



Dialogic[®] 4000 Media Gateway Series SU4.1

Reference Guide

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1. Introduction

About This Manual

The Dialogic® 4000 Media Gateway Series are pre-installed with the Windows versions of the Dialogic® Diva® System Release Software and Dialogic® Diva® SIPcontrol™. 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 "DMG4000 Gateways" is used herein to refer collectively to the Dialogic® 4000 Media Gateway Series and the term "DMG4000 Gateway" is used herein to refer a gateway in the Dialogic® 4000 Media Gateway Series.

For a list of Diva SIPcontrol features, see [Dialogic® Diva® SIPcontrol Features](#).

For a list of Diva Media Board features, see [Dialogic® Diva® Media Board Features](#).

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2. Dialogic® 4000 Media Gateway Features

Dialogic® Diva® SIPcontrol Features

This topic groups Diva SIPcontrol features into the following categories:

- [General features](#)
- [Call handling features](#)
- [Media processing features](#)
- [Supported RFCs](#)
- [Enhanced routing](#)
- [Enhanced address manipulation](#)

General Features

- Support for Microsoft® Lync™ Server 2010
- Noise suppression support
- Echo cancellation selectable via GUI
- Forward the display name from SIP to Q.SIG and vice versa
- Interoperability with Dialogic® Host Media Processing (HMP) software 3.0WIN and 3.1LIN
- Configuration via the Diva SIPcontrol web interface
- Standard web browsers can be used for configuring. Diva SIPcontrol has been tested with the following browsers:
- Microsoft® Internet Explorer® Version 7 and 8
- Mozilla Firefox version 3.6.x
- Remote configuration of Diva SIPcontrol from any computer in the network. The configuration may be encrypted.
- Cause codes: Configurable translation of ISDN cause code to SIP response code and vice versa; consequently, Diva SIPcontrol can adapt to the specific behavior of the PSTN, PBX, and/or SIP peer.
- Configuration changes during runtime: Modify most parameters of Diva SIPcontrol without the need to restart the service; active calls are not affected by configuration updates and continue undisturbed.
- Support for the North American numbering plan: The configuration of multiple area codes is handled as local. Therefore, the Diva SIPcontrol 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 Diva SIPcontrol. 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 Features](#) for supported codecs.

- Support for Proxy and Registrar authentication.
- Support for registering Diva SIPcontrol as an e-phone gateway.
- Support for early media. Early media is supported to and from the PSTN due to Any-to-Any routing, if the line protocol supports it. A call using early media does not need to have a SIP leg.
- Configuration of Diva Media Board parameters via the Diva web interface
- 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 Features

- SIP methods: ACK, BYE, INVITE, NOTIFY, REFER, CANCEL, OPTIONS, PRACK
- Configurable IP transport layer TCP, UDP, or TLS
- Support for TLS encryption and authentication
- Support for SRTP (secure Real-time Transport Protocol)
- Support for SIPS (Secure SIP)
- 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)
- SIP-side Call Transfer as transfer target (C-party) and as transferee (A-party)
- PSTN-side incoming Call Diversion
- Message Waiting Activation / Deactivation*
- Support for SIP 302 REDIRECT, which works as follows: a SIP call is redirected per 302 Redirect with new IP port numbers (as used with Microsoft® Office Communications Server 2007, Microsoft® Office Communications Server 2007 R2, and Microsoft® Exchange Server 2007 without changing the IP address)
- Support for SIP Refer: a SIP call is redirected per SIP Refer to a new SIP target, using Replaces and Referred By
- SIP Session Timer (RFC 4028)
- Simplified Number Normalization based on PSTN connection parameters
- Number Manipulation using Regular Expressions

* NOTIFY in combination with SUBSCRIBE is 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 Diva SIPcontrol to use SUBSCRIBE.

Media Processing Features

- 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**
 - sRTP
- RTP dynamic payload audio/telephony event
- RTP profile RTP/AVP
- DTMF via RTP payload/telephony event (RFC 2833 or RFC 4733)
- PSTN-side fax tone detection via RTP event (RFC 2833 or RFC 4733)
- 128 ms Echo Cancellor supported on all boards, and additionally, 256 ms EC supported on those boards listed in MultiPRI boards section.
- 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

*For G.729, you need to purchase and activate a license before you can use it. For more information, see the *Dialogic® 4000 Media Gateway Series Quickstart Guide*, which is available at <http://www.dialogic.com/manuals/dmg30004000>. G.729 is only available on Dialogic® Diva® Multiport V-PRI Media Boards.

**iLBC is only available on Diva Multiport V-PRI Media Boards. On Dialogic® Diva® V-4PRI/E1/T1-120 PCIe HS boards and Dialogic® Diva® V-8PRI/E1/T1-240 PCIe FS boards, up to 18 channels for each PRI port are supported.

Supported RFCs

- RFC 2617 - HTTP Digest Authentication
- RFC 2833 - RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC 3261 - Session Initiation Protocol
- RFC 3262 - Reliability of Provisional Responses in Session Initiation Protocol (SIP)
- RFC 3264 - An Offer/Answer Model with Session Description Protocol
- RFC 3265 - SIP-specific Event Notification
- RFC 3326 - The Reason Header Field for the Session Initiation Protocol (SIP)
- RFC 3389 - RTP Payload for Comfort Noise
- RFC 3398 - ISDN to SIP mapping
- RFC 3420 - Internet Media Type message/sipfrag
- RFC 3515 - REFER method
- RFC 3550 - Realtime Transport Protocol (RTP)

- RFC 3551 - RTP/AVP profile
- RFC 3711 - The Secure Real-time Transport Protocol (SRTP)
- RFC 3842 - Message Waiting Indication for SIP
- RFC 3891 - SIP "Replaces" header
- RFC 3892 - SIP Referred - By Mechanism
- RFC 3951 - Internet Low Bit Rate Codec (iLBC)
- RFC 3952 - Real-time Transport Protocol (RTP) Payload Format for internet Low Bit Rate Codec (iLBC) Speech
- RFC 3960 - Early Media and Ringing Tone Generation in the Session Initiation Protocol (SIP), only gateway model
- RFC 4028 - Session Timers in SIP
- RFC 4497 - Interworking between SIP and QSIG
- RFC 4566 - Session Description Protocol (SDP)
- RFC 4568 - SDP Security for Media Streams
- RFC 4733 - RTP Payload for DTMF Digits, Telephony Tones, and Telephony Signals
- Draft: Diversion Indication in SIP (draft-levy-sip-diversion-08)

Enhanced Routing

- Support for Any-to-Any routing (SIP to PSTN, PSTN to SIP, SIP to SIP, PSTN to PSTN)
- 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
- Routing based on Active Directory information

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 topic groups Diva Media Board features into the following categories:

- [General features](#)
- [DSP-based features](#)
- [Fax and modem features](#)
- [Q.SIG features](#)
- [Channelized T1 features](#)

Note: This list of Diva Media Board features includes only features relevant for the DMG4000 Gateways. If you are interested in the complete list of features, see the *Dialogic® Diva® System Release Reference Guide*, which is 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 \(robbed bit signaling\)](#))
- 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 the Q.SIG protocol (see [Q.SIG features](#))
- 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 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 Multipoint 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
- Tests have been conducted for the various switch types

For more information, see "Supplementary services" in the *Dialogic® Diva® System Release Reference Guide*, which is 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

DMG 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/Solutions/Unified-Communications/microsoft-pbx-interop-and-config-guides.aspx> provide guidance and configuration information for many different PBX vendors and models.

3. Initial Configuration and License Activation

Preparing for Configuration

The initial configuration allows the unit to be configured using a remote desktop connection.

The DMG4000 Gateway is initially configured with:

- Ethernet port 1: with fixed IP address 192.168.1.1
- Ethernet port 2: with fixed IP address 192.168.2.1 on the older DMG4000 model; enabled with DHCP on the newer DMG4000 model
- Ethernet port 3 and 4: enabled with DHCP, only available on older DMG4000 models

Note: The above numbering of the Ethernet ports refers to the numbering at the rear panel of the DMG4000 Gateway and not to the numbering displayed in the Control Panel of your operating system.

- User name: Administrator
- Password: Dialogic (Change the password after the initial log on.)

Before you start to configure your DMG4000 Gateway, it is necessary to gather some information about the environment in which it will be used.

Collect the following information:

1. In case the gateway is to be connected to a PBX: Which vendor and which PBX type will be used?
2. What protocol will be used on the PSTN/PBX side of the gateway? Examples are EuroISDN (ETSI-DSS1), Q-SIG, and DMS100.

License Activation

Diva SIPcontrol includes a license for two channels that can be used for testing and evaluating Diva SIPcontrol.

You must activate a license if you need more than the two channels with Diva SIPcontrol, or if you want to use G.729 speech compression or fax offered with the installed Diva Media Board. During the activation process of the license, you need to choose a Diva Media Board to which the license should be bound. After having activated the license for this Diva board, the license cannot be transferred to be used with another Diva board.

Notes:

- Diva SIPcontrol licenses need to be activated via the Diva SIPcontrol web interface, as described under [To Activate a License File](#).
- Licenses for G.729 and fax can be uploaded and activated via the Diva SIPcontrol web interface and Diva Configuration Manager. See the Dialogic® Diva® Configuration Manager Online Help for more information.
- Fax is needed to enable T.38 FoIP support in Diva SIPcontrol. It needs to be licensed only for Dialogic® Diva® PRI Media Boards with multiple ports.
- The Dialogic® Host Media Processing (HMP) Software licenses for SIP channels are also valid for SIPcontrol, but they require the Dialogic HMP software to be installed on the same system as Diva SIPcontrol.

After purchasing the license, you will need to generate and activate it to unlock functionality in the product.

To activate your license key, you need the following information:

- [Device Unique ID \(DUID\)](#)
- [Proof of Purchase Code \(PPC\)](#)

Once you have both, the DUID and PPC, visit the Dialogic® Diva® Activation site to register your PPC together with the DUID, and you will receive your license file. Activate this license file in the Diva SIPcontrol web interface. For more information, see [To Activate a License File](#).

Device Unique ID (DUID)

The DUID binds the installed Diva SIPcontrol software to your computer (PC fingerprint).

To obtain the DUID:

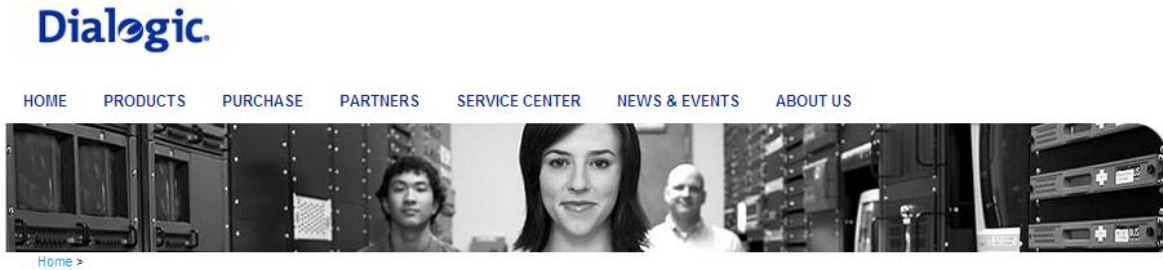
1. Click **Start > Programs > Dialogic Diva > SIPcontrol Configuration** to open the Diva SIPcontrol web interface.
2. Click **License Management** on the left side of the Diva SIPcontrol web interface to open the **License Status** dialog.
3. In the **License Status** dialog, copy the DUID number of the Diva Media Board you want to activate to the clipboard.
4. If you need to do web activation using another computer, open an editor, paste the DUID, and save the file.

Proof of Purchase Code (PPC)

When you purchase a Diva SIPcontrol license, you will receive a PPC either in printed form or via email. By registering this PPC, you represent and warrant that you lawfully purchased the license.

To Register Your PPC and DUID

1. Open the following web site: <http://www.dialogic.com/activate>.
2. Enter your PPC and click **Check**:



Dialogic Activation

PPC

Enter the PPC which you received after placing an order, either in a printed certificate, or by email.

The PPC is a string of letters and digits similar to this: DSIP10000101A160F886F6E4D0D9C8

Build on Dialogic

3. If your PPC is valid, the following web site will open:

Dialogic Activation

PPC

DSSC150000411197BDF6B1BDA6A3C2 Items

Qty	Code	Name
1	DM5-060	SIPcontrol, 30-day demo, 30-channels

DUID

The DUID is a long string of digits and numbers that you can see

- in Tools -> License Manager in the Dialogic Configuration Manager under Windows
- in License Management in the web configuration interface under Linux

Depending on your hardware and software, you may see multiple DUIDs listed.
Enter a DUID starting with the letter N

Email Address

The email address that you enter here will be used for delivery of your license. If you choose the "Activate and Down load" method the license will also be delivered by email for backup purposes.

Comment

You can enter a comment here which will appear in the licence file.

By pressing the "Activate and Email" button the license file will be delivered to you by email. By pressing the "Activate and Download" button you can download the license file to your computer.

4. Paste your Device Unique ID (DUID) that you saved earlier, and enter your email address to which the license file should be sent.
5. Click **Activate and Email** to generate the license file that will be sent to the email address you have entered.
6. Save the license file and activate it. For more information, see [To Activate a License File](#).

To Activate a License File

Note: The date set in the system settings of your computer must be correct. Otherwise, you cannot add your license file.

Follow these steps to activate a license file:

1. Click **License Management** on the left side of the Diva SIPcontrol web interface to open the **License Status** dialog:

The image shows two screenshots of the Diva SIPcontrol web interface. The top screenshot is titled "Diva SIPcontrol" and displays the "License Status" dialog. It shows the board information: "Board[1]: Serial Nr: 1033 DUID: N4FYVJPFBXHTSQWFJYFVG774AQ". Below this, it lists "Installed SIPcontrol Features:" and "Channels: 48". At the bottom, there is a "License files:" label, a text input field, a "Browse..." button, and an "Upload" button. The bottom screenshot is titled "Boards" and shows the "Boards" section. It displays the board information: "Board[1]: Serial Nr: 1033 UID: H04F9EFF20E00000000000003247". Below this, there is a "License files:" label, a text input field, a "Browse..." button, and an "Upload" button.

2. Upload the license file:
 - To upload a SIPcontrol license, click **Browse** in the Diva SIPcontrol section, locate the directory in which you saved the SIPcontrol license file, and click **Upload**.
 - To upload a fax or codec license, click **Browse** in the Boards section, locate the directory in which you saved the fax or codec license, and click **Upload**.

4. Dialogic® Diva® Media Board Configuration

Dialogic® Diva® Media Board Configuration

Since Diva SIPcontrol version 1.8, Diva Media Boards can be configured via the Diva web interface. The configuration via the Diva web interface can be accessed and updated remotely. The classic configuration via the Diva Configuration Manager is also available, but it can only be accessed from the computer on which the Diva System Release software is installed. Any changes will be reflected in both configuration tools, meaning that if you change a parameter in the Diva web interface, the change is automatically done in the Diva Configuration Manager as well (and vice versa). The update of the configuration between both tools will only take effect after you have saved the configuration, and, in case of the Diva Configuration Manager, after you activated the configuration. For more information about activating the configuration, see the Diva Configuration Manager Online Help.

You can find information about the Diva web interface in [Dialogic® Diva® SIPcontrol™ Configuration](#) and in [Configuration Tips and Hints](#).

Dialogic® Diva® Media Board Configuration via the Dialogic® Diva® Web Interface

To configure Diva media boards via the Diva web interface, follow these steps:

1. Click **Start > Programs > Dialogic Diva > SIPcontrol Configuration**.
2. In the Diva web interface, click **Board configuration** on the left hand side. A page displaying all installed Diva Media Boards opens:



3. To configure Diva Media Board parameters, either click the board name or click the down arrow and select **Configuration**. A page displaying the basic parameters will open:

Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 1, SN: 1398

Parameter	Value
D-Channel Protocol:	Europe/other countries, Euro-ISDN (ETSI-DSS1) - (ETSI)
Interface Mode/Resource Board:	TE - mode
Direct Dial In (NT2):	Yes
DDI Number Length:	3
DDI Collect Timeout:	0
DDI Special Number:	
Layer 1 Framing:	National default (default)
Layer 2 Connect Mode:	Permanent
Voice Companding:	Protocol default
View Extended Configuration	No

Save Cancel

Depending on the selected D-channel protocol, an additional menu opens in which you can configure protocol-specific parameters.

- You can also configure extended parameters that depend on the D-channel protocol you selected. To configure those parameters, select **Yes** under **View Extended Configuration**. An additional menu will open:

Extended Parameter	Value
TEI Value:	0 (standard) ▾
Source Of Local Tones (BUSY, ALERTING, ...):	Tones provided by ISDN equipment (default) ▾
Trunk Operation Mode:	Standard (default) ▾
Fraction Starts At Channel:	1 ▾
ETSI Call Transfer:	Default ▾
Deflection Mode:	Deflection (default) ▾
ETSI Message Waiting:	Default ▾
Extended Voice Processing	
DTMF Clamping:	Off ▾
Audio Recording Automatic Gain Control (AGC):	Off ▾
Echo Canceller Tail Length:	128 ms (default) ▾
Suppression of ambient noise:	Off ▾
Part 68 Voice Signal Limiter:	Protocol default ▾
Redirecting Number Emulation:	Disabled (default) ▾

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

The Board Monitor

If you click **Board monitor** on the left hand side, a page opens that allows you to control the current status and configuration of the installed Diva Media Boards, read internal board trace buffers (XLOG), and gain access to the management interface of Diva Media Boards and drivers:

Configuration

Board configuration

SIPcontrol configuration

System

System control

Status

Board monitor ▶

View report

Debugging














Licensing

License management

Available Diva Boards

Number	Board Name	SN	Line	Mgmt
1	Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 1	1398		
2	Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 2	1398		

If you click the icon below the **Mgmt** column in the **Available Diva Boards** section, the management interface browser opens:

Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 1, SN:1398			
Type	Name	Value	Operation
 DIR	-		Refresh
 UINT	CardType	<input type="text" value="79"/>	
 HINT	MIF Version	<input type="text" value="0x00000117"/>	
 ASCIIZ	Build	<input type="text" value="TE_DMLT_Build 110-6. f"/>	
 UINT	Events Down	<input type="text" value="0"/>	
 DIR	Info		Read
 DIR	Config		Read
 DIR	Statistics		Read
 DIR	State		Read
 DIR	StateT		Read
 DIR	Trace		Read
 DIR	Test		Read
 DIR	Debug		Read

The management interface browser allows you to navigate through the management interface directories, and to read, write, and execute management interface variables using the buttons in the **Operation** column.

