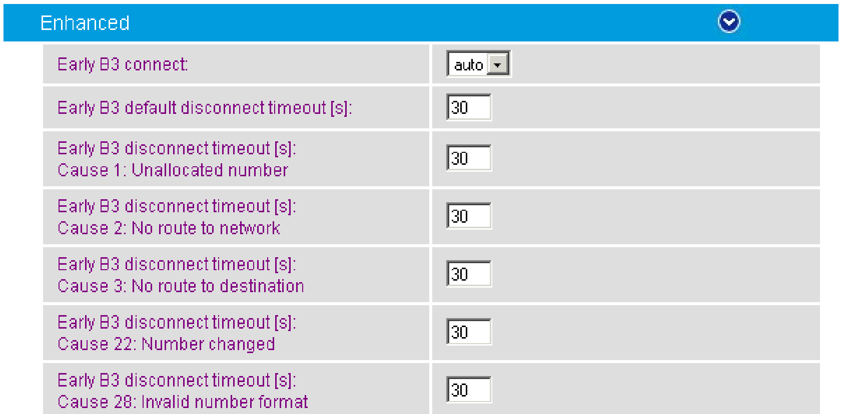
Address map outbound: Select the name of a regular expression list to be applied on calls sent out by this interface. See Address Maps on page 35 for more information about setting up a regular expression list. If you upgraded from Dialogic® Diva® SIPcontrol™ Software version 1.5 or 1.5.1, an address map is automatically generated here to provide the same number processing behavior in the current Diva SIPcontrol software version as in former Diva SIPcontrol software versions. If you used regular expressions in Diva SIPcontrol software version 1.5.1, they will be included in this address map as well, unless they cannot be converted to the new scheme. In this case, the entry **<Use Windows Registry values>** is available. The Diva SIPcontrol software will then use the regular expressions defined in the registry keys that were used by the Diva SIPcontrol software 1.5.1.

Regular expressions may be used to add or remove dial prefixes required by a PBX or to rewrite public phone numbers of different number ranges into a common format. See the Examples on page 65 for more information.

Enhanced

Here you may configure the settings for Early media support. Early media refers audio and video data that is exchanged before a session is accepted by the called user. It may be unidirectional or bidirectional, and can be generated by the calling party, the called party, or both. Typical examples of early media generated by the called party are ringing tone and announcements (e.g., queuing status). Early media generated by the calling party typically consists of voice commands or DTMF tones to drive interactive voice response (IVR) systems.

You may configure the parameters shown in the graphic and explained below:



Early B3 connect: With this parameter you can determine if early media should be enabled on this controller (EarlyB3) or whether early media should be enabled even if no "inband tones available" signal is received from PSTN (EarlyB3ForceMedia).

The following values determine if EarlyB3 and EarlyB3ForceMedia are enabled or not:

Value	EarlyB3	EarlyB3Force Media
auto	enabled	not enabled
on	enabled	enabled
off	not enabled	not enabled

The default value is **auto**.

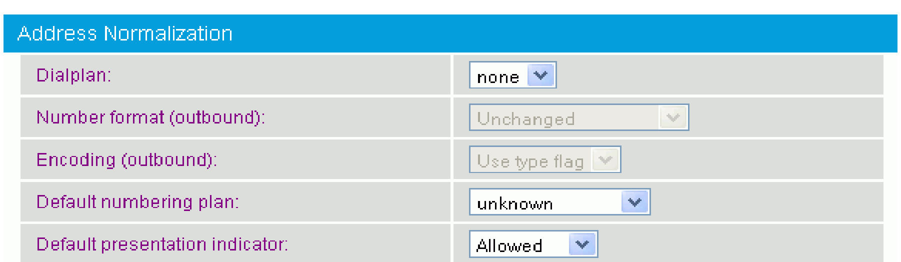
EarlyB3 default disconnect timeout [s]: Specifies the disconnect timeout value for early media calls to the PSTN, depending on the received cause value. The disconnect timer is released if a call to the PSTN is terminated before the receiver answers the call. This allows the caller to listen to a network announcement describing the reason for the failure (e.g., "The number you have dialed is not available. Please try again later.")

Default: **30** seconds

EarlyB3 disconnect timeout [s] Cause <x>:<reason for disconnect timeout>: With these parameters, you may define the disconnect timeout for the different disconnect timeout reasons. The default value for each reason is **30** seconds.

Address Normalization

You may configure the parameters shown in the graphic and explained below:



Dialplan: Select the local dialplan to be used by the dialplan module of the Dialogic® Diva® SIPcontrol™ Software. The selected dialplan applies only to this controller.

In most cases, the PSTN interfaces within the system share a common dialplan of the local environment, but configuring the dialplan per controller allows for handling variants, e.g., if the controllers are connected to different PBXs or if one controller is directly connected to the public network.

Configure the local dialplan as described under Dialplans on page 32 before you select it here.

Number format (outbound): This parameter determines the shortest format allowed in calls sent out by this interface. You may modify this parameter only if you selected a dialplan from the drop down menu. The following options are available:

Unchanged: The number signaled in the SIP message will be used unchanged for dialing.

International number: The number is always converted to an international number, including country and area code.

National number: The number is converted to a national number unless it is an international number with a different country code.

Extension: The number is reduced as possible. An internal number is reduced to its extension only.

For more information about number formats, see How Numbers Are Processed on page 63.

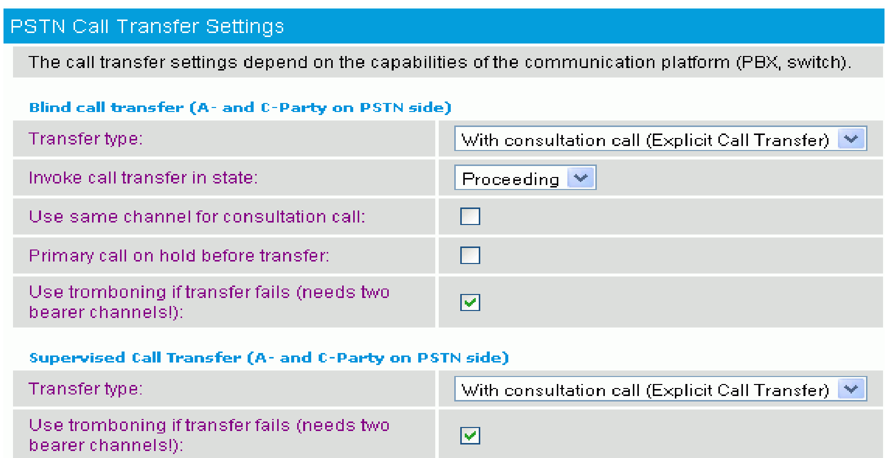
Encoding (outbound): Determines if numbers in calls sent out by this interface should either be encoded as unknown number with national or international prefix digits, or as national or international number with type flags.

Default numbering plan: Change this setting only if the PBX rejects calls from the Diva SIPcontrol software despite the dialed number being correct. This might occur if, for example, the signaled numbering plan is not supported.

Default presentation indicator: If no presentation is specified via address rewriting, the presentation indicator to set on calling party number for calls to ISDN. Select here, whether the calling party number should be shown or not.

PSTN Call Transfer Settings

Some Call Transfer options can be configured in the **Blind Call Transfer** section and in the **Supervised Call Transfer** section.



Transfer type: The following options are available:

Without consultation call (Call Deflection): The call is transmitted automatically.

With consultation call (Explicit Call Transfer): After the transfer to the destination party, the channel is freed. The transfer may be announced or unannounced.

With consultation call via tromboning: The call transfer is emulated. Two B-channels are blocked during the call transfer.

Complete transfer in state: The blind call transfer is typically handled via an implicit call to the transfer destination. Once this call reaches the state specified via the option **Invoke Call Transfer in state**, the call transfer is completed. Default setting is **Connected**. If the calling party should hear the ring back tone from the transfer destination, this parameter must be set to **Proceeding** or **Alerting**.

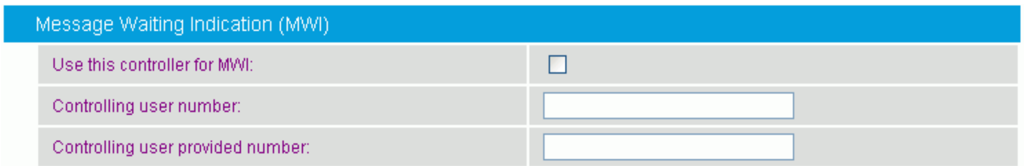
Use same channel for implicit call: The B-channel used for the primary call is used for the consultation call as well. This requires that the option **Hold primary call before transfer** is enabled. For Dialogic® Diva® Analog Media Boards and protocols using inband signaling, this option must be enabled.

Primary call on Hold before transfer: Select this option to place the primary call on hold before a call to the transfer destination is initiated.

Use tromboning if transfer fails (needs two bearer channels): Select this option if the Call Transfer should be emulated in case it could not be transferred with **Call Deflection** or **Explicit Call Transfer**.

Message Waiting Indication (MWI)

You may configure the parameters shown in the graphic and explained below:



Use this controller for MWI: The controller to use for MWI needs to be connected to a PBX port, which allows for updating of the message waiting indication.

Controlling user number: A PBX typically requests an authentication to allow for updating of the message waiting indication. This authentication is done by a **Controlling user number**. The administrator of the PBX can provide this number.

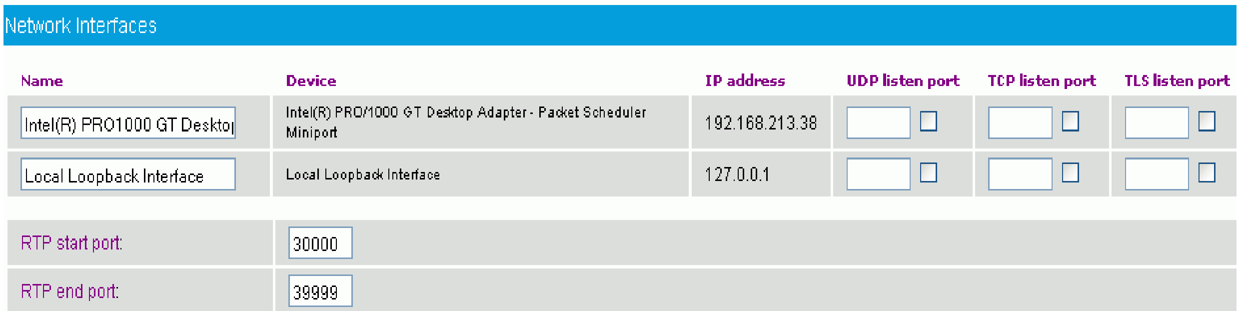
Controlling user provided number: The **Controlling user provided number** (CUPN) is the ISDN number provided by the controlling user, e.g., the ISDN number of the originating user of the indicated message. Few PBXs (e.g., Nortel) require the CUPN. The administrator of the PBX can provide more information.

Network Interfaces

The **Network Interfaces** configuration allows for configuring the global network parameters of the Dialogic® Diva® SIPcontrol™ software, such as the IP addresses and the ports on which the Diva SIPcontrol software will be listening. The Diva SIPcontrol software supports only a single IP address.

To open the online help for a specific parameter, click the parameter and a window with the help text will pop up.

You may configure the parameters shown in the graphic and explained below:



Name — Displays the name of the installed Ethernet adapter. The preset designation may be replaced with a unique identifier, such as "Internal Network".

Device — Displays the complete description of the installed Ethernet adapter assigned by the operating system.

IP address — Displays the IP address of the computer on which the Diva SIPcontrol software is installed.

UDP listen port — If you use UDP as IP protocol for calls from SIP, enable the check box to display the standard port number 5060. This standard port can be used if no other SIP application is running on the same computer as the Diva SIPcontrol software. Note that you may only enable one network interface.

TCP listen port — If you use TCP as IP protocol for calls from SIP, enable the check box to display the standard port number 5060. This standard port can be used if no other SIP application is running on the same computer as the Diva SIPcontrol software. Note that you may only enable one network interface.

TLS listen port — If you use TLS for encrypted calls, enable the check box to display the standard port number 5061. You may change the port number, but it must not be the same as the **TCP Listen Port** number. Note that you may only enable one network interface. If you use TLS, you need to upload security certificates and set the cipher level in the Security Profiles on page 30.

RTP start port — Defines the lowest port of the range in which the Diva SIPcontrol software sends and receives RTP streams. Change this value only if problems occur.

RTP end port — Defines the highest port of the range in which the Diva SIPcontrol software sends and receives RTP streams. Change this value only if problems occur.

SIP Peers

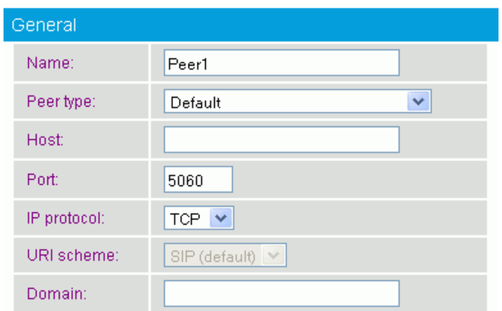
A SIP peer is a specific endpoint to and from which the Dialogic® Diva® SIPcontrol™ software will establish calls. The peer-specific settings may be used to adapt the Diva SIPcontrol software's behavior towards this peer.

To add a SIP peer, click the **Add** button. To change the settings for the enabled SIP peer, click the **Details** button on the right hand side. To open the online help for a specific parameter, click the parameter and a window with the help text will pop up. The following menus are available for configuration:

- General below
- Enhanced on page 23
- Security on page 25
- Session Timer on page 25
- Address Normalization on page 26
- Authentication on page 26

General

You may configure the parameters shown in the graphic and explained below:



Name: Enter a name for the SIP peer. A SIP peer is a specific endpoint to and from which the Diva SIPcontrol software establishes the calls.

Peer type: Some SIP peers need a specific peer, such as a Microsoft® Exchange Server, to work properly with the Diva SIPcontrol software. If this is the case for your configuration, select the specific SIP peer. If not, select **Default**.

Host: Enter the host name or IP address of the peer. The name must be resolvable by local name resolution. During the establishment of a call, the host name is sent by this peer exactly as entered here, unless an address map applies that converts the host name in a different format. For more information about name resolution, see the Windows® documentation.

Port: Displays the SIP port on which the remote peer is listening. The default is 5060, which is the standard port for SIP.

IP protocol: Select the IP protocol to be used for calls to this peer. Calls from this peer are accepted with all protocols and on all ports/addresses configured in Network Interfaces as described on page 21.

If you selected **MS Exchange 2007** or **MS OCS2007 / Mediation Server** as **Peer type**, set the protocol to **TCP**. If you selected **e-phone**, set the protocol to **UDP**.

URI scheme: This option is only available if you selected **TLS** as **IP protocol**.

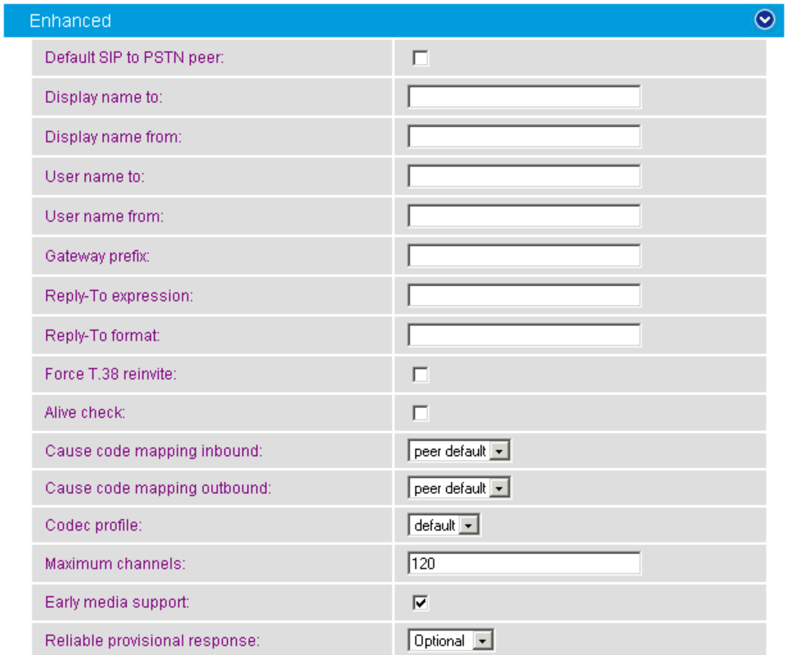
Calls are transmitted via various proxy servers. Some of them do not transmit the calls as encrypted calls. If you select **SIP (default)**, you allow that calls are transmitted via such proxy servers.

To make sure that a call is sent encrypted to the proxy of the remote side, select **SIPS** (secure SIP). If the call is routed via a proxy server that is not able to route the call encrypted, it rejects the call and the call is sent to another proxy until it can be transmitted.

Domain: Enter the domain name, e.g., dialogic.com, or the IP address. The domain name must comply with the DNS rules. The domain name entry here is only needed if the SIP peer does not use its hostname as source domain when it places a call.

Enhanced

You may configure the parameters shown in the graphic and explained below:



Default SIP to PSTN peer: Enable this option if the selected peer type should be used as default peer. Calls from unconfigured SIP peers will be assigned to this peer, and therefore are handled with these settings. If several peers are configured as default, the Diva SIPcontrol software takes the first to transmit the call.

Display name to: Enter the name that is to be sent in the "To" header of the INVITE message on calls from the PSTN to SIP.

Display name from: Enter the name that is to be sent in the "From" header of the INVITE message on calls from the PSTN to SIP. To send the calling party number include an asterisk (*) in the display name. For instance, if the display name is "Dialogic *" and the calling number is 123, then the remote side receives "Dialogic 123". To include an asterisk in the display name, enter "*". To include a backslash enter "\\".

User name to: You may enter a user name in front of the host name, e.g., thomas@dialogic.com. The user name is needed for the default route when no called party number is transmitted, e.g., for Dialogic® Diva® Analog Media Boards.

If a call from SIP does not contain a user name, the name entered here is transmitted as calling party number to the PSTN.

User name from: Enter the user name that is added to the SIP address when a number from the PSTN is suppressed. You may also enter the complete SIP address consisting of <username>@<local-IP/hostname>.

If a call from SIP does not contain a user name, the name entered here is transmitted as called party number to the PSTN.

Gateway prefix: You can configure this parameter only if you selected **e-phone** as **Peer type** in the **Edit SIP Peer Configuration** window.

This prefix is added at the beginning of the address in the "Reply-To" and "Contact" headers, which are copies of the "From" address. If this string is not empty, the parameter "phone-context" will be added in both headers.

Reply-To expression: You can configure this parameter only if you selected **e-phone** as **Peer type** in the **Edit SIP Peer Configuration** window.

Enter the expression that may be necessary for the e-phone server to handle the call. Normally, this is necessary to omit the 0 (zero) for external calls and to manipulate the address so the e-phone server is able to call back.

Reply-To format: You can configure this parameter only if you selected **e-phone** as **Peer type** in the **Edit SIP Peer Configuration** window.

Enter the format that may be necessary for the e-phone server to handle the call. Normally, this is necessary to omit the 0 (zero) for external calls and to manipulate the address so the e-phone server is able to call back.

Force T.38 reinvite: Some peers do not switch the media channel to T.38 if they receive a fax call, e.g., if they do not evaluate the fax calling tone. If you select this option, the Dialogic® Diva® SIPcontrol™ software tries to initiate the media channel switch.

Alive check: If you select this option, the failover procedure is expedited because the Diva SIPcontrol software does not wait for a call time-out if a peer does not respond. To achieve this, the Diva SIPcontrol software sends "pings" periodically to the peer via OPTIONS requests. If the peer does not send a valid answer, it will be treated as "inactive" and no calls will be routed to this peer until the peer responds to the "pings" again. In this case, the Diva SIPcontrol software will automatically direct calls to this peer again.

Cause code mapping inbound: Select the cause code mapping for calls coming from this SIP peer that you configured under Cause Code Maps as described on page 37.

Cause code mapping outbound: Select the cause code mapping for calls to this SIP peer that you configured under Cause Code Maps as described on page 37.

Codec profile: Select the codec list that you configured under Codec Profiles on page 38. If you do not select a list, an internal default list is used with the following default priority order:

1. G.711A
2. G.711u
3. G.726 (16, 24, 32, and 40 kbps)
4. G.729, if licensed
5. iLBC, if available on the used Dialogic® Diva® Media Board
6. GSM-FR
7. DTMF via RFC2833 (no real codec, but internally handled as codec)
8. T.38, if supported by the used Diva Media Board

In calls from SIP to the PSTN, the first codec of the PSTN device is applied that is also in the default codec list of the Diva SIPcontrol software.

Maximum channels: Specifies the number of channels that this SIP peer is able to handle at the same time. This setting is used by the Diva SIPcontrol software to distribute calls in a load-balancing scenario and to avoid speech quality degradation and/or call failures at the peer due to overload conditions.

Early media support: Specifies whether the peer supports early media for calls to PSTN. For non-human peers this should be disabled.

Reliable provisional response: SIP defines two types of responses, provisional and final. Provisional responses provide information on the progress of the request processing and final responses transmit the result of the request processing.

This parameter specifies whether reliable provisional responses (RFC3262) should be used.

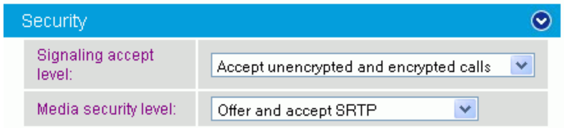
The following values are available:

Disabled: Reliable provisional response is not used.

Optional: Reliable provisional response may be used.

Required: Reliable provisional response is mandatory.

Security



You may configure the parameters shown in the graphic and explained below:

Signaling accept level: This parameter defines, how the call information should be accepted. To accept encrypted calls, you need to activate TLS as listen port in the Network Interfaces configuration.

Accept unencrypted calls only: Only signaling sent with TCP or UDP is accepted. Any encrypted signaling is rejected.

Accept encrypted and unencrypted calls: All calls are accepted, independent from the encryption mode.

Accept encrypted calls only: Only signaling with TLS is accepted; unencrypted signaling is rejected.

Accept encrypted call with SIPS URI only: Only signaling encrypted with the URI scheme secure SIP is accepted. Calls sent with TLS encryption are rejected.

Media security level: The Secure Real-time Transport Protocol (SRTP) authenticates packets and encrypts data and thus adds security to the voice stream. SRTP should be used together with TLS.

No SRTP: The voice stream is not secured with SRTP.

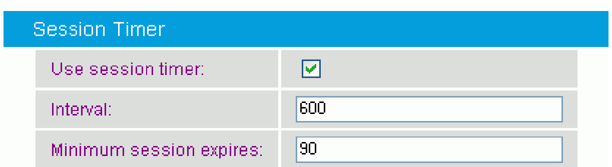
Offer and accept SRTP: The voice stream is secured with SRTP, if possible.

Require SRTP for encrypted calls: Calls via TLS need to use SRTP, otherwise they are rejected.

Note: If you select **Require SRTP for encrypted calls**, calls without SRTP are still allowed via UDP or TCP, unless **Signaling accept level** does not allow calls via UDP or TCP.

Session Timer

You may configure the parameters shown in the graphic and explained below:



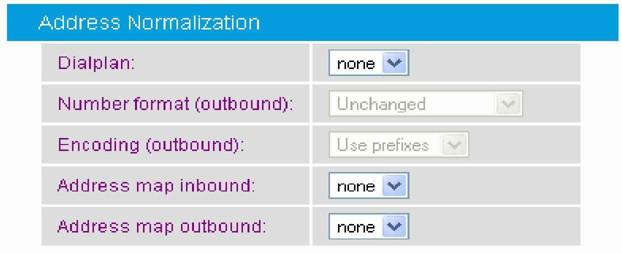
Use session timer: Activates session monitoring via SIP session timers using the time-out values given here. Refer to RFC4028 for details.

Interval: If **Use session timer** is enabled, you may set a time-out in seconds until a call is considered to be aborted. Refreshes are normally performed after the first half of the interval has elapsed. The minimum value is 90 seconds. The default value is 600 seconds.

Minimum session expires: If **Use session timer** is enabled, you may set a time in seconds between two session refresh messages that the Diva SIPcontrol software will accept. The minimum value is 90 seconds.

Address Normalization

You may configure the parameters shown in the graphic and explained below:



Dialplan: Select the local dialplan to be used by the dialplan module of the Dialogic® Diva® SIPcontrol™ Software. Configure the local dialplan under Dialplans as described on page 32 before you select it here.

The dialplan selected here applies only to outgoing calls.

Number format (outbound): This parameter determines the shortest format allowed that is sent in calls to this SIP peer. You may modify this parameter only if you selected a dialplan from the drop down menu. The following options are available:

Unchanged: The number signaled in the SIP message will be used unchanged for dialing.

International number: The number is always converted to an international number, including country and area code.

National number: The number is converted to a national number unless it is an international number with a different country code.

Extension: The number is reduced as much as possible. An internal number is reduced to its extension only.

For more information about number formats, see Number Processing With The Dialogic® Diva® SIPcontrolTM Software on page 50.

Encoding (outbound): Determines if numbers in calls to this SIP peer should either be encoded as unknown number with national or international prefix digits or as national or international number with type flags.

Address map inbound: Name of the regular expressions list applied to the addresses received on calls from this SIP peer. See Address Maps on page 35 for more information about setting up a regular expression list.

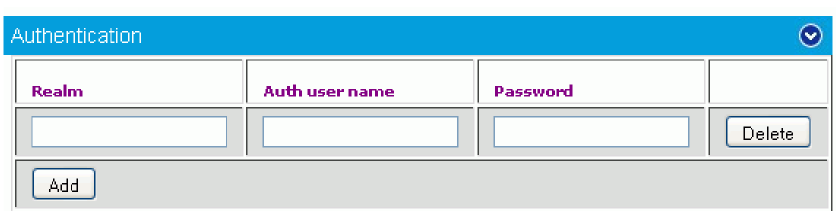
Regular expressions may be used to add or remove dial prefixes required by a PBX or to rewrite public phone numbers of different number ranges into a common format. See the Examples on page 52 for more information.

Address map outbound: Select the name of a regular expression list to be applied on calls to this SIP peer. See Address Maps on page 35 for more information about setting up a regular expression list.

Regular expressions may be used to add or remove dial prefixes required by a PBX or to rewrite public phone numbers of different number ranges into a common format. See the Examples on page 65 for more information.

Authentication

You may configure the parameters shown in the graphic and explained below:



Realm: A realm is a protection domain with its own user names and passwords. Enter the realm used by the SIP peer for authentication. The realm entered here needs to be the same as the realm of the endpoint.

Auth User Name: Enter a user name to be used with this realm.

Password: Enter the password to be used with this realm.

Routing

The **Routing** configuration defines the destination to which incoming calls are forwarded. Possible criteria that may determine the destination are:

- called, calling, and redirected number or SIP address of a call for which the redirected number is only available for calls originating in the PSTN,
- the source where a call originated, i.e., a PSTN interface name or a specific SIP peer,
- the current channel allocation across a set of several possible destinations in a load-balancing environment, and
- the current status of a destination. See Call Processing With The Dialogic® Diva® SIPcontrolTM Software on page 46 for more information.

To add a routing, click the **Add** button. To change the settings for the enabled routing, click the **Details** button on the right hand side. Since routes are processed in their configured order, the first matching route takes the call. To change the order, click the "arrow up" and "arrow down" buttons. To open the online help for a specific parameter, click the parameter and a window with the help text pops up.

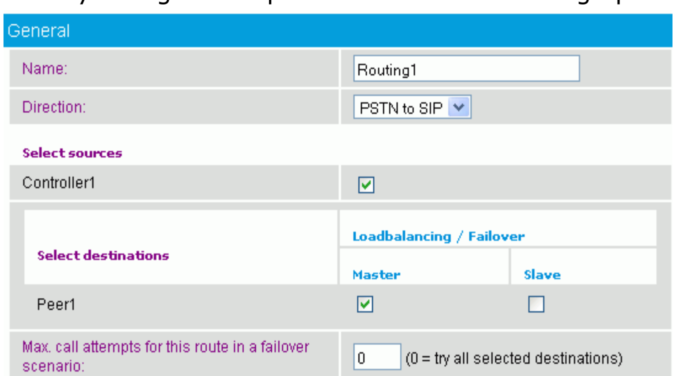
For more information about possible routing configurations, see the Routing examples in the Dialogic® Diva® SIPcontrol™ Software Reference Guide.

The following menus are available for configuration:

- General below
- Address Normalization For Condition Processing (Using Source Dialplan) on page 28
- Conditions on page 28
- Address Manipulation on page 29

General

You may configure the parameters shown in the graphic and explained below:



Name: Enter a unique name for the route, e.g., "Calls to MS Exchange Server".

Direction: Select if this route is for calls from SIP to PSTN or vice versa.

Select Sources: Depending on the selected direction, this part either lists the configured PSTN interfaces or SIP peers. The route will only be considered for a call if the call originated from a selected source.

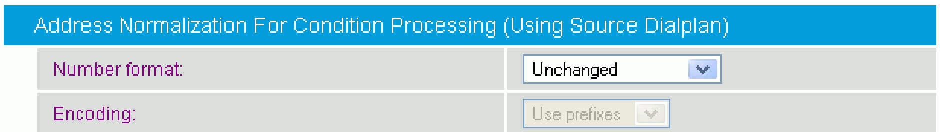
Note: A source may be selected even if it is currently disabled. In this case, the call will already have been rejected before the route is queried. At least one source interface is required for the route.

Select Destinations: You may select the possible destinations for the route, i.e., the set of CAPI controllers or SIP peers to which the call may be routed. The master or slave setting allows for configuring priorities. The Dialogic® Diva® SIPcontrol™ Software will always try to establish a call to one of the masters first and considers the slaves only if all masters have failed or could not accept calls due to their call load.

Max. call attempts for this route in a failover scenario: Enter the number of times the Diva SIPcontrol software should try to call the recipient in a failover environment. If you enter 0 (zero), the Diva SIPcontrol software tries all selected destinations. A value of 1 disables the failover functionality and tries only the first destination of a route.

Address Normalization For Condition Processing (Using Source Dialplan)

You may configure the parameters shown in the graphic and explained below:



Number format: This parameter determines the shortest format allowed in calls using this route. You may modify this parameter only if you selected a dialplan from the drop down menu. The following options are available:

Unchanged: The number signaled in the SIP message will be used unchanged for dialing.

International number: The number is always converted to an international number, including country and area code.

National number: The number is converted to a national number unless it is an international number with a different country code.

Extension: The number is reduced as much as possible. An internal number is reduced to its extension only.

For more information about number formats, see Number Processing With The Dialogic® Diva® SIPcontrolTM Software on page 50.

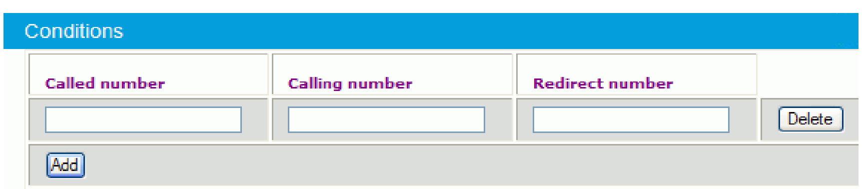
Encoding: Determines if numbers in calls using this route should either be encoded as unknown number with national or international prefix digits or as national or international number with type flags.