The View Report Option

If you click **View report**, the Actual Board Statistics page opens:

Configuration

Board configuration

SIP control configuration

SBA configuration

System

System control

Status

Board monitor

View report

Licensing

License management

Actual Board Statistics

No	Board Name	SN	Layer 1 Layer 2	Successful Calls (IN/OUT)	Abandoned Calls (IN/OUT)	Failed Calls (IN/OUT)	Details
All Boards Summary				41 (22/19)	10 (0/10)	13923 (6910/7013)	
1	<div>Analog-8 PCI v1</div> <div> <div>Line 1</div> <div>Line 2</div> <div>Line 3</div> <div>Line 4</div> <div>Line 5</div> <div>Line 6</div> <div>Line 7</div> <div>Line 8</div> </div>	1004	<div>Down</div> <div>Down</div> <div>Down</div> <div>Down</div> <div>Down</div> <div>Down</div> <div>Idle</div> <div>Idle</div>	0 (0/0)	0 (0/0)	0 (0/0)	
2	<div>V-4PRI/E1/T1-120 PCI v1 - PORT 1</div> <div></div>	1001	<div>Activated</div> <div>UP</div>	19 (1/18)	6 (0/6)	6981 (6898/83)	
3	<div>V-4PRI/E1/T1-120 PCI v1 - PORT 2</div> <div></div>	1001	<div>Activated</div> <div>UP</div>	22 (21/1)	1 (0/1)	6913 (12/6901)	
4	<div>V-4PRI/E1/T1-120 PCI v1 - PORT 3</div> <div></div>	1001	<div>Activated</div>	0 (0/0)	0 (0/0)	0 (0/0)	
5	<div>V-4PRI/E1/T1-120 PCI v1 - PORT 4</div> <div></div>	1001	<div>Activated</div>	0 (0/0)	0 (0/0)	0 (0/0)	
6	<div>BRI-2 PCIe v2</div> <div></div>	6683	<div>Down</div> <div>Down</div>	0 (0/0)	0 (0/0)	0 (0/0)	
7	<div>BRI-2 PCIe v2</div> <div></div>	1559	<div>Down</div> <div>Down</div>	0 (0/0)	0 (0/0)	28 (0/28)	
65	<div>softIP</div>		<div>Activated</div> <div>UP</div>	0 (0/0)	3 (0/3)	1 (0/1)	

Stop Auto Update

The Actual Board Statistics page displays the status of all Diva Media Boards and the cumulative statistics for the active Diva Media Boards, including the number of failed, successful, and abandoned calls. An abandoned call is a call that the caller ends before the call was connected. For example, an abandoned call occurs when a call goes into the ringing state, and the caller hangs up the phone before the called party replies with a "busy," "rejected," or "connected" signal.

A colored symbol in the Board Name column illustrates the line plug status:

- Red: No signal; system is inactive.
- Yellow: Remote synchronization error.
- Green: System is active and functioning normally.

Click the plug status symbol to see additional information about the Layer 1 alarm state. For example:

- Green symbol on Analog board: "Cable detected"
- Green symbol on ISDN board: "Layer 2 connected"
- Red symbol on Analog board: "No cable detected"
- Red symbol on ISDN board: "Layer 1 error"

If there is a Remote Yellow, Blue, or Red alarm condition, clicking the plug status symbol shows a typical E1/T1 alarm message for the alarm. If two alarm conditions occur at the same time, only the most critical alarm is displayed. For example, if a Red alarm condition occurs at the same time as a Yellow alarm condition, only the Red alarm is displayed. If Layer 2 is up, then Layer 1 is also up.

Displaying Cumulative Call and Fax Statistics

To display cumulative call and fax statistics for all Diva Media Boards, click the magnifying glass at the end of the All Boards Summary row on the Actual Accumulated Board Statistics page. The Actual Accumulated Board Statistics page opens.

Actual Accumulated Board Statistics	
Call Statistics Total (In / Out)	
Successful Calls	41 (22 / 19)
Abandoned Calls	10 (0 / 10)
Failed Calls	13923 (6910 / 7013)
User Busy	0 (0 / 0)
No Answer	0 (0 / 0)
Wrong Number	0 (0 / 0)
Out Of Order	0 (0 / 0)
Incompatible Dst	12 (12 / 0)
Call Reject	6898 (6898 / 0)
Ignored	0 (0 / 0)
Other	7013 (0 / 7013)
FAX Statistics Total (In / Out)	
Successful FAX	8
Failed FAX	0
Total Pages	8 (4 / 4)
Stop Auto Update	

The displayed information includes the total number of successful, abandoned, and failed calls, a summary of the most frequent call disconnect causes, the total number of successful and failed fax calls, and the total number of fax pages received and sent. (In the parentheses of the Total Pages field, the number of received faxes is on the left and the number of sent faxes is on the right.) The information contained in the report originates from the management interface of the Diva Media Boards.

Displaying Board-Specific Details

To display details for a specific board, click the magnifying glass at the end of the associated row on the Actual Accumulated Board Statistics page. The displayed information includes the port status, channel usage, board temperature, and call and fax statistics. The information contained in the report originates from the management interface of the Diva Media Boards.

Details View for PRI or BRI Boards

The following screenshot shows the details view for a PRI board:

■ Board:2 V-4PRI/E1/T1-120 PCI v1 - PORT 1 SN: 1001 >>

Port Status		Board Temperature >>		
Identify		↓ Actual	37°C	98°F
Layer 1	Activated	↓ Maximum	41°C	105°F
Layer 2	UP	↓ Minimum	31°C	87°F
Actual Connections >>		Layer 1 Statistics >>		
Channel Usage		Framing Errors		
 1 31		CRC4 Error		
Actual Numbers		Frame Slips		
1 (0 / 1)		undefined		
Call Statistics Total (In / Out) >>		D-Channel Statistics >>		
Successful Calls	19 (1 / 18)	X-Errors		
Abandoned Calls	6 (0 / 6)	R-Errors		
Failed Calls	6981 (6898 / 83)	0		
User Busy	0 (0 / 0)	0		
No Answer	0 (- / 0)	FAX Statistics Total (In / Out) >>		
Wrong Number	0 (0 / 0)	Successful Faxes		
Out Of Order	0 (0 / -)	Failed Faxes		
Incompatible Dst	0 (0 / -)	Total Pages		
Call Reject	6898 (6898 / 0)	4 (0 / 4)		
Ignored	0 (0 / -)	Modem >>		
Other Causes	83 (0 / 83)	B-Channel Statistics >>		

Stop Auto Update

In the details view for PRI or BRI boards:

- The Channel Usage display is correlated to the number of available channels. In this example, because the PRI Board has 31 channels, the Channel Usage field shows 31 lamps — one for each channel.
- The bracket symbol in the Port Status area shows the number of ports for the BRI or PRI board. The highlighted port is associated with the Layer 1 and Layer 2 statistics displayed below the bracket symbol.

Details View for Analog Boards

The following screenshot shows the details view for an analog board:

Board:1 Analog-8 PCI v1 SN: 1004 >>

Parameter	Line 1	Line 2	Line 3	Line 4	Line 5	Line 6	Line 7	Line 8	Summary
Identify									
Layer 1	No Cable Detected	No Cable Detected	No Cable Detected	No Cable Detected	No Cable Detected	No Cable Detected	Idle	Idle	
Board Temperature	Not Available								
Actual Connections >>									
State									
Direction	-	-	-	-	-	-	-	-	
Call Statistics Total (In / Out) >>									
Successful Calls	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)
Abandoned Calls	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)
Failed Calls	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)
User Busy	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)
No Answer	0 (- / 0)	0 (- / 0)	0 (- / 0)	0 (- / 0)	0 (- / 0)	0 (- / 0)	0 (- / 0)	0 (- / 0)	0 (0 / 0)
Wrong Number	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)
Out Of Order	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / 0)
Incompatible Dst	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / 0)
Call Reject	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)
Ignored	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / -)	0 (0 / 0)
Other	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)
Line Statistics									
X-Errors	0	0	0	0	0	0	0	0	0
R-Errors	0	0	0	0	0	0	0	0	0
FAX Statistics Total (In / Out) >>									
Successful FAX	0	0	0	0	0	0	0	0	0
Failed	0	0	0	0	0	0	0	0	0
Total Pages	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)	0 (0 / 0)

Supported Switch Types and Supported PBXs

Diva Media Boards currently support the following switch types:

Public Line ISDN Protocols

EMEA PRI and BRI

- 1TR6 (legacy Germany and old PBXs)
- ETSI Australia variant (On Ramp ETSI)
- ETSI (Europe, Africa)
- ETSI Hong Kong variant
- ETSI Serbia variant
- ETSI Taiwan variant
- ETSI New Zealand variant
- INS-Net 64 / 1500 (Japan)
- VN4 (legacy France, old PBXs)
- VN6 (current France)

Line Side E1

- Australian P2
- Ericsson
- Melcas
- NEC
- Nortel

R2 CAS (E1 only)

- Argentina
- Brazil
- China
- India
- Indonesia
- Korea
- Mexico
- Philippines
- Thailand
- Venezuela

USA PRI and BRI

- 5ESS Custom (AT&T)
- 5ESS Ni Avaya (Lucent)
- DMS 100 (Nortel)
- EWSD (Siemens)

USA T1/PRI

- 4ESS
- T1 RBS

Carrier Grade

ITU-T ISUP SS7

POTS

Worldwide POTS

PBX Protocols

- Generic QSIG T1 and E1

Note: The Generic QSIG switch type can be used for the majority of PBXs

- ETSI

Note: Many European PBXs use the regular ETSI protocol (PRI and BRI).

Specific Major PBX Types

- Alcatel 4200
- Alcatel 4400
- Alcatel 4410
- ASCOM Ascotel 2020
- ASCOM Ascotel 2030
- ASCOM Ascotel 2050
- ASCOM Ascotel 2060
- DeTeWe OpenCOM 1000
- Ericsson MD110/BP250
- GPT Realitis iSDX
- Lucent Definity
- Matracom 6500
- Nortel Meridian
- Nortel opt11 Rev23

Dialogic 4000 Media Gateway Series SU4.1 Reference Guide

- Siemens Hicom 150
- Siemens Hicom 300
- Siemens Hipath 3000
- Siemens Hipath 4000
- Tenovis QSig

For a list of PBXs that are currently supported and tested with gateways from the different Dialogic® Media Gateway Series, see <http://www.dialogic.com/Solutions/Unified-Communications/microsoft-pbx-interop-and-config-guides.aspx>.

5. Dialogic® Diva® SIPcontrol™ Configuration

Dialogic® Diva® SIPcontrol™ Configuration

This chapter describes how to configure Diva SIPcontrol. It provides configuration tips and hints, includes general information about each configuration, and gives an overview of the configurable Diva SIPcontrol parameters. The configuration of the Diva Media Boards is described in [Dialogic® Diva® Media Board Configuration](#).

About Dialogic® Diva® SIPcontrol™ Configuration

Diva SIPcontrol is configured via the Diva SIPcontrol web interface.

Opening the Dialogic® Diva® SIPcontrol™ Web Interface

To open the Diva SIPcontrol web interface, click **Start > Programs > Dialogic Diva > SIPcontrol Configuration**. By default, access to the web interface is only allowed from localhost (127.0.0.1), and the port number to which the server is listening is set to 10005.

If you need to access the configuration via remote access, you must set a password. To do so:

1. Click **System control** on the left side of the web interface.

The System control window appears:

The screenshot displays the Diva SIPcontrol web interface with four main sections:

- System status management**: Contains a label "SIPcontrol:" followed by "Stop" and "Restart" buttons.
- Webserver certificate management**: Contains two rows for "Server certificate file:" and "Server private key file:", each with a text input field and a "Browse..." button. Below these is an "Install selected files" button.
- Password**: Contains three rows for "Please enter current password:", "Please enter new password:", and "Please retype new password:", each with a text input field. Below these is a "Change Password" button.
- Last operation log**: A section header at the bottom of the visible area.

- In the **Password** section, enter the current and new passwords, and then retype the new password. A password must be seven digits or longer, and the use of non-alphanumeric characters for a password is discouraged.
 - Click **Change Password**.
2. If necessary, open the port in the local firewall settings. To do this under Windows® Vista, Windows® 7, Windows Server® 2008, or Windows 2008® R2, open the command prompt with elevated rights.

- Enter the following command on a 64-bit operating system:

```
netsh advfirewall firewall add rule name="Diva Webserver" dir=in  
action=allow program="%ProgramFiles% (x86) \  
Diva Server\DivaWebConfig.exe" protocol=tcp
```

- Enter the following command on a 32-bit operating system:

```
netsh advfirewall firewall add rule name="Diva Webserver" dir=in  
action=allow program="%ProgramFiles% \  
Diva Server\DivaWebConfig.exe" protocol=tcp
```

You can now access the Diva SIPcontrol web interface on any of the IP addresses of the computer where SIPcontrol is installed, and then you can configure the settings according to your needs.

Dialogic® Diva® SIPcontrol™ Configuration Sections

Diva SIPcontrol configuration is divided into the following sections:

- [PSTN Interfaces](#)
- [Network Interfaces](#)
- [SIP Peers](#)
- [Routing](#)
- [Security Profiles](#)
- [LDAP](#)
- [Dialplans](#)
- [Address Maps](#)
- [Cause Code Maps](#)
- [Codec Profiles](#)
- [Registrations](#)
- [Logging and Diagnostics](#)

Configuration Tips and Hints

This section contains useful information about SIPcontrol configuration:

- Changes to the configuration will only take effect after you click **Activate Configuration** at the bottom of each configuration page.
- The settings will be lost if you close the Diva SIPcontrol web interface without having saved the configuration at the bottom of each configuration page.

- A restart of Diva SIPcontrol is recommended if you change the IP address or the port on which SIPcontrol is listening. If you do not restart, Diva SIPcontrol 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 restart the Dialogic® Diva® WebConfig service, in the SIPcontrol web interface, click **System control** on the left hand side, and then click **Restart** in the **System status management** section.
- Diva SIPcontrol provides a secure configuration via the web interface (HTTPS). The default port for HTTPS is 10006. Diva SIPcontrol provides a default certificate, but for security reasons you can install your own webserver certificates. To install a webserver certificate and corresponding key file, upload and install these files in the **Webserver certificate management** section under **System Control**.
- To use TLS for SIP calls, you need to upload the certificates as described under Security Profiles and enable the TLS port as described under [Network Interfaces](#).
- To open the online help for a specific parameter, click the parameter, and a window with the help text will appear.

Configuring Dialogic® Diva® SIPcontrol™

At a minimum, a Diva SIPcontrol configuration must contain the following components:

- At least one enabled network interface
- At least one enabled SIP peer
- At least one route for PSTN to SIP calls and another route for SIP to PSTN calls

There are four ways to configure Diva SIPcontrol:

- [Use the Configuration Wizard](#).
- [Load an existing configuration profile](#)
- [Import an existing configuration file](#)
- [Configure Diva SIPcontrol manually](#)

Using the Configuration Wizard

The easiest way to configure Diva SIPcontrol is to use the Diva SIPcontrol configuration wizard. The wizard provides a step-by-step interface that helps you generate configurations for the following use cases:

- Empty configuration that resets existing Diva SIPcontrol web interface settings
- Simple configuration for a general purpose gateway
- DMG4000 hybrid gateway
- DMG4000 Survivable Branch Appliance

The configuration prompts for the minimum required parameters.

To use the configuration wizard, follow these steps:

1. In the **Overview** section of the Diva SIPcontrol Configuration page, click **Start Configuration Wizard**.
The configuration wizard asks whether you want the wizard to delete all unsaved configuration changes.
2. Click **OK**.
3. Follow the configuration wizard prompts.

Loading an Existing Configuration Profile

Diva SIPcontrol configurations can be saved on the server as a configuration profile. You can load an existing configuration profile to use the saved configuration settings.

To load a configuration profile, follow these steps:

1. From the web interface, click **SIPcontrol configuration**.
The SIPcontrol Configuration page appears.
2. In the **Overview** section of the SIP Configuration page, access the Config.-Profiles field, and select the configuration profile file you want to load:

Overview	
Configuration	<input type="button" value="Activate"/> <input type="button" value="Discard"/> <input type="button" value="Start Configuration Wizard"/>
Config.-Profiles	<div> <input type="text" value="SBA_Use_Case"/> <input type="button" value="Load into GUI..."/> <input type="button" value="Delete..."/> <input type="button" value="Export to file..."/> </div> <div> <input type="button" value="Save GUI settings..."/> <input type="text"/> <input type="button" value="Browse..."/> <input type="button" value="Import from file..."/> </div>
PSTN Interfaces	Controller1, Controller2
Network Interfaces	Intel(R) PRO1000 EB Network (TCP/TLS)
Peers	Lync, ATA2201, ATA2202
Routing	Block ATA numbers from PSTN, PSTNtoSIP, Call transfer routing by Lync, To ATA1, To ATA2, Lync-to-PSTN, From ATA
Dialplans	US Dialplan
Address Maps	FromATA.1
Codec Profiles	Lync-Codecs, ATACodecs

3. Click **Load into Gui**.
A confirmation message appears, warning you that current GUI settings will be overwritten.
4. Click **OK** on the message box to complete the load process.
5. Click **Activate Configuration** at the bottom of the SIPcontrol Configuration page to use the loaded configuration.

Importing an Existing Configuration File

Diva SIPcontrol configurations can be exported to a file on the computer running the browser. You can import an exported configuration file to use the saved configuration settings.

To import a configuration file, follow these steps:

1. From the web interface, click **SIPcontrol configuration**.

The SIPcontrol Configuration page appears.

2. In the **Overview** section of the SIP Configuration page, access the Config.-Profiles field, click **Browse**, and locate the configuration (.cfg) file you want to import:

Overview	
Configuration	<input type="button" value="Activate"/> <input type="button" value="Discard"/> <input type="button" value="Start Configuration Wizard"/>
Config.-Profiles	<div> <input type="text"/> <input type="button" value="Load into GUI..."/> <input type="button" value="Delete..."/> <input type="button" value="Export to file..."/> </div> <div> <input type="button" value="Save GUI settings..."/> <input type="text" value="C:\Documents and Settings"/> <input type="button" value="Browse..."/> <input type="button" value="Import from file..."/> </div>
PSTN Interfaces	Controller1, Controller2
Network Interfaces	Intel(R) PRO1000 EB Network (TCP/TLS)
Peers	Lync, ATA2201, ATA2202
Routing	Block ATA numbers from PSTN, PSTNtoSIP, Call transfer routing by Lync, To ATA1, To ATA2, Lync-to-PSTN, From ATA
Dialplans	US Dialplan
Address Maps	FromATA.1
Codec Profiles	Lync-Codecs, ATACodecs

3. Click **Import from file**.

A confirmation message appears, warning you that current GUI settings will be overwritten.

4. Click **OK** on the message box to complete the import process.
5. Click **Activate Configuration** at the bottom of the SIPcontrol Configuration page to use the loaded configuration.

Configuring Diva SIPcontrol Manually

To configure Diva SIPcontrol settings manually:

1. From the Diva SIPcontrol web interface, click **SIPcontrol configuration**.

The SIPcontrol Configuration page appears:

2. Configure each of the sections on the SIPcontrol Configuration page:
 - To expand a section, click on it.
 - To close a section, click the left arrow at the top right of the section.
 - To save the new settings, click **Activate Configuration** at the bottom of each configuration page.

The portion of this chapter starting with [PSTN Interfaces](#) describes the settings in each section of Diva SIPcontrol web interface.

Saving Configuration Settings

Once you configure Diva SIPcontrol as desired, you can save the configuration settings for future use. There are two ways to save the configuration settings:

- Save the settings as a configuration profile on the server.
- Save the settings by exporting them to a file on the computer running the browser.

Saving Configuration Settings as a Configuration Profile

To save the current configuration settings as a configuration profile on the server, follow these steps:

1. Configure Diva SIPcontrol by following the instructions in this chapter.
2. In the **Overview** section of the SIP Configuration page, access the Config.-Profiles field, and click **Save GUI settings**.

The Explorer User Prompt window appears.

3. Enter a name for the saved profile, and click **OK**.

The profile name now appears in the Config.-Profiles listbox.

Exporting Configuration Settings

To export current configuration settings to the computer running the browser, follow these steps:

1. Configure Diva SIPcontrol by following the instructions in this chapter.
2. In the **Overview** section of the SIP Configuration page, access the Config.-Profiles field, and click **Export to file**.

The File Download window appears, and asks whether you want to open or save the file.

3. Click **Save**.

The Save As window appears.

4. Locate the directory where you want to save the file, enter a file name, and click **Save**. Diva SIPcontrol uses the *.cfg* extension for exported files.

Deleting a Configuration Profile

To delete a configuration profile, follow these steps:

1. In the **Overview** section of the SIP Configuration page, access the Config.-Profiles field, and select the profile you want to delete.
2. Click **Delete**.

A confirmation message appears.

3. Click **OK** on the confirmation message to delete the selected profile.

PSTN Interfaces

This section describes Diva SIPcontrol's PSTN interface related settings, e.g., which lines are used by Diva SIPcontrol 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® Media Board Configuration](#).

At least one PSTN interface must be enabled for Diva SIPcontrol to be able to work. Disabled PSTN interfaces are ignored for both inbound and outbound calls. For each line, you can select a dialplan that you can configure as described in [Dialplans](#).

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 appear.

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

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

- [General](#)
- [Enhanced](#)
- [Address Normalization](#)
- [Message Waiting Indication \(MWI\)](#)

General

You can configure the following parameters in the **General** section when you create or modify a PSTN interface:

General	
Hardware description:	Dialogic Diva PRI/E1/T1-8 PCI v3 SN: 1302
PSTN interface number:	1
Name:	<input type="text" value="Controller1"/>
Address map inbound:	<input type="text" value="none"/> ▼
Address map outbound:	<input type="text" value="none"/> ▼

Hardware description: Displays the installed 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 Diva Media Board. The name can be modified in order to display the purpose of the interface or the name of the PBX to which the interface is connected.

Address map inbound: Select the name of a regular expression list to be applied on calls received on this interface. See [Address Maps](#) for more information about setting up a regular expression list. If you upgraded from Diva SIPcontrol 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 version as in former Diva SIPcontrol versions. If you used regular expressions in Diva SIPcontrol 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. Diva SIPcontrol will then use the regular expressions defined in the registry keys that were used by Diva SIPcontrol 1.5.1.

Regular expressions can be used to add or remove dial prefixes required by a PBX or to rewrite public phone numbers with different number ranges into a common format. See [Address Map Examples](#) 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](#) for more information about setting up a regular expression list. If you upgraded from Diva SIPcontrol 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 version as in former Diva SIPcontrol versions. If you used regular expressions in Diva SIPcontrol 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. Diva SIPcontrol will then use the regular expressions defined in the registry keys that were used by Diva SIPcontrol 1.5.1.

Regular expressions can 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 [Address Map Examples](#) for more information.

Enhanced

In the **Enhanced** section, you can configure the settings for early media support. Early media refers to audio and video data that is exchanged before a session is accepted by the called user. It can be unidirectional or bidirectional, and can be generated by the calling party, 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 can configure the following parameters in the **Enhanced** section when you create or modify a PSTN interface:

Enhanced

Early B3 connect:	<input type="text" value="auto"/>
Disable progress:	<input type="checkbox"/>
Early B3 default disconnect timeout [s]:	<input type="text" value="30"/>
Early B3 disconnect timeout [s]: Cause 1: Unallocated number	<input type="text" value="30"/>
Early B3 disconnect timeout [s]: Cause 2: No route to network	<input type="text" value="30"/>
Early B3 disconnect timeout [s]: Cause 3: No route to destination	<input type="text" value="30"/>
Early B3 disconnect timeout [s]: Cause 16: Normal call clearing	<input type="text" value="30"/>
Early B3 disconnect timeout [s]: Cause 22: Number changed	<input type="text" value="30"/>
Early B3 disconnect timeout [s]: Cause 28: Invalid number format	<input type="text" value="30"/>

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 an "inbound tones available" signal is not received from the PSTN.

The following values determine whether EarlyB3 and EarlyB3ForceMedia are enabled:

Value	EarlyB3	EarlyB3ForceMedia
auto	enabled	not enabled
on	enabled	enabled
off	not enabled	not enabled

The default value is **auto**.

Disable progress

Disables the sending of PROGRESS messages to ISDN if a 183 Session Progress message is received from the SIP peer.

This field is disabled by default.

**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.")

The default value is **30** seconds. This value also applies to all causes not listed in the **Enhanced** section.

**EarlyB3
disconnect
timeout [s]
Cause
<x>:<reason
for
disconnect
timeout>:**

With these parameters, you can define the disconnect timeout for the different disconnect timeout reasons. The default value for each reason is **30** seconds.

Address Normalization

You can configure the following parameters in the **Address Normalization** section when you create or modify a PSTN interface:

Address Normalization	
Dialplan:	none
Type of number (outbound):	Unchanged
Encoding (outbound):	Use type flag
ISDN numbering plan - Default:	unknown
Presentation indicator - Default:	Allowed
Internal interface:	<input type="checkbox"/>

Dialplan:

Select the local dialplan to be used by the dialplan module of Diva SIPcontrol. 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](#) before you select it here.

Type of number (outbound):	<p>This parameter determines the shortest format allowed in calls sent out by this interface. You can modify this parameter only if you selected a dialplan from the drop down menu. The following options are available:</p> <p>Unchanged: The number type signaled on the received call request or the type previously set via an inbound dialplan or address map will be used unchanged for dialing.</p> <p>International number: The number is always converted to an international number, including country and area code.</p> <p>National number: The number is converted to a national number unless it is an international number with a different country code.</p> <p>Extension: The number is reduced as much as possible. An internal number is reduced to its extension only.</p> <p>For more information about number formats, see Overview of How Call Address are Processed.</p>
Encoding (outbound):	<p>This parameter determines if numbers in calls sent out by this interface should be encoded as unknown numbers with national or international prefix digits, or as national or international numbers with type flags.</p>
ISDN numbering plan - Default	<p>Change this setting only if the PBX rejects calls from Diva SIPcontrol despite the dialed number being correct. This might occur, for example, if the signaled numbering plan is not supported.</p>
Presentation indicator - Default	<p>If no presentation is specified via address rewriting, this parameter specifies the presentation indicator to set on the calling party number for calls to ISDN. The presentation indicator determines whether the calling party number is shown or hidden from the called user.</p> <p>This default does not apply to PSTN-PSTN calls, unless the known presentation indicator is explicitly removed via an address map.</p>

Internal interface:

This setting controls the usage of the outside access digit by the dialplan in conjunction with this interface. If no outside access digit is configured in the dialplan, this setting has no relevance. Basically, this setting controls whether the outside access digit is expected in the called or calling number depending on the call direction.

- If this setting is enabled, the outside access digit is expected in the called number for calls received on this interface and in the calling number for calls sent by this interface.
- If this setting is disabled, the outside access digit is expected in the calling number for calls received on this interface and in the called number for calls sent by this interface.

In most cases this setting is directly related to the NT/TE mode of the interface. If the interface is in NT mode, this setting usually needs to be enabled. If the interface is in TE mode or an FXO board is used, the setting usually needs to be disabled.

If the Internal Interface option is enabled, calls to the connected network will have a calling number with an outside access digit, unless the calling party has an internal number or the number is converted to the number type format instead of a number with a dialing prefix.

Also, the dialplan expects that calls from the connected network have a called number with outside access digit, unless the called party has an internal number. This expectation can be disabled in dialplan configuration, if necessary.

If the option is disabled, calls to the connected network will have a called number with an outside access digit, unless the called party has an internal number or the number is converted to the number type format instead of a number with dialing prefix.

Also, the dialplan expects that calls from the connected network have a calling number with outside access digit, unless the calling party has an internal number. This expectation can be disabled in dialplan configuration, if necessary.

Message Waiting Indication (MWI):

You can configure the following parameters in the **Message Waiting Indication (MWI)** section when you define or modify a PSTN interface:

Message Waiting Indication (MWI)	
Use this controller for MWI:	<input type="checkbox"/>
Controlling user number:	<input type="text"/>
Controlling user provided number:	<input type="text"/>

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 it to update 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, that is, the ISDN number of the originating user of the indicated message. Few PBXs 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 Diva SIPcontrol, such as the IP addresses and the ports on which Diva SIPcontrol will be listening. Diva SIPcontrol 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 appear.

You can configure the following parameters when you define or modify a network interface:

Network Interfaces					
Name	Device	IP address	UDP listen port	TCP listen port	TLS listen port
Intel(R) PRO1000 GT Desktop	Intel(R) PRO1000 GT Desktop Adapter - Packet Scheduler Miniport	192.168.213.38	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
Local Loopback Interface	Local Loopback Interface	127.0.0.1	<input type="checkbox"/>	<input type="checkbox"/>	<input type="checkbox"/>
RTP start port:	<input type="text" value="30000"/>				
RTP end port:	<input type="text" value="39999"/>				

Name

Displays the name of the installed Ethernet adapter. The preset designation can 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 Diva SIPcontrol is installed.

UDP listen port

If you use UDP as the 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 Diva SIPcontrol. Note that you can only enable one network interface.

TCP listen port	If you use TCP as the 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 Diva SIPcontrol. Note that you can 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 can change the port number, but it must not be the same as the TCP Listen Port number. Note that you can only enable one network interface. If you use TLS, you need to upload security certificates and set the cipher level, as described in Security Profiles.
RTP start port	Defines the lowest port of the range in which Diva SIPcontrol sends and receives RTP streams. Change this value only if problems occur.
RTP end port	Defines the highest port of the range in which Diva SIPcontrol 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 Diva SIPcontrol will establish calls. The peer-specific settings can be used to adapt Diva SIPcontrol'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 appear. The following menus are available for configuration:

- [General](#)
- [Enhanced](#)
- [Security](#)
- [Session Timer](#)
- [Address Normalization](#)
- [Authentication](#)

General

You can configure the following parameters in the **General** section when you define or modify a SIP Peer:

General	
Name:	<input type="text" value="Peer1"/>
Peer type:	<input type="text" value="Default"/> ▼
Host:	<input type="text"/>
Port:	<input type="text" value="5060"/>
IP protocol:	<input type="text" value="TCP"/> ▼
URI scheme:	<input type="text" value="SIP (default)"/> ▼
Domain:	<input type="text"/>

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

Peer type: SIPcontrol needs special workarounds to work properly with some SIP peers, such as Microsoft® Exchange Server. 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 to 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](#).

If you selected:

- **MS Exchange 2007** or **MS OCS 2007/2007 R2 - Mediation Server** as the **Peer type**, set the protocol to **TCP**.
- **MS Lync 2010 - Mediation Server** as the **Peer type**, set the protocol to **TCP** or **TLS**.
- **e-phone**, as the **Peer type**, set the protocol to **UDP**.

URI scheme: This option is only available if you selected **TLS** as the **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 calls to be transmitted via 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 the source domain when it places a call.

Enhanced

You can configure the following parameters in the **Enhanced** section when you create or modify a SIP Peer:

Enhanced	
Default peer for received SIP calls:	<input type="checkbox"/>
Display name to:	<input type="text"/>
Display name from:	<input type="text"/>
User name to:	<input type="text"/>
User name from:	<input type="text"/>
Gateway prefix:	<input type="text"/>
Reply-To expression:	<input type="text"/>
Reply-To format:	<input type="text"/>
Alive check:	<input type="checkbox"/> <input type="text" value="0"/> seconds (0=auto)
Disconnect tone support:	<input type="checkbox"/>
Support MWI (Message waiting):	<input checked="" type="checkbox"/>
Cause code mapping inbound:	peer default ▾
Cause code mapping outbound:	peer default ▾
Codec profile:	default ▾
Maximum channels:	<input type="text" value="480"/>
Early media support:	<input checked="" type="checkbox"/>
Reliable provisional response:	Optional ▾

Default peer for received SIP calls	<p>Enable this option if the selected peer type should be used as the default peer. Calls from unconfigured SIP peers will be assigned to this peer, and therefore are handled with these settings. This option can also be changed via the Default Peer option on the main SIPcontrol page.</p> <p>Note: When a peer is selected as the default, the previously selected default peer is automatically unselected.</p>
Display name to:	Enter the name to be sent in the "To" header of the INVITE message on calls from the PSTN to SIP.
Display name from:	<p>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 "\\".</p>
User name to:	<p>You can 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 Diva Analog Media Boards.</p> <p>If a call from SIP does not contain a user name, the name entered here is transmitted to the receiver as the calling party number. This applies to all references to PSTN in this section. (The opposite side can either be PSTN or SIP.)</p>
User name from:	<p>Enter the user name that is added to the SIP address when a number from the PSTN is suppressed. You can also enter the complete SIP address consisting of <username>@<local-IP/hostname>.</p> <p>If a call from SIP does not contain a user name, the name entered here is transmitted to the PSTN as the called party number.</p>
Gateway prefix:	<p>You can configure this parameter only if you selected e-phone as Peer type in the Edit SIP Peer Configuration window.</p> <p>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.</p>
Reply-To expression:	<p>You can configure this parameter only if you selected e-phone as Peer type in the Edit SIP Peer Configuration window.</p> <p>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.</p>

Reply-To format:	<p>You can configure this parameter only if you selected e-phone as Peer type in the Edit SIP Peer Configuration window.</p> <p>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.</p>
Alive check:	<p>If you select this option, the failover procedure is expedited, because Diva SIPcontrol does not wait for a call time-out if a peer does not respond.</p> <p>To achieve this, Diva SIPcontrol 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.". In this case, Diva SIPcontrol will automatically direct calls to this peer again.</p>
Disconnect tone support:	<p>If the remote side is able to provide inband tones or signals on disconnect, check here to play those inband tones to the SIP peer instead of terminating the SIP call immediately. The SIP call ends either by the client sending a BYE or after the Disconnect Timer of the PSTN interface ends (normally with "Normal call clearing").</p> <p>Normally this option is set only if the peer is a human talker.</p>
Support MWI (Message waiting)	<p>If enabled (the default), the SIP peer is able to receive Message Waiting Indication (MWI) requests via a SIP NOTIFY message from SIPcontrol.</p> <p>If disabled, the destination for MWI requests is chosen from the routing table. If no alternate destinations have been configured, the request is declined.</p>
Cause code mapping inbound:	<p>Select the cause code mapping for calls coming from this SIP peer that you configured under Cause Code Maps.</p>
Cause code mapping outbound:	<p>Select the cause code mapping for calls to this SIP peer that you configured under Cause Code Maps.</p>

Codec profile: Select the codec list that you configured under [Codec Profiles](#). 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.729, if licensed*
4. GSM-FR*
5. G.726 (16, 24, 32, and 40 kbps)*
6. Comfort Noise
7. T.38, if supported by the used Diva Media Board
8. DTMF via RFC 2833/RFC 4733 (no real codec, but internally handled as codec)

In calls from SIP to the PSTN, the first codec offered by the peer that is also in the set of supported and available codecs is selected. This can be changed by a manual configuration that is not currently available via the Diva SIPcontrol web interface.

*For Office Communications Server 2007, Office Communications Server 2007 R2, and Lync Server, G.729, GSM-FR, and G.726 are disabled by default.

Maximum channels: Specifies the number of channels that this SIP peer is able to handle at the same time. This setting is used by Diva SIPcontrol 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 the PSTN. For non-human callers, this option 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 (RFC 3262) should be used. The following values are available:

- **Disabled:** Reliable provisional response is not used.
- **Optional:** Reliable provisional response can be used.
- **Required:** Reliable provisional response is mandatory.

Security

You can configure the following parameters in the **Security** section when you define or modify a SIP Peer:

Security	
Signaling accept level:	Accept unencrypted and encrypted calls
Media security level:	Offer and accept SRTP

Signaling accept level:

This parameter defines how 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, regardless of 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.
- **Require STP for all calls:** All calls are established with SRTP only, regardless of the signaling protocol.

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 can configure the following parameters in the **Session Timer** section when you define or modify a SIP Peer:

Session Timer	
Use session timer:	<input checked="" type="checkbox"/>
Interval:	600
Minimum session expires:	90

Use session timer:	Activates session monitoring via SIP session timers using the time-out values given here. Refer to RFC 4028 for details.
Interval:	If Use session timer is enabled, you can 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 can set a time in seconds between two session refresh messages that Diva SIPcontrol will accept. The minimum value is 90 seconds.