Conditions

You may configure certain conditions for a route. If you do not configure any conditions, the route is used as default route.

Note: If prefixes need to match, the digits of the prefix need to be prepended by a caret symbol ("^"); otherwise, these digits would match within the number as well, e.g. 0 would also match 1230@sipcontrol.com.

You may configure the parameters shown in the graphic and explained below:



Called number: If the routing is supposed to be valid only for specific calls, enter the called party number to which the route should apply. The Dialogic® Diva® SIPcontrol™ Software compares the current called party number against the called number entered here. If they do not match, the Diva SIPcontrol software verifies the next routing until it finds a match.

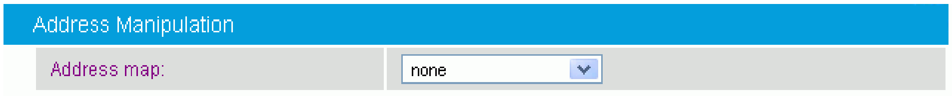
Calling number: If the routing is supposed to be valid only for specific calls, enter the calling party number to which the route should apply. The Diva SIPcontrol software compares the current calling party number against the calling number entered here. If they do not match, the Diva SIPcontrol software verifies the next routing until it finds a match.

Redirect number: If the routing is supposed to be valid only for specific calls, enter the redirecting number to which the route should apply. The Diva SIPcontrol software compares the current redirecting number against the redirect number entered here. If they do not match, the Diva SIPcontrol software verifies the next routing until it finds a match.

Note: A route can only be matched if the three condition parts (called number, calling number, and redirect number) match their call address counterpart in any of the lines. Empty condition entries always match, i.e., a line with the three condition parts left empty will always apply, thus working as a default route.

Address Manipulation

You may configure the parameter shown in the graphic and explained below:



Address Map: If a route matches, the address manipulation setting allows for modifying the call addresses according to your needs. For example, if calls with the called party number starting with "9" should be directed to a specific peer, it might be desirable to remove this digit. This can be done with a special address map configured. Note that you need to configure the address map as described under Address Maps on page 35 before you may select it here.

Security Profiles

When you use the Transport Layer Security (TLS) protocol for secure communication, you need to set various security settings.

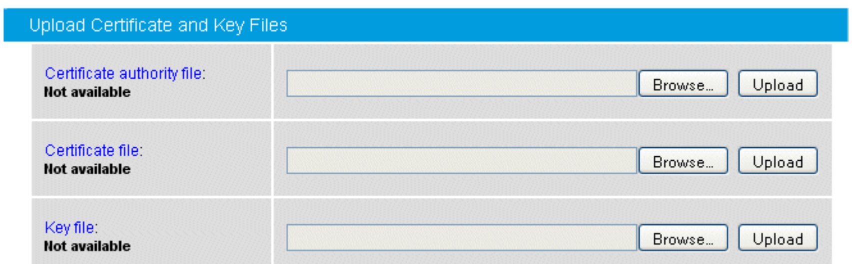
The following menus are available for configuration:

- Upload Certificate And Key Files below
- Global Security Parameters on page 31

Upload Certificate And Key Files

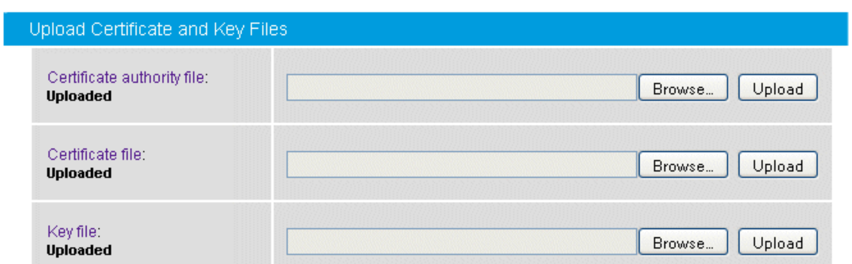
For authentication and data encryption, certificates need to be installed on the computer with the Diva SIPcontrol software and on remote computers. When a secure domain is opened, server and client authenticate each other with a so called "SSL handshake". With this handshake, the identity of a user is certified and the user can be trusted. All necessary certificates are provided by a Certificate Authority (CA) and they are issued for one domain name. For test purposes or internal usage, you can also create and sign your own self-signed certificate, e.g., with one of the many tools available on the internet, just google for "self-signed certificate" and you will find a list of possible tools. But you need to be aware that self-signed certificates do not provide the same security as CA-signed certificates. All files need to be in "pem" format, that means base-64-encoded.

The screen below shows the web interface with no certificates uploaded.



To upload a certificate:

1. Click **Browse**, in the **File Upload** window go to the folder where the certificate file is located, and click **Open**.
2. In the Diva SIPcontrol software web interface click **Upload**. After the certificates are uploaded web interface looks like this:



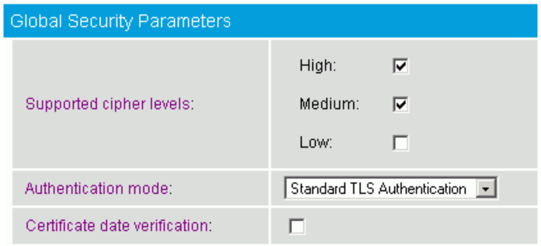
Certificate authority file: This file is the root certificate, which is used to sign a certificate. It is only needed for MTLS or TLS authentication.

With this file, the CA ensures that the public key contained in the certificate belongs to the server stated in the certificate.

Certificate file: This file is also generated from the CA and it contains the public key of the server on which the Diva SIPcontrol software is installed. This file is used for encrypting of information.

Key file: This file contains the private key for each endpoint, and it is used for decrypting of information. The key file must not be password protected.

Global Security Parameters



Supported cipher levels:

Cipher is an algorithm for encrypting and decrypting data. During the SSL handshake between client and server, the cipher level is negotiated. A low cipher level should only be used for systems that do not transmit any important information.

High: This currently means cipher suites with key lengths larger than 128 bits, and some with 128-bit keys.

Medium: Currently some suites using 128-bit encryption.

Low: Currently suites using 64- or 56-bit encryption algorithms but excluding export cipher suites.

Authentication mode:

Select how the server-client authentication should be handled.

Mutual Authentication: MTLS is used by Microsoft® Office Communications Server (OCS) 2007 Server roles and by Microsoft® Exchange 2007 UM role to communicate with each other. In this mode, both peers need to authenticate each other and both client and server exchange certificates.

For connecting to Microsoft® OCS 2007 R2 Mediation Server via TLS, use Standard TLS authentication mode. For a direct connection to Microsoft® Exchange 2007 UM role via TLS, use MTLS authentication mode.

Standard TLS Authentication: This is the normal authentication mode, in which the client asks the server for authentication to ensure a secure connection to the correct server.

No Authentication: In this mode, neither the server nor the client need to proof its authentication.

The default setting is: **Standard TLS Authentication**.

Certificate date verification:

If enabled, the expiration date of the peer certificate is verified. If the certificate is expired, an informational message is displayed and the call is aborted.

Dialplans

With help of the local phone settings, the Dialogic® Diva® SIPcontrol™ Software is able to convert a received call address to a normalized form, e.g., the E.164 format. This does not only ease the definition of subsequent conditions or maps, but it also converts the call to the format as required by the receiver.

The dialplan module supports the following features:

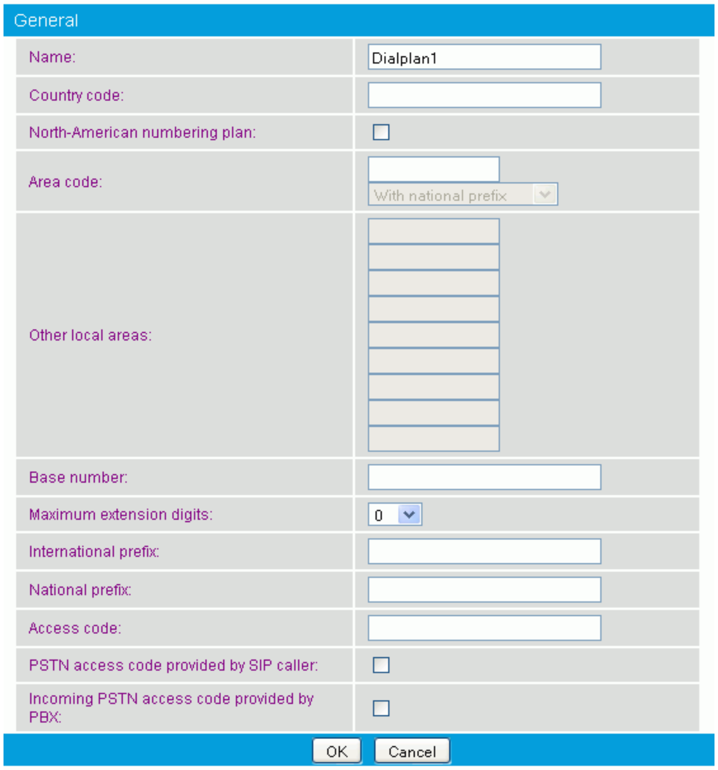
- Number expansion and reduction: called, calling, and redirected numbers are converted to one of the following formats: international, national, local, or internal (extension-only) format; for each format, either prefix digits or digital number type flags may be used.
- Adding and removing of the line access code: If not present, dialed numbers are automatically prepended by the digit(s) needed to get access to the public telephone network.
- Support for North American numbering plan: Up to 10 area codes may be configured to be treated differently. For example, in many areas dialing into neighboring areas requires to not dial a long-distance prefix.

Important information about the outside access digit configuration

- Configure the outside access digit only if there is a PBX between the PSTN and the Diva SIPcontrol software, and if this PBX requires the outside access digit for external calls. If you need to configure the outside access digit, also configure the following related options:
  - **Incoming PSTN access code provided by the PBX**: This option defines if the Diva SIPcontrol software expects the outside access digit in the calling number in external calls from the PBX.
    The PBX normally prepends the outside access digit to the calling number of incoming external calls in order to enable callback functionality at internal phones. If this is the case, enable this option.
  - **PSTN access code provided by the SIP caller**: This option defines if the Diva SIPcontrol software expects the outside access digit in the called number of external calls from SIP to the PSTN.
    It is normally required to prepend the outside access digit to call an external number from an internal phone, in this case, these are phones on the SIP side; however, in some configurations this is not required, such as a configuration that is part of the North American numbering plan (NANP), where an internal number can be identified based on its length. If it is possible to identify an internal call purely by the length of the called number, this option can be disabled. In all other configurations with outside access digits this option has to be enabled.
    It is recommended to have this option enabled in dialplans with the outside access digit set.
- The Diva SIPcontrol software's number normalization function does not remove outside access digits as a PBX can do for external calls. If the Diva SIPcontrol software needs to behave like a PBX with an outside access digit for external calls, use the Address Map functionality in combination with a Routing module.

To add a dialplan, click the **Add** button. To change the configuration settings, click the **Details** button on the right hand side. To open the online help for a specific parameter, click the parameter and a window with the help text pops up.

You may configure the parameters shown in the graphic and explained below:



Name: Enter a name to easily identify the dialplan, e.g., Stuttgart office.

Country code: Enter the country code without any prefixes of the country in which the computer with the installed Dialogic® Diva® SIPcontrol™ Software is located, e.g. 1 for US or 49 for Germany.

North-American numbering plan: Select this option if the North American numbering plan (NANP) is needed for your configuration. With the NANP, a city can have more than one area code, consequently it is not evident how to dial a number in the same city. The Diva SIPcontrol software allows you to enter various area codes that are considered local and should be called without long-distance prefix. See **Area code** and **Other local areas** for more information.

Area code: If you do not use the North American numbering plan (NANP), enter the area code without the leading zero here. If the NANP is needed for your configuration, enter the code for the home area here and enter the codes for the other local areas in **Other local areas**.

If you need to use NANP, you can choose between the following number transmission methods:

With national prefix: The long-distance code is added to the number.

Local: The number is transmitted without any area code.

Without national prefix: The number is transmitted without the long-distance prefix.

Other local areas: You may enter various area codes that are considered local and should be called without the long-distance prefix. This is the case in some countries where the North American numbering plan is deployed, e.g., in the USA. With the NANP a city can have more than one area code, consequently it is not clear how to dial a number in the same city.

Base number: Enter your subscriber or trunk number without country and area code. If you use MSNs, leave this field empty and enter the length of the MSNs in **Maximum extension digits**.

Maximum extension digits: Specify the maximum number of extension digits. Use the "arrow up" and "arrow down" buttons to do so.

International prefix: Enter the international prefix for your country, e.g., 00.

National prefix: Enter the digits of the national prefix, e.g., 0 in Germany.

Access code:	Enter the digits that are needed to get access to the public network, e.g., 9.
PSTN access code provided by the SIP caller:	Select this option if the SIP caller has to provide the access code. If the length of the called number is not sufficient to identify it as an internal number, activate this option to avoid ambiguous numbers. This is usually the case if you are not using the North American Numbering Plan (NANP).
Incoming PSTN access code provided by the PBX:	Select this option if the PBX adds the access code to the calling number for incoming external calls.

Address Maps

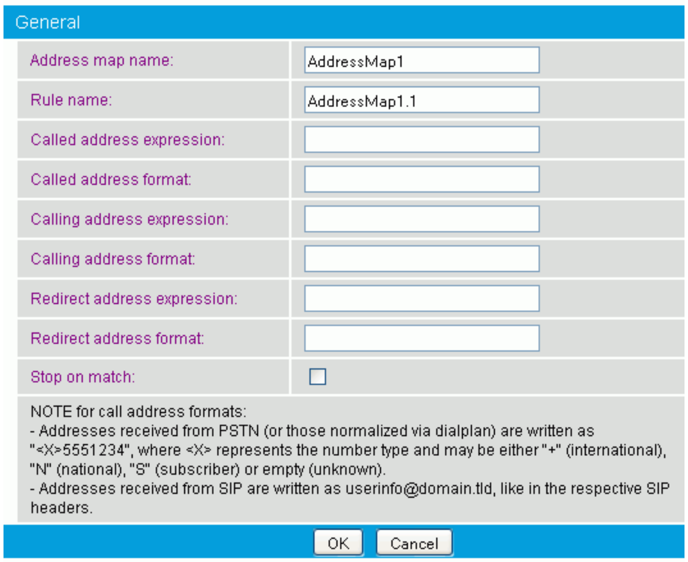
In general, address maps should be used for cases that are not covered by the dialplan. Possible scenarios are:

- set the calling number to that of the central office on SIP-to-PSTN calls,
- change the called extension to another value if an employee left,
- remove trunk prefixes while routing to a global voicemail server.

Each address map consists of a number of rules that are checked and applied from first to last until a matching rule is found that has the **Stop on match** option enabled. A rule matches only if all three expressions of that rule match. The order of the address maps is not important, but the order of the rules within a map is significant and can therefore be changed with the "arrow down" and "arrow up" buttons.

To add an address mapping, click the **Add** button. To change the settings for each address mapping, click the **Details** button on the right hand side. To open the online help for a specific parameter, click the parameter and a window with the help text pops up.

You may configure the parameters shown in the graphic and explained below:



Address map name: Enter a name for the address map that helps you remember the purpose of the map. This name is shown in other menus where an address map may be selected.

Note: The name may be edited only during the creation of a map.

Rule name: Enter a name for the rule of the map, e.g., "Remove 9 from all incoming calls".

Called address expression: If the regular expression entered here matches a called address, the format string is applied to the result. See Number Processing With The Dialogic® Diva® SIPcontrolTM Software on page 50 for more information on regular expressions.

Called address format: If the address format entered here matches a called address, the format string is applied to the result. See Number Processing With The Dialogic® Diva® SIPcontrolTM Software on page 50 for more information on regular expressions and formats.

Calling address expression: If the regular expression entered here matches a calling address, the format string is applied to the result. See Number Processing With The Dialogic® Diva® SIPcontrolTM Software on page 50 for more information on regular expressions.

Calling address format: If the address format entered here matches a calling address, the format string is applied to the result. See Number Processing With The Dialogic® Diva® SIPcontrolTM Software on page 50 for more information on regular expressions and formats.

Redirect address expression: If the regular expression entered here matches a redirected address, the format string is applied to the result. See Number Processing With The Dialogic® Diva® SIPcontrolTM Software on page 50 for more information on regular expressions.

Redirect address format:	If the address format entered here matches a redirected address, the format string is applied to the result. See Number Processing With The Dialogic® Diva® SIPcontrolTM Software on page 50 for more information on regular expressions and formats.
Stop on match:	If all expressions match all addresses of a call, this flag determines if the Dialogic® Diva® SIPcontrol™ Software should continue to search for matching rules. If set, the address matching is aborted.

Note: If expressions should match from the beginning, prepend the caret symbol ("^") at the beginning of the expression, for example:

Number: 1234567

Expression: ^123

Format: 4567

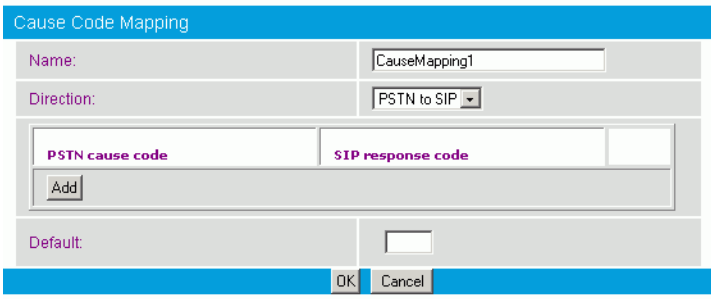
Result: 45674567

Cause Code Maps

Depending on the type of SIP peer selected, different default mapping tables are used to adapt SIPcontrol's responses to the values expected by that peer.

If the internal default mapping table provided by the Dialogic® Diva® SIPcontrol™ Software does not fulfill your needs, e.g., because your local PBX uses non-standard cause codes, you may configure your own cause code mapping table, which will be checked before the default table is. See Cause Code Mapping With The Dialogic® Diva® SIPcontrolTM Software on page 54 for the cause/response code mapping table. If you create your own cause code mapping table, make sure to select it in the **SIP Peer Configuration** under Enhanced.

To add a cause code, click the **Add** button. To change the settings, click the **Details** button on the right hand side. To open the online help for a specific parameter, click the parameter and a window with the help text will pop up. You may configure the parameters shown in the graphic and explained below:



Name — Enter a name to easily identify the cause code mapping table.

Direction — Select the direction for which this table is used:

- Select **PSTN to SIP** to configure mappings of PSTN cause codes to SIP response codes. This mapping is used if a call from a SIP endpoint to a PSTN endpoint cannot be completed.
- Select **SIP to PSTN** to configure mappings of SIP response codes to PSTN cause codes. This mapping is used if a call from a PSTN endpoint to a SIP endpoint cannot be completed.

PSTN Cause Code — Enter the PSTN cause code equivalent to the SIP response code entered in this menu. The PSTN cause code is also known as Q.850 cause code. Valid values are 1 to 127.

SIP Response Code — Enter the SIP response code equivalent to the PSTN cause code entered in this menu. The values are only valid in the range from 400 to 699.

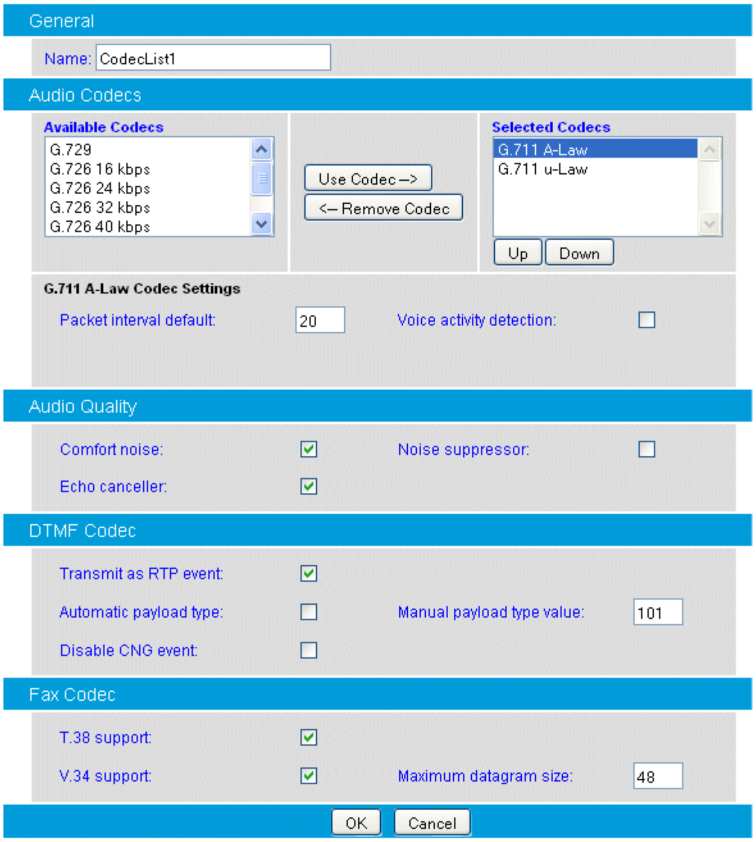
Default — Enter the cause or response code that the Dialogic® Diva® SIPcontrol™ software should use per default if no mapping for the received cause or response code is specified in this table.

Note: If this value is not configured and no mapping for the received cause or response code is specified in this table, the Diva SIPcontrol software's internal default mapping table will be used.

Codec Profiles

To configure the codec list, click the **Add** button. To change the settings, click the **Details** button on the right hand side. If you create a codec profile, make sure to select it in the **SIP Peer Configuration** under Enhanced.

To open the online help for a specific parameter, click the parameter and a window with the help text will pop up. You may configure the parameters shown in the graphic and explained below:



Name: Enter a name to easily identify the codec list. You may select the codec list in the SIP Peer configuration.

Available Codecs: This list includes all available codecs. If you want to use a certain codec, select it and click **Use Codec**. The codec will be moved to the **Selected Codecs** list. The G.729 codec can only be used after you have purchased and activated a license. See **License Activation** in the Dialogic® 4000 Media Gateway Series Quickstart Guide for more information.

Selected Codecs: By default, the G.711 A-law and G.711 μ-law codecs are selected. If you want to delete a certain codec, select it and click **Remove Codec**. The codecs are used according to their position in the list, with the first codec being the first to be used. To change the order, use the **Up** and **Down** buttons.

Packet interval default: Interval between RTP packets in an RTP stream. Also known as packetization time or RTP frame size.

Voice activity detection: If you activate voice activity detection, silence during a conversation is detected and the data rate is reduced.

Comfort noise support: If you enable the comfort noise feature and the voice activity detection (VAD) is active on your system, packets with low artificial background noise are sent to fill periods of total silence. Among others, total silence in digital transmissions can have the unwanted effect that the called party may think that the transmission has been lost and hang up prematurely.

Noise suppressor: Enable this parameter if you want to use the noise suppressor functionality.

Echo canceller: If you enable this parameter, the audio echo canceller is active.

Note: The echo canceller is activated as long as only one used codec has this parameter enabled.

Transmit DTMF as RTP event:	With RTP events, DTMF and fax tones can be sent and received as digital notifications instead of audio signals.
Automatic payload type:	G.726, iLBC, and DTMF have a dynamic RTP payload. If you select this option, the Dialogic® Diva® SIPcontrol™ software sets the values automatically. Only if the endpoint cannot handle the automatically set value, enter it manually under **Manual payload type value**.
Manual payload type value:	Some endpoints expect a certain payload type value. You can enter any value between 96 and 127. In calls from SIP to the PSTN, the Diva SIPcontrol software uses the value suggested by the endpoint. Generally, this parameter is left at its default value.
Disable CNG event:	This parameter defines which RTP events should NOT be supported locally, i.e., which events should be transported inband. Unsupported events are also negotiated as not available to the peer. Each bit set defines the respective event as being NOT supported, with bit 0 of the first array entry representing event 0, and bit 8 of the 32nd entry representing event 255. See http://www.iana.org/assignments/audio-telephone-event-registry/audio-telephone-event-registry.xhtml for a list of possible events. Only events 0-15 (DTMF "0"-"9", "*", "#", "A"-"D"), and 36 (CNG) are supported; all other event bits are set to 1.
T.38 Support:	T.38 is a protocol that enables fax transmissions on the IP network in real time. Enable this option if T.38 fax should be supported. Note that this feature is supported on Dialogic® Diva® Media Boards with multiple ports only after activating the respective license. See **License Activation** in the Dialogic® 4000 Media Gateway Series Quickstart Guide for more information.
V.34 Support:	The V.34 fax transmission protocol allows facsimiles to be transmitted at a maximum speed of 33.600 bps. Enable this option if V.34 should be supported. Note that this feature is supported on Diva Media Boards with multiple ports only after activating the respective license. See **License Activation** in the Dialogic® 4000 Media Gateway Series Quickstart Guide for more information.
Maximum datagram size:	This value defines the maximum amount of data that can be transmitted in one T.38 packet. Some endpoints are limited to packets of a certain size. You can enter a value between 32 and 192. Default is 48 bytes.

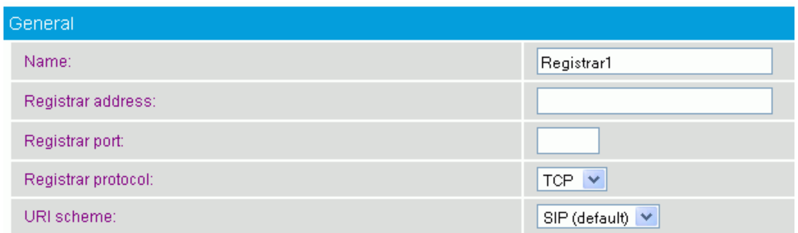
Registrations

SIP devices can communicate directly if the URL of both devices is known, but in general, SIP gateways are used in a network to enable functionalities such as routing, registration, authentication, and authorization.

Registration at a registrar server can be useful because in many cases, only the SIP address of a user is known but the location (SIP address of the device) is unknown or may change. A registrar server keeps track of the location of user agents from which the registrar server has received REGISTER requests. Thus, only the SIP address of the user needs to be sent to the registrar server, which then returns one or more contact addresses of the user.

If the Dialogic® Diva® SIPcontrol™ Software is configured to use a registrar server, it registers with the server as soon as it is active. Thus, all local addresses configured for registration are registered with the server. You may use either a private registrar service or a public registrar server.

To configure a registrar server, click the **Add** button. To change the settings, click the **Details** button on the right hand side. To open the online help for a specific parameter, click the parameter and a window with the help text will pop up. You may configure the parameters shown in the graphic and explained below:



Name: Enter a name for the registrar configuration.

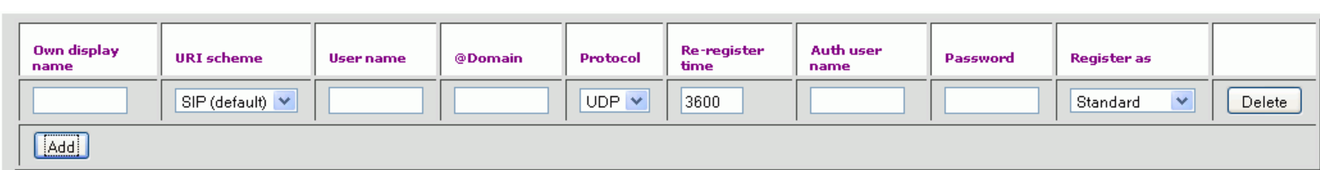
Registrar address: Enter the IP address or the hostname of the registrar server.

Registrar port: Enter the port number of the registrar server. Usually, the registrar server is listening on port 5060.

Registrar protocol: Select the protocol the registrar server uses.

URI scheme: This option is only available if you selected **TLS** as **Registrar protocol**.
Calls are transmitted via various proxy servers. Some of them do not transmit the calls as encrypted calls. If you select **SIP (default)**, you allow that calls are transmitted via such proxy servers.
To make sure that a call is sent encrypted to the proxy of the remote side, select **SIPS** (secure SIP). If a call is routed via a proxy server that is not able to route the call encrypted, it rejects the call and the call is send to another proxy until it can be transmitted.

To configure the settings for each user that should register at the same registrar server, click **Add** and configure the following parameters:



Own display name: Enter the name that should be displayed at the registrar server.

URI scheme: Select either **SIP(default)** or **SIPS** as URI scheme.

User name: Enter the name or number that the Diva SIPcontrol software uses to register at the registrar server.

Domain: Enter the domain name of the registrar server.

Protocol: Select **UDP** if you register as e-phone gateway.

Re-register time: Enter the re-register time in seconds. This is the time the registration to the registrar server remains valid. After this time has elapsed, the SIP stack service would need to re-register to be available again. The default value is 3600 seconds.

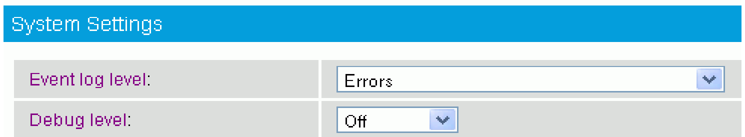
Auth user name: Enter a user name for authentication at the registrar server.

Password: Enter your password for authentication at the registrar server.

Register as: Leave the setting at the default value **Standard**. Select **e-phone GW** only if you use e-phone and you want the Dialogic® Diva® SIPcontrolTM Software to function as gateway for e-phone.

System Settings

You may configure the parameters shown in the graphic and explained below:



Event Log Level: A computer with the Diva SIPcontrol software installed may write different types of events into the System Event Log. The details for each event log are described in Event Logging With The Dialogic® Diva® SIPcontrolTM Software on page 60.

Debug Level: The debug level setting may be used for debugging and tracing purposes. During normal operation, it should be set to **Off** to lessen the effect on system performance.

Data Security Overview

Since version 2.0, the Dialogic® Diva® SIPcontrol™ software provides additional security options for transmitted and received data:

- Secure HTTP: You may use Secure HTTP (HTTPS) to transmit data between the web-based configuration interface of the Diva SIPcontrol software and your web browser.
- TLS: The Transport Layer Security (TLS) protocol may be used to encrypt and authorize SIP messages.
- Secure RTP: The Secure Real-time Transport Protocol (SRTP) may be used for encrypting the data of the actual conversation.

Note: The HTTPS and TLS protocols require digital identity Certificates (e.g., public key certificates).

Secure HTTP

HTTP is a protocol that transmits data between the web-based configuration interface of the Diva SIPcontrol software and your web browser. Even though the HTTP interface has access security (via a password), the transmitted data is not entirely secure. The data is transmitted as clear text and thus it is possible for the transmission to be intercepted and, in turn, for the data to be read.

HTTPS uses HTTP over an encrypted Secure Sockets Layer (SSL) or Transport Layer Security (TLS) connection and with a different default port than HTTP.