Address Normalization

You can configure the following parameters in the **Address Normalization** section when you define or modify a SIP Peer:

Address Normalization	
Dialplan:	none ▼
Number format (outbound):	Unchanged ▼
Encoding (outbound):	Use prefixes ▼
Address map inbound:	none ▼
Address map outbound:	none ▼

- Dialplan:** Select the local dialplan to be used by the dialplan module of Diva SIPcontrol. Configure the local dialplan as described in [Dialplans](#), 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 can 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 Modification Using Address Maps](#).

Encoding (outbound):	Determines if numbers in calls to this SIP peer should either be encoded as unknown numbers with national or international prefix digits or as national or international numbers with type flags.
Address map inbound:	<p>Name of the regular expressions list applied to the addresses received on calls from this SIP peer. See Address Maps for more information about setting up a regular expression list.</p> <p>Regular expressions can 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 Number Modification Using Address Maps for more information.</p>
Address map outbound:	<p>Select the name of a regular expression list to be applied on calls to this SIP peer. See Address Maps for more information about setting up a regular expression list.</p> <p>Regular expressions can 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 Number Modification Using Address Maps for more information.</p>

Authentication

You can configure the following parameters in the **Authentication** section when you define or modify a SIP Peer:

Authentication			
Realm	Auth user name	Password	
<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
<input type="button" value="Add"/>			

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 can 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 from which 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.
- The current status of a destination. See [How Calls Are Processed](#) for more information.
- The result of an Active Directory query.

To add a route, click the **Add** button. To change the settings for the enabled route, 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 appears.

For more information about possible route configurations, see [Routing Examples](#).

The following menus are available for configuration:

- [General](#)
- [Address Normalization For Condition Processing \(Using Source Dialplan\)](#)
- [Conditions](#)
- [Address Manipulation](#)

General




You can configure the following parameters in the **General** section when you define or modify a route:

General		
Name:	<input type="text" value="Routing1"/>	
Type: Name	Source	Destination
PSTN: Controller1	<input type="checkbox"/>	<input type="text" value="-"/> ▼
PSTN: Controller2	<input type="checkbox"/>	<input type="text" value="-"/> ▼
SIP: SBA	<input type="checkbox"/>	<input type="text" value="-"/> ▼
Max. call attempts for this route in a failover scenario:	<input type="text" value="0"/> (0 = try all selected destinations)	

Name:	Enter a unique name for the route, e.g., "Calls to MS Exchange Server".
Source:	<p>Select either the configured PSTN interfaces or SIP peer as a source. The route will only be considered for a call if the call originated from a selected source.</p> <p>You can select the same interface as a source and destination, e.g., if a call from the PSTN should be routed back to the PSTN.</p> <p>Calls arriving at disabled sources are immediately rejected without querying any route.</p> <p>At least one source interface is required for the route.</p>
Destination:	The master or slave option in the dropdown menu allows for configuring priorities. Diva SIPcontrol 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.
Maximum call attempts for this route in a failover scenario:	<p>Enter the maximum number of destinations in a route that Diva SIPcontrol should call in a failover environment. If you enter 0 (zero), Diva SIPcontrol tries all selected destinations. A value of 1 disables the failover functionality and tries only the first destination of a route.</p> <p>If LDAP is used and the LDAP query requires various call attempts, this counts only as one call attempt in a failover environment, because the LDAP queries are not counted as call attempts here but in a different instance.</p>

Address Normalization For Condition Processing (Using Source Dialplan)

You can configure the following parameters in the **Address Normalization For Condition Processing (Using Source Dialplan)** section when you define or modify a route:

Address Normalization For Condition Processing (Using Source Dialplan) 	
Disable inbound and outbound dialplan:	<input type="checkbox"/>
Number format:	Unchanged 
Encoding:	Use prefixes 

Disable inbound and outbound dialplan:

This parameter defines whether the interface-specific inbound and outbound dialplans should be disabled for the route being defined. If you select this parameter, neither the inbound nor outbound dialplan is applied, and only the address maps are used for the numbers.

Disable this parameter for emergency numbers such as 911 in the U.S. and 110 in Germany, because these numbers can falsely be converted to E.164 using a dialplan. To be sure that special numbers like emergency numbers pass unchanged, you should also define special break-out rules for these numbers in the address maps.

Note: If Disable inbound and outbound dialplan is enabled, the Number format and Encoding parameters will not be evaluated.

Number format:

This parameter determines the shortest format allowed in calls using this route. If the source interface of the call has no dialplan assigned, this setting has no effect. The following options are available:

Unchanged: The number signaled in the received call request 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 Modification Using Address Maps](#).

Encoding:

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

Conditions

You can configure certain conditions for a route. 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, so a line with three condition parts left empty will always apply, thus working as a 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 can configure the following parameters in the **Conditions** section when you create or modify a route:

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Extended Condition	Item	Called	Calling	Redirect	Action
<input checked="" type="checkbox"/>	Address	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="button" value="delete row"/>
	Name	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="button" value="<<< less"/>
	Number type	<input checked="" type="checkbox"/> Unknown Type <input checked="" type="checkbox"/> International <input checked="" type="checkbox"/> National <input checked="" type="checkbox"/> Network specific <input checked="" type="checkbox"/> Subscriber <input checked="" type="checkbox"/> Abbreviated	<input checked="" type="checkbox"/> Unknown Type <input checked="" type="checkbox"/> International <input checked="" type="checkbox"/> National <input checked="" type="checkbox"/> Network specific <input checked="" type="checkbox"/> Subscriber <input checked="" type="checkbox"/> Abbreviated	<input checked="" type="checkbox"/> Unknown Type <input checked="" type="checkbox"/> International <input checked="" type="checkbox"/> National <input checked="" type="checkbox"/> Network specific <input checked="" type="checkbox"/> Subscriber <input checked="" type="checkbox"/> Abbreviated	
	Numbering plan	<input checked="" type="checkbox"/> Unknown Type <input checked="" type="checkbox"/> ISDN/telephony E.164 <input checked="" type="checkbox"/> National standard <input checked="" type="checkbox"/> Private	<input checked="" type="checkbox"/> Unknown Type <input checked="" type="checkbox"/> ISDN/telephony E.164 <input checked="" type="checkbox"/> National standard <input checked="" type="checkbox"/> Private	<input checked="" type="checkbox"/> Unknown Type <input checked="" type="checkbox"/> ISDN/telephony E.164 <input checked="" type="checkbox"/> National standard <input checked="" type="checkbox"/> Private	
	Presentation		<input checked="" type="checkbox"/> Allowed <input checked="" type="checkbox"/> Restricted <input checked="" type="checkbox"/> Number not available <input checked="" type="checkbox"/> Undefined	<input checked="" type="checkbox"/> Allowed <input checked="" type="checkbox"/> Restricted <input checked="" type="checkbox"/> Number not available <input checked="" type="checkbox"/> Undefined	
	Screening Indicator		<input checked="" type="checkbox"/> Not screened <input checked="" type="checkbox"/> Verified and passed <input checked="" type="checkbox"/> Verified and failed <input checked="" type="checkbox"/> Network provided	<input checked="" type="checkbox"/> Not screened <input checked="" type="checkbox"/> Verified and passed <input checked="" type="checkbox"/> Verified and failed <input checked="" type="checkbox"/> Network provided	
<input type="button" value="Add new condition"/>					

Extended Condition

Determines whether additional route conditions appear when you click the **more >>>** button in the Action column. This parameter is enabled by default.

If **Extended Condition** is not enabled, 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 Called, Calling, and Redirect condition entries always match, i.e., a line with the three condition parts left empty will always apply, thus working as a default route.

If **Extended Condition** is enabled, 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, **and** if the route matches at least one of the enabled conditions in each category. For example, suppose the Called condition is set to ^123 and the National and Network specific extended conditions are enabled for the called number. A called number that starts with 123 and is a national number will match, but a called number that starts with 123 and that is not a national or network-specific number will not match.

Empty Called, Calling, and Redirect condition entries work like this when there are extended conditions:

- If the Called and Calling fields are empty and there are **no** extended conditions enabled for those fields, then the called and calling numbers will always match the route conditions.
- If the Called and Calling fields are empty and there **are** extended conditions enabled for those fields, then the called and calling numbers will match the route conditions if they match at least one of the extended conditions.
- If the Redirect field is empty, then the redirect number will always match the redirect route condition, whether or not there are extended conditions enabled for the Redirect field.

**Address
Called:**

If the route is supposed to be valid only for specific called numbers, enter the regular expression to which the route should apply. A regular expression is a pattern that can include numbers, alphabetic characters, special characters, and wildcards.

Diva SIPcontrol attempts to match the regular expression entered here to the called number transmitted on call setup by the remote device. If there is a match, Diva SIPcontrol selects the associated route. If there is no match, Diva SIPcontrol repeats this process with the next route until it finds a match.

**Address
Calling:**

If the route is supposed to be valid only for specific calling numbers, enter the regular expression to which the route should apply. A regular expression is a pattern that can include numbers, alphabetic characters, special characters, and wildcards.

Diva SIPcontrol attempts to match the regular expression entered here to the calling number transmitted on call setup by the remote device. If there is a match, Diva SIPcontrol selects the associated route. If there is no match, Diva SIPcontrol repeats this process with the next route until it finds a match.

**Address
Redirect:**

If the route is supposed to be valid only for specific redirect numbers, enter the regular expression to which the route should apply. A regular expression is a pattern that can include numbers, alphabetic characters, special characters, and wildcards.

Diva SIPcontrol attempts to match the regular expression entered here to the redirect number transmitted on call setup by the remote device. If there is a match, Diva SIPcontrol selects the associated route. If there is no match, Diva SIPcontrol repeats this process with the next route until it finds a match.

The following additional parameters appear when **Extended Condition** is enabled:

Name Called

If the route is supposed to be valid only for specific called numbers, enter a name for the called numbers to which the route should apply.

Name Calling

If the route is supposed to be valid only for specific calling numbers, enter a name for the calling numbers to which the route should apply.

Name Redirect If the route is supposed to be valid only for specific redirect numbers, enter a name for the redirect numbers to which the route should apply.

The following additional parameters appear when **Extended Condition** is enabled and you click the **more >>> button**:

- Number type** The calling, called, and/or redirect numbers to which the route should apply must match one of the enabled number type conditions.
- For example, if National and Network specific are enabled in the Calling column, then the calling number in the route must be either a national or network-specific number type.
- Numbering plan** The calling, called, and/or redirect numbers to which the route should apply must match one of the enabled numbering plan conditions. For example, if ISDN/telephony E.164 and National standard are enabled in the Called column, then the called number in the route must use either an ISDN/telephony E.164 or national standard numbering plan.
- Presentation** The calling and/or redirect numbers to which the route should apply must match the enabled presentation conditions.
- Screening indicator** The calling and/or redirect numbers to which the route should apply must match the enabled screening indicator conditions.

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

The address modified by an address map in a routing condition is usually formatted as it was on arrival. If the address originated from a SIP peer, there will be a host/domain name attached to the address.

Address Manipulation

You can configure the following parameter in the **Address Manipulation** section when you create or modify a route:

Address Manipulation	
Address map:	<input type="text" value="none"/>

Address Map: If a route matches, the address manipulation setting allows you to modify 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 configured address map. You need to configure the address map as described under [Address Maps](#) before you can select it here.

Security Profiles

For authentication and data encryption, certificates need to be installed on the computer on which Diva SIPcontrol is installed and on remote computers. For detailed information on using, generating, and installing certificates, see the following topics in the Data Security section:

- [Using Certificates for Authentication and Data Encryption](#)
- [Generating Private Key Files and Certificates](#)
- [Uploading the Certificate Authority, Certificate, and Key Files to Dialogic® Diva® SIPcontrol](#)
- [Example of Creating a Private Key File and Certificate Request](#)

Uploading a Certificate or Key File

To set up or modify a security profile, click the **Details button** in the **Security Profiles** section. To open the online help for a specific parameter, click the parameter, and a window with the help text will appear.

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

Upload Certificate and Key Files	
Certificate authority file: Not available	<input type="text"/> <input type="button" value="Browse..."/> <input type="button" value="Upload"/>
Certificate file: Not available	<input type="text"/> <input type="button" value="Browse..."/> <input type="button" value="Upload"/>
Key file: Not available	<input type="text"/> <input type="button" value="Browse..."/> <input type="button" value="Upload"/>

To upload a certificate or key file:

1. Click the **Browse** button next to the type of file you want to upload.
2. In the **Choose File to Upload** window locate the certificate or key file, and click **Open**.

- In the Diva SIPcontrol web interface, click **Upload**. After the certificates are uploaded, the information below the certification files changes from "Not available" to "Uploaded". You will not see the paths to the directory in which the files are stored:

Upload Certificate and Key Files	
Certificate authority file: Uploaded	<input type="text"/> <input type="button" value="Browse..."/> <input type="button" value="Upload"/>
Certificate file: Uploaded	<input type="text"/> <input type="button" value="Browse..."/> <input type="button" value="Upload"/>
Key file: Uploaded	<input type="text"/> <input type="button" value="Browse..."/> <input type="button" value="Upload"/>

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 Diva SIPcontrol is installed. This file is used for encrypting information.

Key file:

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

Global Security Parameters

You can configure the following parameters in the **Global Security Parameters** section when you set up a security profile:

Global Security Parameters	
Host name:	<input type="text"/> must match 'CommonName' of certificate!
Supported cipher levels:	High: <input checked="" type="checkbox"/> Medium: <input checked="" type="checkbox"/> Low: <input type="checkbox"/>
Authentication mode:	<input type="text" value="Standard TLS Authentication"/>
Certificate date verification:	<input type="checkbox"/>
<input type="button" value="OK"/> <input type="button" value="Cancel"/>	

Host name	The common name used in the certificate to identify the Diva SIPcontrol host machine.
Supported cipher levels:	<p>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.</p> <ul style="list-style-type: none"> • High: This currently means cipher suites with key lengths larger than 128 bits, and some with 128-bit keys. • Medium: This currently means cipher suites using 128-bit encryption. • Low: This currently means cipher suites using 64- or 56-bit encryption algorithms but excluding export cipher suites.
Authentication mode:	<p>Select how the server-client authentication should be handled:</p> <ul style="list-style-type: none"> • Mutual Authentication: MTLS is used by Microsoft® Office Communications Server 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. <p>For connecting to Microsoft® Office Communications Server 2007 R2 Mediation Server via TLS, use Standard TLS authentication mode. For connecting to Microsoft® Lync Server or Exchange 2007 UM role via TLS, use MTLS authentication mode.</p> <ul style="list-style-type: none"> • 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 needs to prove its authentication. <p>The default setting is Standard TLS Authentication.</p>
Certificate date verification:	<p>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.</p>

LDAP

The Lightweight Directory Access Protocol (LDAP), is an application protocol that programs use for querying information from a server. The protocol runs over TCP/IP. Deployments today tend to use Domain Name System (DNS) names for structuring the topmost levels of the hierarchy. LDAP servers index all the data in their entries, and "filters" can be used to select the person or group for which you are looking. LDAP is appropriate for any kind of directory-like information, where fast lookups and less-frequent updates are the norm. For example, when you use Microsoft® Outlook and search the address book for a colleague, you access the Microsoft® Active Directory database via LDAP.

How to Use LDAP to Access Active Directory for Routing Calls via Diva SIPcontrol

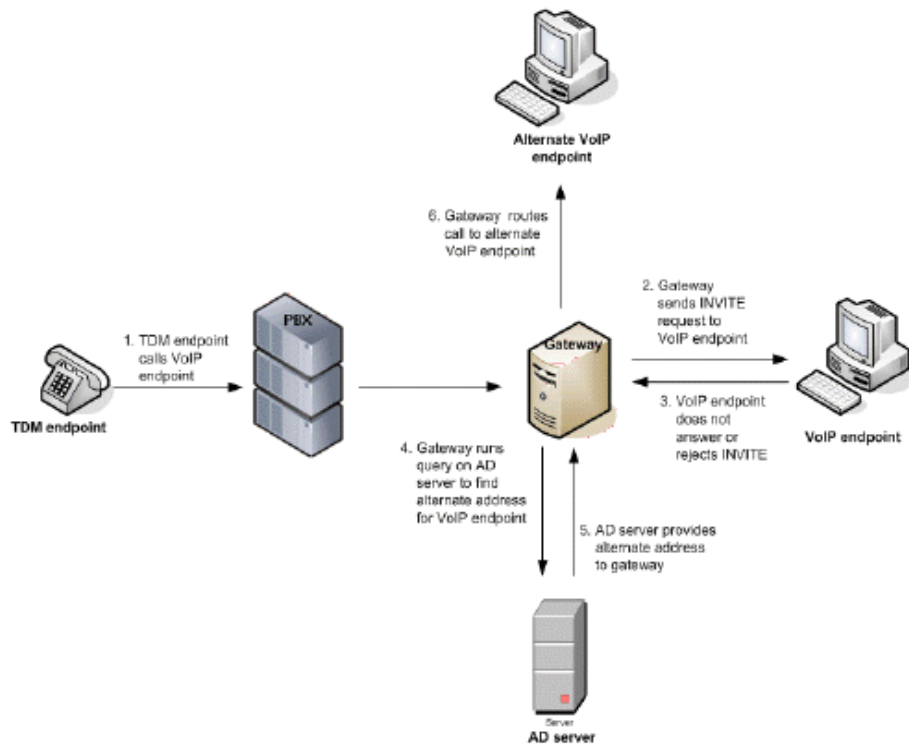
You can enable LDAP functionality via Diva SIPcontrol web configuration. When LDAP is activated, Diva SIPcontrol will query the server on startup and store the query results internally for a faster lookup. In a default configuration, this internal storage will be updated once a day to reflect changes on the LDAP database. If you use LDAP, you need to configure two routes for one LDAP call:

- One route should contain the LDAP destination.
- The other route should contain the final destination.

The order of the routes is irrelevant, but it is important to configure the first route with the conditions needed to avoid recursion.

Use Case for LDAP

In this scenario, the gateway is connected between the PSTN or PBX and a VoIP endpoint. Additionally, the gateway receives a call from the TDM network that is intended for the VoIP endpoint. However, the VoIP endpoint does not respond to the original INVITE request or it responds with a failure. The gateway must query the Active Directory server to determine if there is an alternate address (i.e. phone number, etc.) where the VoIP endpoint can be contacted, and if so, the gateway must route the call to the alternate address.



To add LDAP query settings, click the **Add** button in the **LDAP** section. To change the settings for each LDAP, click the **Details** button on the right hand side.

LDAP Query

You can configure the following parameters in the **LDAP Query** section when you create or modify an LDAP:

LDAP Query	
Name:	QUERY1
Search Attribute:	telephoneNumber
Result (1st):	
Result (2nd):	
Result (3rd):	

Name Enter a name to easily identify the LDAP query.

Search attribute: Select the attribute to search for:

- **telephoneNumber:** The primary office number.
- **homePhone:** The primary home/private number.
- **mobile:** The primary mobile number.
- **facsimileTelephoneNumber:** The primary fax number.
- **proxyAddresses:** E-mail address attributes that can have either format SMTP:bob@domain.de or sip:bob@domain.de.

The following attributes are used by Office Communications Server 2007, Office Communications Server 2007 R2, and Lync Server 2010:

- **msRTCSIP-PrimaryUserAddress:** This attribute contains the SIP address of a given user.
- **msRTCSIP-Line:** Refers to a user's primary office number as specified in the Active Directory msRTCSIP-line attribute, which the Microsoft® Office Communications Server uses to perform reverse number lookup to obtain all the user's SIP endpoints.

Result: The list defines the attributes for which to search. The called address searched for will be replaced by the contents of the result attribute with the highest priority value. If there is more than one attribute with that priority, the call will be forked and several simultaneous call attempts will be made. The attributes in result 2 and 3 are sequentially searched only if the address with the highest priority fails.

LDAP Domain

LDAP Domain

Domain:	<input type="text"/>		
Base Search DN:	<input type="text"/>		
Search Scope:	base (search the object itself) ▼		
Server Address	Server Port	User Name	Password
<input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="button" value="Add"/>			

You can configure the following parameters in the **LDAP Domain** section when you define or modify an LDAP:

- Domain:** Enter the domain name of the LDAP server, e.g., dialogic.com.
- Base Search DN:** This is the starting point in the Active Directory hierarchy at which your search will begin, e.g.,
ou=EMEA,ou=corp,dc=dialogic,dc=com.
- Search Scope:** The LDAP search scope indicates the set of entries at or below the base search DN that can be considered potential matches for a search operation.
- There are three search scope values:
- **base (search the object itself):** This specifies that the search should only be performed against the entry specified as the base search DN. No entries below it will be considered. Use this option if the base search DN is close to the data to be searched for, because this way desired data can be found quickly.
 - **one level (search the object's immediate children):** This specifies that the search operation should only be performed against entries that are immediate subordinates of the entry specified as the base search DN. Neither the base entry itself nor the entries below the immediate subordinates of the search base entry are included.
 - **subtree (search the object and all its decendants):** This specifies that the search operation should be performed against the base search DN itself and all of its subordinates.
- Server Address:** Enter the IP address of the Active Directory server. This entry is mandatory, because Diva SIPcontrol does not use a default server.
- You can enter the server address either as an IP address or as an FQDN, e.g., 11.11.11.11 or ldap.dialogic.com.
- It is possible to configure multiple servers for an LDAP query. In this case, SIPcontrol will use the second server if the first one fails.
- Server Port:** Enter the port to which the server is listening. The default value is **389**.
- and LDAPS 636. If you set the port to 0, Diva SIPcontrol will select a port automatically.
- If you use an indexed database, such as Microsoft® Active Directory, set the port to 3268 to speed up LDAP queries.

User Name: Enter a user name. This can be a DN, UPN (User Principal Name, e.g., name@domain.name), Windows NT style username (e.g., domain\username), or another name that the directory server will accept as an identifier.

In some cases, it is possible to connect to an LDAP server without a user name and password. If it is not possible, you can create a dummy user for this gateway task.

Password: Enter a password for the user account.

Note: Diva SIPcontrol currently supports Simple authentication, which means that the password is transmitted in clear text over the network. The password also is stored and processed locally in clear text.

It is recommended that you use a separate user account with restricted permissions for Diva SIPcontrol access.

LDAP Cache

You can configure the following parameters in the **LDAP Cache** section when you define or modify an LDAP:

LDAP Cache	
Cached Entries:	<input type="text" value="5000"/>
Prefetched Entries:	<input type="text" value="5000"/>
Refresh Timeout (sec):	<input type="text" value="14400"/>

Cached Entries Number of local cache entries resident in the system memory. Valid range is from 100 to 1000000. Default is **5000** entries.

Prefetched Entries Number of LDAP entries loaded into the local LDAP cache during startup. Valid range is from 0 (disabled) to the number of Cached Entries. Default is **5000** entries.

Refresh Timeout (sec) Amount of time allowed to elapse from the time an LDAP entry is cached to the time the cached entry will no longer be used. If a cached entry is queried after the refresh timeout elapses, the entry will be retrieved from the LDAP server and cached again. If the LDAP server is not reachable during a query, the older cached value remains in the cache and will be used for the query.

Valid range is from 0 to $2^{32} - 1$ seconds (limited by a 32 bit variable).

Default is **14400** seconds (4 hours)

Dialplans

With help of the local phone settings, Diva SIPcontrol is able to convert a received call address to a normalized form, e.g., the E.164 format. This not only eases the definition of subsequent conditions or maps, but it also converts the call to the format 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 can be used.
- Adding and removing the line access code: If not present, dialed numbers are automatically prepended by the digits needed to access the public telephone network.
- Support for the North American numbering plan: Up to 10 area codes can be configured to be treated differently. For example, in many areas, dialing into neighboring areas does not require dialing a long-distance prefix.

For more information about how dialplans work, see [Number Normalization Based on a Dialplan](#).

Important information about the outside access digit configuration

Configure the outside access digit only if there is a PBX between the PSTN and Diva SIPcontrol, 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 Diva SIPcontrol expects the outside access digit in the calling number of 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 caller:** This option defines whether Diva SIPcontrol expects the outside access digit in the called number of external calls. It is normally required to prepend the outside access digit to call an external number from an internal phone. 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.

Enable this option if the internal users that use SIPcontrol for external calls prepend the outside access digits in their calls; otherwise disable it.

Note: Diva SIPcontrol's number normalization function does not remove outside access digits as a PBX can do for external calls. If Diva SIPcontrol 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 will appear.

You can configure the following parameters in the **General** section when you create or modify a dialplan:

General

Name:	<input type="text" value="Dialplan1"/>
Country code:	<input type="text"/>
North-American numbering plan:	<input type="checkbox"/>
Area code:	<input type="text"/> With national prefix ▾
Other local areas:	<input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/>
Base number:	<input type="text"/>
Maximum extension digits:	<input type="text" value="0"/> ▾
International prefix:	<input type="text"/>
National prefix:	<input type="text"/>
Access code:	<input type="text"/>
PSTN access code provided by SIP caller:	<input checked="" type="checkbox"/>
Incoming PSTN access code provided by PBX:	<input checked="" type="checkbox"/>
Keep SIP URI Domain and Parameters:	<input type="checkbox"/>

OK

Cancel

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

Country code: Enter the country code without any prefixes for the country in which the computer with the installed Diva SIPcontrol 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. Diva SIPcontrol 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:	<p>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.</p> <p>If you need to use the NANP, you can choose between the following number transmission methods:</p> <p>With national prefix: The long-distance code is added to the number.</p> <p>Local: The number is transmitted without any area code.</p> <p>Without national prefix: The number is transmitted without the long-distance prefix.</p>
Other local areas:	You can 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 a 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 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 PBX:

Select this option if the PBX adds the access code to the calling number for incoming external calls.

Keep SIP URI Domain and Parameters:

(Relevant for calls from SIP only)

If this option is enabled, then the SIP URI domain and parameters are kept as a suffix of the resulting, normalized address. This may have an impact on the address maps and conditions applied after normalization. In the case of a SIP-to-SIP call, the host part of the SIP URI is forwarded to the other SIP peer.

If this option is disabled (the default), then the number normalization process removes the SIP URI domain and parameters from the resulting address, leaving only the SIP user part in the resulting address.

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 conditions 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 be changed with the "arrow down" and "arrow up" buttons in Microsoft® Internet Explorer® or the **Up** and **Down** buttons in Mozilla Firefox.

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 will appear.

You can configure the following parameters when you create or modify an address map:

General	
Address map name:	<input type="text" value="AddressMap1"/>
Rule name:	<input type="text" value="AddressMap1.1"/>
Stop on match:	<input type="checkbox"/>
Enhanced configuration:	<input checked="" type="checkbox"/>

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 can be selected.

Note: The name can be edited only during the creation of an address map.

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

Stop on match: This flag determines whether Diva SIPcontrol should continue to search for matching rules when all expressions match all addresses of a call. If set, the address matching is aborted when there is a match.

Enhanced configuration This flag determines whether Diva SIPcontrol uses the new interface for configuring address maps. The new interface handles the flag and number portions of a SIP address separately. For example, with the SIP address +17166391234@dmg.dialogic.com, the flag portion is + and the number portion is **17166391234**.

In addition, the new interface supports the ability to specify the number type, numbering plan, and presentation indicator directly, while the older version supported prefixing the address with "+," "N", and "S", respectively, to specify a limited subset of number types.

Leave this option enabled (the default), unless you need to use the old interface for compatibility reasons. (For example, if you need to import address maps from an earlier SIPcontrol version.) For address maps created in earlier versions of SIPcontrol, this option is disabled by default.

Note: In the old interface, number type flags from digital networks, e.g., ISDN or SS7, are converted into special prefixes on the SIP side. Therefore, the following address formats apply only to the old interface:

- + 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

The following fields appear when the **Enhanced configuration** option is enabled:

Called address rules

	Condition	Result	
Address:	<input type="text"/>	<input type="text"/>	Fields for the number portion
Name:	<input type="text"/>	<input type="text"/>	
Number type:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> International <input checked="" type="checkbox"/> National <input checked="" type="checkbox"/> Network specific <input checked="" type="checkbox"/> Subscriber <input checked="" type="checkbox"/> Abbreviated	<input type="text" value="No change"/>	Fields for the flag portion
Numbering plan:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> ISDN/telephony E.164 <input checked="" type="checkbox"/> National standard <input checked="" type="checkbox"/> Private	<input type="text" value="No change"/>	

Calling address rules

	Condition	Result	
Address:	<input type="text"/>	<input type="text"/>	Fields for the number portion
Name:	<input type="text"/>	<input type="text"/>	
Number type:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> International <input checked="" type="checkbox"/> National <input checked="" type="checkbox"/> Network specific <input checked="" type="checkbox"/> Subscriber <input checked="" type="checkbox"/> Abbreviated	<input type="text" value="No change"/>	Fields for the flag portion
Numbering plan:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> ISDN/telephony E.164 <input checked="" type="checkbox"/> National standard <input checked="" type="checkbox"/> Private	<input type="text" value="No change"/>	
Presentation:	<input checked="" type="checkbox"/> Allowed <input checked="" type="checkbox"/> Restricted <input checked="" type="checkbox"/> Number not available <input checked="" type="checkbox"/> Undefined	<input type="text" value="No change"/>	
Screening Indicator:	<input checked="" type="checkbox"/> Not screened <input checked="" type="checkbox"/> Verified and passed <input checked="" type="checkbox"/> Verified and failed <input checked="" type="checkbox"/> Network provided	<input type="text" value="No change"/>	

Redirect address rules		
	Condition	Result
Address:	<input type="text"/>	<input type="text"/>
Name:	<input type="text"/>	<input type="text"/>
Number type:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> International <input checked="" type="checkbox"/> National <input checked="" type="checkbox"/> Network specific <input checked="" type="checkbox"/> Subscriber <input checked="" type="checkbox"/> Abbreviated	No change
Numbering plan:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> ISDN/telephony E.164 <input checked="" type="checkbox"/> National standard <input checked="" type="checkbox"/> Private	No change
Presentation:	<input checked="" type="checkbox"/> Allowed <input checked="" type="checkbox"/> Restricted <input checked="" type="checkbox"/> Number not available <input checked="" type="checkbox"/> Undefined	No change
Screening Indicator:	<input checked="" type="checkbox"/> Not screened <input checked="" type="checkbox"/> Verified and passed <input checked="" type="checkbox"/> Verified and failed <input checked="" type="checkbox"/> Network provided	No change

Fields for the number portion

Fields for the flag portion

OK Cancel

Address Checks or manipulates the current address (normally a number for a PSTN address or a SIP-URI for a SIP address). To remove the part matched by the condition, set the Result field to empty. To check for a specific condition and keep the original address, set the Result field to **\$&**.

Name Checks or manipulates the current name. To remove the part matched by the condition, set the Result field to empty. To check for a specific condition and keep the original address, set the Result field to **\$&**.

Number type Checks or manipulates the number type of the current address. For unmodified SIP addresses, the number type is "unknown". To check for a number type without changing it, set the Result field to **No change**.

Numbering plan	Checks or manipulates the numbering plan of the current address. For unmodified SIP addresses, the numbering plan is Unknown . To check for a numbering plan without changing it, set the Result field to No change .
Presentation	Checks or manipulates the presentation indicator. For unmodified SIP addresses, the presentation is Undefined . To check for a presentation without changing it, set the Result field to No change .
Screening indicator	Checks or manipulates the screening indicator. For unmodified SIP addresses, the value is Not screened . To check for a screening indicator value without changing it, set the Result field to No change .

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

The address modified by an address map is usually formatted as it was on arrival. If the address originated from a SIP peer, there will be a host/domain name attached to the address.

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 Diva SIPcontrol does not fulfill your needs, e.g., because your local PBX uses non-standard cause codes, you can configure your own cause code mapping table, which will be checked before the default table is checked. See [Cause Code Mapping](#) for the cause/response code mapping table. If you create your own cause code mapping table, make sure to select it in the **SIP Peers** section under [Enhanced](#).

To add a cause code, open the **Cause Code Maps** section and 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 appear. You can configure the following parameters when you create or modify a cause code:

Cause Code Mapping	
Name:	CauseMapping1
Direction:	PSTN to SIP
PSTN cause code	SIP response code
Add	
Default:	
OK Cancel	

- 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.
- Note:** For SIP to SIP calls or PSTN to PSTN calls, the original cause code is preserved, so mapping is unnecessary.
- 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.
- Note:** For SIP to SIP calls or PSTN to PSTN calls, the original cause code is preserved, so mapping is unnecessary.
- Default** Enter the cause or response code that Diva SIPcontrol 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, Diva SIPcontrol's internal default mapping table will be used.

Codec Profiles

To configure a codec profile, 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 Peers** section, under **Enhanced**. To open the online help for a specific parameter, click the parameter, and a window with the help text will appear.

You can configure the following parameters when you create or modify a codec profile:

General

Name:

Audio Codecs

Available Codecs

G.729
G.726 16 kbps
G.726 24 kbps
G.726 32 kbps
G.726 40 kbps

Use Codec -->

<-- Remove Codec

Selected Codecs

G.711 A-Law
G.711 u-Law

Up Down

G.711 A-Law Codec Settings

Packet interval default:

Voice activity detection: ☐

Comfort Noise Generation: ☐

Audio Quality

Support comfort noise payload: ☒

Noise suppressor: ☐

Echo canceller: ☒

DTMF Codec

Transmit as RTP event: ☒

Automatic payload type: ☒

Disable CNG event: ☐

Manual payload type value:

Fax Codec

T.38 support: ☒

V.34 support: ☒

Maximum datagram size:

OK Cancel

Name:	Enter a name to easily identify the codec profile. You can select the codec profile in the SIP Peers section.
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 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 Generation:	<p>If you enable this parameter, packets with low artificial background noise are sent to fill periods where no data packets are received from the SIP peer. This helps prevent the other party from thinking the transmission has been lost (because of the silence) and hanging up prematurely.</p> <p>Note: Support for this feature depends on the type of Diva Media Board present in the system. If the hardware does not support this feature, the setting is ignored.</p>
Support comfort noise payload:	<p>If you enable this parameter and VAD is configured for the codec used for the call, periods of silence will be replaced by sending a special "comfort noise" signal to the SIP peer. This allows a supporting SIP device to generate appropriate artificial background noise in order to remove the impression of the call being interrupted.</p> <p>If the SIP peer does not support this type of event, this setting has no effect.</p>
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 for calls to or from the PSTN.
Transmit as RTP event:	This option enables DTMF and fax tones to be sent and received as RTP events instead of audio signals.
Automatic payload type:	G.726, iLBC, RT Audio, and DTMF have a dynamic RTP payload. If you enable this option, Diva SIPcontrol sets the values automatically. 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, Diva SIPcontrol uses the value suggested by the endpoint. Generally, this parameter is left at its default value.
Disable CNG event:	Select this option to transmit the CNG event as an in-band audio signal instead of an RTP event according to RFC 4733. Note: This option is only available if the option Transmit as RTP event is enabled.
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 Diva Media Boards with multiple ports only after activating the respective license. See License Activation 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 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 can 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 for the user.

If Diva SIPcontrol 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 can use either a private registrar service or a public registrar server.

To configure a registrar server, open the **Registrations** section and 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 appear.

You can configure the following parameters when you create or modify a registration:

General	
Name:	<input type="text" value="Registrar1"/>
Registrar address:	<input type="text"/>
Registrar port:	<input type="text"/>
Registrar protocol:	TCP ▼
URI scheme:	SIP (default) ▼

- 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 calls to be 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 sent 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	URI scheme	User name	@Domain	Protocol	Re-register time	Auth user name	Password	Register as	
<input type="text"/>	SIP (default) ▼	<input type="text"/>	<input type="text"/>	UDP ▼	3600	<input type="text"/>	<input type="text"/>	Standard ▼	Delete
<input type="button" value="Add"/>									

- 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 Diva SIPcontrol uses to register at the registrar server.

@Domain:	Enter the domain name of the registrar server.
Protocol:	Select UDP if you register as an e-phone gateway.
Re-register time:	Enter the re-register time in seconds. This is the time for which 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 Diva SIPcontrol to function as a gateway for e-phone.

Logging and Diagnostics

You can configure the following parameters shown in **Logging and Diagnostics** section:

Logging and Diagnostics	
Event log level:	Errors
Debug level:	Off
XML Configuration file:	Show
<div> <div>Activate Configuration</div> <div>Discard Configuration</div> </div>	

Event Log Level:	A computer with Diva SIPcontrol installed can write different types of events into the System Event Log. The details for each event log are described in Event Logging .
Debug Level:	The debug level setting can be used for debugging and tracing purposes. During normal operation, it should be set to Off to lessen the effect on system performance.
XML configuration file:	Shows the configuration in raw format. This option is used only by Dialogic Support.