As an example, if a message containing a request to change a password was captured by a third party, the third party could log on to the Diva SIPcontrol software web interface and change the configuration. HTTPS encrypts and authenticates HTTP data, and thus the data is no longer transmitted as clear text and is not easily readable.

HTTPS requires two actions by the user:

- Both the Diva SIPcontrol software and the computer on which the web browser used to connect to the Diva SIPcontrol software via HTTPS is running must be configured with the proper certificate.
- When accessing the Diva SIPcontrol software web interface, use https:// instead of the non-secure http:// followed by the URL of the PC on which the Diva SIPcontrol software is installed.

TLS

SIP (Session Initiation Protocol) is a signaling protocol used for VoIP calls over the Internet. SIP messages contain information such as call-party information, call media type, whether it is a secure call, and if so, what encryption algorithm is used, etc. SIP can be carried by UDP, TCP, or TLS transports. Both UDP and TCP transport data in clear text. As a result, UDP and TCP can easily be monitored by a third party. TLS, on the other hand, carries SIP data in a secure way by encrypting the data and authenticating the transport connections. Authentication provides that you are talking to the intended peer. For authentication purposes, you need to install Certificates as described in Security Profiles on page 30 and enable TLS as transport protocol, as described in Network Interfaces on page 21.

Secure RTP

Once a Voice over IP (VoIP) call is established, voice data is transported in packets with the Real-time Transport Protocol (RTP). The voice data can be easily extracted from RTP packets and replayed using commercially available software. SRTP adds security by encrypting voice data and authenticating packets. Digital identity certificates are not required, the parameters are negotiated during call initiation time. SRTP mode is activated typically in combination with TLS, but in some cases (e.g., testing, intranet connections only) it is useful to allow SRTP also without TLS being activated.

For encryption and decryption of data, SRTP uses ciphers. The two parties involved in a conversation must be "compatible" in the sense that each party understands the other party's cipher requirements and supports them. The Diva SIPcontrol software supports the following ciphers: DH, ADH, AES (128-256 bits), 3DES (64 bits), DES (64 bits), RC4 (64bytes), RC4 (256 bytes), MD5, SHA1.

SRTP can be set for each SIP peer in the Security configuration, as described on page 25. The cipher level can be set in the Global Security Parameters as described on page 31.

Certificates

For authentication and data encryption, certificates need to be installed on the computer on which the Diva SIPcontrol software is installed and on remote computers. When a secure domain is opened, server and client authenticate each other with a so called "SSL handshake". With this handshake, the identity of a user is certified and it is assured that the user can be trusted. All necessary certificates should be provided by a Certificate Authority (CA), and they are issued for one domain name. For test purposes or internal usage, you can also create and sign your own self-signed certificate, but you need to be aware that self-signed certificates do not provide the same security as CA-signed certificates. Also, many web browsers check if the certificate is signed by a CA, and, if it is not, a warning message will pop up asking whether the user really wants to trust that web site, which can make the user feel insecure.

Using certificates with Microsoft® Office Communications Server (OCS) 2007

Microsoft® OCS 2007 requires that:

- server certificates contain one or more CRL (Certificate Revocation List) distribution points.

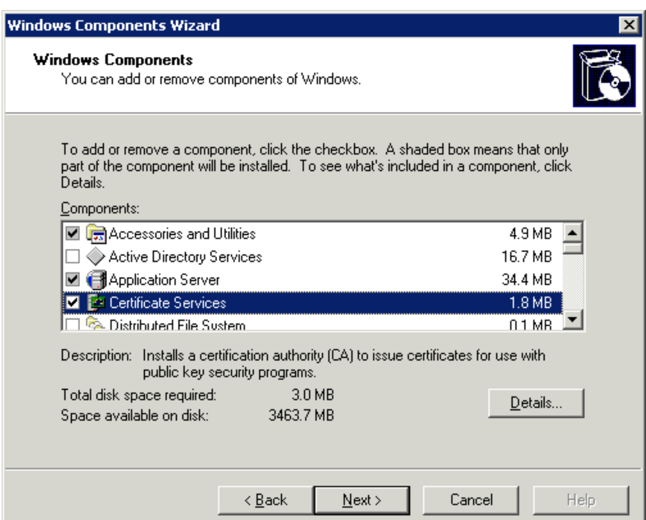
  CRL distribution points are locations from which CRLs can be downloaded to verify that the certificate has not been revoked since the time it was issued.
- server certificates support EKU (Enhanced Key Usage).

  EKUs are needed for server authentication and ensure that the certificate is valid only for the purpose of authenticating servers. This EKU is essential for MTLS (Mutual TLS).
- the gateway server certificate has a FQDN (Fully Qualified Domain Name) either in the Certification field CN (Common Name) / SN (Subject Name) or SAN (Subject Alternative Name) or both.

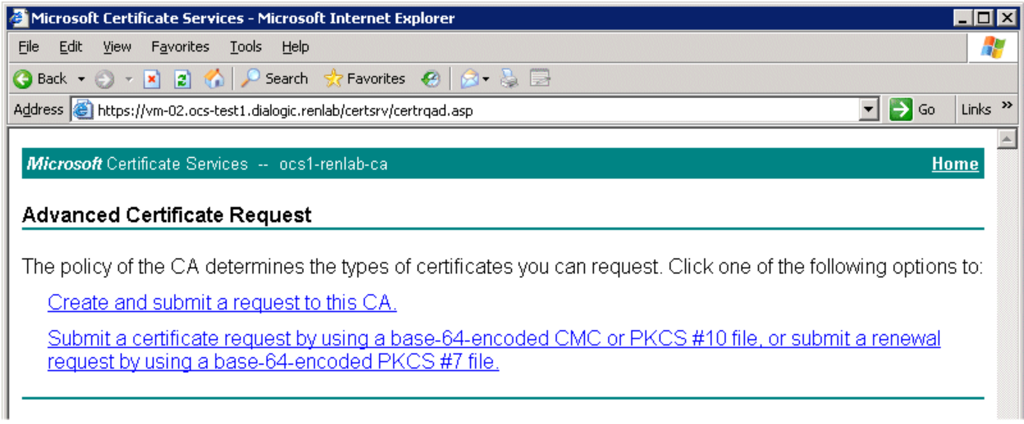
How to generate certificates using Microsoft® Certificate Services and upload them in the Diva SIPcontrol web interface

Microsoft® Certificate Services is a component of the Microsoft® Windows Server® operating system. On Microsoft® Windows Server® 2003, it can be installed through the Windows® Component Wizard.

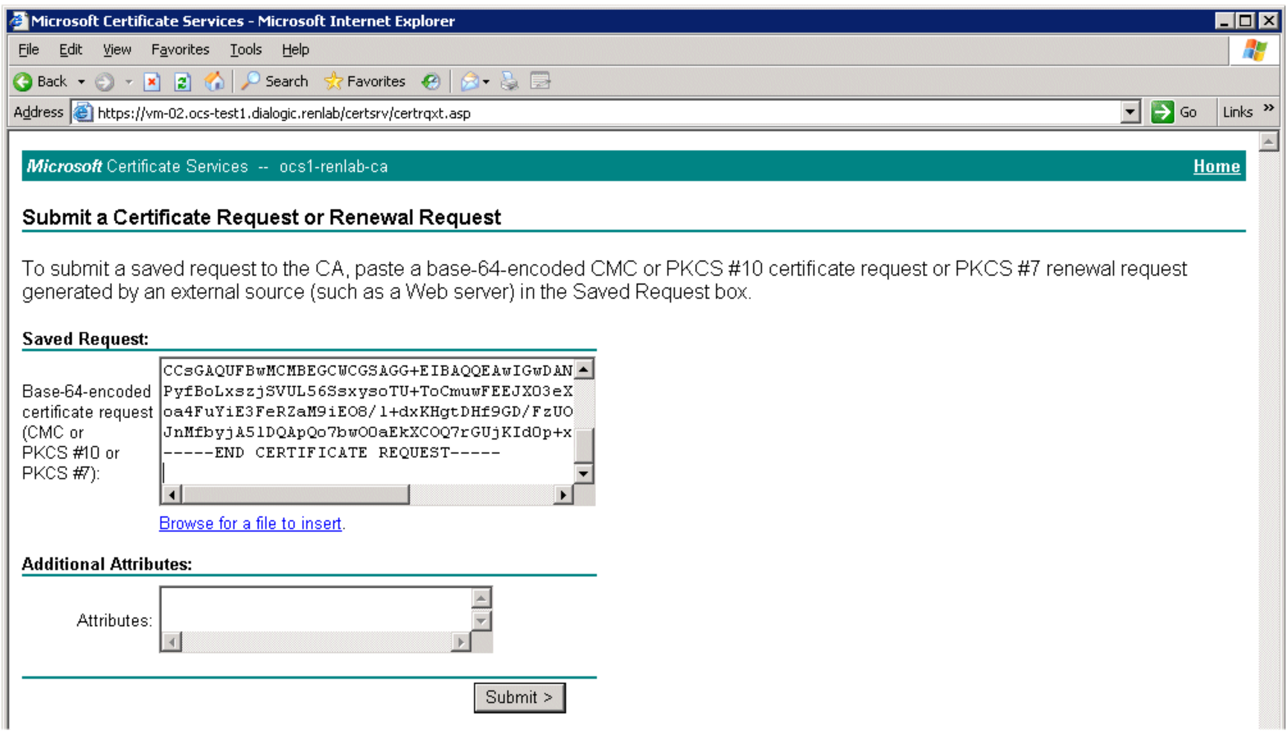
Note: Do not install the Microsoft® Certificate Services on your Dialogic® 4000 Media Gateway, but on a separate computer.



1. Create a key file and a certificate request with a third party program.
2. On your Microsoft® Certificate Services web site, got to **Advanced Certificate Request** and select the second option to submit a base-64-encoded request.

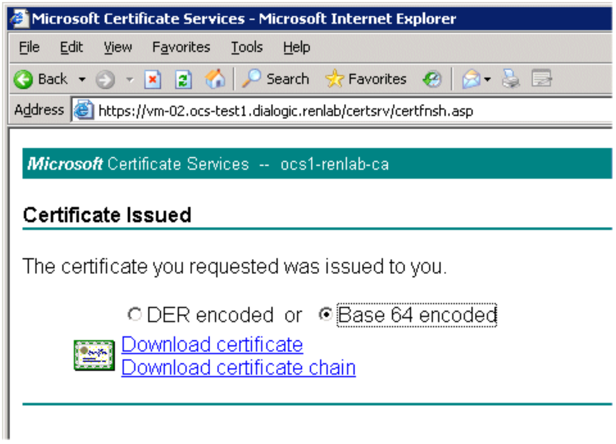


3. Open the key file with Wordpad, select the contents, paste it into the Microsoft® Certificate Services web site, and click **Submit**.

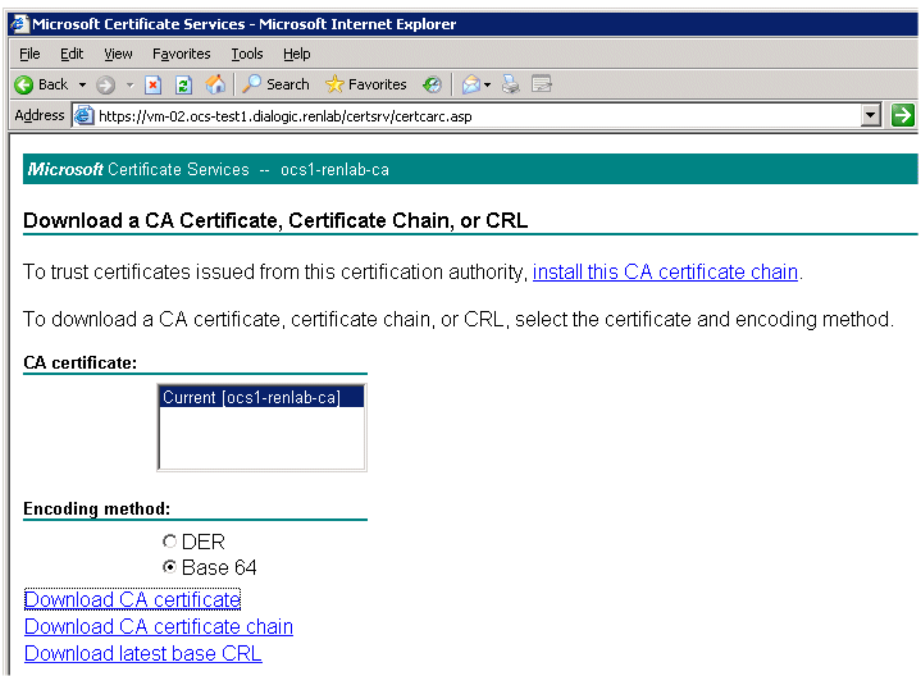


4. Go to the Microsoft® Certification Authority Management console and sign the certificate request.

5. Go to the Microsoft® Certificate Services download page for signed certificates, select **Base 64 encoded**, and click **Download certificate**.



6. Go to the Microsoft® Certificate Services download page for the CA certificate, select **Base 64**, and click **Download CA certificate**.



7. Upload the key file, the certificate file, and the certificate authority file. To do so, open the Diva SIPcontrol web interface and on the left hand side click **SIPcontrol configuration**.

8. Click **Security Profiles** and then **Details**.

9. Next to **Key file**, click **Browse**, go to the folder where you stored the file, and click **Open** to upload it.

10. Repeat the procedure in step 9 for the certificate file and the certificate authority file.

11. In **Authentication mode**, select how the server-client authentication should be handled.

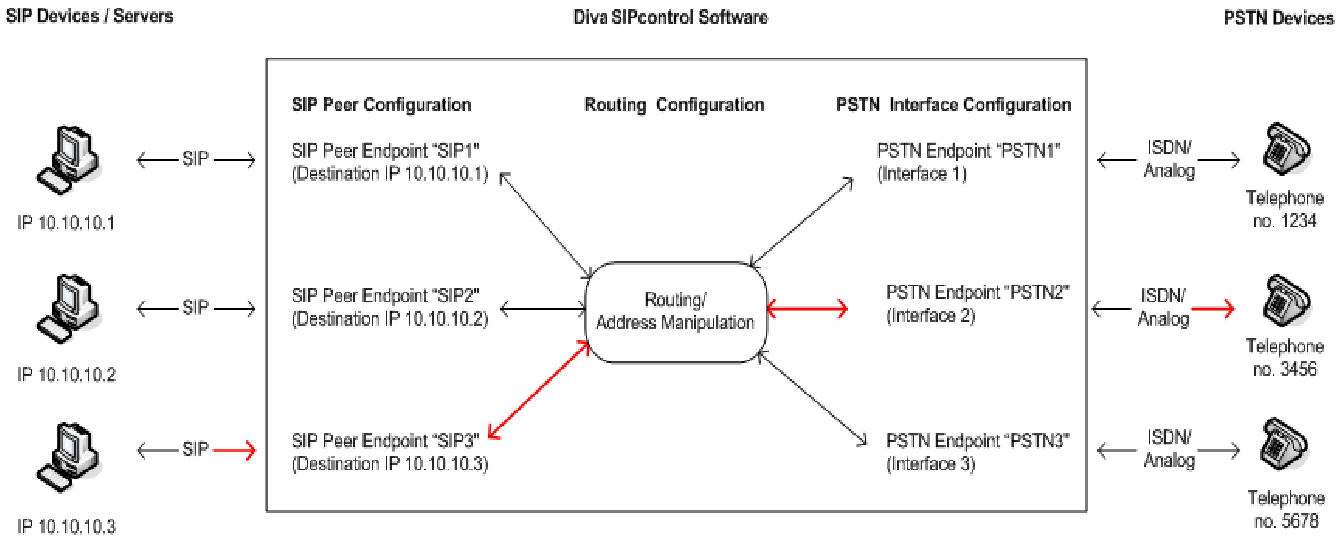
12. Click **OK** to close the **Security Profiles** window.

13. At the bottom of the Diva SIPcontrol web interface click **Save** to save the configuration.

14. Since changes in the Security Profiles require a restart, click **Service status** at the left hand side of the web interface and then click **Restart SIPcontrol**.

Call Processing With The Dialogic® Diva® SIPcontrol™ Software

The Dialogic® Diva® SIPcontrol™ Software uses an endpoint-based approach to process calls, which means that every PSTN interface and every configured SIP peer is considered as a single endpoint. The endpoint saves the Diva SIPcontrol software settings for the respective PSTN interface or SIP peer. Each call originates at a specific endpoint (on the SIP side after assigning the SIP call request to one of the configured peers) and needs a route to find its designated endpoint (the destination). Thus, the most simple configuration needs one PSTN endpoint, one SIP peer, and one route as shown in red in the graphic below.



This graphic shows that an endpoint is only a virtual object of a real device. The endpoint saves the settings for the corresponding device. For example, if a call should be routed from SIP device 3 to PSTN device 2 as marked red in the graphic, then:

- The settings of SIP device 3 need to be configured as SIP peer endpoint in the **SIP Peer Configuration**,
- the settings PSTN device 2 needs to be configured as PSTN endpoint in the **PSTN Interface Configuration**, and
- the condition "called address is 3456" needs to be configured in the **Routing Configuration** to route the call to the correct device.

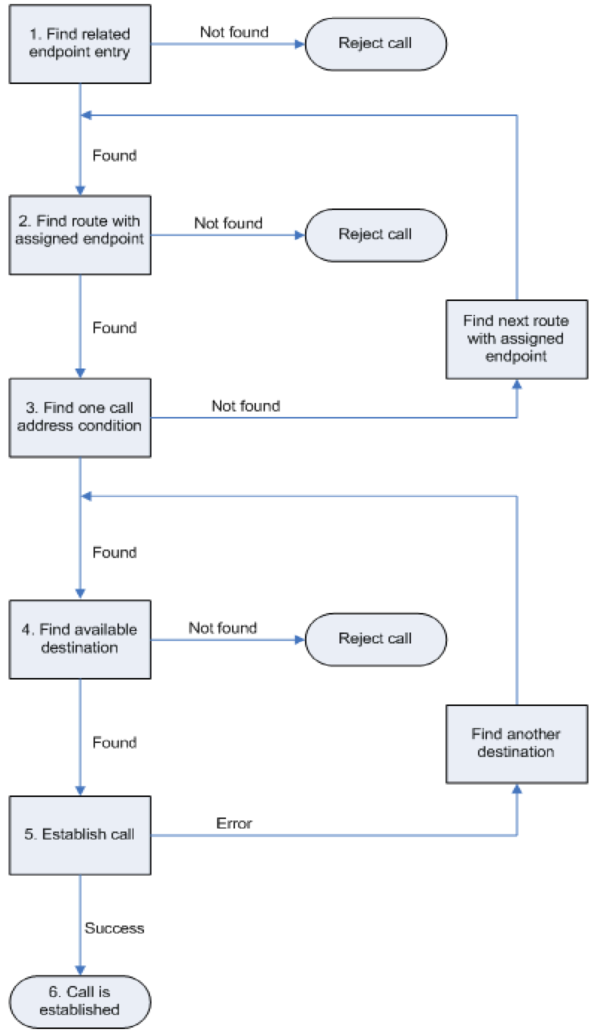
If you have for example a SIP or PSTN device 4 with no endpoints configured in the Diva SIPcontrol software, then you cannot establish a call, because the Diva SIPcontrol software will not know the settings of the device.

The PSTN endpoint is found via its controller number. On the SIP side, multiple SIP peers may connect via the same network interface. Therefore, the assignment is more complex:

1. The host/domain name and port number of the received "FROM" header is compared against the SIP peer settings.
2. If no host matches, the same address is compared against the "Domain" parameters of the SIP peers.
3. If no match is found, the Diva SIPcontrol software looks for a SIP peer with the **Default SIP to PSTN Peer** option enabled.
4. If the call cannot be assigned, regardless of whether the call originated in the PSTN or SIP network, the call is rejected.

Every route defines only one direction. Therefore, at least two routes are needed to support both PSTN-to-SIP and SIP-to-PSTN connections. The basic call (without address manipulation) is processed as follows:

1. Find and assign an endpoint for an incoming call request (PSTN: lookup by CAPI controller number; SIP: lookup by "From" address of received message).
2. Go sequentially through the list of routes and find the first route that has this endpoint defined in its configured sources list.
3. Determine whether at least one call address condition of this route matches simultaneously the called, calling, and redirected addresses of the call request; if not, find another route.
4. If any route condition matches, verify in the list of configured destinations which one is the most preferred. This is done based on settings. See Some things to know about call processing below for more information.
5. Try to establish the call via this destination. If the destination is unavailable or rejects the call, try the next destination of the route. Note that the call will be aborted immediately if a cause code is received that signals final failure, e.g., user busy or unallocated number.
6. The call is established.



Some things to know about call processing

- Each route may point to several destinations, between which the Diva SIPcontrol software chooses according to the following settings (in decreasing order of importance):
  - availability (destination enabled),
  - alive state of destination (if enabled to be verified),
  - priority (Master/Slave),
  - channel load quota (a factor calculated by comparing used vs. total supported channels).
- For each call only one route is chosen. Even if another route would also match the call criteria, only the first matching route is ever evaluated. Therefore, default routes should be created carefully.
- Load balancing/failover is only performed between the destinations of a single route.
- Routes without any conditions always match (as long as the source endpoint is listed in route sources).

Emergency calls

In many environments, certain numbers, e.g., 110/112 in Germany or 911 in the U.S., have to be handled differently from others. For example, they might need to be dialed without any access digit.

This can be achieved by creating an additional route from any configured SIP peers to one or more PSTN interfaces and setting the called address expression to the emergency number(s). The route should be placed at the top position in the list. Should there be a dialplan and/or address map configured for the respective PSTN interfaces, it may be necessary to add another regular expression to the address maps of the interfaces to handle those calls.

Routing conditions

The Dialogic® Diva® SIPcontrol™ Software organizes the conditions of a route in a list. Each list entry consists of different expressions for called, calling, and redirected address. The route matches only if all three expressions simultaneously match the respective call addresses. Empty expressions are considered to match, so there is no need to add wildcards into unused expressions. As a result, if a call should match either a called address or a calling number, two list entries have to be created, with called expression in the first and calling expression in the second row. If both have to match concurrently, both expressions have to be entered into the same list entry.

Call Address Processing With The Dialogic® Diva® SIPcontrol™ Software

The call addresses provided by the caller may be modified at different stages of the call processing within the Diva SIPcontrol software. The reason for multiple manipulation is that it allows for modifying the address where it is needed, which means that more complex environments can be configured with less effort since data does not need to be entered redundantly at different places. It also makes it easier to "team" SIP peers or PSTN interfaces with different settings.

The Diva SIPcontrol software converts addresses automatically, without any intervention from the user. This means that the Diva SIPcontrol software adds or removes a special prefix to a number with a known number type, e.g. "+" for international numbers, when converting between a number and an address. See Common formats: on page 52 for a list of prefixes.

Note: Number type flags from digital networks, e.g., ISDN or SS7 are converted into special prefixes on the SIP side. International numbers get a "+" prefix, national numbers get an "N" prefix, and subscriber numbers get an "S" prefix.

The automatic conversions are done for calling numbers, called numbers, and redirected numbers.

Possible scenarios

- At a PSTN interface, a line access digit must be prepended in order to call to the public network, while another PSTN interface is directly connected and does not need an access digit.

  Solution: Add a regular expression to outbound address map of the first interface.
- All calls to a number beginning with "9" shall be routed to one specific SIP peer while removing this digit.

  Solution: Manipulate the called number in the route. This way the SIP peer may also receive calls to other numbers (via other routes) without having to deal with different number formats.
- SIP peer "A" needs the dialed numbers to be formatted in E.164 format, while SIP peer "B", which is in load-balancing or fail-over partnership with "A", needs it in an extension-only format.

  Solution: Define different number formats in the SIP peer settings.
- SIP peer "A" is located at a different location than SIP peer "B", e.g., London and Stuttgart. Therefore, both need different location settings regarding country and area codes, etc.

  Solution: Create different dialplans and assign each dialplan to one SIP peer.

How addresses are manipulated

Note: Each step is optional.

1. Save the inbound call addresses as "A".
2. Apply the "address map inbound" of the endpoint assigned to the call setup request to "A", resulting in "B".
3. To check the first route: apply the number format settings of the route together with the dialplan of the source endpoint to the call addresses "B", resulting in "C".
4. Check the route as described in the route processing section (5) against addresses "C". If the route does not match, discard the changes and try the next route with "B" again.
5. If the route matches, apply the route address map to the addresses "C", resulting in "D".
6. After selecting one of the destinations of the route, normalize the addresses "D" using the dialplan and number format of the destination endpoint, resulting in addresses "E".
7. Apply the outbound address map of the destination endpoint to "E", giving the effective call addresses "F" sent to the destination.
8. If the call to the selected destination endpoint fails and if there are other endpoints in a fail-over configuration, start with step 6 again with the respective settings of the next endpoint.

Address Map Processing With The Dialogic® Diva® SIPcontrol™ Software

Address maps are processed as follows:

1. Get the first map rule of the address map.
2. Verify if called, calling, and redirect expression each match the respective part of the call addresses (or are empty). If not, verify the next map rule.
3. If all three expressions match, apply each format string of the rule to the respective address match.
4. If the option **Stop on match** is enabled, stop processing. Otherwise, continue with the next rule as described in step 2.

Number Processing With The Dialogic® Diva® SIPcontrol™ Software

The Dialogic® Diva® SIPcontrol™ software provides two mechanisms for number processing. Both mechanisms can be used together:

1. Number normalization based on a dialplan as described below.
2. Number modification using regular expressions as described on page 51.

Number normalization based on a dialplan

The number normalization based on a dialplan can work in an environment in which the Diva SIPcontrol software is connected to a private SIP network and a public switched telephone network (PSTN), optionally with a PBX between the PSTN and the Diva SIPcontrol software. If the Diva SIPcontrol software is used as a gateway between a private circuit switched network and a public SIP-based network, the number normalization function of the Diva SIPcontrol software should not be used.

The Diva SIPcontrol software also supports dialplans using the North American numbering plan (NANP). See North-American numbering plan: on page 33 for more information.

The number normalization is done in two steps:

1. The received called, calling and redirected numbers are analyzed based on the dialplan configured for the PSTN Interface or SIP Peer.
2. The number is converted into the configured target format. Six target formats are available:
   - **International number with prefixes**: All numbers are converted to an international number with the prefix for international calls and, if required, an outside access digit.
   - **International number with number type**: All numbers are converted to an E.164 number with the number type flag set to "international" ("+" is used in SIP addresses).
   - **National number with prefixes**: If possible, all numbers are converted to a national number with the prefix for national calls and an outside access digit, as required. Exception: Numbers with different country code will be converted to an international number with prefix for international calls and outside access digit, if required.
   - **National number with number type**: If possible, all numbers are converted to a national number with the number type flag set to "national". Exception: Numbers with different country code will be converted to an international number with number type set to "international". **Note:** This target format should not be used for calls to SIP networks.
   - **Extension only with prefixes**: All numbers are reduced as much as possible; only the required prefixes are prepended.
   - **Extension only with number type**: All numbers are reduced as much as possible. Instead of prefixes the appropriate number type is set. **Note:** This target format should not be used for calls to SIP networks.

Important information about the outside access digit configuration

- Configure the outside access digit only if there is a PBX between the PSTN and the Diva SIPcontrol software, and if this PBX requires the outside access digit for external calls. If you need to configure the outside access digit, also configure the following related options:
  - **Incoming PSTN access code provided by the PBX**: This option defines whether the Diva SIPcontrol software expects the outside access digit in the calling number in external calls from the PBX.
    The PBX normally prepends the outside access digit to the calling number in incoming external calls in order to enable callback functionality at internal phones. If this is the case, enable this option.
  - **PSTN access code provided by the SIP caller**: This option defines whether the Diva SIPcontrol software expects the outside access digit in the called number in external calls from the SIP side to the PSTN.
    It is normally required to prepend the outside access digit to call an external number from an internal phone. In this case, these are phones on the SIP side. However, in some configurations this is not required, especially in a configuration that is part of the North-American numbering plan (NANP), where an internal number can be identified based on its length. If it is possible to identify an internal call purely by the length of the called number, this option can be disabled. In all other configurations with outside access digit this option has to be enabled.
    It is recommended to have this option enabled in dialplans with outside access digit.
- The Diva SIPcontrol software's number normalization function does not remove outside access digits as a PBX can for external calls. If the Diva SIPcontrol software needs to behave like a PBX with an outside access digit for external calls, use the Address Map functionality in combination with a Routing module.

Number modification using regular expressions

The Dialogic® Diva® SIPcontrol™ software organizes regular expressions into address maps, and each endpoint or route may be assigned one map. Each address map contains a number of regular expressions together with the respective output format string that ensures that virtually every required manipulation scheme can be configured.

By using separate address maps, instead of rules embedded into the routes and endpoints, it is possible to share the same settings across different objects. For example, if several PSTN interfaces are connected to the same PBX, they will most probably be configured with the same settings and, therefore, can share an address map that the Diva SIPcontrol software lets you assign for each individual controller.

The Diva SIPcontrol software uses the style of regular expressions used by Perl. Most tutorials and how-to's covering Perl regular expressions can apply to the Diva SIPcontrol software.

Common expressions:

Character	Meaning
.	Matches any character
^	Matches the beginning of a number only
$	Matches the end of a number
\+	Matches the plus sign ("+")
*	Matches any number of occurrences of the previous character
{n}	Matches the previous character exactly n times
{n,m}	Matches the previous character between n and m times, both inclusive
( )	Marks a sub-expression to be referenced in format string and also groups sets of characters
|	Alternate operator, matches either the left or right sub-expression
[ ]	Matches any character given within the square brackets, i.e [123] matches either 1, 2, or 3, but not 4, 5, or 123.

Common formats:

Character	Meaning
0-9,+	Inserts the respective character into the output
(?n(digits))	Inserts the digits given only if the n^{th} sub-expression of the expression matched
$&	Outputs what matched the whole expression
$n	Outputs the n^{th} matched sub-expression
+	Indicates an international number type, if it is the first character in the string
N	Indicates a national number type, if it is the first character in the string
S	Indicates a subscriber number type, if it is the first character in the string
$(S)	Inserts the current calling (source) number
$(D)	Inserts the called (destination) number
$(R)	Inserts the first redirected number
$(R2)	Inserts the second redirected number
$(Rn)	Inserts the n^{th} redirected number (up to the 9th)

Examples

Note: In all examples, the hyphen ("-") is only used for clarification. It must not be included either in the dialed numbers or in the configured expressions and formats.

The examples may be used for calling or called number normalization for both the inbound and outbound directions.

Omit the prefix digits

Task: A leading "33" prefix should be removed from the number.

Example: 33-444-5555 should be converted to 444-5555.

Expression entry: ^33

Format entry: (none)

Note: If the number does not start with "33", it passes unchanged.

Add the prefix digits

Task: The number needs the leading prefix "9".

Example: 444-5555 should go out as 9-444-5555.

Expression entry: .*

Format entry: 9$&

Replace the international number type by prefix

Task: A call that is indicated as an international call should be placed with prefixes instead.