6. Data Security

Data Security Overview

Since version 2.0, Diva SIPcontrol provides the following security options for transmitted and received data:

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

This section describes the use of the Secure HTTP, TLS, and Secure RTP protocols. It also describes how to generate, install, and use private key files and certificates.

Secure HTTP

HTTP is a protocol that transmits data between the web-based configuration interface of Diva SIPcontrol 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.

For 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 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 Diva SIPcontrol and the computer on which the web browser used to connect to Diva SIPcontrol via HTTPS is running must be configured with the proper certificate.
- When accessing the Diva SIPcontrol web interface, use `https://<IP-address-or-URL-of-Diva-Webserver>:10006/` instead of `http://<IP-address-or-URL-of-Diva-Webserver>:10005/`.

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 helps to ensure that you are talking to the intended peer. For authentication purposes, you need to install [Certificates](#), as described in Security Profiles, and enable TLS as the transport protocol, as described in [Network Interfaces](#).

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, and 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. Diva SIPcontrol 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. For information, see Security Profiles. The cipher level can be set in the [Global Security Parameters](#).

Using Certificates for Authentication and Data Encryption

For authentication and data encryption, certificates need to be installed on the computer on which Diva SIPcontrol 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, e.g., with one of the many tools available on the internet. Search 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. Also, many web browsers check if the certificate is signed by a CA, and, if it is not, a warning message will appear asking whether the user really wants to trust that web site, which can make the user feel insecure.

Certificate files can be generated in different formats, e.g., .pem, .der, .cer, or .pfx. All files need to be in "pem" format (base64 encoded) in order to be used by Diva SIPcontrol.

A default certificate is provided with the software, but for security reasons, you should install your own web server certificate.

Note for CER files: CER files can be renamed to .pem directly if they are base64 encoded. No bag attribute lines and/or additional CR and empty lines are allowed. If CER files are ASN.1 coded, they need to be converted to with a converter tool.

Note for PFX files: The PFX or PKCS#12 format is a binary format for storing the server certificate, any intermediate certificates, and the private key in one encryptable file. When converting a PFX file to PEM format, tools like OpenSSL will put all the certificates and the private key into a single file. You will need to open the file in a text editor and copy each certificate and private key (including the BEGIN/END statements) to its own individual text file and save them as certificate.cer, CACert.cer, and privateKey.key respectively.

How to Retrieve Keys and Certificates from a PFX File for Use in Diva SIPcontrol

In the following procedure openssl is used as example converter tool.

1. Export the private key file from the PFX file:

```
openssl pkcs12 -in filename.pfx -nocerts -out protected-key.pem
```

2. Remove the passphrase from the private key as required by Diva SIPcontrol:

```
openssl rsa -in protected-key.pem -out key.pem
```

3. Export the certificate file from the PFX file:

```
openssl pkcs12 -in filename.pfx -clcerts -nokeys -out cert.cer
```

4. Export the Root CA certificate file from the PFX file:

```
openssl pkcs12 -in filename.pfx -cacerts -nokeys -out cacert.cer
```

Using Certificates with Microsoft® Office Communications Server 2007

Microsoft® Office Communications Server 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. The CRL distribution point is an extension within the digital certificate that can be used if the CA (certification authority) in your PKI (Public Key Infrastructure) has a CRL distribution point.

- 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 an FQDN (Fully Qualified Domain Name), either in the Certification field CN (Common Name) / SN (Subject Name) or SAN (Subject Alternative Name), or both.

Using Certificates with Microsoft® Lync™ Server 2010

Lync Server requires that the gateway server certificate must contain the FQDN configured for the gateway in the Lync Topology Builder. This FQDN must be specified in the CN or SAN. Alternatively, it can be specified in both locations.

Generating Private Key Files and Certificates

Microsoft® Active Directory Certificate Services is a role of the Windows Server 2008 operating system. On Windows Server 2008, it can be installed through the Add Roles Wizard. On Windows Server 2003, this service is a component and can be installed through the Windows Component Wizard.

Note: Do not install the Microsoft Active Directory Certificate Services on your DMG4000 Gateway. Install it on a separate computer.

To use Microsoft Active Directory Certificate Services to generate private key files and certificates for the DMG4000 Gateway, follow these steps:

1. Create a private key file and a certificate request file with a third party program. For an example, see below.

Example of Creating a Private Key File and Certificate Request

The following example shows how to create a security certificate using openssl:

1. (If you use the openssl that was preinstalled on the DMG 4000 Gateway, you can skip this step.) Download and install openssl:
<http://gnuwin32.sourceforge.net/packages/openssl.htm>
2. Create a folder to hold the key file and certificate request; for example:
`c:\Keys\SBA1`.
3. In Windows Explorer, search for the *openssl.conf* file, and make note of the directory path for the file. On the DMG4000 Gateway SU4.1, the *openssl.conf* file is in the `C:\Program Files (x86)\GnuWin32\share\openssl.cnf` directory.
4. Execute openssl req commands to request both a private key file and a certificate request file. When you execute these commands, you must use the `-config` option to point to *openssl.conf*; otherwise you will get an error.

For example, the following commands request a private key file named `priv.cer` and a certificate request file named `request.csr`. This example uses the default install location of OpenSSL. You can copy these commands if you want to use the same install location.

```
C:\Program Files (x86)\GnuWin32\bin>openssl req -new -nodes -keyout c:\keys\sba1\priv.cer -out c:\keys\sba1\request.csr -config "C:\Program Files (x86)\GnuWin32\share\openssl.cnf"
```

Output like the following appears:

```
Loading 'screen' into random state - done
Generating a 1024 bit RSA private key
.....+++++
.....+++++
writing new private key to 'c:\keys\sba1\priv.cer'
-----
You are about to be asked to enter information that will be incorporated
into your certificate request.
What you are about to enter is what is called a Distinguished Name or a DN.
There are quite a few fields but you can leave some blank
For some fields there will be a default value,
If you enter '.', the field will be left blank.
-----
```

5. Enter values for the requested fields. The values you enter make up the Distinguished Name (DN) of the CA certificate. The value for Common Name is the most important value, and it must be the exact FQDN. Leave the values for the 'extra attributes' blank.

For example:

```
Country Name (2 letter code) [AU]:US
State or Province Name (full name) [Some-State]:NY
Locality Name (eg, city) []:Buffalo
Organization Name (eg, company) [Internet Widgits Pty Ltd]:Dialogic
Organizational Unit Name (eg, section) []:Dialogic Research
Common Name (eg, YOUR name) []:sba1.training.com
Email Address []:
Please enter the following 'extra' attributes
to be sent with your certificate request
A challenge password []:
An optional company name []:
```

If the openssl requests are successful, the CA places two security files into the directory you created in Step 2 (c:\Keys\SBA1).

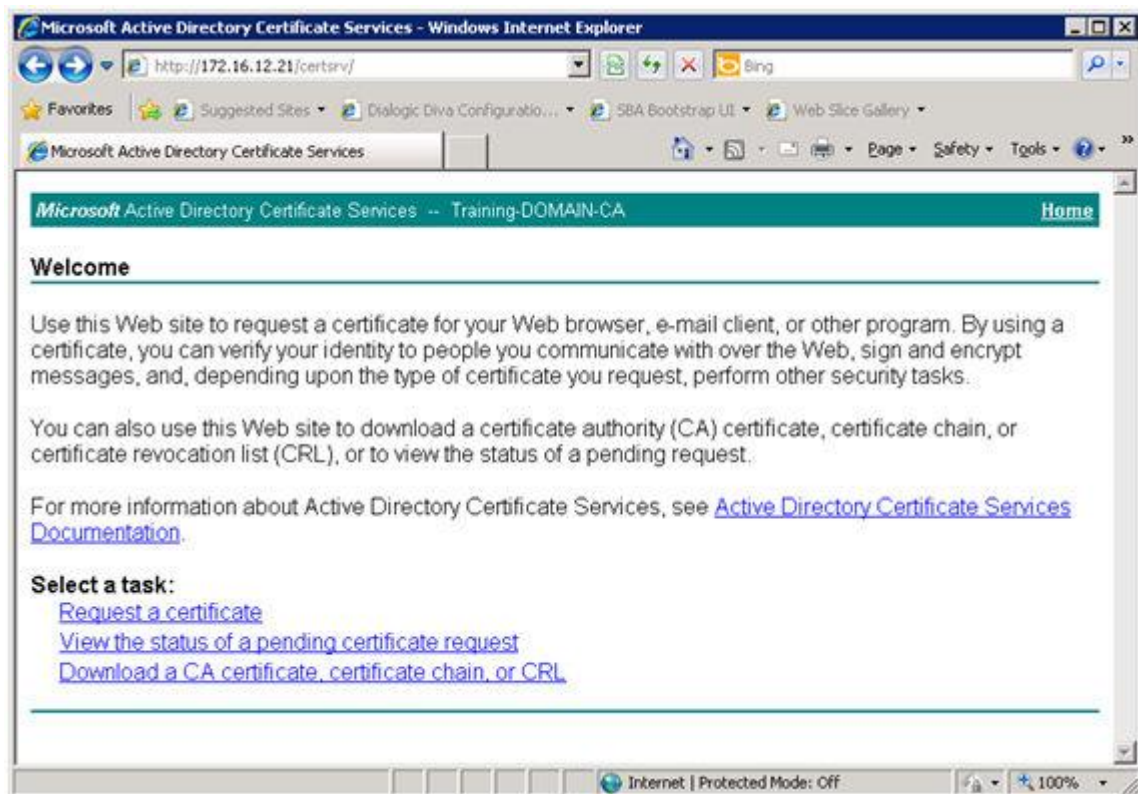
2. Access the Microsoft® Active Directory Certificate Services website from any machine in the domain where the Lync Front End Server is installed. The domain and IP address will vary, depending on the installation. For example:

```
http://domain/certsrv
```

```
http://172.16.12.21/certsrv
```

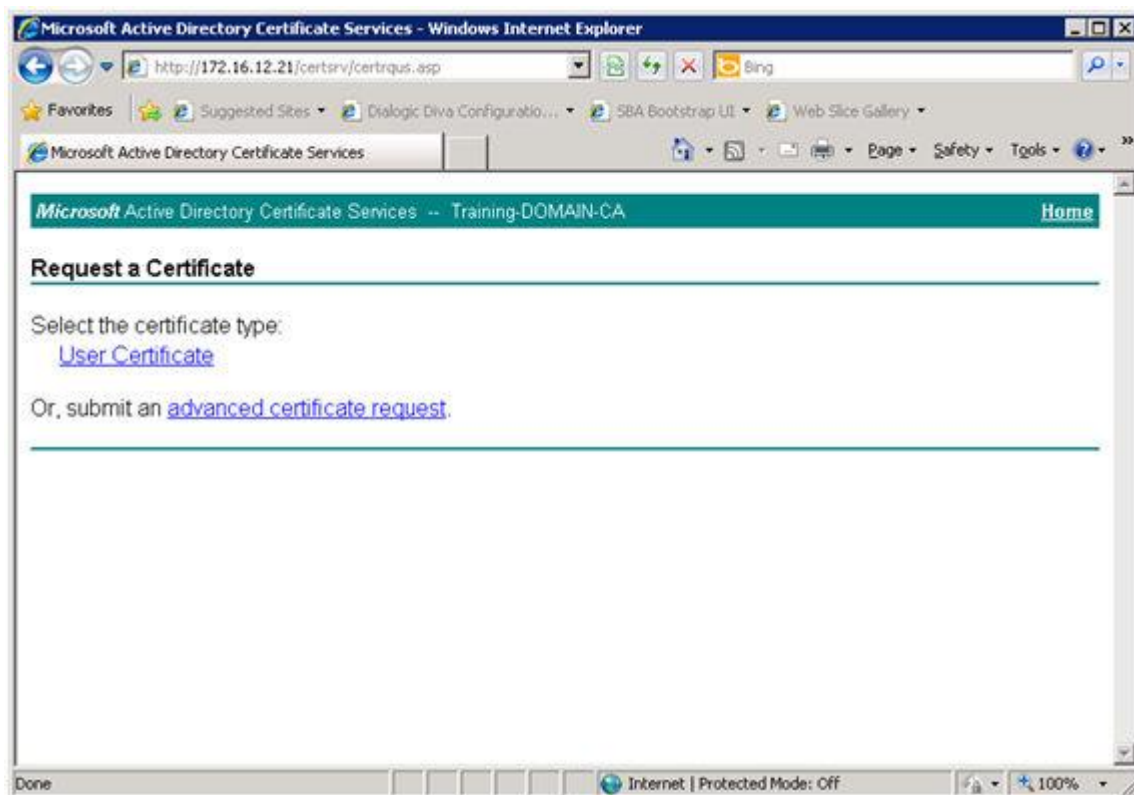
3. Log into the Microsoft Active Directory Certificate Services website with Administrator rights.

The Microsoft Active Directory Certificate Services website appears:



4. Click **Request a certificate**.

The Request a Certificate page appears:

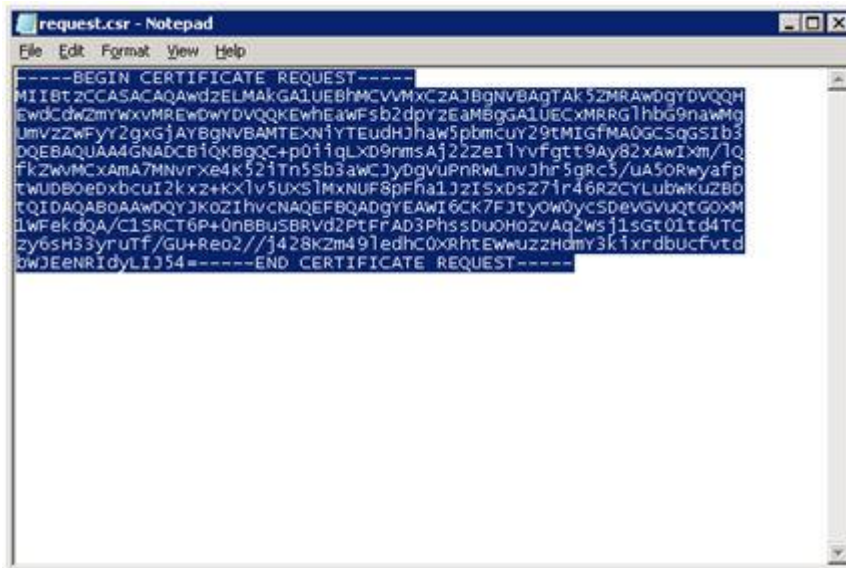


5. Click **advanced certificate request**.

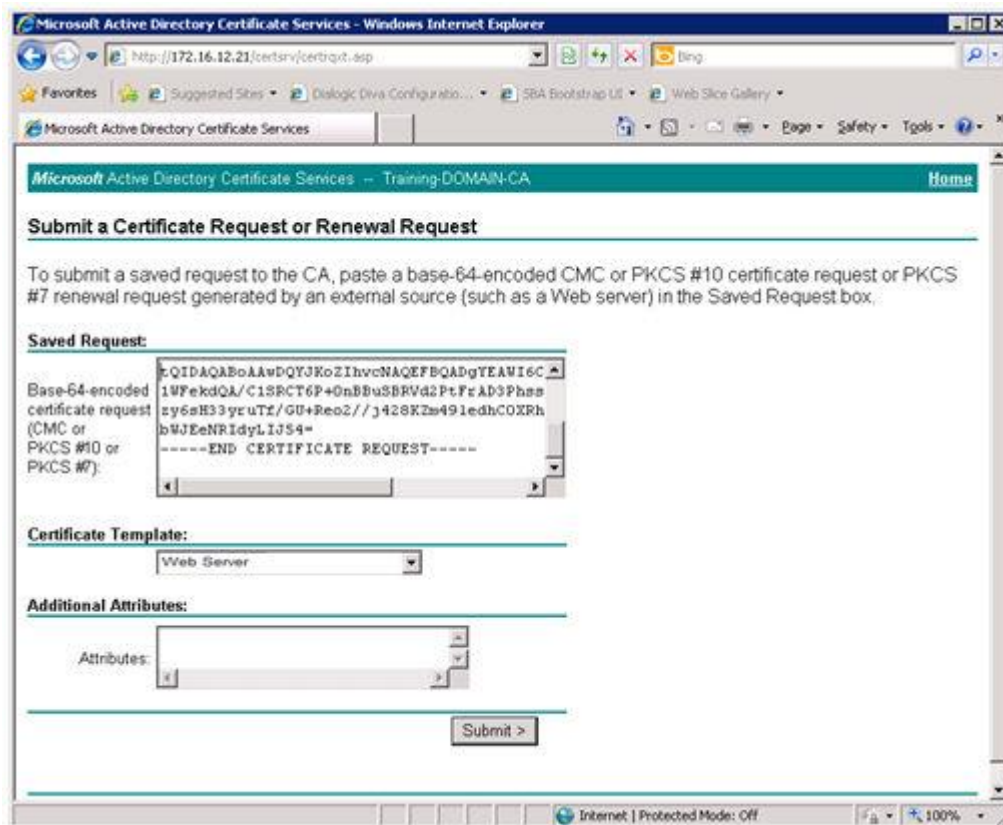
The Advanced Certificate Request page appears:



6. Select the second option, **Submit a certificate request by using a base-64-encoded CMC or PKCS #10 file, or submit a renewal request by using a base-64-encoded PKCS #7 file**.
7. Open the certificate request file with Wordpad. (In the following example, this file is called *request.csr*, and it resides in the *c:\keys\sba1 directory*.) Select all file contents (everything between BEGIN CERTIFICATE REQUEST to END CERTIFICATE REQUEST), as shown below:



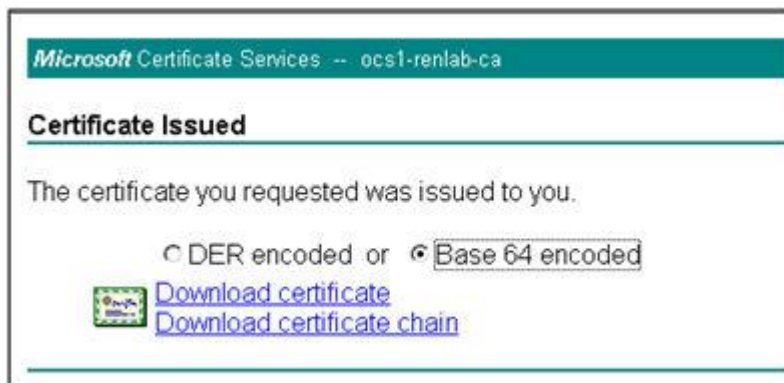
8. Paste the contents into the Saved Request section of the Microsoft Active Directory Certificate Services website, and select Web Server in the Certificate Template field:



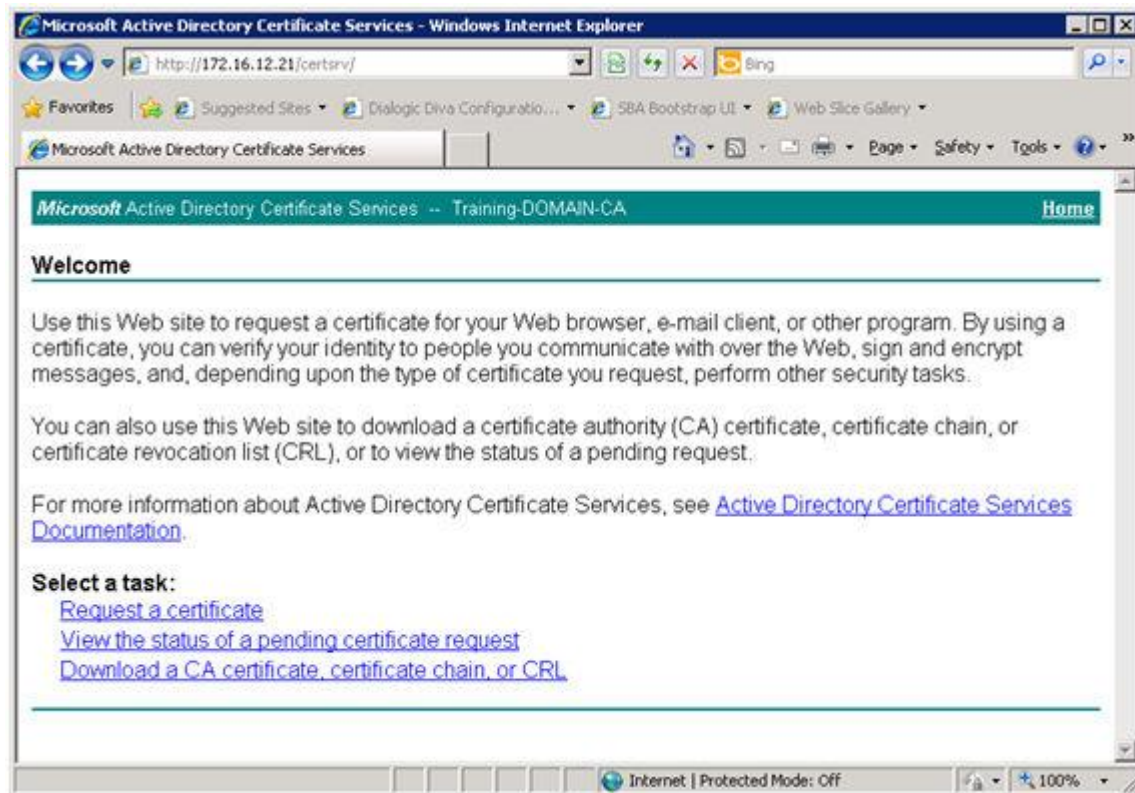
9. Click **Submit**.

If the certificate creation is successful, the Microsoft Active Directory Certificate Services download page appears.

10. On the Microsoft Active Directory Certificate Services download page for signed certificates, select **BASE 64 encoded**, and click **Download certificate**:



11. In the File Download dialog box, select Save This File to Disk and click OK. The saved file is the Certificate File for Diva SIPcontrol.
12. Go back to the Microsoft Active Directory Certificate Services home page:



13. Click **Download a CA certificate, certificate chain, or CRL**.
The Download a CA Certificate, Certificate Chain, or CRL page appears:
14. Select **Base 64**, and click the text in the CA certificate field.
15. Save the downloaded file. The saved file is the Certificate Authority file (*certAuth.cer*).

Uploading the Certificate Authority, Certificate, and Key Files to Dialogic® Diva® SIPcontrol™

Once you generate the key files and certificate request file, upload them to Diva SIPcontrol, as described in the steps below:

1. Click **Start > Programs > Dialogic Diva > SIPcontrol Configuration** to access the Diva SIPcontrol web interface. By default, access to the web interface is only allowed from local host (127.0.0.1), and the port number to which the server is listening is set to 10005.

The Diva web interface login page appears.

2. In the Password field, enter Dialogic.

The Dialogic® Diva® Configuration page appears:



3. Click **SIPcontrol configuration** on the left hand side.

The SIPcontrol Configuration page appears.

4. Click **Security Profiles** (about halfway down the page), and then click Details to open the Security Profiles options:

Upload Certificate and Key Files	
Certificate authority file: Not available	<input type="text"/> Browse... Upload
Certificate file: Not available	<input type="text"/> Browse... Upload
Key file: Not available	<input type="text"/> Browse... Upload

Global Security Parameters	
Host name:	<input type="text"/> must match 'CommonName' of certificate!
Supported cipher levels:	High: <input checked="" type="checkbox"/>
	Medium: <input checked="" type="checkbox"/>
	Low: <input type="checkbox"/>
Authentication mode:	Standard TLS Authentication ▼
Certificate date verification:	<input type="checkbox"/>

OK Cancel

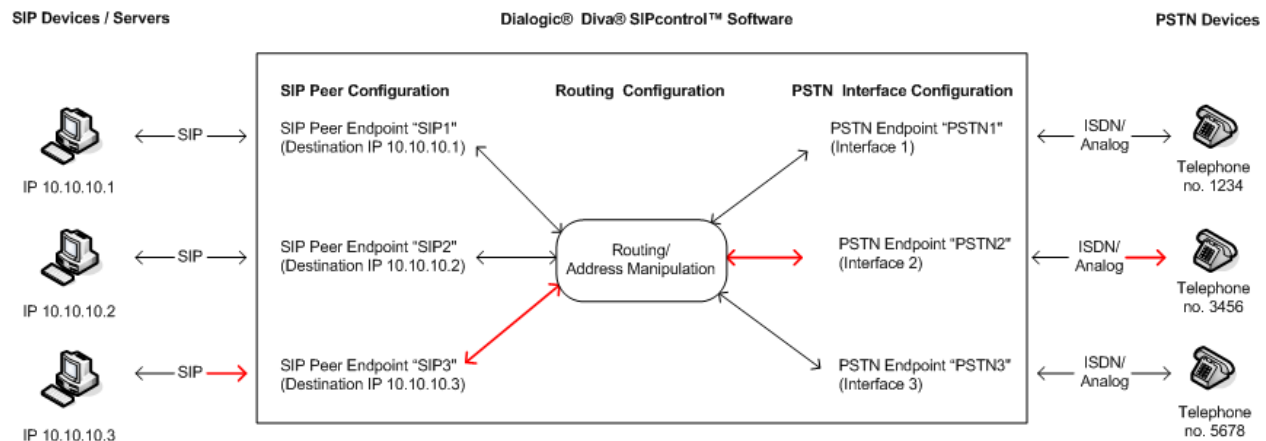
5. In the Certificate authority file field, use **Browse** to locate the Certificate Authority file (*certAuth.cer*).
6. Click **Upload** to upload the *certAuth.cer* file to Diva SIPcontrol.
A message box appears.
7. Click **OK** to upload the file.
8. Repeat Steps 5 – 7 for the Certificate file and Key file.
9. In the Authentication mode field (in the Global Security Parameters section), select how the server-client authentication should be handled.
10. Click **OK** at the bottom of the page to close the Security Profiles window.
11. At the bottom of the SIPcontrol main page, click Activate Configuration to save the configuration.
12. Restart Diva SIPcontrol to use the new configuration. Click **System Control** on the left side of the web interface, and then click **Restart** in the SIPcontrol field.

7. How Calls are Processed

How Calls are Processed

In the following discussion, SIP and PSTN endpoints/interfaces are interchangeable, and they may be used on no, one, or two sides of the call.

Diva SIPcontrol 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 holds Diva SIPcontrol 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 two endpoints of any type 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 holds 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 a SIP peer endpoint in the **SIP Peers** section.
- Settings PSTN Device 2 needs to be configured as a PSTN endpoint in the **PSTN Interfaces** section.
- Condition "called address is 3456" needs to be configured in the **Routing** section to route the call to the correct device.

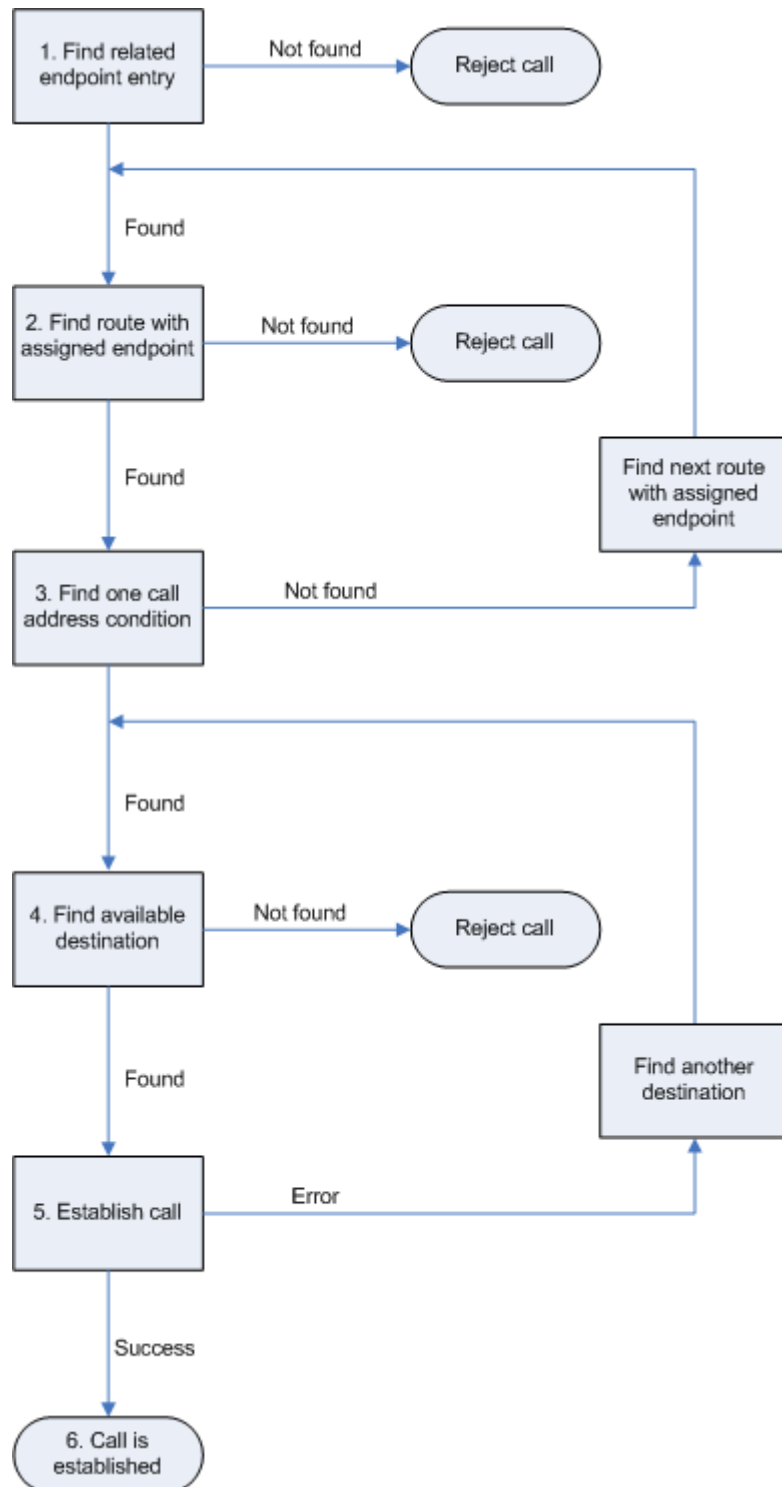
For example, if you have a SIP or PSTN Device 4 with no endpoints configured in Diva SIPcontrol, you cannot establish a call, because Diva SIPcontrol will not know the settings of the device.

The PSTN endpoint is found via its controller number. On the SIP side, multiple SIP peers can 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, Diva SIPcontrol looks for a SIP peer with the **Default SIP 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 [Information 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.



Information about Call Processing

- Each route can point to several destinations, between which Diva SIPcontrol 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 also matches the call criteria, only the first matching route is ever evaluated. Therefore, default routes should be created carefully and located at the end of the routing table, if appropriate.
- 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 condition 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

Diva SIPcontrol 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.

Routing Examples

This section describes the configuration of four possible routing scenarios:

- [Direct Routing between One PSTN Interface and One SIP Peer](#)
- [Connecting Two SIP Peers to Two PSTN Interfaces Exclusively](#)
- [Connecting Two SIP Peers to the Same PSTN Interface](#)
- [Load Balancing or Failover between Two SIP Peers](#)

Direct Routing between One PSTN Interface and One SIP Peer

If you choose to route all calls from the PSTN to the same SIP peer, and calls from that SIP peer to the PSTN, configure the parameters as follows. For this configuration, no address rewriting is done:

1. Under **PSTN Interfaces**, enable and configure all PSTN interfaces connected to a PBX. Confirm each dialog box with **OK**.
2. Under **SIP Peers**, create a SIP peer with the necessary settings, and make sure that the option **Default peer for received SIP calls** is enabled. Confirm with **OK**.
3. Under **Routing**, create Route 1 and do the following:
 - Select each required PSTN interface as a source peer.
 - Select the SIP peer configured in Step 2 as a Master destination.
 - Set the **Number format** field to **Unchanged**.
 - Confirm with **OK**.
4. Under **Routing**, create Route 2 and do the following:
 - Enable the SIP peer configured in Step 2 as a source peer.
 - Enable all required PSTN interfaces as Master destinations.
 - Set the **Number format** field to **Unchanged**.
 - Confirm with **OK**.
5. Save the configuration in the main configuration interface.

Direct Routing between Two SIP Peers

If you choose to route all calls from the PSTN to the same SIP peer, and calls from that SIP peer to the PSTN, configure the parameters as follows. For this configuration, no address rewriting is done:

1. Under **SIP Peers**, create one SIP peer with the necessary settings, and make sure that the option **Default peer for received SIP calls** is enabled. Confirm with **OK**.
2. Under **SIP Peers**, create the other SIP peer with the necessary settings. Confirm with **OK**.
3. Under **Routing**, create Route 1 and do the following:
 - Select the SIP Peer configured in Step 1 as a source peer.
 - Select the SIP peer configured in Step 2 as a Master destination.
 - Set the **Number format** field to **Unchanged**.
 - Confirm with **OK**.
4. Under **Routing**, create Route 2 and do the following:
 - Select the SIP peer configured in Step 2 as a source peer.
 - Select the SIP peer configured in Step 1 as a Master destination.
 - Set the **Number format** field to **Unchanged**.
 - Confirm with **OK**.
5. Save the configuration in the main configuration interface.

Connecting Two SIP Peers to Two PSTN Interfaces Exclusively

If you choose to connect two SIP peers to two PSTN interfaces, so that each SIP peer can use one interface exclusively, then carry out the following configuration steps. The procedure is similar if you need to configure more PSTN interfaces, e.g., three PSTN interfaces to three SIP peers.

1. Under **PSTN Interfaces**, enable and configure the two PSTN interfaces. Confirm with **OK**.
2. Under **SIP Peers**, create both SIP peers and make sure the entry in **Domain** exactly matches the domain used by the SIP peer in its SIP address for outgoing calls. Do not enable the option **Default peer for received SIP calls** for any of these peers. Confirm with **OK**.
3. Under **Routing**, create Route 1 and do the following:
 - Enable the first PSTN interface as a source peer.
 - Enable the first SIP peer configured in Step 2 as a Master destination.
 - Confirm with **OK**.
4. Under **Routing**, create Route 2. Then, repeat Step 3 for the second PSTN interface and the second SIP peer.
5. Under **Routing**, create Route 3 and do the following:
 - Enable the first SIP peer configured in Step 2 as a source peer.
 - Enable the first PSTN interface as a Master destination.
 - Confirm with **OK**.
6. Under **Routing**, create Route 4. Then, repeat Step 5 for the second PSTN interface and the second SIP peer.
7. Save the configuration in the main configuration interface.

Connecting Two SIP Peers to the Same PSTN Interface

If you want to connect two SIP peers to the same PSTN interface so that all calls from the PSTN are sent to the first SIP peer if the numbers begin with "1" and to the second peer if the numbers begin with "2", configure the parameters as follows:

1. Under **PSTN Interfaces**, enable and configure the PSTN interface. Confirm with **OK**.
2. Under **SIP Peers**, create both SIP peers and make sure the entry in **Domain** matches exactly the domain used by the SIP peer in its SIP address for outgoing calls. Do not enable the option **Default peer for received SIP calls** for any of these peers. Confirm with **OK**.
3. Under **Routing**, create Route 1 and do the following:
 - Enable the first PSTN interface as a source peer.
 - Enable the first SIP peer configured in Step 2 as a Master destination.
 - Under **Conditions**, click **Add** and set the **Called address** to "1.*".
 - Confirm with **OK**.
4. Under **Routing**, create Route 2 and repeat Step 3 for the second SIP peer with the only difference that the called address condition for this route is "2.*".

5. Under **Routing**, create Route 3 and do the following:
 - Enable both SIP peers as source peers.
 - Enable the first PSTN interface as a Master destination.
 - Confirm with **OK**.
6. Save the configuration in the main configuration interface.

If calls other than those beginning with 1 or 2 should also be directed to one peer, remove the condition from the respective PSTN to SIP route and move the route to the end of the list.