Example: The number +1-472-333-7777 should be dialed as 011-472-333-7777

Expression entry: ^\+

Format entry: 01

Replace the international dial prefix by number type

Task: A call that has an international dial prefix should be placed with an international number type instead of the prefix.

Example: The number (01)1-472-333-7777 should be dialed as +1-472-333-7777

Expression entry: ^01

Format entry: +

Replace an extension by another

Task: Calls for specific extensions should be indicated with other extensions.

Example: The extension 1111 should be replaced by 2222, and extension 3333 by extension 4444.

First expression entry: 1111(@.*)?$

First format entry: 2222

Stop on Match: true

Second expression entry: 3333(@.*)?$

Second format entry: 4444

Stop on Match: true

Note: This example applies only for calls from the SIP to the PSTN.

Replace the "N" in a national number

The "N" can be set to signal a number as national number.

Task: Replace the "N" in a national number with the national prefix.

Example: N123-45678 should be signaled as 0123-45678

Expression: ^N

Format entry: 0

Display the "user=phone" parameter without E.164

The "user=phone" parameter is set automatically if the number is a valid "tel:" URI. The number is either in E.164 format or has the "phone-context=XXX" parameter added. If you need the "user=phone" without E.164, you need to provide the phone-context parameter.

Task: Display "user=phone" parameter without E.164 and provide phone-context parameter.

Example: Present the phone number +1(123)727-0203 without E.164.

Expression: ^(.*)

Format entry: $1;phone-context=+1(123)$1

Cause Code Mapping With The Dialogic® Diva® SIPcontrol™ Software

If the Dialogic® Diva® SIPcontrol™ software uses Microsoft® Office Communications Server 2007 as SIP peer, the cause/response code tables are used as specified by Microsoft. See Default cause code mapping for Microsoft® Office Communications Server (OCS) 2007 SIP peers on page 57 for a detailed list of cause/response codes.

If the Diva SIPcontrol software does not use Microsoft® OCS 2007, the default cause/response code mapping is used. See Default cause code mapping below for a detailed list of cause/response codes.

Default cause code mapping

The Dialogic® Diva® SIPcontrol™ software includes a default cause/response code mapping table that includes the most common cause codes according to RFC 3398 and RFC 4497. If you need to define a cause code mapping other than in the table, you can configure it in the Cause Code Mapping Configuration.

For ISDN to SIP code mappings, see ISDN cause code to SIP response code below.

For SIP to ISDN code mappings, see SIP response code to ISDN cause code on page 55.

ISDN cause code to SIP response code

ISDN cause code	Description	SIP response code forwarded to the SIP peer	Description
1	Unallocated number	404	Not found
2	No route to specified transit network	404	Not found
3	No route to destination	404	Not found
16	Normal call clearing	603	Decline (The PBX of Philips sends this code during call set-up if the user rejects the call.)
17	User busy	486	Busy here
18	No user response	603	Decline (The PBX of Philips sends this code during call set-up if the user rejects the call.)
19	No answer from the user	480	Temporarily unavailable
20	Subscriber absent	480	Temporarily unavailable
21	Call rejected	603	Decline
22	Number changed	410	Gone
23	Redirection to new destination	410	Gone
26	Non-selected user clearing	404	Not found
27	Destination out of order	502	Bad gateway
28	Address incomplete	484	Address incomplete
29	Facility rejected	501	Not implemented
31	Normal, unspecified	480	Temporarily unavailable
34	No circuit available	503	Service unavailable
38	Network out of order	503	Service unavailable
41	Temporary failure	503	Service unavailable
42	Switching equipment congestion	503	Service unavailable
47	Resource unavailable	503	Service unavailable

ISDN cause code	Description	SIP response code forwarded to the SIP peer	Description
55	Incoming class barred within Closed User Group (CUG)	403	Forbidden
57	Bearer capability not authorized	403	Forbidden
58	Bearer capability not presently available	503	Service unavailable
63	Service or option not available, unspecified	488	Not acceptable here
65	Bearer capability not implemented	488	Not acceptable here
69	Requested Facility not implemented	501	Not implemented
70	Only restricted digital available	488	Not acceptable here
79	Service or option not implemented	501	Not implemented
87	User not member of Closed User Group (CUG)	403	Forbidden
88	Incompatible destination	503	Service unavailable
102	Recover on Expires timeout	504	Server time-out
111	Protocol error	500	Server internal error
127	Interworking, unspecified	500	Server internal error
Any code other than listed above:		500	Server internal error

SIP response code to ISDN cause code

SIP response code from the SIP peer	Description	ISDN cause code	Description
400	Bad Request	41	Temporary failure
401	Unauthorized	21	Call rejected
402	Payment required	21	Call rejected
403	Forbidden	21	Call rejected
404	Not found	1	Unallocated number
405	Method not allowed	63	Service or option unavailable
406	Not acceptable	79	Service/option not implemented
407	Proxy authentication required	21	Call rejected
408	Request timeout	41	Temporary failure
410	Gone	22	Number changed
413	Request entity too large	63	Service or option unavailable
414	Request-URI too long	63	Service or option unavailable
415	Unsupported media type	79	Service/option not implemented
416	Unsupported URI scheme	79	Service/option not implemented
420	Bad extension	79	Service/option not implemented
421	Extension required	79	Service/option not implemented

SIP response code from the SIP peer	Description	ISDN cause code	Description
423	Interval too brief	63	Service or option unavailable
480	Temporarily unavailable	19	No answer from user
481	Call/transaction does not exist	41	Temporary failure
482	Loop detected	25	Exchange routing error
483	Too many hops	25	Exchange routing error
484	Address incomplete	28	Invalid number format (address incomplete)
485	Ambiguous	1	Unallocated number
486	Busy here	17	User busy
488	Not acceptable here	65	Bearer capability not implemented
500	Server internal error	41	Temporary failure
501	Not implemented	79	Service/option not implemented
502	Bad gateway	38	Network out of order
503	Service unavailable	63	Service or option unavailable
504	Server time-out	41	Temporary failure
505	Version not supported	79	Service/option not implemented
513	Message too large	63	Service or option unavailable
600	Busy everywhere	17	User busy
603	Decline	21	Call rejected
604	Does not exist anywhere	1	Unallocated number
606	Not acceptable	65	Bearer capability not implemented
Any code other than listed above:		31	Normal, unspecified

Default cause code mapping for Microsoft® Office Communications Server (OCS) 2007 SIP peers

The Dialogic® Diva® SIPcontrolTM software includes a default cause/response code mapping table for Microsoft® OCS 2007 SIP peers that includes the most common (as of the date of publication of this document) cause codes according to RFC 3398 and RFC 4497. If you need to define a cause code mapping other than in the table, you can configure it in the Cause Code Mapping Configuration.

For ISDN to SIP code mappings, see Microsoft® OCS 2007 ISDN cause code to SIP response code below.

For SIP to ISDN code mappings, see Microsoft® OCS 2007 SIP response code to ISDN cause code on page 58.

Microsoft® OCS 2007 ISDN cause code to SIP response code

ISDN cause code	Description	SIP response code forwarded to Microsoft® OCS 2007	Description
1	Unallocated number	404	Not found
2	No route to specified transit network	404	Not found
3	No route to destination	404	Not found
16	Normal call clearing	603	Decline (The PBX of Philips sends this code during call set-up if the user rejects the call.)
17	User busy	486	Busy here
18	No user response	408	Request timeout
19	No answer from the user	480	Temporarily unavailable
20	Subscriber absent	480	Temporarily unavailable
21	Call rejected	603	Decline
22	Number changed	410	Gone
23	Redirection to new destination	410	Gone
26	Non-selected user clearing	404	Not found
27	Destination out of order	502	Bad gateway
28	Address incomplete	484	Address incomplete
29	Facility rejected	501	Not implemented
31	Normal, unspecified	480	Temporarily unavailable
34	No circuit available	503	Service unavailable
38	Network out of order	503	Service unavailable
41	Temporary failure	503	Service unavailable
42	Switching equipment congestion	503	Service unavailable
47	Resource unavailable	503	Service unavailable
55	Incoming class barred within Closed User Group (CUG)	403	Forbidden
57	Bearer capability not authorized	403	Forbidden
58	Bearer capability not presently available	503	Service unavailable
65	Bearer capability not implemented	488	Not acceptable here
69	Requested Facility not implemented	501	Not implemented
70	Only restricted digital available	488	Not acceptable here

ISDN cause code	Description	SIP response code forwarded to Microsoft® OCS 2007	Description
79	Service or option not implemented	501	Not implemented
87	User not member of Closed User Group (CUG)	403	Forbidden
88	Incompatible destination	400	Bad request
102	Recover on Expires timeout	504	Server time-out
111	Protocol error	500	Server internal error
127	Interworking, unspecified	500	Server internal error
Any code other than listed above:		500	Server internal error

Microsoft® OCS 2007 SIP response code to ISDN cause code

SIP response code from Microsoft® OCS 2007	Description	ISDN cause code	Description
400	Bad Request	41	Temporary failure
401	Unauthorized	21	Call rejected
402	Payment required	21	Call rejected
403	Forbidden	21	Call rejected
404	Not found	1	Unallocated number
405	Method not allowed	63	Service or option unavailable
406	Not acceptable	79	Service/option not implemented
407	Proxy authentication required	21	Call rejected
408	Request timeout	102	Recovery on timer expiry
410	Gone	22	Number changed
413	Request entity too large	127	Interworking, unspecified
414	Request-URI too long	127	Interworking, unspecified
415	Unsupported media type	79	Service/option not implemented
416	Unsupported URI scheme	127	Interworking, unspecified
420	Bad extension	127	Interworking, unspecified
421	Extension required	127	Interworking, unspecified
423	Interval too brief	127	Interworking, unspecified
480	Temporarily unavailable	18	No user responding
481	Call/transaction does not exist	41	Temporary failure
482	Loop detected	25	Exchange routing error
483	Too many hops	25	Exchange routing error
484	Address incomplete	28	Invalid number format (address incomplete)
485	Ambiguous	1	Unallocated number

SIP response code from Microsoft® OCS 2007	Description	ISDN cause code	Description
486	Busy here	17	User busy
488	Not acceptable here	65	Bearer capability not implemented
500	Server internal error	41	Temporary failure
501	Not implemented	79	Service/option not implemented
502	Bad gateway	38	Network out of order
503	Service unavailable	41	Temporary failure
504	Server time-out	102	Recovery on timer expiry
505	Version not supported	127	Interworking, unspecified
513	Message too large	127	Interworking, unspecified
600	Busy everywhere	17	User busy
603	Decline	21	Call rejected
604	Does not exist anywhere	1	Unallocated number
606	Not acceptable	65	Bearer capability not implemented
Any code other than listed above:		31	Normal, unspecified

Event Logging With The Dialogic® Diva® SIPcontrol™ Software

A computer with the Diva SIPcontrol software installed may write the following types of events into the System Event Log:

- Errors
- Warnings
- Informational messages

You can view the events in the Windows® Event Viewer. To do so, click **Programs** > **Settings** > **Control Panel** > **Administrative Tools**. In the **Administrative Tools** window, double-click **Event Viewer** and then **Application**, where the Diva SIPcontrol software stores the events.

Errors

An error is a significant problem such as loss of data or loss of functionality. For example, if a service fails to load, an error event will be logged.

See below for possible error events. Variables are enclosed in angle brackets. Parameters enclosed in square brackets are optional:

Event ID	Event Text	Event Description
2000	Service could not start. <Reason>	The <Reason> is text that explains why the service could not start.
2001	Service could not stop. <Reason>	The <Reason> is text that explains why the service could not stop.
2002	Updating configuration failed. <Reason>	The new configuration could not be activated, probably due to invalid configuration data.
2003	Cannot bind to IP address. <IP address>:<port> [<protocol>].	The service cannot be bound to the IP address.
2004	TLS initialization failed, call attempt aborted.	The configured TLS settings are invalid, or a required file is missing. For calls to SIP only: the call is aborted unless an alternate destination without TLS encryption is available.

Warnings

A warning is an event that is not necessarily significant but may indicate a possible future problem.

See the following table for possible warnings. Variables are enclosed in angle brackets:

Event ID	Event Text	Event Description
3000	SIP peer <Host Name> is not available.	The SIP peer does not respond to keep-alive check requests, and has therefore been marked as inactive. It will receive no calls from SIPcontrol until the ongoing keep-alive check receives valid responses.
3001	Cannot process call from <Calling Number> to <Called Number>. No more licenses available.	The number of currently active calls has reached the number of licensed channels and a further call has been declined thereof. The <Calling Number> and <Called Number> of the PSTN call are inserted as signaled from the line.
3002	Cannot process outgoing PSTN call to <Called Number> from <Calling Number>. No free PSTN channel available.	The <Called Number> and <Calling Number> are inserted. It can be a PSTN or SIP address.
3003	Call transfer to <Called Number> failed. <Optional Reason>	The <Called Number> is the PSTN-based number. The reason is optional and may contain any text.
3004	Registration to <Registrar Host Name> with user<User Host Name> failed.	The Registration to a Registrar with the user to register failed.
3005	SIP peer <Host Name> is available again.	An inactive SIP peer is alive again (has responded to alive check request)

3006	Cannot process call from <Calling Address> to <Called Address>. Codec negotiation failed.	A call could not be established because non of the audio codecs support by and allowed for the SIP peer could be used for the call and no alterntive targets were available.
3007	Can not establish TLS connection to <address>: <Reason>.	No TLS connection could be established to the SIP peer. <Optional Reason> gives more details if available.
3008	TLS certificate verification failed with error <OpenSSL errorcode>.	The TLS certificate presented by the peer could not be verified successfully. The error code is the value returned by the TLS library.
3009	TLS Data Error	An error occuring during TLS data processing. The trace may give additional information.

Informational messages

Informational messages refer to successful operation events such as starting or stopping the service:

See the following for informal events. Variables are enclosed in angle brackets:

Event ID	Event Text	Event Description
4000	Service started.	Service has been started successfully.
4001	Service stopped.	Service was requested to stop or shutdown, and did so successfully.
4002	Configuration successfully updated.	Called when service configuration has been successfully updated.
4003	Call from <Calling Number> to <Called Number> established.	The <Calling Number> and the <Called Number> are inserted. The Number can be a PSTN or SIP address.
4004	Call from <Calling Number> to <Called Number> disconnected.	The <Calling Number> and the <Called Number> are inserted. The Number can be a PSTN or SIP address.
4005	Call from <Calling Number> successfully transferred to <Called Number>.	The <Calling Number> is the calling number. The <Called Number> is the number of the transfer destination.
4006	Registration to <Registrar Host Name> with user<User Host Name> is successful.	The registration to a registrar with the user to register is successful.
4008	Cannot process call from <Calling Number> to <Called Number>, <Reason>.	The <Calling Number> and <Called Number> are inserted, the SIP or Q.850 cause code text is inserted at runtime. Different reasons (busy, rejected,…) are translated to runtime.
4009	Available/changed licensed channels <Licensed channels>.	List the amount of licensed channels. If no license file is read, the default is "8" licensed channels. Issued if the licensed amount changes, e.g., after a new license file has been installed.
4010	Available/changed PSTN channels <PSTNChannels>.	Gives the amount of available channels to the telephone network. Called if the number changes due to configuration updates or controllers being enabled/disabled.

Configuration Scenarios

The following scenarios describe common configurations for a Dialogic® 4060 Media Gateway with one Dialogic® Diva® V-2PRI/E1/T1 Media Board installed. All scenarios can also be applied to a Dialogic® 4120 Media Gateway. The scenarios are based on the Dialogic® Media Gateway being operated in Germany.

- Use cases for Microsoft® Office Communications Server (OCS) 2007 below
- Use case for Microsoft® Exchange Server 2007 on page 90

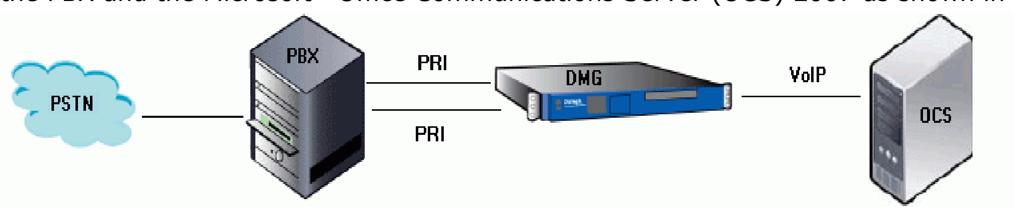
Use cases for Microsoft® Office Communications Server (OCS) 2007

The following are use cases for the Dialogic® Media Gateway and the OCS 2007:

1. Using the Dialogic® 4060 Media Gateway between the PBX and Microsoft® Office Communications Server 2007: Both lines of the Diva Media Board are connected between the PBX and the Microsoft® Office Communications Server (OCS) 2007.
2. Using the Dialogic® 4060 Media Gateway between PBX/PSTN and the Microsoft® Office Communications Server 2007: One line of the Diva Media Board is connected to the PBX and one line is connected directly to the PSTN. The Dialogic® Media Gateway is connected to the Microsoft® Office Communications Server (OCS) 2007.
3. Using the Dialogic® 4060 Media Gateway between the PSTN and PBX/Microsoft® Office Communications Server 2007: One line of the Diva Media Board is connected to the PBX and one line is connected to the PSTN. The Dialogic® Media Gateway is connected between the PSTN and the PBX. The Microsoft® Mediation Server is installed on the Dialogic® Media Gateway.

Using the Dialogic® 4060 Media Gateway between the PBX and Microsoft® Office Communications Server 2007

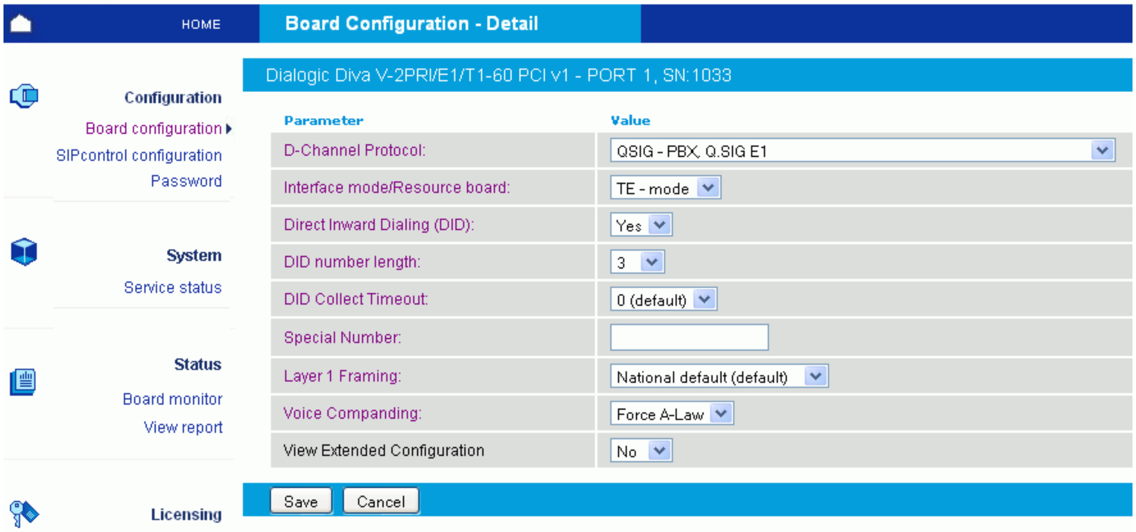
This configuration scenario describes the necessary steps to configure the Dialogic® Media Gateway between the PBX and the Microsoft® Office Communications Server (OCS) 2007 as shown in the graphic.



1. Open the Dialogic® Diva® SIPcontrol™ Software web interface to configure the required parameters. To do so, click **Start** > **Programs** > **Dialogic Diva** > **SIPcontrol Configuration**.
2. In the Diva SIPcontrol software web interface, click **Board configuration** on the left hand side to open the **Available Diva Boards** page.

Click either the board icon or the name of the Dialogic® Diva® Media Board to open the **Board Configuration - Detail** page.

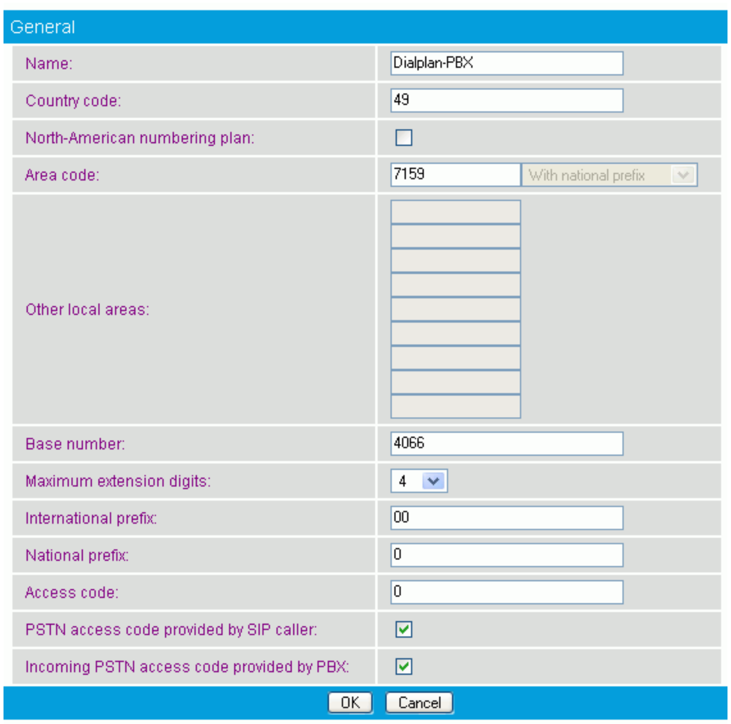
3. Configure the **D-Channel Protocol** of the PBX. In the example, **QSIG-PBX, Q.SIG E1** is selected. Click **Save**.



4. Repeat step 2 and 3 for the other PRI line.

5. In the Diva SIPcontrol web interface, click **SIPcontrol configuration** on the left hand side to open the **SIPcontrol Configuration** page. For this configuration scenario, a dialplan, the PSTN interface, the network interface, a SIP peer, and a routing need to be configured.

6. Open the **Dialplan Configuration**, click **Add**, and enter the following parameters:



- **Name**: Enter a unique name to easily identify the dialplan.
- **Country code**: Enter the country code of the country in which the Dialogic® Media Gateway is located.
- **Area code**: Enter the area code of the region in which the Dialogic® Media Gateway is located.
- **Base number**: Enter the subscriber or trunk number.
- **Maximum extension digits**: Select the maximum number of extension digits that are provided.
- **International prefix**: Enter the international prefix of the country in which the Dialogic® Media Gateway is located.
- **National prefix**: Enter the national prefix that needs to be dialed for long distance calls within the country in which the Dialogic® Media Gateway is located.
- **Access code**: Enter the digit that is necessary to access the public network.
- Enable the options **PSTN access code provided by SIP caller** and **Incoming PSTN access code provided by PBX**.

Click **OK** to save the settings and to close the window.

7. Under **PSTN Interface Configuration**, configure both lines of the Dialogic® Diva® Media Board with the same settings. To do so, click **Details** at the right and select the **Dialplan** you configured above and set **Number format (outbound)** to **Extension** and **Encoding (outbound)** to **Use prefixes**. Leave the remaining parameters at their default values.

General	
Hardware description:	Dialogic Diva V-2PRI/E1/T1 - PORT 1 SN: 1033
PSTN interface number:	1
Name:	Controller1
Address map inbound:	none
Address map outbound:	none
Address Normalization	
Dialplan:	Dialplan-PBX
Number format (outbound):	Extension
Encoding (outbound):	Use prefixes
Default numbering plan:	unknown
Default presentation indicator:	Allowed

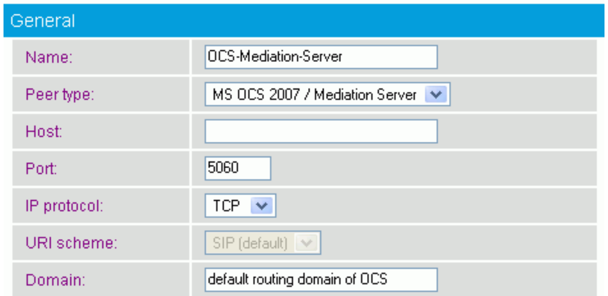
Click **OK** to save the settings and to close the window.

8. Configure the second line with the same settings.
9. Under **Network Interface Configuration**, enable your Ethernet adapter and set the **SIP Listen Port** to **9803**.

Network Interfaces

Name	Device	IP address	UDP listen port	TCP listen port	TLS listen port
Intel(R) PRO1000 GT Desktop	Intel(R) PRO/1000 GT Desktop Adapter - Packet Scheduler Miniport	192.168.213.38	9803 ☑	9803 ☑	5061 ☐
Local Loopback Interface	Local Loopback Interface	127.0.0.1	5060 ☐	5060 ☐	0 ☐

10. Open the **SIP Peer Configuration**, click **Add**, and configure the following parameters:

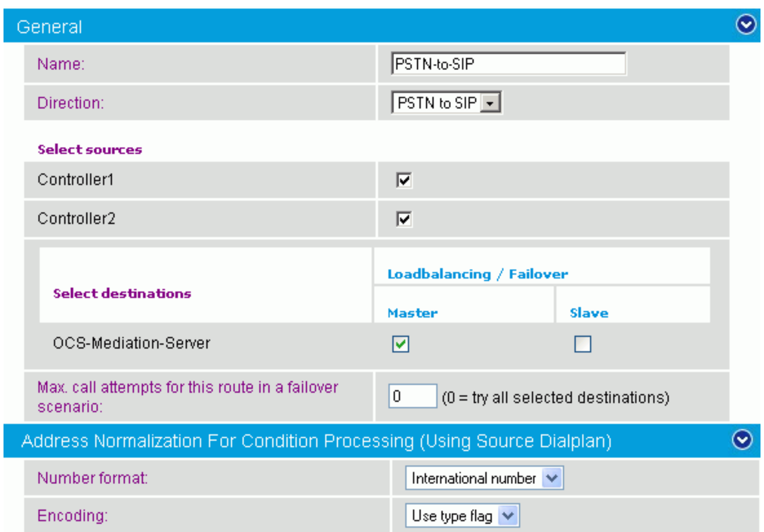


Under **Edit SIP Peer Configuration**, configure the following parameters:

- **Name**: Enter a unique name to easily identify the SIP peer.
- **Peer type**: Select **MS OCS 2007/ Mediation Server** from the dropdown menu.
- **Host**: Enter the IP address or host name of the host PC.
- **Domain**: For the correct domain entry, see the configuration of your Microsoft® Office Communications Server.

Click **OK** to save the settings and to close the window.

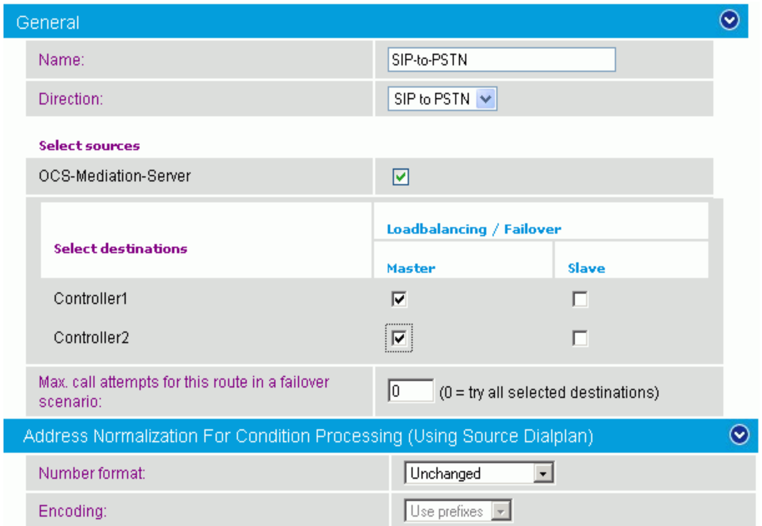
11. Create two routings, one for each direction (PSTN to SIP and SIP to PSTN). To configure the routing from PSTN to SIP, open the **Routing Configuration**, click **Add**, and configure the following parameters:



- **Name**: Enter a unique name to easily identify the routing.
- **Direction**: Select **PSTN to SIP** from the dropdown menu.
- **Select sources**: Select both controllers of the Diva Media Board.
- **Select destinations**: Select the above configured SIP peer as master destination.
- **Number format**: Select **International number** from the dropdown menu.
- **Encoding**: Select **Use type flag** from the dropdown menu.

Click **OK** to save the settings and to close the window.

12. For the routing from SIP to PSTN, click **Add** again, and configure the following parameters:



- **Name**: Enter a unique name to easily identify the routing.
- **Direction**: Select **SIP to PSTN** from the dropdown menu.
- **Select sources**: Select the above configured SIP peer as source.
- **Select destinations**: Select both controllers of the Diva Media Board.

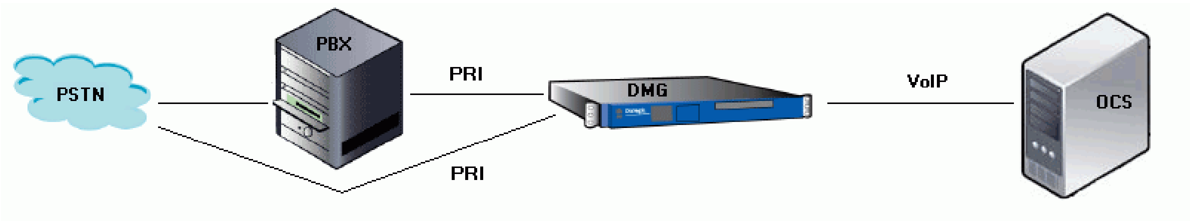
Click **OK** to save the settings and to close the window.

13. Click **Save** in the main configuration page to save the settings and to activate the changes.

14. Configure the Microsoft® Office Communications Server 2007 as described in the Dialogic® 4000 Media Gateway Quickstart Guide.

Using the Dialogic® 4060 Media Gateway between PBX/PSTN and the Microsoft® Office Communications Server 2007

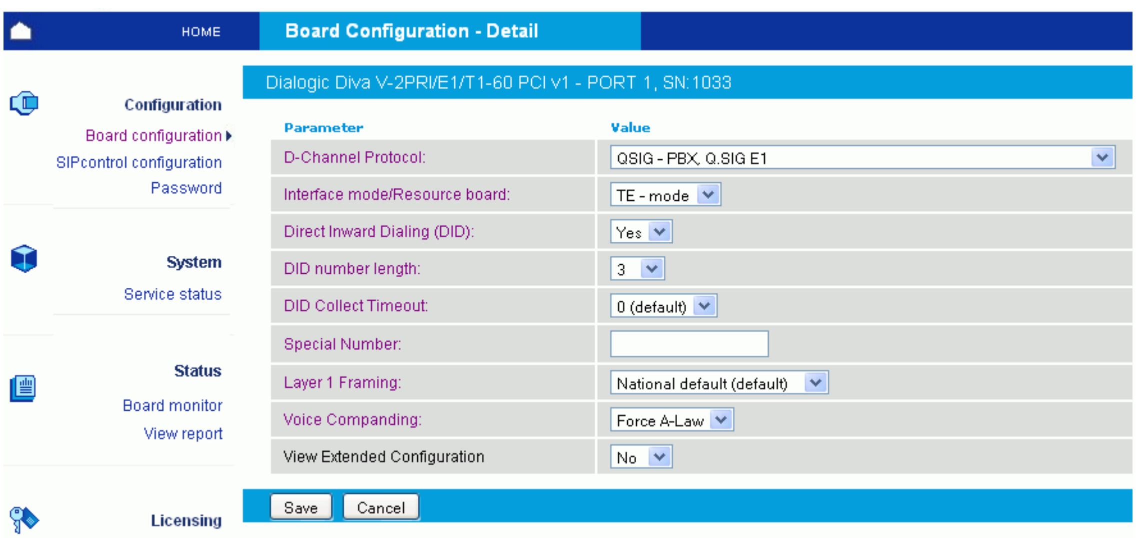
This configuration scenario describes the necessary steps if one line of the Dialogic® Media Gateway is connected to the PBX and one line is connected directly to the PSTN as shown in the graphic.



1. Open the Dialogic® Diva® SIPcontrol™ Software web interface to configure the required parameters. To do so, click **Start** > **Programs** > **Dialogic Diva** > **SIPcontrol Configuration**.
2. In the Diva SIPcontrol software web interface, click **SIPcontrol** on the left hand side to open the **Board Configuration** page.

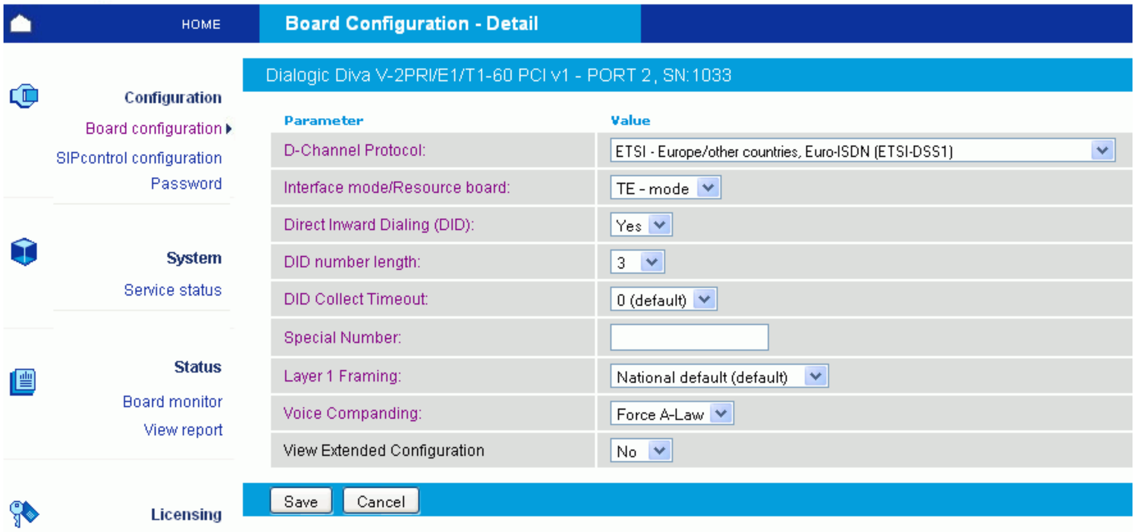
Click either the board icon or the name of the first Dialogic® Diva® Media Board line to open the **Board Configuration - Detail** page.

3. Configure the **D-Channel Protocol** of port 1. In the example, **QSIG-PBX, Q.SIG E1** is selected. Click **Save**.



4. Click **Board configuration** again and select port 2 of your Diva Media Board to open the **Board Configuration - Detail** page again.

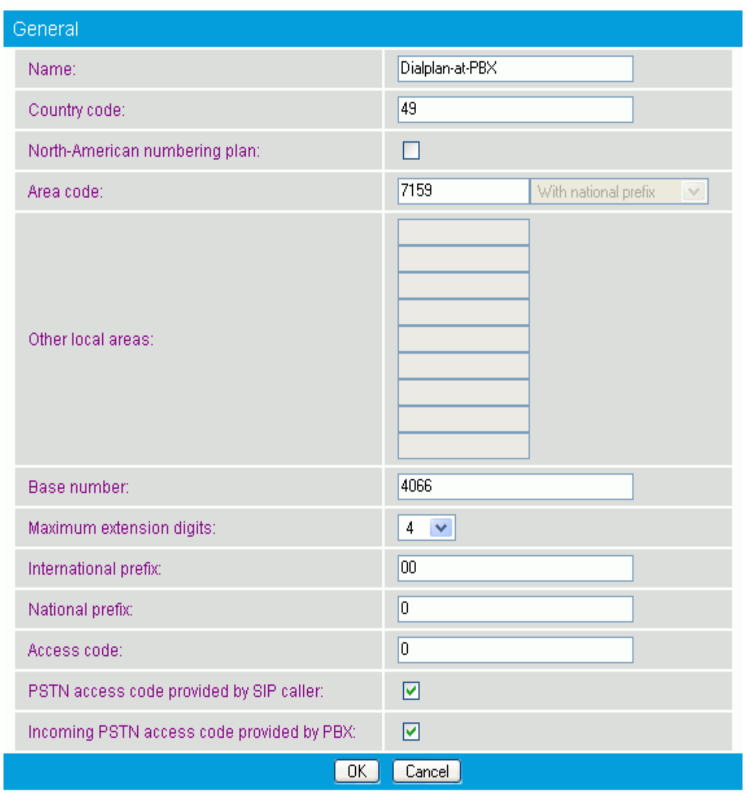
5. Configure the **D-Channel Protocol** of the PRI line connected to the PSTN. In the example, **ETSI-Europe/other countries, Euro-ISDN (ETSI-DSS1)** is selected. Click **Save**.



6. In the Diva SIPcontrol web interface, click **SIPcontrol configuration** on the left hand side to open the **SIPcontrol Configuration** page.

   For this configuration scenario, two dialplans, the two PSTN interfaces, the network interface, a SIP peer, an address map, and three routings need to be configured.

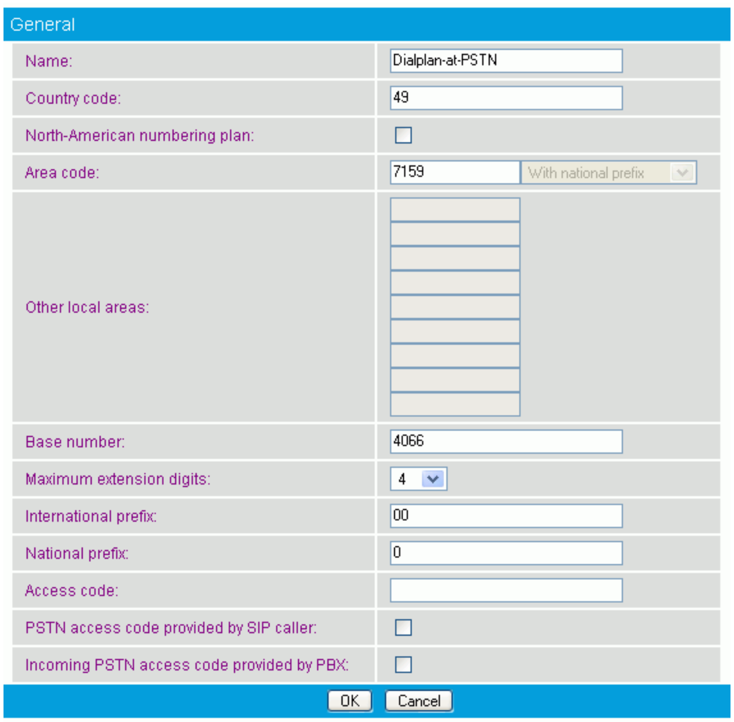
7. Configure two dialplans; one for the line connected to the PBX and one for the line connected directly to the PSTN. To do so, open the **Dialplan Configuration**, click **Add**, and configure the following parameters for the PBX dialplan:



- **Name**: Enter a unique name to easily identify the dialplan.
- **Country code**: Enter the country code of the country in which the Dialogic® Media Gateway is located.
- **Area code**: Enter the area code of the region in which the Dialogic® Media Gateway is located.
- **Base number**: Enter the subscriber or trunk number.
- **Maximum extension digits**: Select the maximum number of extension digits that are provided.
- **International prefix**: Enter the international prefix of the country in which the Dialogic® Media Gateway is located.
- **National prefix**: Enter the national prefix that needs to be dialed for long distance calls within the country in which the Dialogic® Media Gateway is located.
- **Access code**: Enter the digit that is necessary to access the public network.
- Enable the options **PSTN access code provided by SIP caller** and **Incoming PSTN access code provided by PBX**.

Click **OK** to save the settings and to close the window.

8. To configure the PSTN dialplan, click **Add** again, and configure the following parameters.

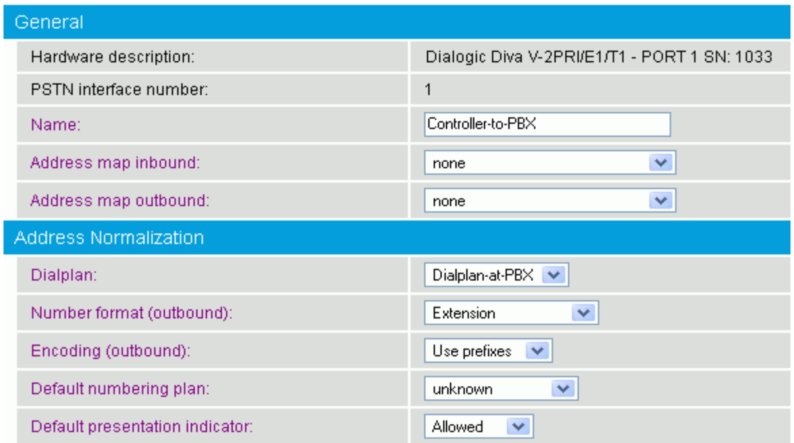


- **Name**: Enter a unique name to easily identify the dialplan.
- **Country code**: Enter the country code of the country in which the Dialogic® Media Gateway is located.
- **Area code**: Enter the area code of the region in which the Dialogic® Media Gateway is located.
- **Base number**: Enter the subscriber or trunk number.
- **Maximum extension digits**: Select the maximum number of extension digits that are provided.
- **International prefix**: Enter the international prefix of the country in which the Dialogic® Media Gateway is located.
- **National prefix**: Enter the national prefix that needs to be dialed for long distance calls within the country in which the Dialogic® Media Gateway is located.

Click **OK** to save the settings and to close the window.

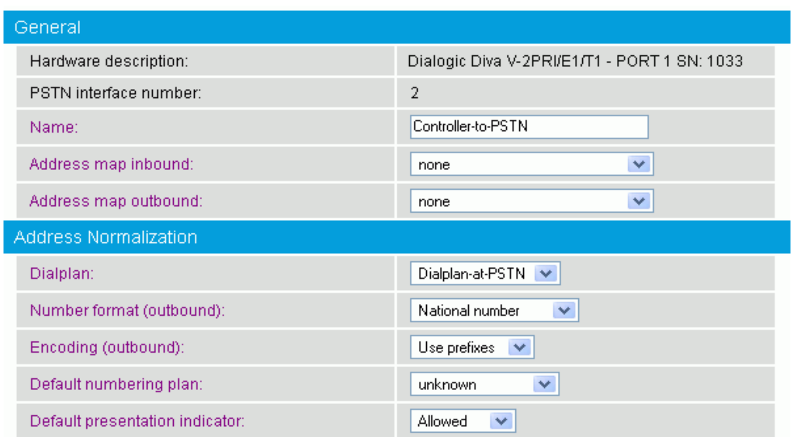
9. Under **PSTN Interface Configuration**, configure controller 1 with the PBX-specific settings and controller 2 with the PSTN-specific settings. The controller number that you configure with the PBX-specific settings needs to correspond to the line number in the Diva Configuration Manager that you configured with the switch type of your PBX. The same is true for the controller with the switch type of the PSTN line.

- To configure the PBX-specific parameters, click **Details** at the right of the controller connected to the PBX. Select the **Dialplan** you configured for the PBX, set **Number format (outbound)** to **Extension**, and **Encoding (outbound)** to **Use prefixes**.



Click **OK** to save the settings and to close the window.

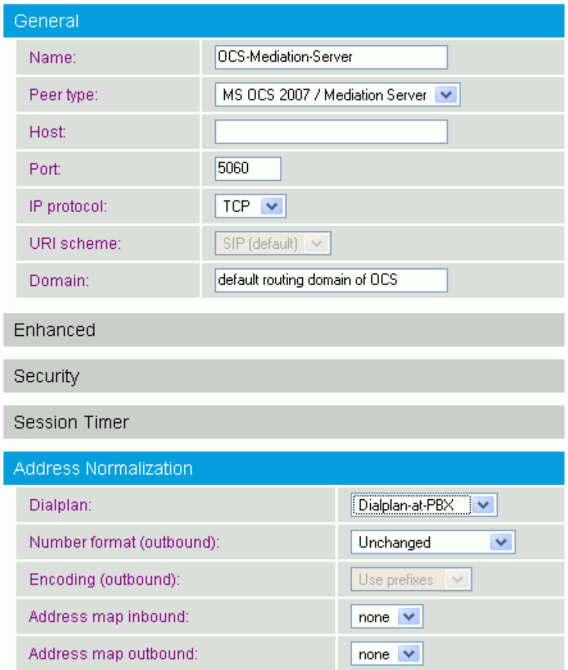
- To configure the PSTN-specific parameters, click **Details** at the right of the controller connected to the PSTN line. Select the **Dialplan** you configured for the PSTN, set **Number format (outbound)** to **National Number**, and **Encoding (outbound)** to **Use prefixes**.



Click **OK** to save the settings and to close the window.

10. Under **Network Interface Configuration**, enable your Ethernet adapter and as **SIP Listen Port** enter **9803**.

11. Open the **SIP Peer Configuration**, click **Add**, and configure the following parameters:



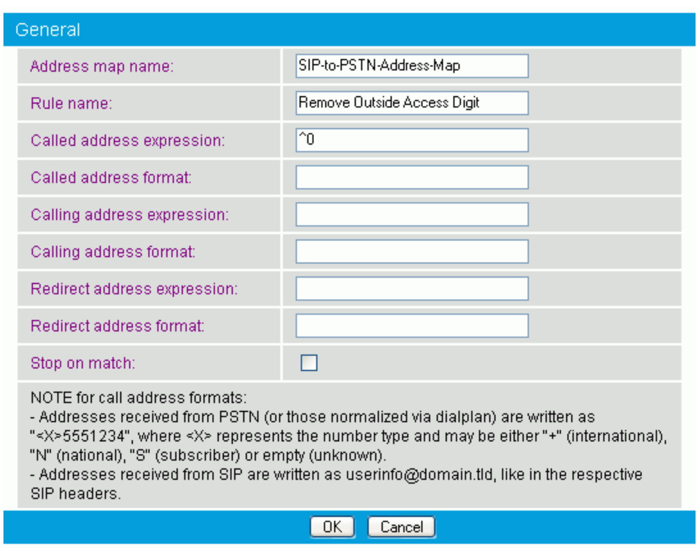
Under **Edit SIP Peer Configuration**, configure the following parameters:

- **Name**: Enter a unique name to easily identify the SIP peer.
- **Peer type**: Select **MS OCS 2007/ Mediation Server** from the dropdown menu.
- **Host**: Enter the IP address or host name of the host PC.
- **Domain**: For the correct domain entry, see the configuration of your Microsoft® Office Communications Server 2007.

Under **Address Normalization Configuration**, select the dialplan you configured for the controller connected to the PBX.

Click **OK** to save the settings and to close the window.

12. Create an address map for the SIP to PSTN direction to remove the outside access digit. To do so, open the **Address Map Configuration**, click **Add**, and configure the following parameters:



- **Address map name**: Enter a descriptive name of the address map.
- **Rule name**: Enter a name that explains the address map rule.
- **Called address expression**: Enter the expression as displayed in the graphic above to remove the outside access digit.

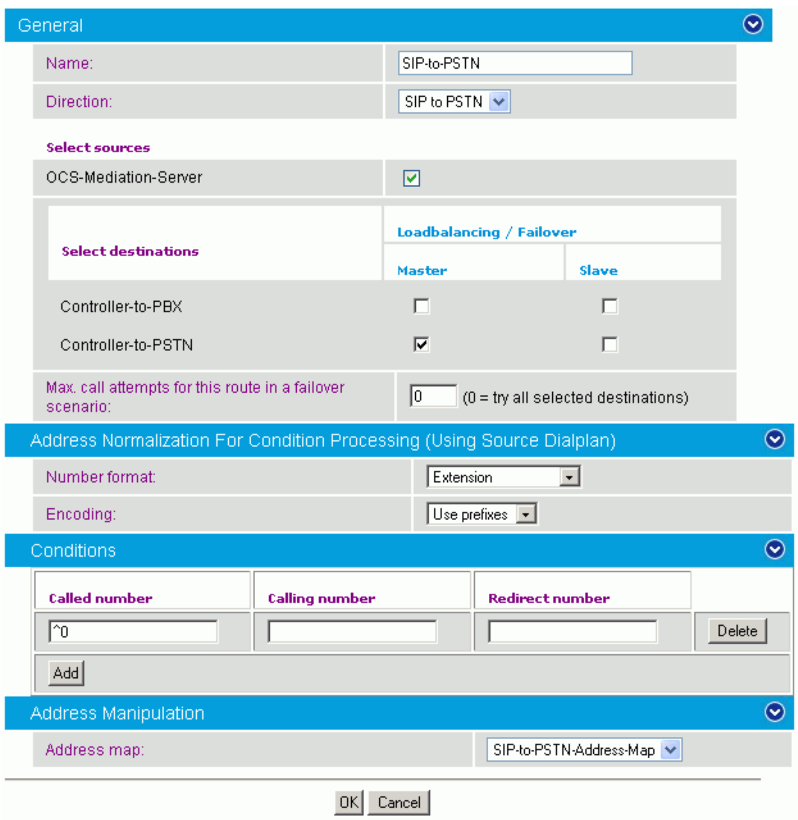
Click **OK** to save the settings and to close the window.

13. Create three routings:

- one from the Microsoft® Office Communications Server (OCS) 2007 directly to the PSTN,
- one from the Microsoft® OCS 2007 via the PBX to the PSTN, and
- one from the PSTN/PBX via the Dialogic® Media Gateway to Microsoft® OCS 2007.

In this scenario, the order of the routings is important because only one routing will be configured with a condition. In this example, the routings are configured in the correct order but it is also explained how to change the order in case you configured them differently.

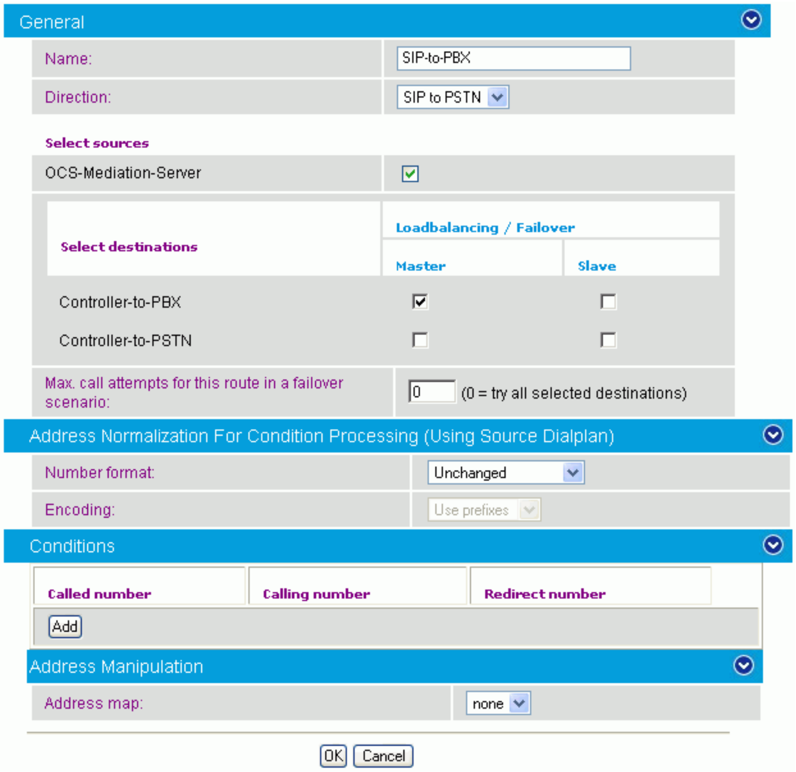
14. Create the route from Microsoft® OCS 2007 to the PSTN first. To do so, open the **Routing Configuration**, click **Add** and configure the following parameters:



- **Name**: Enter a unique name to easily identify the routing.
- **Direction**: Select **SIP to PSTN** from the dropdown menu.
- **Select sources**: Select the above configured SIP peer as master destination.
- **Select destinations**: Select the controller you configured for the PSTN.
- **Number format**: Select **Extension** from the dropdown menu.
- **Encoding**: Select **Use prefixes** from the dropdown menu.
- Under **Conditions**, click **Add**, and in **Called number** enter **^0** to omit the outside access digit.

Click **OK** to save the settings and to close the window.

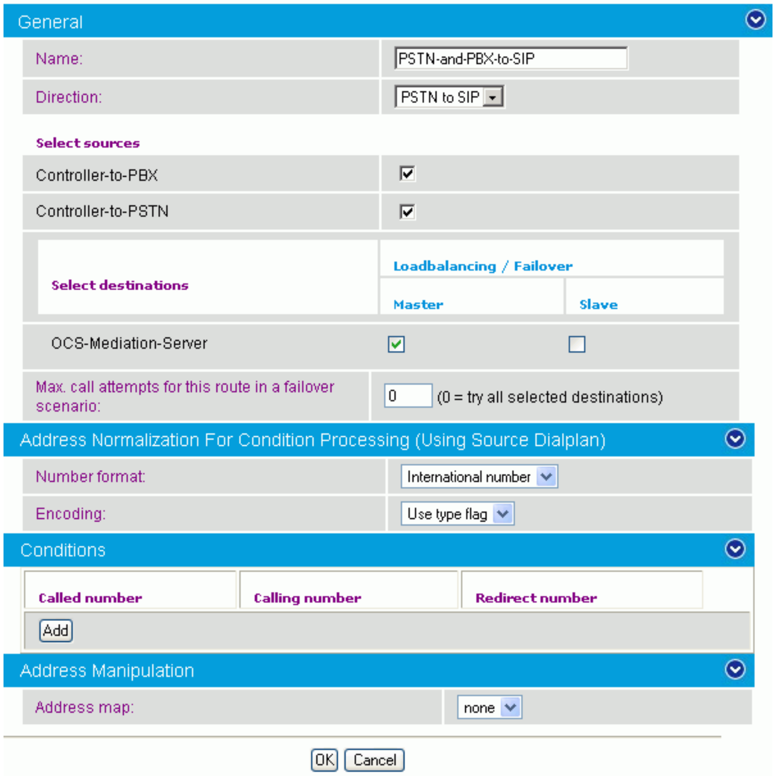
15. Click **Add** again, and configure the route from Microsoft® OCS 2007 to the PBX.



- **Name**: Enter a unique name to easily identify the routing.
- **Direction**: Select **SIP to PSTN** from the dropdown menu.
- **Select sources**: Select the above configured SIP peer as master destination.
- **Select destinations**: Select the controller you configured for the PBX.

Click **OK** to save the settings and to close the window.

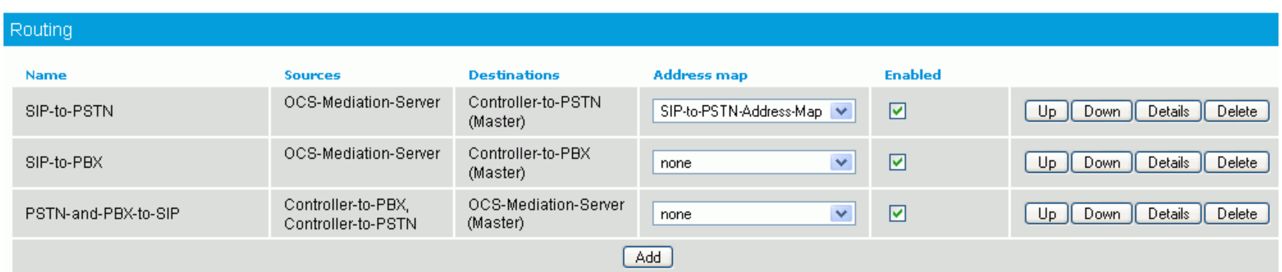
16. Click **Add** again to configure the third route. Configure the following parameters:



- **Name**: Enter a unique name to easily identify the routing.
- **Direction**: Select **PSTN to SIP** from the dropdown menu.
- **Select sources**: Select both controllers of the Diva Media Board.
- **Select destinations**: Select the above configured SIP peer as master destination.
- **Number format**: Select **International number** from the dropdown menu.
- **Encoding**: Select **Use type flag** from the dropdown menu.

Click **OK** to save the settings and to close the window.

17. To change the order of the routings in the main configuration page, click the arrow up or arrow down buttons. The order needs to be the same as shown in this graphic:

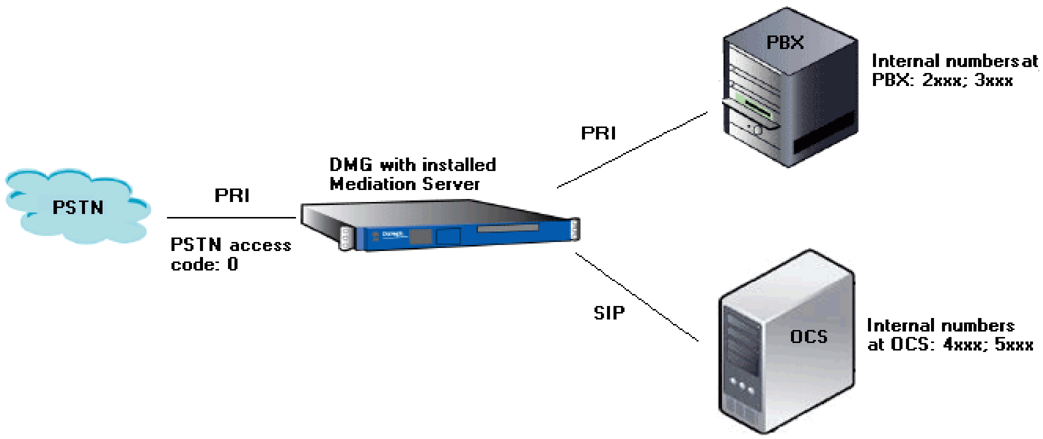


18. Click **Save** in the main configuration page to save the settings and to activate the changes.

19. Configure the Microsoft® Office Communications Server 2007 as described in the Dialogic® 4000 Media Gateway Quickstart Guide.

Using the Dialogic® 4060 Media Gateway between the PSTN and PBX/Microsoft® Office Communications Server 2007

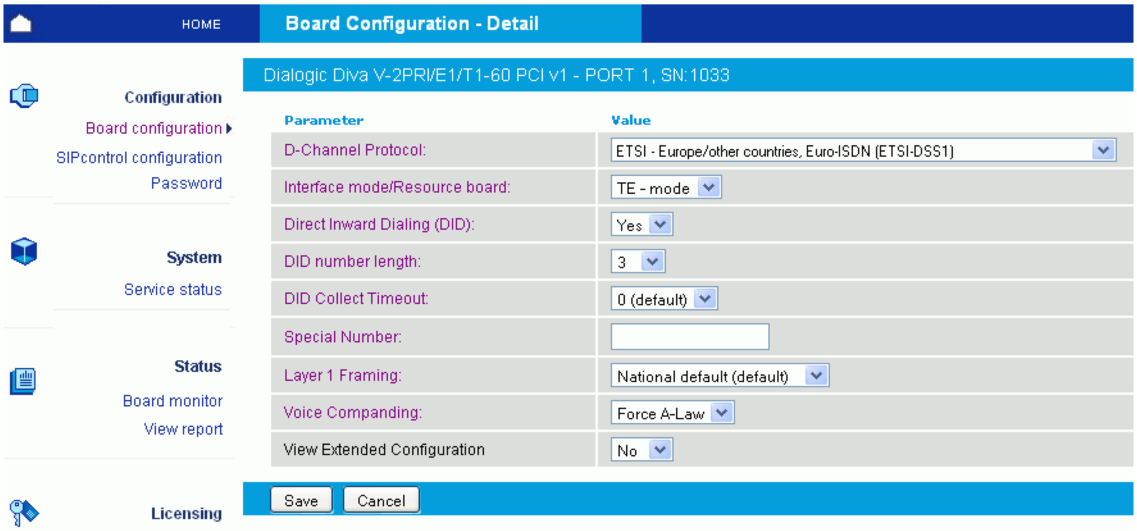
This configuration scenario describes the necessary steps if the Dialogic® Media Gateway is connected between the PSTN and PBX/Microsoft® Office Communications Server (OCS) 2007. This way, the Diva SIPcontrol software can also route calls from the PBX to the PSTN, and vice versa. One PRI line is connected to the PBX, which is configured behind the gateway, and one PRI line is connected directly to the PSTN. The Microsoft® Mediation Server is installed on the gateway. The PBX is configured for the extensions starting with 2 or 3. The Microsoft® OCS 2007 is configured for the extensions starting with 4 or 5. The Diva SIPcontrol software expects a PSTN access code in calls from Microsoft® OCS 2007 or from the PBX to the PSTN; in this case it is an additional 0 (zero). The Diva SIPcontrol software is configured to remove the access code before forwarding the call to the PSTN. For convenience, the Diva SIPcontrol software is also configured to add the outside access code to the calling number in calls coming from the PSTN.



1. Open the Dialogic® Diva® SIPcontrol™ software web interface to configure the required parameters. To do so, click **Start** > **Programs** > **Dialogic Diva** > **SIPcontrol Configuration**.
2. In the Diva SIPcontrol software web interface, click **Board configuration** on the left hand side to open the **Available Diva Boards** page.

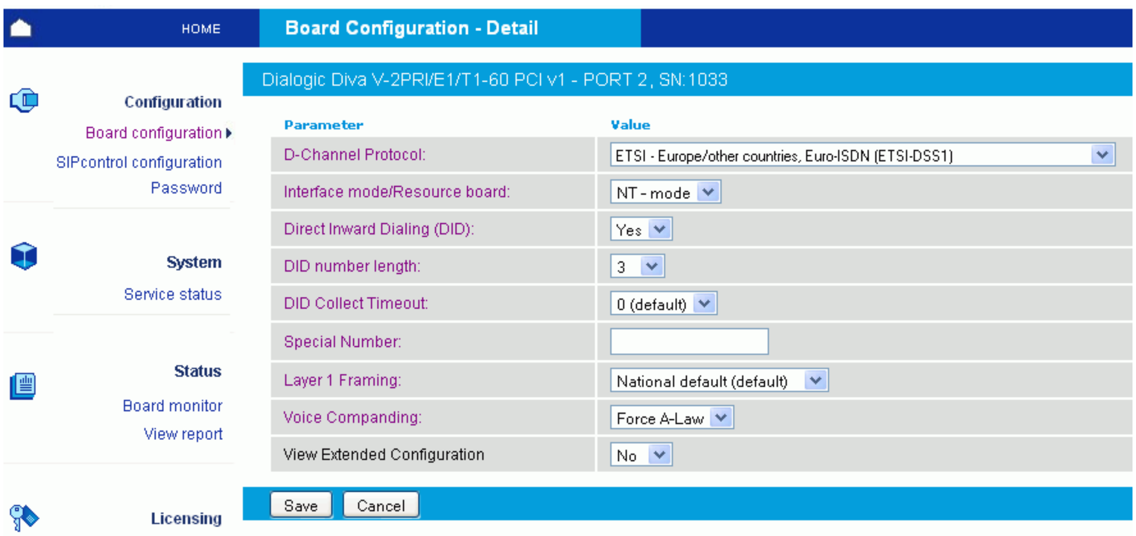
Click either the board icon or the name of the first Dialogic® Diva® Media Board line to open the **Board Configuration - Detail** page.

3. Configure the **D-Channel Protocol** of the PRI line connected to the PSTN. In the example, **ETSI-Europe/other countries, Euro-ISDN (ETSI-DSS1)** is selected. Click **Save**.



4. Click **Board configuration** again and select port 2 of your Diva Media Board to open the **Board Configuration - Detail** page for this board.

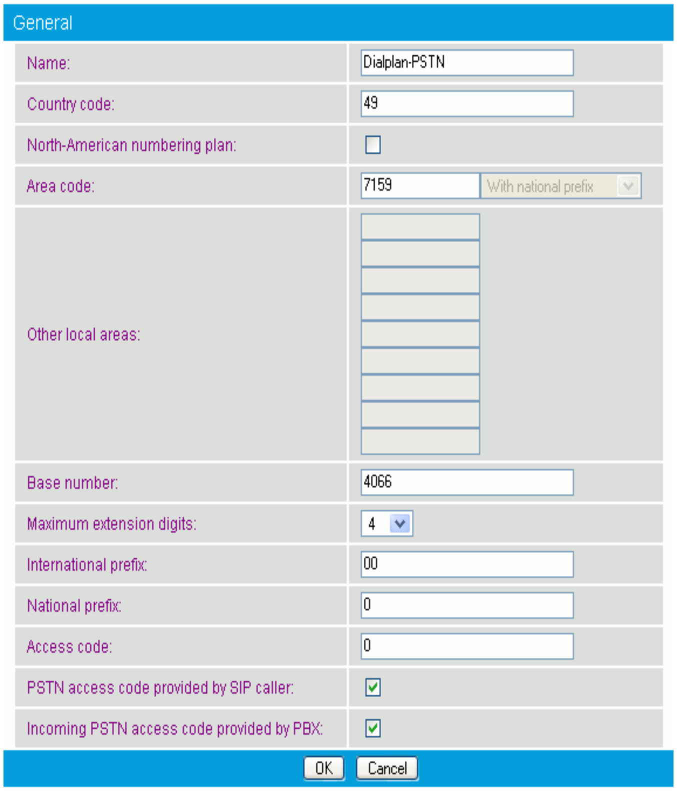
5. Configure the **D-Channel Protocol** and the **Interface mode/Resource board** of the PRI line connected to the PBX. In the example, **ETSI-Europe/other countries, Euro-ISDN (ETSI-DSS1)** and **NT - mode** are selected. Click **Save**.



6. In the Diva SIPcontrol web interface, click **SIPcontrol configuration** on the left hand side to open the **SIPcontrol Configuration** page.

   For this configuration scenario, a dialplan, two address maps, the two PSTN interfaces, the network interface, a SIP peer, and eight routings need to be configured.

7. Open the **Dialplan Configuration**, click **Add**, and configure the following parameters for the dialplan:

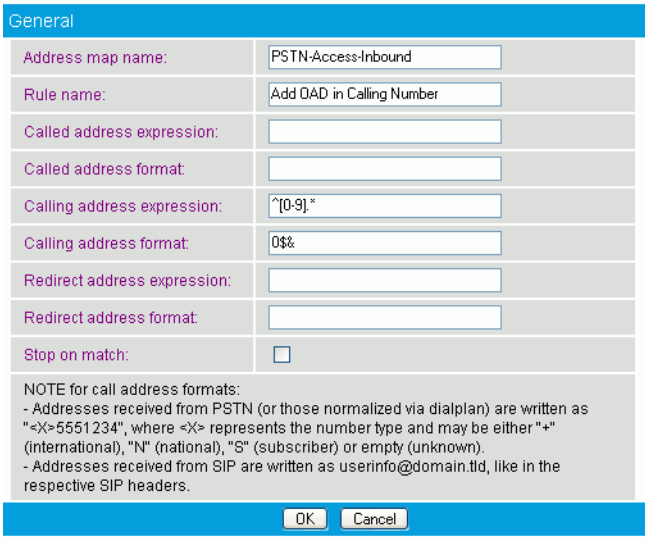


- **Name**: Enter a unique name to easily identify the dialplan.
- **Country code**: Enter the country code of the country in which the Dialogic® Media Gateway is located.
- **Area code**: Enter the area code of the region in which the Dialogic® Media Gateway is located.
- **Base number**: Enter the subscriber or trunk number.
- **Maximum extension digits**: Select the maximum number of extension digits that are provided.
- **International prefix**: Enter the international prefix of the country in which the Dialogic® Media Gateway is located.
- **National prefix**: Enter the national prefix that needs to be dialed for long distance calls within the country in which the Dialogic® Media Gateway is located.
- **Access code**: Enter the digit that is necessary to get access to the public network.
- Enable the options **PSTN access code provided by SIP caller** and **Incoming PSTN access code provided by PBX**.

Click **OK** to save the settings and to close the window.

8. Create two address maps, one each for incoming and outgoing calls. The address maps are necessary to either add or omit the OAD (Outside Access Digit) in the calling number.

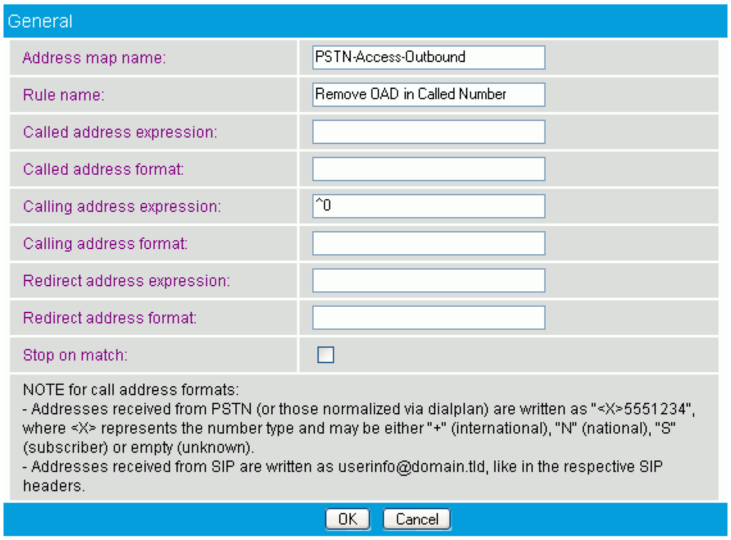
- To create the address map for incoming calls, open the **Address Map Configuration**, click **Add**, and configure the following parameters:



  - **Address map name**: Enter a name for the incoming call address map.
  - **Rule name**: Enter a name that describes the address map rule. This name will be displayed on the main configuration page.
  - **Calling address expression**: Enter the calling address expression as shown in the graphic above. With this expression, the OAD will be added to incoming calls.
  - **Calling address format**: Enter the calling address format as shown in the graphic above.

Click **OK** to save the settings and to close the window.

- To create the second address map, click **Add** again, and configure the following parameters:

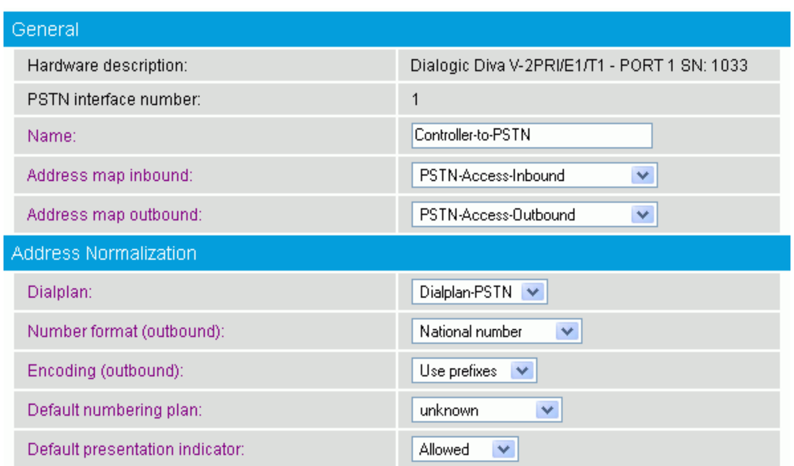


  - **Address map name**: Enter a name for the outgoing call address map.
  - **Rule name**: Enter a name that describes the address map rule. This name will be displayed on the main configuration page.
  - **Calling address expression**: Enter the calling address expression as shown in the graphic above. With this expression, the OAD will be removed from outgoing calls.

Click **OK** to save the settings and to close the window.

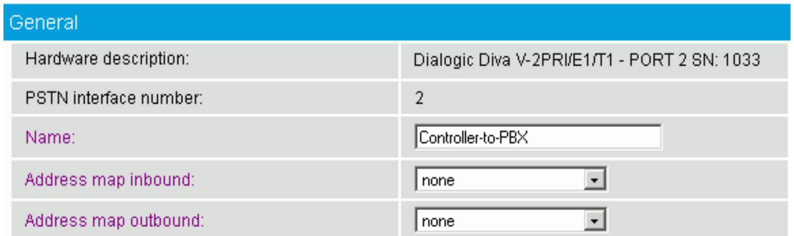
9. Under **PSTN Interface Configuration**, configure controller 1 with PBX-specific settings and controller 2 with PSTN-specific settings. The controller number you configure with PBX-specific settings needs to correspond to the line number in the Diva Configuration Manager that you configured with the switch type of your PBX. The same is true for the controller with the switch type of the PSTN line.

- To configure PSTN-specific parameters, click **Details** at the right of the controller connected to the PSTN line. Select the inbound and outbound address map that you configured for access from the PSTN. Select the **Dialplan** you configured for the PSTN, set **Number format (outbound)** to **National Number**, and **Encoding (outbound)** to **Use prefixes**.



Click **OK** to save the settings and to close the window.

- To identify the controller that is connected to the PBX later in the configuration, click **Details** at the right of the respective controller. Enter a unique name for this controller, e.g., "Controller-to-PBX", as in the graphic below. Leave the remaining parameters at their default values.



General	
Hardware description:	Dialogic Diva V-2PRI/E1/T1 - PORT 2 SN: 1033
PSTN interface number:	2
Name:	Controller-to-PBX
Address map inbound:	none
Address map outbound:	none

Click **OK** to save the settings and to close the window.

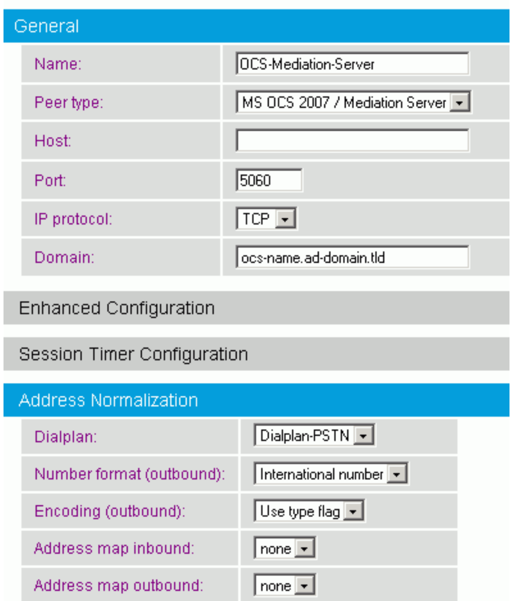
10. Under **Network Interface Configuration**, enable your Ethernet adapter and set the **SIP Listen Port** to **9803**.

Network Interfaces

Name	Device	IP address	UDP listen port	TCP listen port	TLS listen port
Intel(R) PRO1000 GT Desktop	Intel(R) PRO/1000 GT Desktop Adapter - Packet Scheduler Miniport	192.168.213.38	9803	9803	5061
Local Loopback Interface	Local Loopback Interface	127.0.0.1	5060	5060	0

11. Create two SIP peers, one for the PSTN to the Microsoft® Mediation Server installed on the Dialogic® Media Gateway and one for the PSTN to PBX connection.

- To configure the first SIP peer, open the **SIP Peer Configuration**, click **Add**, and enter the following parameters:



General	
Name:	OCS-Mediation-Server
Peer type:	MS OCS 2007 / Mediation Server
Host:	
Port:	5060
IP protocol:	TCP
Domain:	ocs-name.ad-domain.tld

Enhanced Configuration

Session Timer Configuration

Address Normalization	
Dialplan:	Dialplan-PSTN
Number format (outbound):	International number
Encoding (outbound):	Use type flag
Address map inbound:	none
Address map outbound:	none

Under **Edit SIP Peer Configuration**, configure the following parameters:

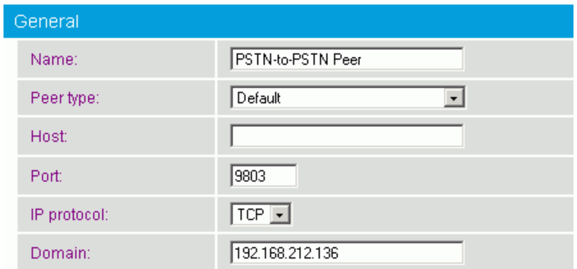
- **Name**: Enter a unique name to easily identify the SIP peer.
- **Peer type**: Select **MS OCS 2007/ Mediation Server** from the dropdown menu.
- **Host**: Enter the IP address or host name of the host PC.
- **Domain**: For the correct domain entry, see the configuration of your Microsoft® Office Communications Server 2007.

Under **Address Normalization Configuration**, configure the following parameters:

- **Dialplan**: Select the dialplan you configured for the controller connected to the PBX.
- **Number format (outbound)**: Select **International number** form the dropdown menu.
- **Encoding (outbound)**: Select **Use type** flag from the dropdown menu.

Click **OK** to save the settings and to close the window.

- To create the second SIP peer, click **Add** again, and configure the following parameters:



Under **Edit SIP Peer Configuration**, configure the following settings:

- **Name**: Enter a unique name to easily identify the SIP peer.
- **Peer type**: Select **Default** from the dropdown menu.
- **Host**: Enter the IP address or host name of the host PC.
- **Domain**: Enter the IP address of the host PC.

Click **OK** to save the settings and to close the menu.

This peer is the loopback SIP peer that is needed to correctly route calls from the PBX to the PSTN.

12. Create eight routings:

- one from the PSTN to Microsoft® Office Communications Server (OCS) 2007,
- one from the PBX to Microsoft® OCS 2007,
- one from Microsoft® OCS 2007 to the PBX,
- one from Microsoft® OCS 2007 to the PSTN,
- two from the PSTN to the PBX, via the PSTN-to-PSTN SIP peer,
- two from the PBX to the PSTN, via the PSTN-to-PSTN SIP peer.

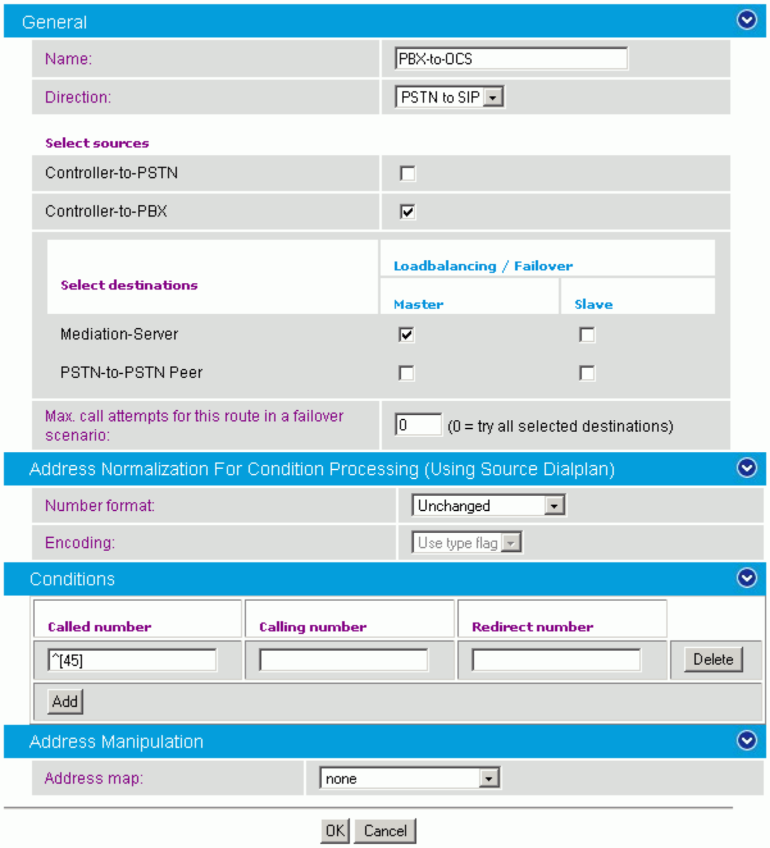
In this scenario, the order of the routings is important because only one routing will be configured with a condition. In this example, the routings are configured in the correct order but it is also explained how to change the order in case you configured them differently.

13. Create the route from the PSTN to Microsoft® OCS 2007 first. To do so, open the **Routing Configuration**, click **Add** and configure the following parameters:

- **Name**: Enter a unique name to easily identify the routing.
- **Direction**: Select **PSTN to SIP** from the dropdown menu.
- **Select sources**: Select the controller you configured for the PSTN as source.
- **Select destinations**: Select the SIP peer you configured for the Microsoft® Mediation Server as **Master** destination.
- **Number format**: Select **Extension** from the dropdown menu.
- **Encoding**: Select **Use prefixes** from the dropdown menu.
- Under **Conditions**, click **Add**, and in **Called number**, enter **^[45]**. Thus, any extension starting with 4 or 5 is routed to the Microsoft® Mediation Server that was configured to manage these extensions.

Click **OK** to save the settings and to close the window.

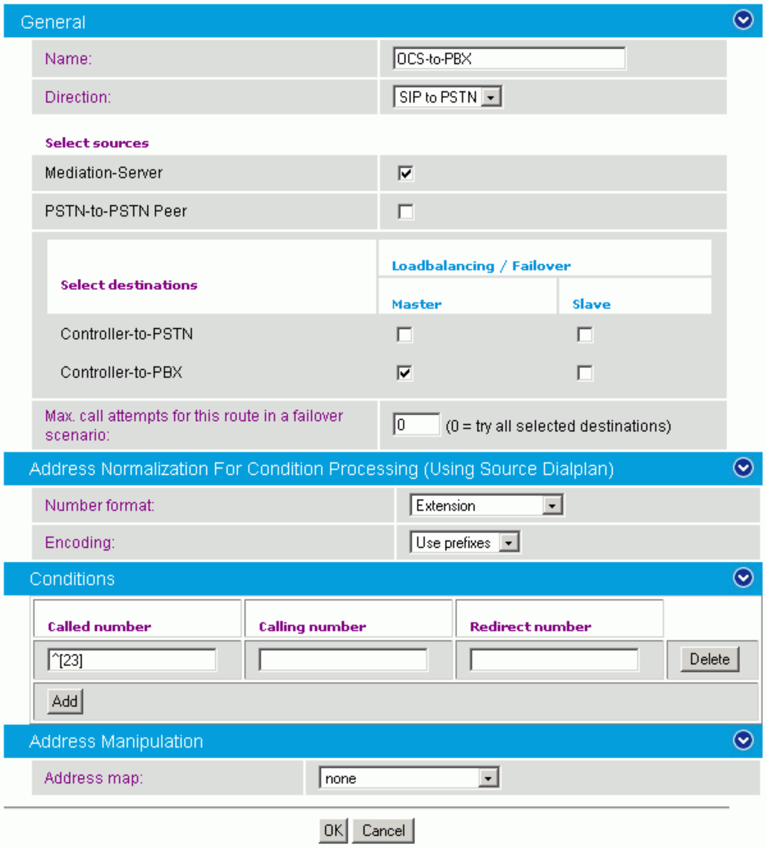
14. Create the second route from the PBX to the Microsoft® OCS 2007. To do so, click **Add** under **Routing Configuration** and configure the following parameters:



- **Name**: Enter a unique name to easily identify the routing.
- **Direction**: Select **PSTN to SIP** from the dropdown menu.
- **Select sources**: Select the controller you configured for the PBX as source.
- **Select destinations**: Select the SIP peer you configured for the Mediation Server as **Master** destination.
- Under **Conditions**, click **Add**, and in **Called number**, enter **^[45]** so that extensions starting with 4 or 5 are routed to the Microsoft® Mediation Server that was configured for these extensions.

Click **OK** to save the settings and to close the window.

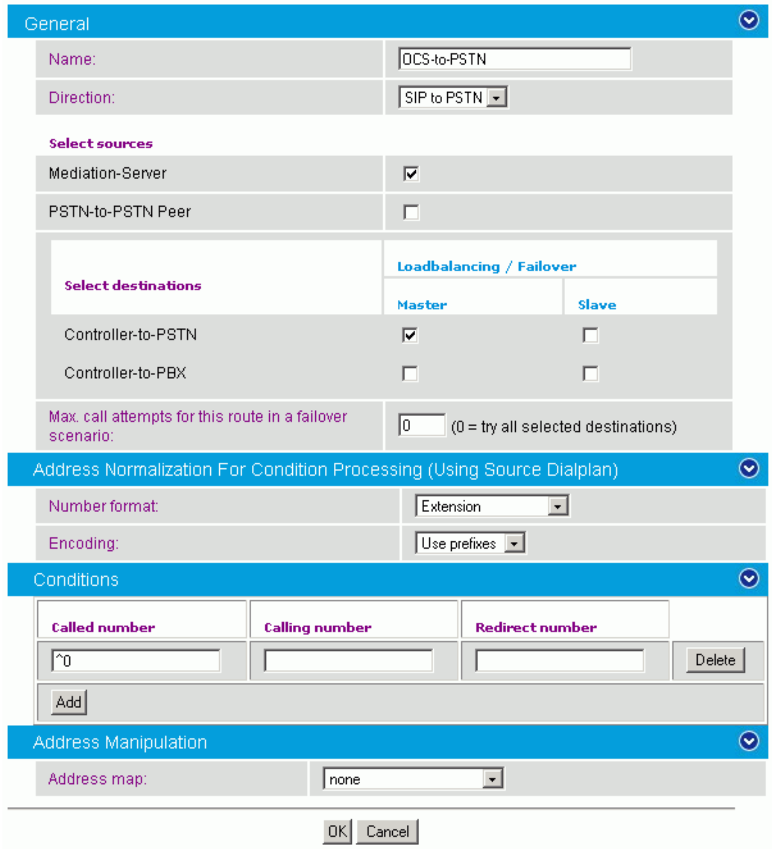
15. Create the third route from the Microsoft® OCS 2007 to the PBX. To do so, click **Add** under **Routing Configuration** and configure the following parameters:



- **Name**: Enter a unique name to easily identify the routing.
- **Direction**: Select **SIP to PSTN** from the dropdown menu.
- **Select sources**: Select the SIP peer you configured for the Microsoft® Mediation Server as source.
- **Select destinations**: Select the controller you configured for the PBX as **Master** destination.
- **Number format**: Select **Extension** from the dropdown menu.
- **Encoding**: Select **Use prefixes** from the dropdown menu.
- Under **Conditions**, click **Add**, and in **Called number**, enter **^[23]** so that extensions starting with 2 or 3 are routed to the PBX that was configured for these extensions.

Click **OK** to save the settings and to close the window.

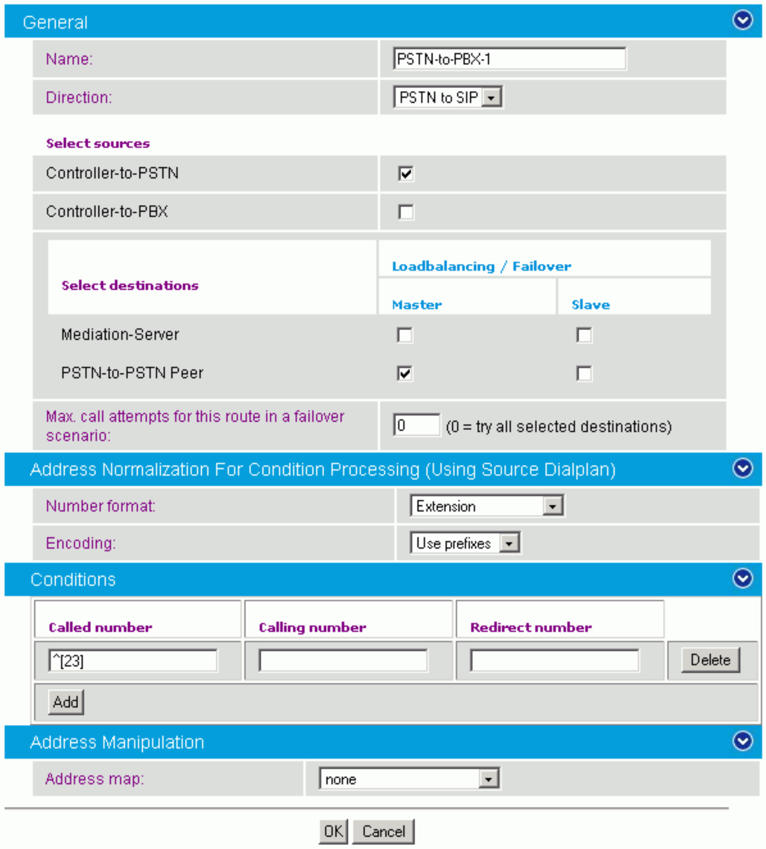
16. Create the fourth route from the Microsoft® OCS 2007 to the PSTN. To do so, click **Add** under **Routing Configuration** and configure the following parameters:



- **Name**: Enter a unique name to easily identify the routing.
- **Direction**: Select **SIP to PSTN** from the dropdown menu.
- **Select sources**: Select the SIP peer you configured for the Microsoft® Mediation Server as source.
- **Select destinations**: Select the controller you configured for the PSTN as **Master** destination.
- **Number format**: Select **Extension** from the dropdown menu.
- **Encoding**: Select **Use prefixes** from the dropdown menu.
- Under **Conditions**, click **Add**, and in **Called number**, enter **^0** to route all calls starting with 0 directly to the PSTN.

Click **OK** to save the settings and to close the window.

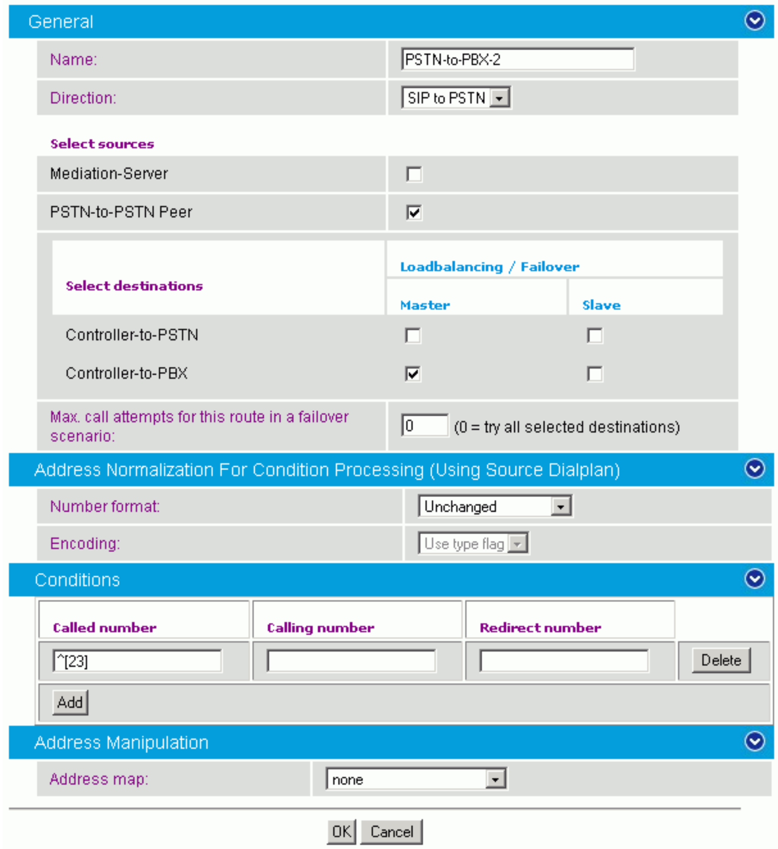
17. Create the fifth route from the PSTN to the PBX. To do so, click **Add** under **Routing Configuration** and configure the following parameters:



- **Name**: Enter a unique name to easily identify the routing.
- **Direction**: Select **PSTN to SIP** from the dropdown menu.
- **Select sources**: Select the controller you configured for the PSTN as source.
- **Select destinations**: Select the SIP peer you configured for the PSTN as **Master** destination.
- **Number format**: Select **Extension** from the dropdown menu.
- **Encoding**: Select **Use prefixes** from the dropdown menu.
- Under **Conditions**, click **Add**, and in **Called number**, enter **^[23]** so that extensions starting with 2 or 3 are routed to the PBX that was configured for these extensions.

Click **OK** to save the settings and to close the window.

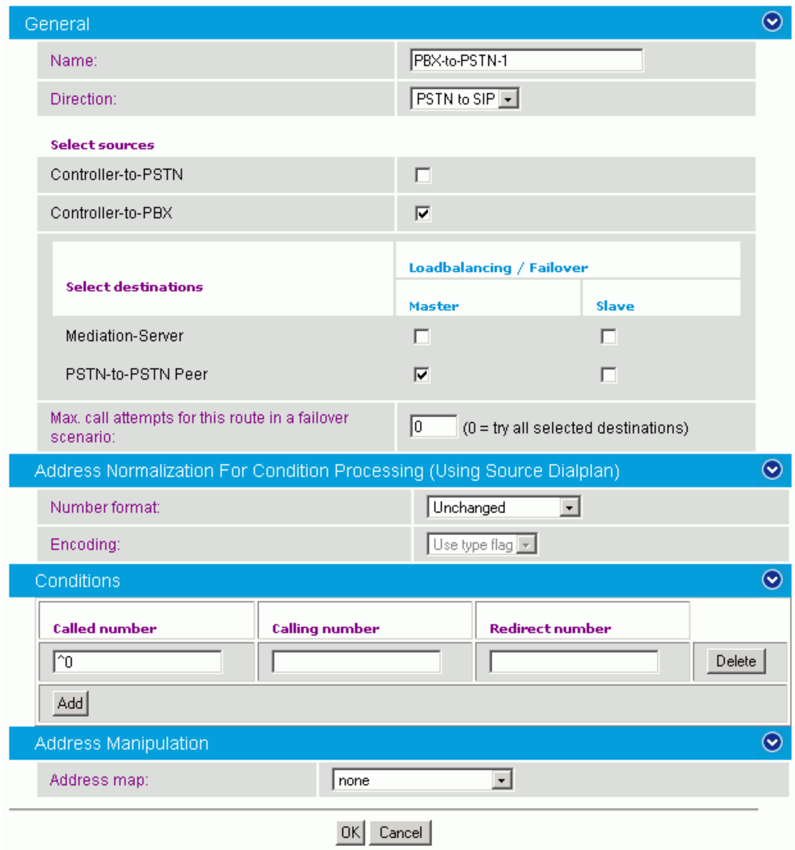
18. Create the sixth route also from the PSTN to the PBX. To do so, click **Add** under **Routing Configuration** and configure the following parameters:



- **Name**: Enter a unique name to easily identify the routing.
- **Direction**: Select **SIP to PSTN** from the dropdown menu.
- **Select sources**: Select the SIP peer you configured for the PSTN as source.
- **Select destinations**: Select the controller you configured for the PBX as **Master** destination.
- Under **Conditions**, click **Add**, and in **Called number**, enter **^[23]** so that extensions starting with 2 or 3 are routed to the PBX that was configured for these extensions.

Click **OK** to save the settings and to close the window.

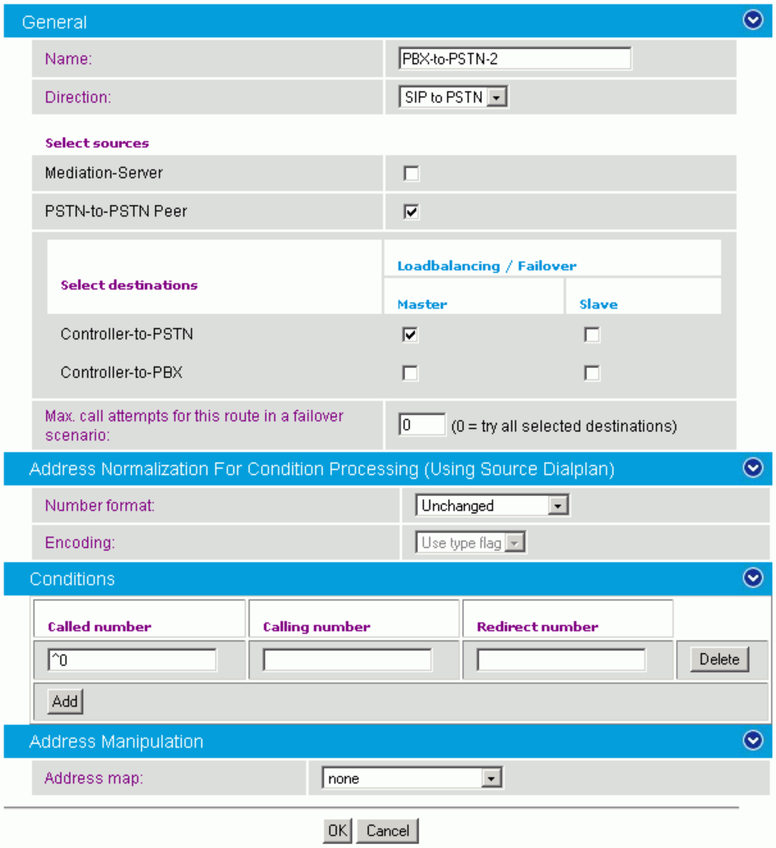
19. Create the seventh route from the PBX to the PSTN. To do so, click **Add** under **Routing Configuration** and configure the following parameters:



- **Name**: Enter a unique name to easily identify the routing.
- **Direction**: Select **PSTN to SIP** from the dropdown menu.
- **Select sources**: Select the controller you configured for the PBX as source.
- **Select destinations**: Select the SIP peer you configured for the PSTN as **Master** destination.
- Under **Conditions**, click **Add**, and in **Called number**, enter **^0** to route all calls starting with 0 to the loopback SIP peer and then to the PSTN.

Click **OK** to save the settings and to close the window.

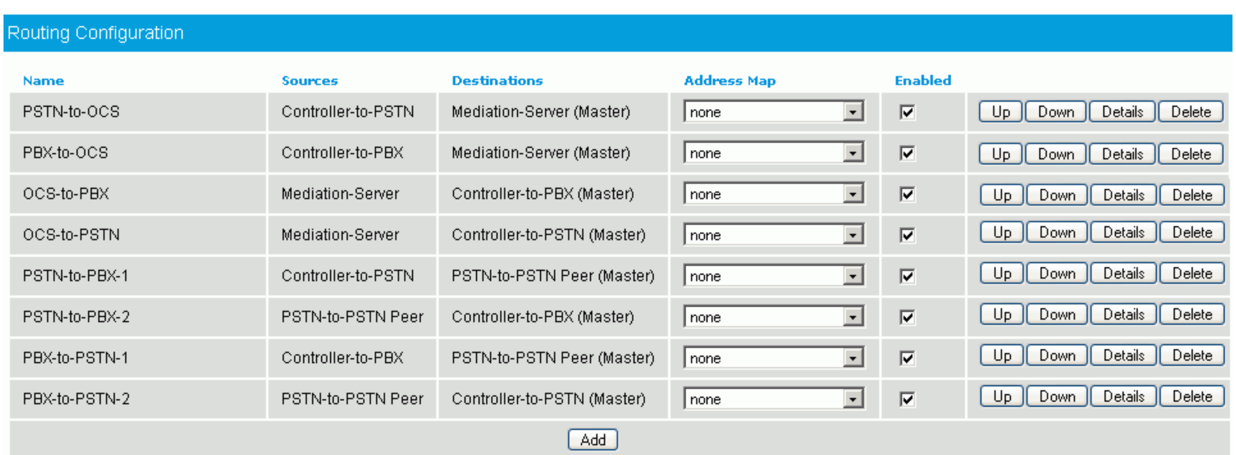
20. Create the eighth route also from the PBX to the PSTN. To do so, click **Add** under **Routing Configuration** and configure the following parameters:



- **Name**: Enter a unique name to easily identify the routing.
- **Direction**: Select **SIP to PSTN** from the dropdown menu.
- **Select sources**: Select the SIP peer you configured for the PSTN as source.
- **Select destinations**: Select the controller you configured for the PSTN as **Master** destination.
- Under **Conditions**, click **Add**, and in **Called number**, enter **^0** to route all calls starting with 0 from the loopback SIP peer directly to the PSTN.

Click **OK** to save the settings and to close the window.

21. To change the order of the routings in the main configuration page, click the arrow up or arrow down buttons. The order needs to be the same, as shown in following graphic:


Routing Configuration

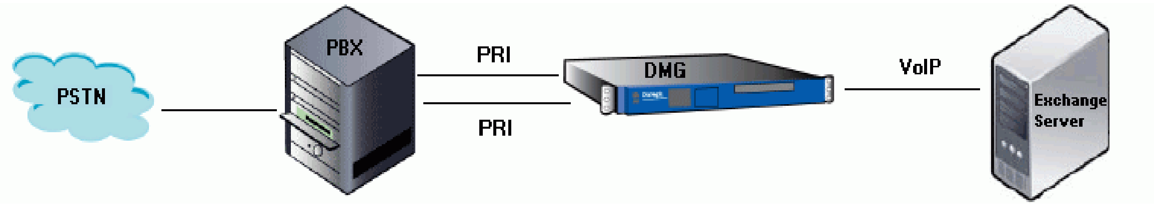
Name	Sources	Destinations	Address Map	Enabled	
PSTN-to-OCS	Controller-to-PSTN	Mediation-Server (Master)	none	☑	Up Down Details Delete
PBX-to-OCS	Controller-to-PBX	Mediation-Server (Master)	none	☑	Up Down Details Delete
OCS-to-PBX	Mediation-Server	Controller-to-PBX (Master)	none	☑	Up Down Details Delete
OCS-to-PSTN	Mediation-Server	Controller-to-PSTN (Master)	none	☑	Up Down Details Delete
PSTN-to-PBX-1	Controller-to-PSTN	PSTN-to-PSTN Peer (Master)	none	☑	Up Down Details Delete
PSTN-to-PBX-2	PSTN-to-PSTN Peer	Controller-to-PBX (Master)	none	☑	Up Down Details Delete
PBX-to-PSTN-1	Controller-to-PBX	PSTN-to-PSTN Peer (Master)	none	☑	Up Down Details Delete
PBX-to-PSTN-2	PSTN-to-PSTN Peer	Controller-to-PSTN (Master)	none	☑	Up Down Details Delete

Add

22. Click **Save** in the main configuration page to save the settings and activate the changes.
23. Configure the Microsoft® Office Communications Server 2007 as described in the Dialogic® 4000 Media Gateway Quickstart Guide.

Use case for Microsoft® Exchange Server 2007

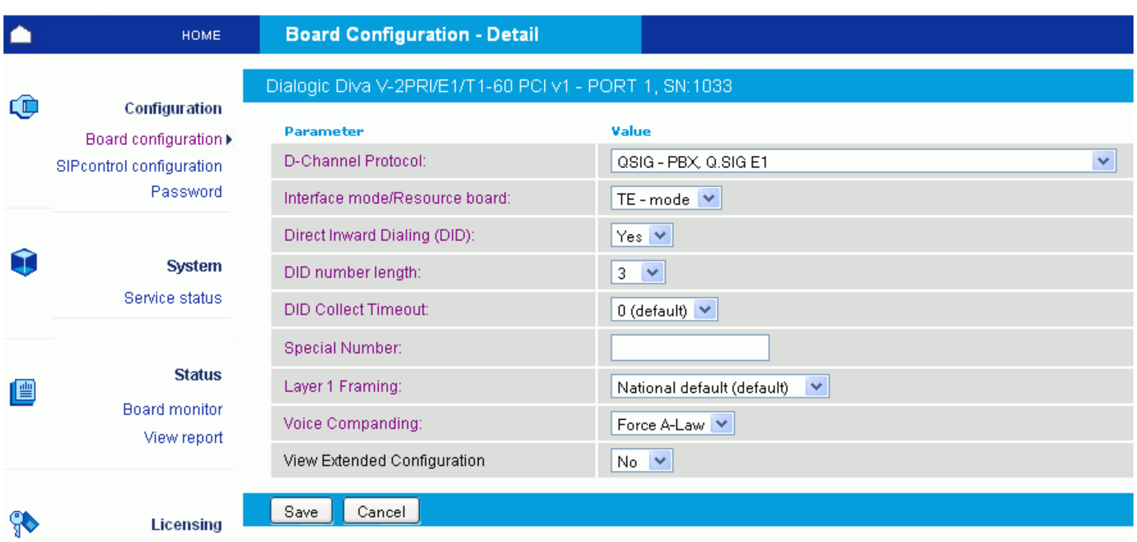
This configuration scenario describes the necessary steps to configure the Dialogic® 4060 Media Gateway between the PBX and the Microsoft® Exchange Server 2007 as shown in the grapic.



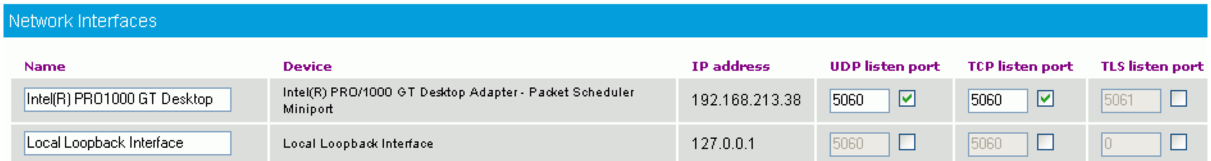
1. Activate the fax license as described in the Dialogic® 4000 Media Gateway Series Quickstart Guide.
1. Activate the fax license as described in **License Activation** in the Dialogic® 4000 Media Gateway Quickstart Guide.
2. Open the Dialogic® Diva® SIPcontrol™ Software web interface to configure the required parameters. To do so, click **Start** > **Programs** > **Dialogic Diva** > **SIPcontrol Configuration**.
3. In the Diva SIPcontrol software web interface, click **Board Configuration** on the left hand side to open the **Available Diva Boards** page.

Click either the board icon or the name of the Dialogic® Diva® Media Board to open the **Board Configuration - Detail** page.

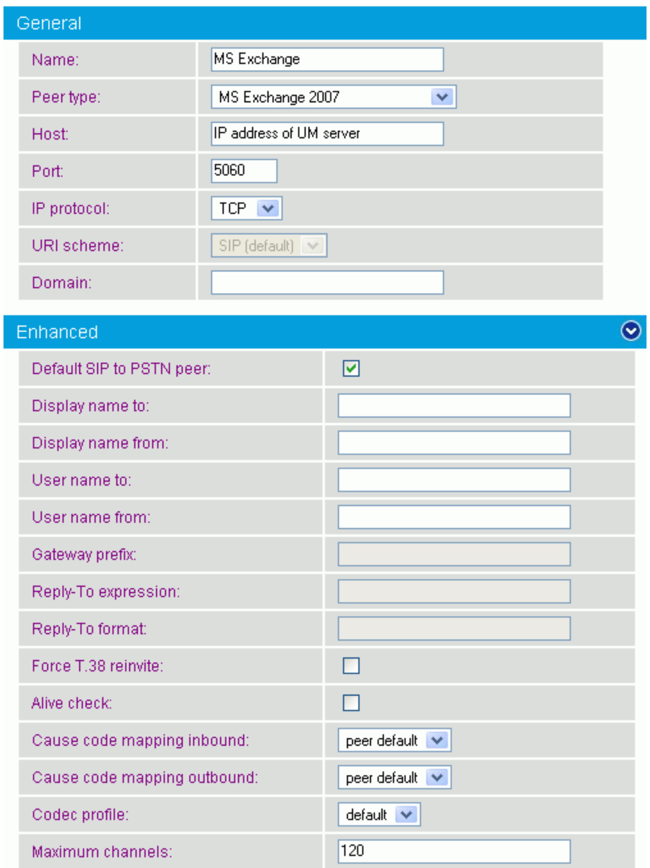
4. Configure the **D-Channel Protocol** of the PBX. In the example, **QSIG-PBX, Q.SIG E1** is selected. Click **Save**.



5. Repeat steps 3 and 4 for the other PRI line.
6. In the Diva SIPcontrol software web interface, click **SIPcontrol configuration** on the left hand side to open the **SIPcontrol Configuration** page. For this configuration scenario, the PSTN interface, the network interface, a SIP peer, and a routing need to be configured.
7. Go to **Network Interface Configuration**, set the **SIP Listen Port** to **5060**, and check the **Enabled** box of your Ethernet adapter.



8. Configure the SIP peer settings. To do so, open the **SIP Peer Configuration**, click **Add**, and configure the following parameters:



Under **Edit SIP Peer Configuration**, configure the following parameters:

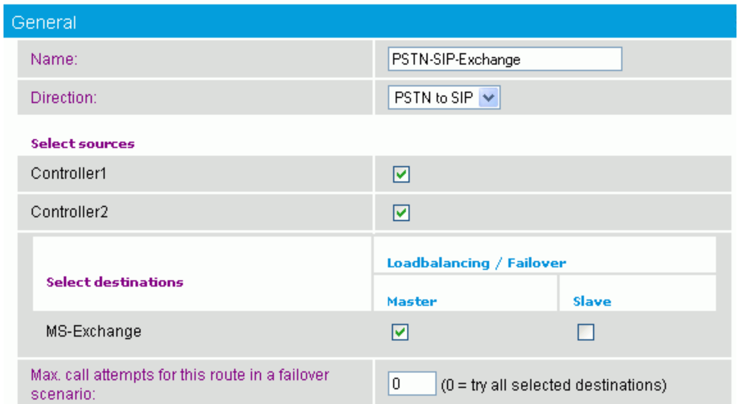
- **Name**: Enter a name for the SIP peer configuration.
- **Peer type**: Select **MS Exchange 2007** as peer type.
- **Host**: Enter the IP address or host name of your Unified Messaging server.

Under **Enhanced Configuration**, enable the **Default SIP to PSTN peer** check box.

Click **OK** to save the settings and to close the window.

9. Configure one routing for the PSTN to SIP and one for SIP to the PSTN. To do so, click **Routing Configuration** and then click **Add** to open the **Edit Routing Configuration** window.

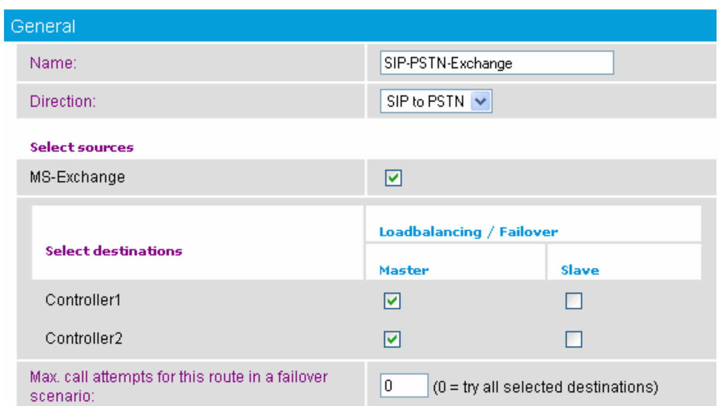
- For the PSTN to SIP route, configure the following parameters:



  - **Name**: Enter a unique name to easily identify the routing.
  - **Direction**: Select **PSTN to SIP** from the dropdown menu.
  - **Select sources**: Select both controllers as source.
  - **Select destinations**: Select the SIP peer as **Master** destination.

  Click **OK** to save the settings and to close the window.

- For the SIP to the PSTN route, click **Add** again and configure the following parameters:



  - **Name**: Enter a unique name to easily identify the routing.
  - **Direction**: Select **PSTN to SIP** from the dropdown menu.
  - **Select sources**: Select both controllers as source.
  - **Select destinations**: Select the SIP peer as **Master** destination.

  Click **OK** to save the settings and to close the window.

10. Click **Save** in the main configuration page to save the settings and activate the changes.

SNMP Support For A Dialogic® Diva® Media Board

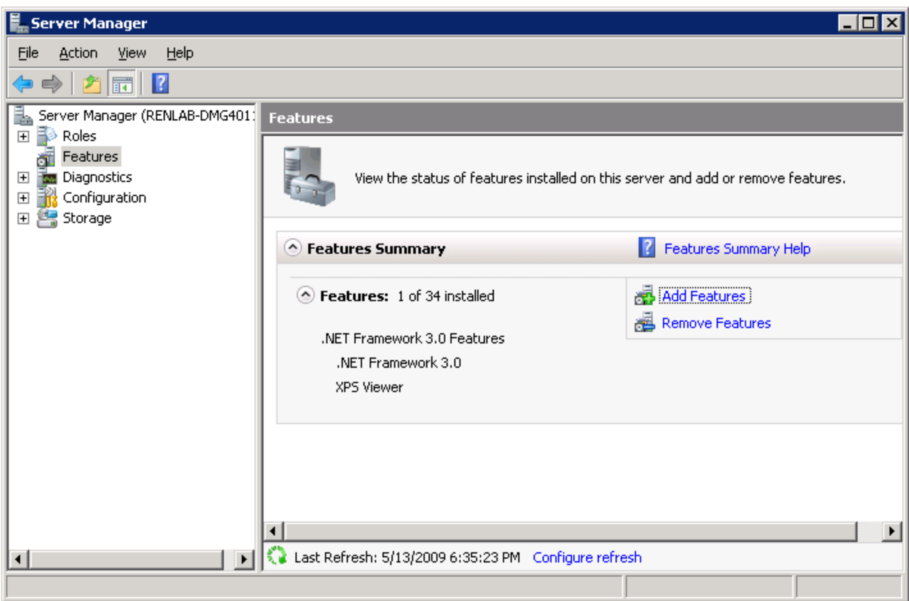
The Windows® implementation of the Simple Network Management Protocol (SNMP) is used to configure remote devices or to monitor network performance, to audit network usage, and to detect network faults or inappropriate access. The SNMP support is only available if the service is installed for your operating system. The output formats are defined in the MIB specification. To see the messages of the SNMP, you need specific SNMP tools that are not part of the Dialogic® Diva® System Release software. To activate the SNMP service, use the Dialogic® Diva® Configuration Manager as described below.

To activate SNMP support for a Dialogic® Diva® Media Board

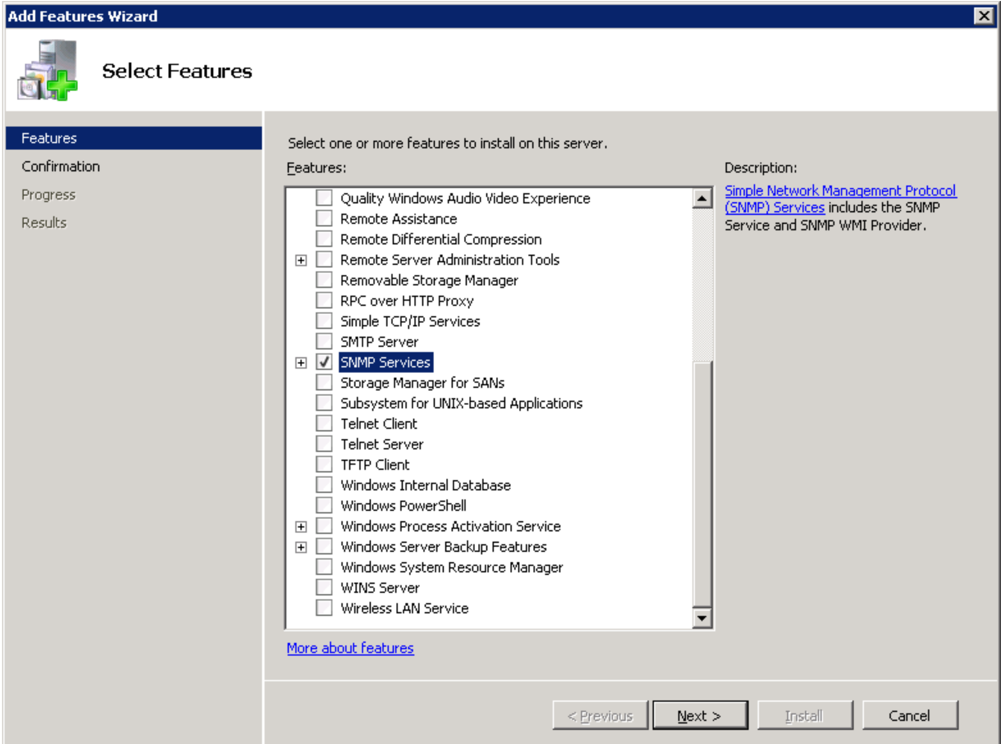
1. Install the Windows® SNMP Service as described below.
2. Add the SNMP Service in the Diva Configuration Manager as described on page 95.
3. Install an SNMP tool, e.g., Net.SNMP (optional, for testing only).
4. Restart your computer.
5. Verify the service status as described on page 95.
6. Verify the function of the SNMP Service as described on page 97.

To install the Windows® SNMP Service under Microsoft® Windows Server® 2008:

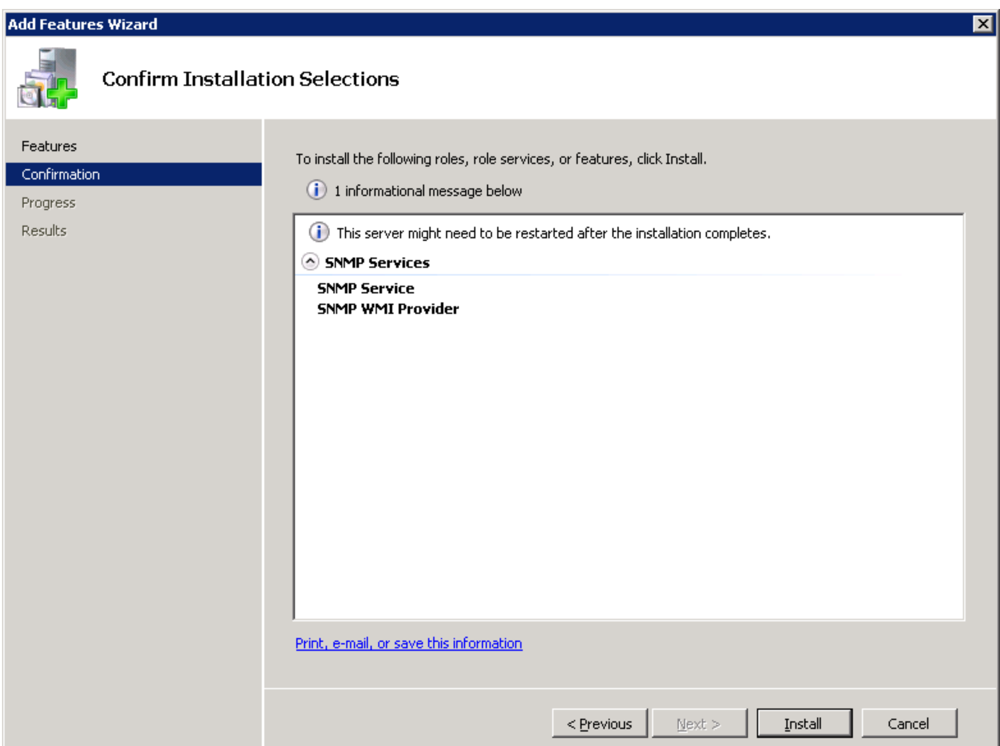
1. Open the Microsoft® Server Manager via **Start** > **Server Manager**.
2. In the **Server Manager** window, go to **Features** and click the **Add Features** link.



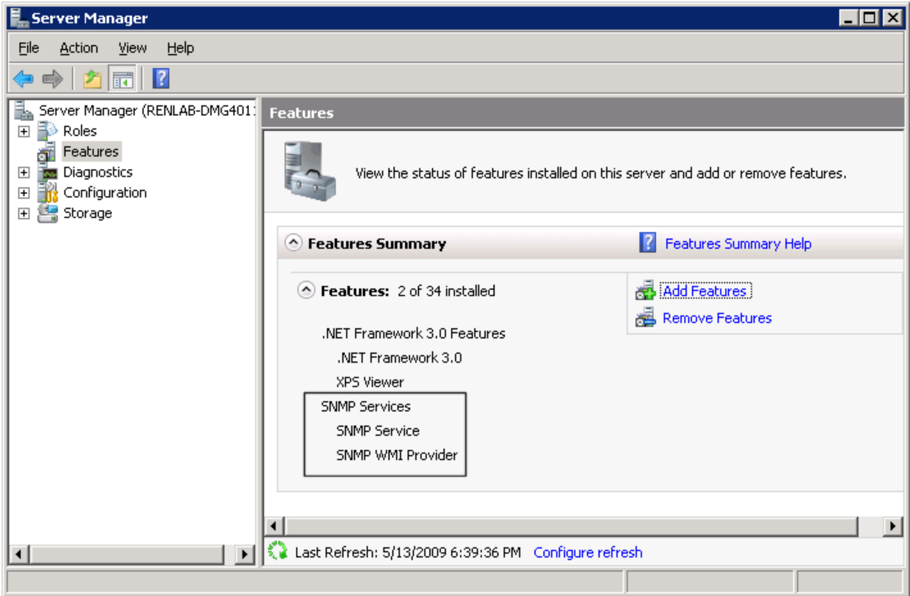
3. In the **Add Features Wizard** window, select **SNMP Services** and click **Next**.



4. Click **Install**, to intall the SNMP Services.



5. After the installation of the SNMP Services has finished, close the **Add Features Wizard** window. You will see the SNMP Services added to the list of installed features in the **Server Manager** window.



You can now add the SNMP Service to the Dialogic® Diva® Configuration Manager as described below.

To add the SNMP Service in the Dialogic® Diva® Configuration Manager

1. Click **Start** > **All Programs** > **Dialogic Diva** > **Configuration Manager** to open the Diva Configuration Manager.
2. In the menu bar, click **Insert** > **SNMP Service**. The SNMP Service is added to the Services layer.
3. Activate the configuration. Once the configuration is activated, the Dialogic® Diva® System Release software validates if Windows® SNMP support is available. If it is not available, an error message is displayed and the SNMP icon is removed from the configuration.

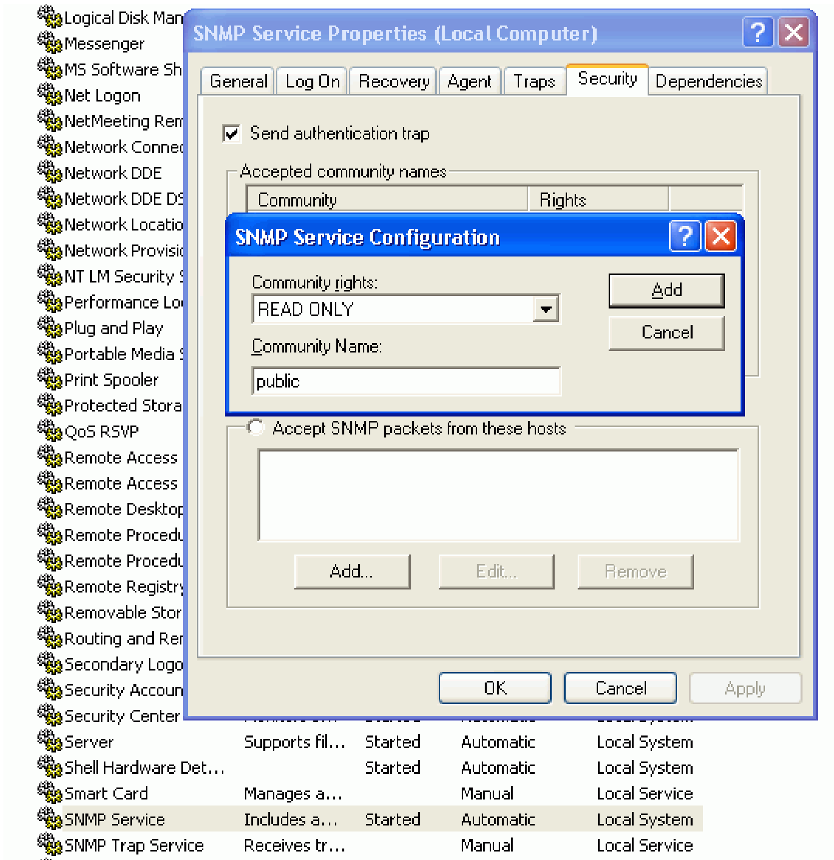
Note: You do not need to connect the SNMP service to any Dialogic® Diva® Media Board. The SNMP is always available for all installed Diva Media Boards.

You can now install the SNMP tool and restart the PC. To install the SNMP tool correctly, consult the documentation of the tool.

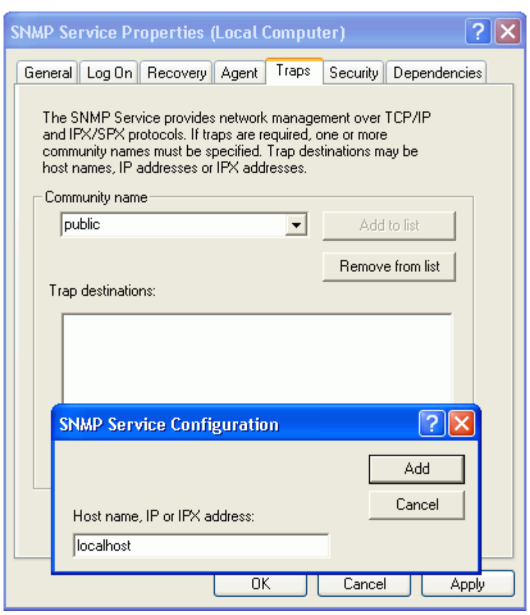
To verify the service status

1. Click **Start** > **Control Panel** > **Administrative Tools** to open the **Administrative Tools** window.
2. In the **Administrative Tools** window, double-click **Services**.
3. In the **Services** window, right-click **SNMP Service** and select **Properties** from the list.

4. In the **Properties** dialog box, click the **Security** tab and under **Accepted community names**, click **Add**. Enter a community name, for instance, public, and for **Community rights**, select **READ ONLY**.



5. Click the **Traps** tab, enter the community name you added in the **Security** tab, and click **Add to list**.
6. Under **Traps destinations**, click **Add**, enter the name or IP address of the host computer, and click **Add**.



7. The host name is added to the list of Trap destinations.
8. Click **OK** to close the dialog box.

9. Restart the SNMP Service. To do so, right-click **SNMP Service** in the **Services** window and select **Restart** from the list.
10. Close the **Services** window.

To verify the function of the SNMP Service

1. Click **Start** > **Run** and type `cmd` to open a DOS window.
2. In the DOS window type: `snmpwalk -v 2c -c public localhost interface | find_"Diva"`

   The result should be similar to the following, which is for a Dialogic® Diva® V-4PRI Media Board:

```
IF-MIB::ifDescr.101 = STRING: Dialogic_Diva_V-4PRI/E1/T1_1030
IF-MIB::ifDescr.133 = STRING: Dialogic_Diva_V-4PRI/E1/T1_1030
IF-MIB::ifDescr.164 = STRING: Dialogic_Diva_V-4PRI/E1/T1_1030
IF-MIB::ifDescr.195 = STRING: Dialogic_Diva_V-4PRI/E1/T1_1030
```

3. In the DOS window type: `snmptrapd -f -L o`

   The result should be similar to the following:

```
2006-01-28 11:14:35 NET-SNMP version 5.2.1.2 Started.
```

You can create an output of traps if you change the status of the layer 1/2, for instance by disconnecting the cable from the Diva Media Board. The result after changing the status of layer 1/2 should be similar to the following:

2006-01-28 11:16:25 localhost [127.0.0.1] (via UDP: [127.0.0.1]:1053) TRAP, SNMP v1, community public

SNMPv2-SMI::enterprises.434.2 Link Up Trap (0) Uptime: 1:16:47.06

IF-MIB::ifIndex.101 = INTEGER: 101

SNMPv2-SMI::enterprises.434.2 Link Down Trap (0) Uptime: 1:16:48.57

IF-MIB::ifIndex.101 = INTEGER: 101

2006-01-28 11:16:26 localhost [127.0.0.1] (via UDP: [127.0.0.1]:1053) TRAP, SNMP v1, community public

SNMPv2-SMI::enterprises.434.2 Link Up Trap (0) Uptime: 1:16:48.81

IF-MIB::ifIndex.101 = INTEGER: 101

Supported MIBs, OIDs, and traps

This section provides information about supported MIBs, OIDs, and traps by the Dialogic® Diva® SNMP service and about the relationship between supported OIDs and Dialogic® Diva® Media Board management interface variables.

MIB-II (RFC 1213/2233)	Path	Description
MIB-II	interfaces.ifTable.ifEntry.	
	ifIndex	Unique index of Dialogic® Diva® interfaces starting with ifIndex-offset + 1 (see option -oN). First, all installed Dialogic® Diva® Media Boards are listed, followed by the available B-channels.
	ifDescr	For Diva Media Boards, the board name and it's serial number are returned. For B-channels, the string "BRI + ifIndex_of_board + number_of_b- channel_on_board" is returned.
	ifType	The type of the interface according to IANA: PRI, BRI, ISDN.
	ifMTU	Since the concept of MTU is not applicable on Dialogic® Diva® interfaces, they return always 0.
	ifSpeed	The maximum interface speed in bps
	ifAdminStatus	Always up
	ifOperStatus	The current operating status of the interface

	ifInBytes, ifInPackets, ifInErrors, ifOutBytes, ifOutPackets, ifOutErrors	For Dialogic® Diva® Media Boards, the added values of the D- and B-channel interface counters are returned. **mantool** reports these values in the following paths "Statistics\\[D|B]-Layer2\\[R|X]-[Bytes|Frames|Errors]". For B-channels, the following values are reported: "State\\B[n]\\L2 Stats\\R- [Bytes|Frames|Errors]".
	ifPhysAddr	Returns vendor-id, PnP-id, serial number of Diva Media Boards formatted as hex string. Returns no information for B-channels.
	LinkUp/LinkDown Traps	For status changes of interfaces a trap is generated that includes the appropriate ifOperStatus varbind. Trap destinations and access parameters must be configured in the underlying master agent (trapsink, etc.).
ISDN-MIB (RFC2127)	transmission.isdnMib.isdnMibObjects.isdnSignalingGroup	
	isdnSignalingGetIndex	Number of possible D-channels (equals number of installed Diva Media Boards)
ISDN-MIB	transmission.isdnMib.isdnMibObjects.isdnBasicRateGroup .isdnBasicRateTable.isdnBasicRateEntry	Dialogic® Diva® BRI Media Boards
	isdnBasicRateIfType	isdns or isdnu (IANA-ifType 75, 76)
	isdnBasicRateLineTopology	pointToPoint or pointToMultipoint
	isdnBasicRateIfMode	TE mode or NT mode
	isdnBasicRateSignalMode	D-channel active or inactive
ISDN-MIB	transmission.isdnMib.isdnMibObjects.isdnBearerGroup.isdnBearerTable.isdnBearerEntry	B-channels
	isdnBearerChannelType	dialup or leased
	isdnBearerOperStatus	idle, active, unknown
	isdnBearerChannelIndex	Index of B-channel per Diva Media Board
	isdnBearerPeerAddress	Remote address
	isdnBearerPeerSubAddress	Remote sub address
	isdnBearerCallOrigin	Answer or originate
	isdnBearerInfoType	Info type as per Q.931 (unrestrictedDigital)
	isdnBearerCallConnectTime	Time measured from start of divasnmpx
DIAL-CONTROL-MIB	transmission.dialControlMib.dialControlMibObjects.callActive.callActiveTable.callActiveEntry	
	callActiveSetupTime	Timeticks at start of call, measured from start of **divasnmpx**.
	callActiveIndex	Unique index
	callActivePeerAddress	Address of remote partner
	callActivePeerSubAddress	Subaddress of remote partner
	callActivePeerId	Always 0 (unknown)
	callActivePeerIfIndex	Always 0 (unknown)

	callActiveLogicalIfIndex	Index of entry in ifTable for the interface used by this call.
	callActiveConnectTime	0 if the call was not connected, otherwise timeticks measured from start of **divasnmpx**.
	callActiveCallState	State of call
	callActiveCallOrigin	Direction of call: Answer or originate
DIAL-CONTROL-MIB (RFC2128)	transmission.dialControlMib.dialControlMibObjects.callHistory	
	callHistoryTableMaxLength	The maximum number of entries in the callHistoryTable (read/write).
	callHistoryRetainTimer	The minimum amount of time in minutes that a callHistoryEntry will be maintained before being deleted.
DIAL-CONTROL-MIB	transmission.dialControlMib.dialControlMibObjects.callHistory.callHistoryTable.callHistoryEntry	
	callHistoryPeerAddress	Address of remote partner
	callHistoryPeerSubAddress	Subaddress of remote partner
	callHistoryPeerId	Always 0
	callHistoryPeerIfIndex	Always 0
	callHistoryLogicalIfIndex	Index of entry in ifTable for the interface used by this call.
	callHistoryDisconnectCause	Reason for disconnecting this call
	callHistoryDisconnectText	empty
	callHistoryConnectTime	Timeticks measured from start of **divasnmpx**.
	callHistoryDisconnectTime	Timeticks measured from start of **divasnmpx**.
	callHistoryCallOrigin	Direction of call: Answer or originate.

The definition for the ISDN-, DIAL-CONTROL-, and DS1-MIB can be imported into any management application to decode the OIDs reported by **divasnmpx**. For net-snmp, simply copy these files to the standard MIB path (usually <%program files%>\netsnmp\share\snmp\mibs) and tell the snmp command line tools to use them by exporting/setting the environment variable "MIBS" with the names of the appropriate MIBs (or simply the keyword ALL), e.g., **Set MIBS=ALL**.

Verify The Line Configuration With The Dialogic® Diva® Line Test Tool

To check the line configuration, use the Dialogic® Diva® Line Test tool available under **Start > Programs** > **Dialogic Diva** > **Line Test**.

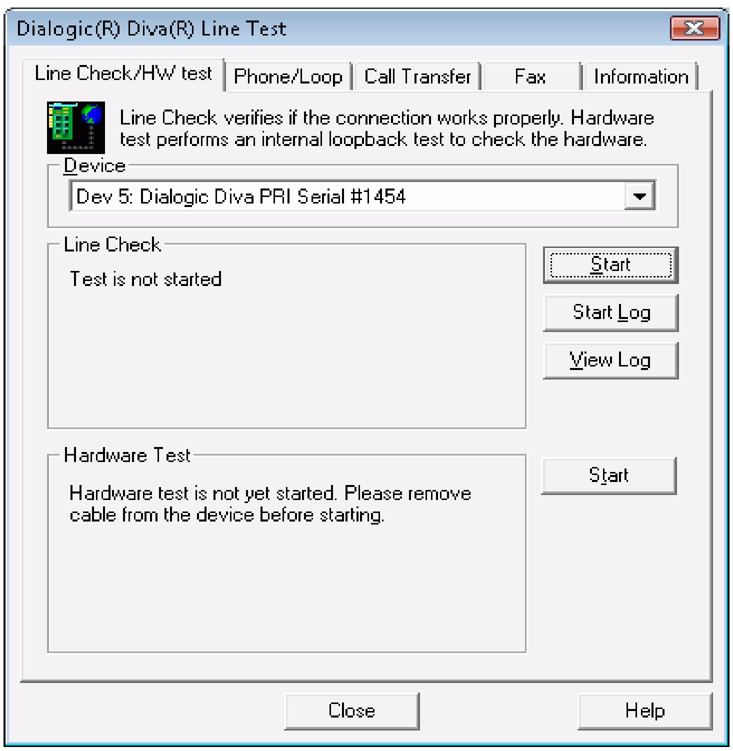
The Diva Line Test tool offers the following tests:

- Line Check: Performs a quick check of your Dialogic® Diva® System Release software installation and the physical connection.
- Hardware test: Performs a test of the physical controller only.
- Phone/Loop test: Performs basic inbound or outbound phone tests to test the connection to other telephones or to itself.
- Call Transfer: Performs different call transfer tests, with the option to choose the transfer type.
- Fax test: Performs basic inbound or outbound fax tests.

The Diva Line Test tool also has a Blink LED button integrated on the information page to easily identify a physical Diva Media Board or the physical line of a controller.

Line Check

1. Open the Diva Line Test tool and click the **Line Check/HW Test** tab.



2. Under **Device** select the line of the Dialogic® Diva® Media Board to test.
3. Click **Start** to begin the test.

   If the line check test reports an error, verify that:

- the cabling is connected correctly.
- the switch type, network type, and ISDN or phone numbers are configured correctly in the Dialogic® Diva® Configuration Manager.
- the SPIDs (Service Profile Identifiers) are configured correctly in the Diva Configuration Manager if you use a North American switch type.
- the Diva Media Board is not conflicting with any other hardware.
- the telco company is not experiencing any issues.

4. If line check reports no issues, and you are still having trouble connecting, there might be a problem in the configuration of the application you are using with your Diva Media Board (such as Dial-Up Networking or fax software). Check the configuration and repeat the test.
5. Click **Stop** to abort the test.
6. If you wish to trace the test for analyzing purposes, you can create a trace file.
7. Click **Close** to end the configuration dialog.

Hardware test

The hardware test performs a test of the controller only. It changes the controller to run under the internal loop-back mode. Starting with the first channel, the hardware test tries to connect/disconnect each channel and stops when the highest channel is tested.

1. Open the Diva Line Test tool and click the **Line Check/HW Test** tab
2. Select under **Device** the line of the Dialogic® Diva® Media Board to test.
3. Click **Start** to begin the test.

   If the hardware test reports an error, verify that:

   - the cable is NOT connected to the board,
   - the Diva Media Board is loaded correctly,
   - the newest Dialogic® Diva® System Release software is installed,
   - the test is successful with a standard protocol like ETSI or NT1.
   - If the test still fails, contact the Dialogic Customer Support personnel. For more information, see Customer Service on page 107.

   Note: To help you as efficiently as possible, the Customer Support personnel will ask you for details of the tests you have conducted and their results. So, be sure you have all information handy when contacting them.
4. Click **Stop** to abort the test.
5. Click **Close** to end the configuration dialog.

Phone/Loop test

The phone test performs an outgoing call to verify the connectivity to another telephone. You can stop the test once the remote phone is ringing. If the phone answers the call, an announcement is played.

The loop test performs an incoming call to itself to test the end-to-end connectivity with an inbound tone test. Since some networks use a different channel allocation (like different Q.SIG versions), it might be possible to connect a call despite the wrong physical channel being in use or not operational at all.

1. Under **Device** select the line of the Dialogic® Diva® Media Board to test.
2. Enter the **Called Party Number** and **Calling Party Number**.
3. To configure advanced settings, click **Advanced**. For more information, see Advanced setup on page 102.
4. If you wish to test incoming calls, select **Loop Test (enables incoming calls)**.

   For incoming calls, the program accepts the call if it is pending and the address is valid or not specified. Then it activates the detection of DTMF tones and an announcement is played. If a DTMF tone is detected, the announcement is stopped and the same DTMF tone is replied with delay to the calling party. The test is passed successfully only if the DTMF tone is received at the calling party.

5. To trace outgoing or incoming calls for analysis, you can create a trace file.
6. After you have entered all necessary information, you can start the test by clicking **Call**.
7. If a issues occurs that you cannot resolve, you may obtain technical support. For more information, see Customer Service on page 107.
8. To abort the test manually, click **Disconnect** during the test.
9. To delete all messages from the status box, click **Clear**.
10. All performed tests are saved in a text file. To view this file, click **History**.

    Note: The information in the history file is overwritten every time you open the Diva Line Test tool. If you want to keep the information of a specific call, save the file in a different folder.

Advanced setup

Normally it is not necessary to change the predefined advanced settings. Change the values only for test purposes. The following settings are available in the **Advanced Setup** dialog box:

1. If you wish to test incoming calls on a specific number, enter it in **Dialed for**. Incoming calls for another number are not accepted.
2. Under **Type of Number** you can choose from the following:
    - **Unknown (Default)**: Use this type of number if you do not know how your PBX is configured or when the PBX has no knowledge of the type of number, e.g., international or national. You need to enter all necessary prefixes.
    - **International**: Select **International** if the PBX understands the number as international. Instead of entering 0049 as international prefix and country code for Germany for example, you need to enter only 49.
    - **National**: Select this type of number if the PBX understands it as national. Instead of entering 0711 as city code for example, you need to enter only 711.
    - **Network**: Select this type of number if you use a network specific coding.
    - **Subscriber**: Select **Subscriber** if your PBX understands that it is a number without country code and city code.
    - **Abbreviated**: Select if you use a quick dialing number to test the line.
3. Under **Number Plan ID** you can choose from the following:
    - **Unknown (Default)**: Change in specific cases and for test purposes only.
    - **ISDN Telephony**: Some countries require an ISDN number plan.
    - **Data**: Select this option if the call you want to test is a specific number plan for a data call.
    - **National**: Select this option if your provider is using a national numbering plan.
    - **Private**: Select this option if your provider or PBX is using a private numbering plan.
4. Under **Presentation Indicator** you can choose from the following:
    - **Allowed**: The calling party number is presented to the called party.
    - **Restricted**: The calling party number is not presented to the called party.
    - **Not Available**: You may want to use this option for test purposes.
5. Under **Screening Indicator** you can choose from the following:
    - **Not screened**: If you specify a calling party number and select this option, the number is not screened.
    - **Verified and passed**: If you specify a number, it is verified by the network. If the number is correct, it is passed to the application.

- **Verified and failed**: If you specify a number, it is verified by the network. If the number is wrong and you have a Point-to-Point configuration, the number is not passed to the application. If you have a Point-to-Multipoint configuration, the wrong number is ignored and the MSN is added.
- **Network-provided**: If you do not specify a number, the network generates one with the bit set to "Network provided".

Call Transfer

This test transfers a call to the configured destination number. In this tab you can select the transfer method, e.g., if a consultation call is used or if the call is put on hold before it is transferred.

Note: To test a call transfer, the supplementary services Call Deflection and Explicit Call Transfer need to be supported.

1. Under **Device** select the line of the Dialogic® Diva® Media Board to test.
2. Enter the number to which the call should be transferred in **Called Party Number** under **Transfer Destination**.
3. To configure the transfer type, consultation call, and completion mode, click **Advanced** under **Transfer Type**. For more information, see Advanced transfer setup on page 103.
4. Click **Start**.
5. If you are prompted in the **Status** box, dial in from another application or phone.
6. The displayed message in the **Status** box notifies you if the test was successful or not. If the test is not terminated automatically, click **Stop**.
7. During the transfer test you can also create a trace file.
8. If you cannot set up a call transfer and you cannot locate the cause of the issue, you may obtain technical support. For more information, see Customer Service on page 107.
9. All performed tests are saved in a text file. To view this file, click **History**.

   Note: The information in the history file is overwritten every time you open the Diva Line Test tool. Thus, if you want to keep the information of a specific call, save the file in a different folder.

Advanced transfer setup

Note: The settings are only valid while the Dialogic® Diva® Line Test tool is opened.

Select under **Transfer Type** how the call should be transferred:

- the call should be transferred without consultation call, or
- the call should be answered and the incoming call should be placed on hold, or
- the call should be answered but the incoming call is not placed on hold.

If the incoming call is put on hold, you can choose under **Consultation Call** whether to transfer the consultation call on the same or on a different line.

Moreover, if you transfer the call with a consultation call, you can decide under **Complete Transfer Type** whether the call is completed in ringing state or after the call is connected.

Fax test

With the fax test you can send or receive a fax document that you can view with a fax viewer.

1. Under **Device** select the line of the Dialogic® Diva® Media Board to test.
2. Enter the **Called Party Number** and **Calling Party Number**.
3. To configure advanced settings, click **Advanced**.
4. If you wish to test incoming fax calls from another device, select **Receive incoming fax**. With this test you can verify if another Diva Media Board, a controller of a Diva multiport Media Board, or the PBX is working correctly.

   If you do not have another application to test incoming fax calls, you can open the Dialogic® Diva® Line Test tool twice and configure one utility as the calling party and the other utility as the called party. For more information, see To set up a test for incoming fax calls below.
5. To trace outgoing or incoming calls for analysis, you can create a trace file. After you traced an incoming or outgoing fax, the option **Play Audio** is available. Click this button to save a wave file of the transmitted or received fax. This might be necessary for analysis.

   Note: You can only save the wave file of the trace as long as the Diva Line Test Tool is opened. Once you close the tool, the unsaved wave file will be deleted.
6. After you have entered all necessary information you can start the test. To do so, click **Call**.
7. If an issue occurs that you cannot resolve, you may obtain technical support. For more information, see Customer Service on page 107.
8. To abort the test manually, click **Disconnect** during the test.
9. To delete all messages from the status box, click **Clear**.
10. All performed tests are saved in a text file. To view this file, click **History**.

    Note: The information in the history file is overwritten every time you open the Diva Line Test tool. If you want to keep the information of a specific call, save the file in a different folder.

To set up a test for incoming fax calls

1. Open two Dialogic® Diva® Line Test tools and click the **Fax** tab.
2. Select the devices to test under **Device**.
3. At one utility enter the **Called Party Number** and **Calling Party Number** under **Call Settings**.
4. At the other utility do not enter any number under **Call Settings** and select **Receive incoming fax** for the fax test.
5. You can also configure advanced settings for both Diva Line Test tools.
6. Click **Call** on the sending Diva Line Test tool to start the test.
7. To see if the fax was transmitted correctly, click **View Fax**.

   Note: If the tool receives a fax more than twice without saving, the fax result file will be overwritten.
8. During the test you can also create a trace.
9. If an issue occurs that you cannot resolve, you may obtain technical support.

To write a message into a trace file

1. Click **Start Log**.
2. Click **Call** or **Start** to start the test call.
3. If the call is finished, click **Stop Log**.
4. Click **View Log** to open the log file in a separate editor. You might want to save the file for analysis.
5. After you traced an incoming or outgoing fax, the option **Play Audio** is available. Click this button to save a wave file of the transmitted or received fax. This might be necessary for analysis.

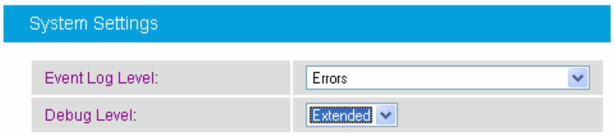
   Note: You can only save the wave file of the trace as long as the Dialogic® Diva® Line Test tool is open. Once you close the tool, the unsaved wave file will be deleted.

Create A Trace With The Dialogic® Diva® Diagnostics Tool

To create a trace for the Dialogic® Media Gateway Series, you need to set the correct debug level in the Dialogic® Diva® SIPcontrol™ software web interface first. Then you can create a trace in the Dialogic® Diva® Diagnostics tool.

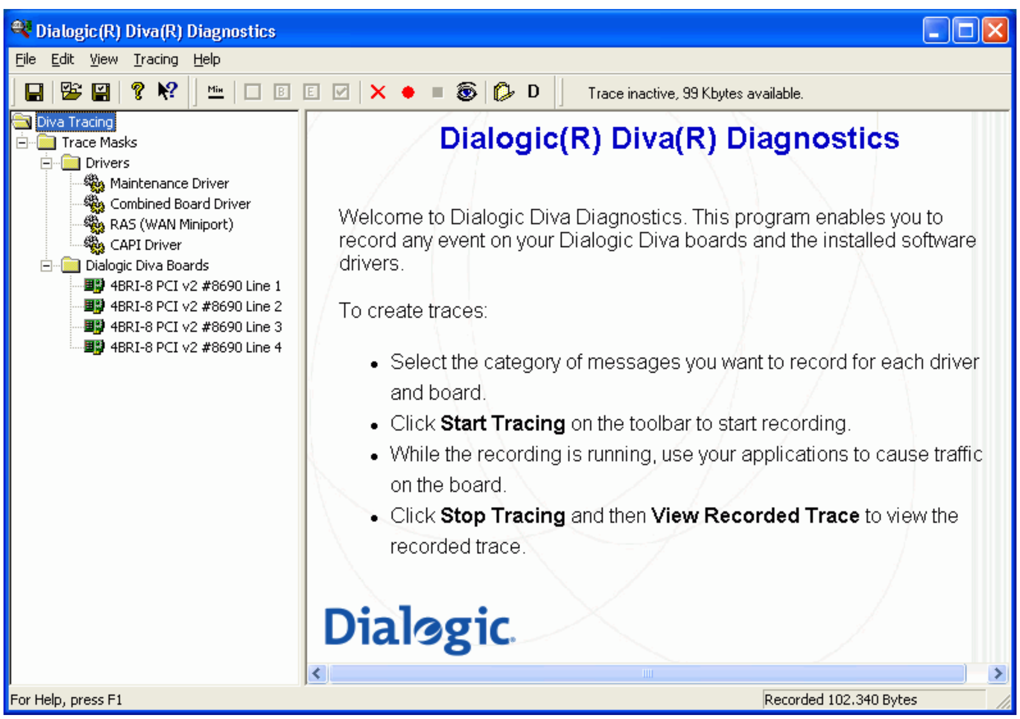
To set the correct debug level in the Dialogic® Diva® SIPcontrol™ software web interface:

1. Click **Start > Programs** > **Dialogic Diva** > **SIPcontrol Configuration** to open the Diva SIPcontrol software web interface.
2. Click **SIPcontrol** on the left hand side to open the configuration interface.
3. In the configuration interface, click **System Settings** and set the **Debug level** to **Extended**.

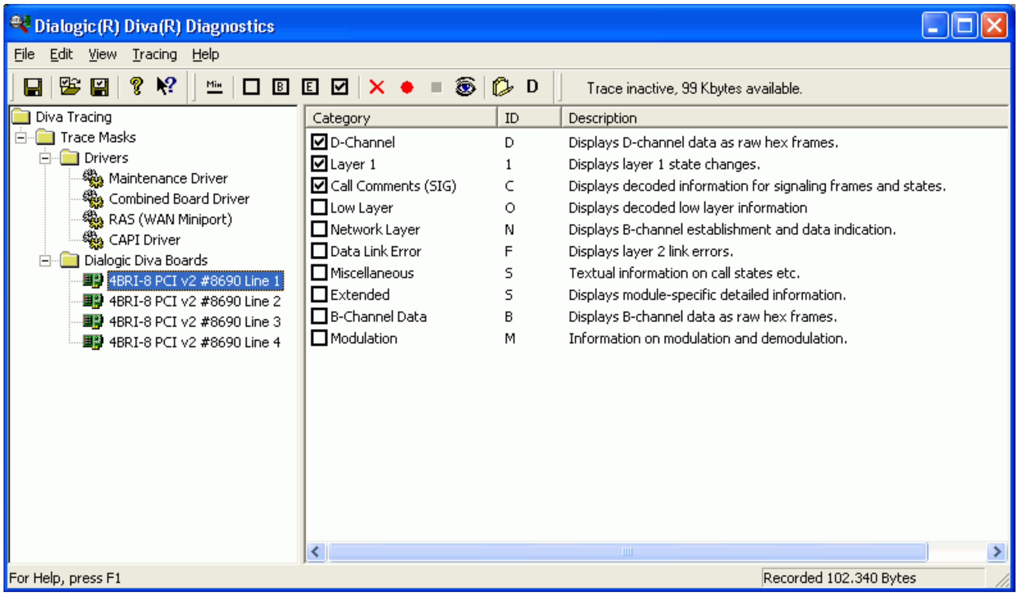


To create a trace with the Diva Diagnostics tool:

1. Click **Start** > **Programs** > **Dialogic Diva** > **Diagnostics**, to open the Diva Diagnostics tool. In the left pane you will see the installed software drivers and the controllers of the installed Dialogic® Diva® Media Boards.



2. Click the controller of the Dialogic® Diva® Media Board for which you want to create a trace. In the right pane, the different trace categories and their descriptions are displayed. Leave the trace settings at their default values.



3. Click **Tracing** > **Start Tracing** or click the start trace button in the toolbar, to start the trace.
4. Use your applications to cause traffic on the board.
5. Click **Tracing** > **Stop Tracing** or click the stop trace button in the toolbar, to stop the trace.
6. Click **View** > **View Recorded Trace** or click the view trace button in the toolbar, to view the trace.
7. You can also save the recorded trace. To do so, click **File** > **Save Recorded Trace** or click the save trace button .
8. In the displayed dialog box, select the folder where you want to save the trace file or create a new folder if required.
9. Enter the file name and click **Save**.

For detailed information about the Diva Diagnostics tool, see the Dialogic® Diva® Diagnostics Online Help under **Help** > **Help Topics**.

Customer Service

Dialogic provides various options and arrangements for obtaining technical support for your Dialogic® product. We recommend that you use the Dialogic® Diva® Support Tools first before contacting your Dialogic supplier. Also we suggest that you visit the Dialogic Services and Support web site, which includes detailed information about a variety of topics. In the unusual case that neither your supplier nor the information on the web site adequately address your support issue, you can contact Dialogic Customer Support.

For more information see:

- Dialogic® Diva® Support Tools
- Dialogic Services and Support web site
- Dialogic Customer Support

Dialogic® Diva® Support Tools

If an issue occurs during the operation of your Dialogic® Diva® product, use the following Dialogic® Diva® Support Tools:

- Dialogic® Diva® Line Test: With the Diva Line Test tool, you can test your hardware and perform simple phone test calls, call transfers, or basic inbound and outbound calls. For more information, see Verify The Line Configuration With The Dialogic® Diva® Line Test Tool on page 100.
- Dialogic® Diva® Diagnostics: With the Diva Diagnostics tool, you can write traces for each Dialogic® Diva® Media Board or driver into a file. For more information, see Create A Trace With The Dialogic® Diva® Diagnostics Tool on page 105.
- Dialogic® Diva® Management tool: With the Diva Management tool, you can view the current status of the connected lines, the active connections, and the history of the connections.

If you cannot solve an issue through use of these tools, contact your Dialogic supplier.

Dialogic Services and Support web site

If your supplier is unable to help you to address your issue, visit the Dialogic Services and Support web site. It contains:

- detailed information about the Dialogic® Pro™ Services (1 or 5 year 24/7 service contracts) at http://www.dialogic.com/support/DialogicPro/
- a help web section for Dialogic® products at http://www.dialogic.com/support/helpweb
- a download section, to install the current version of your software at http://www.dialogic.com/support/software.aspx
- a training section, with information about webinars as well as online and onsite trainings at http://www.dialogic.com/training
- a manuals section, that includes currently available documentation, at http://www.dialogic.com/manuals
- technical discussion forums about different developer-specific Q&A at http://www.dialogic.com/den/groups/developers/default.aspx
- the Dialogic Customer Support web site. For detailed information about how to contact Dialogic Customer Support, see Dialogic Customer Support below.

Dialogic Customer Support

If the information on the Dialogic Services and Support web site is not sufficient to help you solve your problem, contact the Dialogic Customer Support personnel. Please note that when you contact the Customer Support personnel, they may need you to provide one or more of the following:

- A debug trace (see Create A Trace With The Dialogic® Diva® Diagnostics Tool on page 105 for more information),
- A copy of your active Dialogic® Diva® Media Board configuration (see the Dialogic® Diva® Configuration Manager Online Help file - DSMain.chm), and

- A copy of your Dialogic® Diva® SIPcontrol™ configuration. To save a copy, click **Show Configuration** at the bottom of the configuration web interface, copy & paste the contents into a separate file, and save as text file.

See www.dialogic.com/support/contact for details on how to contact Dialogic.