Load Balancing or Failover between Two SIP Peers

To configure two servers for load balancing or failover, follow these steps:

1. Under **PSTN Interfaces**, enable and configure all required PSTN interfaces. Confirm with **OK**.
2. Under **SIP Peers**, create both SIP peers and make sure the entry in **Domain** exactly matches the domain used by the SIP peer in its SIP address for outgoing calls. Do not enable the option **Default peer for received SIP calls** for any of these peers. If you configure a failover, SIP peer 1 (a Master) should have the option **Alive check** enabled. Confirm with **OK**.
3. Under **Routing**, create Route 1 and do the following:
 - Enable the first PSTN interface as a source peer.
 - Enable the first SIP peer configured in Step 2 as a Master destination. For load-balancing configurations, SIP peer 2 should be configured as a Master destination. For failover configurations, it should be configured as a Slave destination.
 - Confirm with **OK**.
4. Under **Routing**, create Route 2 and do the following:
 - Enable both SIP peers as source peers.
 - Enable the first PSTN interfaces as a Master destination.
 - Confirm with **OK**.
5. Save the configuration in the main configuration interface.
6. Under **Routing**, create Route 2 and repeat Step 3 for the second SIP peer with the only difference that the called address condition for this route is **2.***.
7. Under **Routing**, create Route 3 and do the following:
 - Enable both SIP peers as source peers.
 - Enable the first PSTN interface as a Master destination.
 - Confirm with **OK**.
8. Save the configuration in the main configuration interface.

If calls other than those beginning with 1 or 2 should also be directed to one peer, remove the condition from the respective PSTN to SIP route and move the route to the end of the list.

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8. How Call Addresses are Processed

Overview of How Call Addresses are Processed

The call addresses provided by the caller can be modified at different stages of the call processing within Diva SIPcontrol. 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.

Diva SIPcontrol provides two mechanisms for number processing:

- [Number Normalization Based on a Dialplan](#)
- [Number Modification Using Address Maps](#)

Both mechanisms can be used together.

When using a dialplan, Diva SIPcontrol converts addresses automatically, without any intervention from the user. This means that Diva SIPcontrol adds or removes a special prefix to a number with a known number type when converting between a number and an address. The automatic conversions are done for calling numbers, called numbers, and redirected numbers. See [Common Results](#) for a list of prefixes.

For complex conversions, you can configure an address map for Diva SIPcontrol to use when converting addresses.

Diva SIPcontrol uses routing conditions, dialplans, and address maps to manipulate call addresses. For information and examples on how these configuration entities work together, see [How Call Addresses Are Manipulated](#) and [Possible Call Routing Scenarios](#).

Number Normalization Based on a Dialplan

Number normalization based on a dialplan can work on any PRI, BRI, and analog interface, including interfaces in NT mode. It can also work on SIP peers that do not represent a direct or indirect connection to a public SIP trunk.

Diva SIPcontrol supports dialplans using the North American numbering plan (NANP). See [North-American numbering plan](#) for more information.

Depending on how it is defined, a dialplan might affect only the numeric part of an address:

- If the address consists of a number only, then the complete address is used to match a dialplan condition, and the complete address is modified by the matching dialplan.
- If the address is a SIP address with a number in the user part, then only the user part of the address is used to match a dialplan condition, and only the user part of the address is modified by the matching dialplan. For example, for SIP address 12345@domain.tld, only 12345 will be used to match a dialplan condition. If Diva SIPcontrol finds a match, only 12345 will be modified by the matching dialplan.

The **Keep SIP URI Domain and Parameters** option determines whether the SIP URI domain and parameters are used as a suffix of the resulting, normalized address. If this option is enabled, then the SIP URI domain and parameters are kept as a suffix of the resulting, normalized address. Otherwise, the number normalization process removes the SIP URI domain and parameters, leaving only the SIP user part in the resulting address.

Configure the outside access digit only if there is a PBX between the PSTN and Diva SIPcontrol, and if this PBX requires the outside access digit for external calls. For information on configuring the outside access digit, see [Important Information about the Outside Access Digit Configuration](#).

Diva SIPcontrol's number normalization function does not remove outside access digits like a PBX can for external calls. If Diva SIPcontrol needs to behave like a PBX with an outside access digit for external calls, use the address map functionality in combination with a route.

Steps for Number Normalization Based on a Dialplan

Number normalization based on a dialplan 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 result. Six target results 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 a different country code will be converted to an international number with a prefix for international calls and an 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 a different country code will be converted to an international number with the number type set to "international". **Note:** This target result 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 result should not be used for calls to SIP networks.

Number Modification Using Address Maps

Diva SIPcontrol organizes regular expressions into address maps, and each endpoint or route can be assigned one map. Each address map contains a number of regular expressions together with the respective output result strings. This helps to ensure 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 probably be configured with the same settings. Therefore, these PSTN interfaces can share a single address map that Diva SIPcontrol lets you assign for each individual controller.

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

Common Expressions:

Character	Meaning
.	Matches any character
^	Matches the beginning of an address
\$	Matches the end of an address
\+	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-condition to be referenced in result string and also groups sets of characters
	Alternate operator, matches either the left or right sub-condition
[]	Matches any character given within the square brackets, i.e [123] matches either 1, 2, or 3, but not 4, 5, or 123.
(?i)	Considers case for everything after the tag. For example "username@(?i) hostname.tld" matches "username\$HOSTNAME.TLD" and "username@hostname.TLD", but not "USERNAME@hostname.tld".

Common Results

Character	Meaning
0-9, +	Inserts the respective character into the output
(?n(digits))	Inserts the digits given only if the n th sub-condition of the condition matched
\$&	Outputs what matched the whole condition

\$n	Outputs the n th matched sub-condition
\$(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)

Address Map 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 conditions and results.

The examples can 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.

Address Condition: ^33

Address Result: (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.

Address Condition: .*

Address Result: 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

Address Condition: .*

Address Result: 01\$&

Number type Condition: International

Number type Result: Unknown

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

Address Condition: ^01

Address Result: (none)

Number type Condition: Unknown

Number type result: International

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 Address Condition: 1111(@.*)?\$

First Address Result: 2222

Stop on Match: true

Second Address Condition: 3333(@.*)?\$

Second Result Condition: 4444

Stop on Match: true

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

Replace the National Number Type with a Prefix

Task: Replace the National number type in a national number with the national prefix 0.

Example: National number 123-45678 should be signaled as 0123-45678

Address Condition: .*

Address Result: 0\$&

Number type Condition: National

Number type Result: Unknown

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 result 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 727-0203 without E.164. The local area code is +1(123).

Address Condition: ^(.*)

Address Result: \$1;phone-context=+1(123)\$1

How Call Addresses are Manipulated

Diva SIPcontrol uses routing conditions, dialplans, and address maps to manipulate call addresses in the following way:

Note: Each step is optional, depending on the configuration.

1. Saves the inbound call addresses as "A".
2. Applies the "address map inbound" of the endpoint assigned to the call setup request to "A", resulting in "B".
3. To check the first route: applies the number format settings of the route together with the dialplan of the source endpoint to the call addresses "B", resulting in "C".
4. Checks the route against addresses "C". If the route does not match, Diva SIPcontrol discards the changes and tries the next route with "B" again. For information about routes, see [Routing](#).
5. If the route matches, Diva SIPcontrol applies the route address map to the addresses "C", resulting in "D".
6. After selecting one of the destinations of the route, Diva SIPcontrol normalizes the addresses "D" using the dialplan and number format of the destination endpoint, resulting in addresses "E".
7. Applies 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 there are other endpoints in a fail-over configuration, Diva SIPcontrol starts with Step 6 again with the respective settings of the next endpoint.

Possible Call Routing 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 the 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 can 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.

9. Software Uninstallation

Software Uninstallation

Note: If you want to upgrade from Diva SIPcontrol version 1.5.1, DO NOT uninstall the software before you install the current Diva SIPcontrol version, because you might lose some settings, including your regular expressions.

The uninstallation procedure depends on the installed operating system.

To uninstall Diva SIPcontrol under Windows® XP or Windows Server® 2003:

1. Click **Start > Settings > Control Panel**.
2. Double-click **Add or Remove Programs**.
3. In the **Add or Remove Programs** box, select Diva SIPcontrol software and click **Remove**.
4. When you are asked if you want to remove Diva SIPcontrol from your computer, confirm with **Yes**.
Diva SIPcontrol is now uninstalled.
5. If you want to uninstall the Diva System Release software, see the *Dialogic® Diva® System Release Reference Guide*, which is available on the Dialogic web site: www.dialogic.com.

To uninstall Diva SIPcontrol under Windows Vista®, Windows Server® 2008, or Windows® 7:

1. Click **Start > Control Panel > Programs**.
2. Click **Uninstall a program**.
3. In the displayed window, right-click the Diva SIPcontrol software entry and select **Uninstall**.
4. If you are asked either to **Cancel** or **Allow** the uninstallation, click **Allow** to proceed.
5. Diva SIPcontrol is now uninstalled.
6. If you want to uninstall the Diva System Release software, see the *Dialogic® Diva® System Release Reference Guide*, which is available on the Dialogic web site: www.dialogic.com.

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10. Cause Code Mapping

Cause Code Mapping

If Diva SIPcontrol uses Microsoft® Office Communications Server 2007 or Lync Server 2010 as a SIP peer, the cause/response code tables are used as specified by Microsoft. See [Default Cause Code Mapping for Microsoft® Office Communications Server 2007 and Lync Server 2010 Peers](#) for a detailed list of cause/response codes.

If Diva SIPcontrol does not use Microsoft® Office Communications Server 2007 or Lync Server 2010, the default cause/response code mapping is used. See [Default Cause Code Mapping](#) for a detailed list of cause/response codes.

Default Cause Code Mapping

Diva SIPcontrol 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 Maps** section.

For ISDN to SIP code mappings, see [ISDN Cause Code to SIP Response Code](#).

For SIP to ISDN code mappings, see [SIP Response Code to ISDN Cause Code](#).

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

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	503	Service unavailable
127	Interworking, unspecified	503	Service unavailable
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

423	Interval too brief	63	Service or option unavailable
429	Provide Referrer Identity	31	Normal, unspecified
480	Temporarily unavailable	19	No answer from the 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
487	Request Terminated	127	Interworking, unspecified
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 2007 and Lync Server 2010 Peers

Diva SIPcontrol includes a default cause/response code mapping table for Microsoft® Office Communications Server 2007 SIP peers and Lync Server 2010 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 Maps section, as described in [Cause Code Maps](#).

For ISDN to SIP code mappings, see [Microsoft® Office Communications Server 2007 and Lync Server 2010 ISDN Cause Code to SIP Response Code](#).

For SIP to ISDN code mappings, see [Microsoft® Office Communications Server 2007 and Lync Server SIP Response Code to ISDN Cause Code](#).

Microsoft® Office Communications Server 2007 and Lync Server 2010 ISDN Cause Code to SIP Response Code

ISDN cause code	Description	SIP response code forwarded to Microsoft® Office Communications Server 2007 or Lync Server 2010	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
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	503	Service unavailable
127	Interworking, unspecified	500	Server internal error
Any code other than listed above		500	Server internal error

Microsoft® Office Communications Server 2007 and Lync Server SIP Response Code to ISDN Cause Code

SIP response code from Microsoft® Office Communications Server 2007 or Lync Server 2010	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
429	Provide Referrer Identity	31	Normal, 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
486	Busy here	17	User busy
487	Request Terminated	127	Interworking, unspecified
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

Cause Code Mapping

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

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11. Event Logging

Event Logging

A computer with Diva SIPcontrol installed can 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 Diva SIPcontrol 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 can 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 can 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 none of the audio codecs support by and allowed for the SIP peer could be used for the call and no alternative 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 occurring during TLS data processing. The trace can 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.

12. Use Case Examples

Use Case Examples

Diva SIPcontrol is designed to support standard VoIP RFCs, therefore, the usage of Diva SIPcontrol is not limited to the described use case samples below. Diva SIPcontrol is interoperable with many applications, e.g., Asterisk or e-phone.

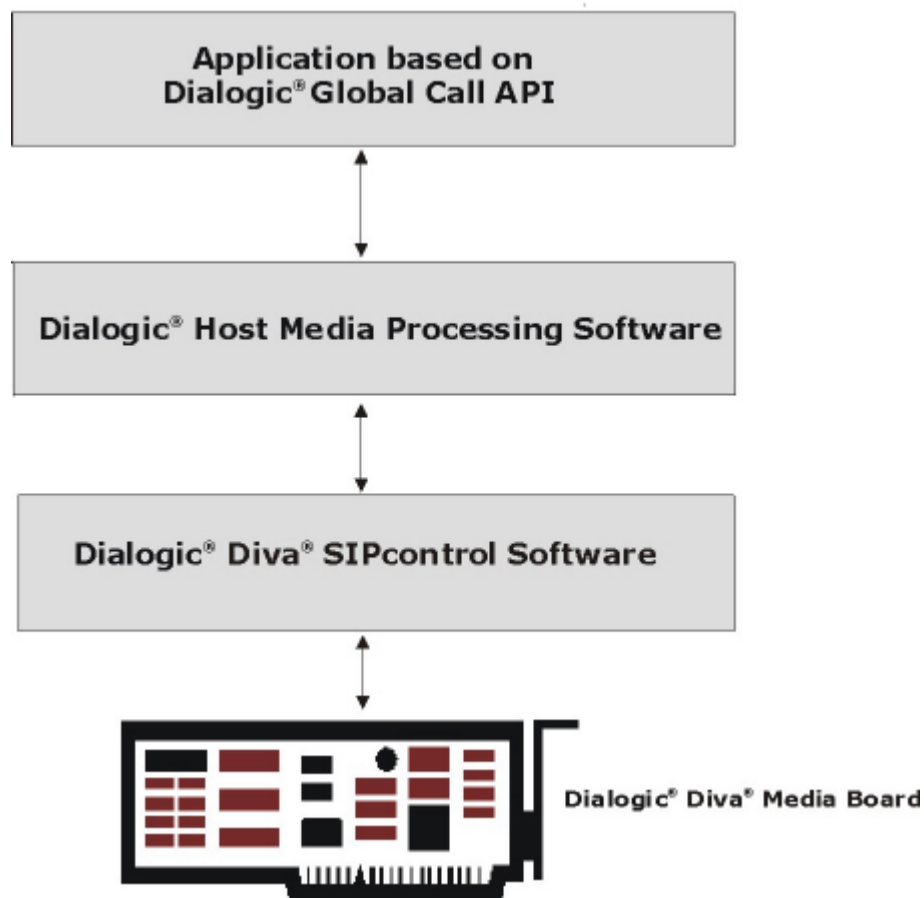
The following scenarios describe how to use the gateway computer with Dialogic® HMP Software, Microsoft® Exchange Server 2007, Microsoft® Office Communications Server 2007, and Microsoft® Lync™ Server 2010.

- [Use Case for Dialogic® HMP Software](#)
- [Use Case for Microsoft® Exchange Server 2007](#)
- [Using the Gateway Computer Between the PBX and Microsoft® Office Communications Server 2007](#)
- [Using the Gateway Computer Between the PBX/PSTN and Microsoft® Office Communications](#)
- [Using the Gateway Computer Between the PSTN and PBX/Microsoft® Office Communications Server 2007](#)
- [Using the Gateway Computer Between the PSTN and Microsoft® Lync™ Server 2010](#)

The gateway computer is a server with a Diva Media Board and Diva SIPcontrol installed. The use cases are based on Diva SIPcontrol version 2.5.

Use Case for Dialogic® HMP Software

This use case describes the usage of the Dialogic® Host Media Processing (HMP) software running on the same computer as the Diva Media Board and Diva SIPcontrol, as shown in the graphic below. However, Diva SIPcontrol also supports the interoperability with HMP over the LAN. The use case is based on HMP version 3.0WIN and 3.1LIN. In order for the application based on the Dialogic® Global Call API to connect with Diva SIPcontrol, it needs to be set to listen on port 5060 and to send SIP messages to the IP address 127.0.0.1 on port 9803.



For this configuration scenario, the network interface, one SIP peer, and two routes need to be configured.

To configure Diva SIPcontrol to function with your Global Call application:

1. Open the Diva SIPcontrol web interface to configure the required settings. To do so, click **Start > Programs > Dialogic Diva > SIPcontrol Configuration**.
2. In the Diva SIPcontrol web interface, click **SIPcontrol configuration** on the left hand side to open the **SIPcontrol Configuration** page.
3. Under **Network Interfaces**, enable the local loopback interface by enabling one or more listen ports, and enter **9803** for the UDP and TCP listen ports:

Network Interfaces					
Name	Device	IP address	UDP listen port	TCP listen port	TLS listen port
Intel(R) PRO1000 GT Desktop	Intel(R) PRO1000 GT Desktop Adapter - Packet Scheduler Miniport	192.168.213.38	5060 <input type="checkbox"/>	5060 <input type="checkbox"/>	5061 <input type="checkbox"/>
Local Loopback Interface	Local Loopback Interface	127.0.0.1	9803 <input checked="" type="checkbox"/>	9803 <input checked="" type="checkbox"/>	0 <input type="checkbox"/>

4. Under **SIP Peers**, click **Add new peer**, and configure the following parameters:

General	
Name:	HMP
Peer type:	Default
Host:	127.0.0.1
Port:	5060
IP protocol:	UDP
URI scheme:	SIP (default)
Domain:	

- **Name:** Enter a unique name to easily identify the SIP peer.
- **Peer type:** Leave the **Default** setting.
- **IP protocol:** Select **UDP**.

In the **Enhanced** section, enable **Default peer for received SIP calls**:

Enhanced	
Default peer for received SIP calls:	<input checked="" type="checkbox"/>
Display name to:	
Display name from:	
User name to:	
User name from:	
Gateway prefix:	
Reply-To expression:	
Reply-To format:	
Alive check:	<input type="checkbox"/> 0 seconds (0=auto)
Disconnect tone support:	<input type="checkbox"/>
Support MWI (Message waiting):	<input checked="" type="checkbox"/>
Cause code mapping inbound:	peer default
Cause code mapping outbound:	peer default
Codec profile:	default
Maximum channels:	480
Early media support:	<input checked="" type="checkbox"/>
Reliable provisional response:	Optional

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

5. Create two routes: one for each direction (SIP to PSTN and PSTN to SIP).
Configure the SIP to PSTN route. To so, open the **Routing** section, click **Add**, and configure the following parameters:

General		
Name:	SIP to HMP	
Type: Name	Source	Destination
PSTN: Controller1	<input type="checkbox"/>	Master
PSTN: Controller2	<input type="checkbox"/>	Master
SIP: HMP	<input checked="" type="checkbox"/>	-
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)	

- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select the configured SIP peer as a source.
- **Destination:** Select the controllers of the Diva Media Board as Master destination

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

Configure the SIP to PSTN route. To do so, click **Add** again, and configure the following parameters:

General		
Name:	HMP to SIP	
Type: Name	Source	Destination
PSTN: Controller1	<input checked="" type="checkbox"/>	-
PSTN: Controller2	<input checked="" type="checkbox"/>	-
SIP: HMP	<input type="checkbox"/>	Master
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)	

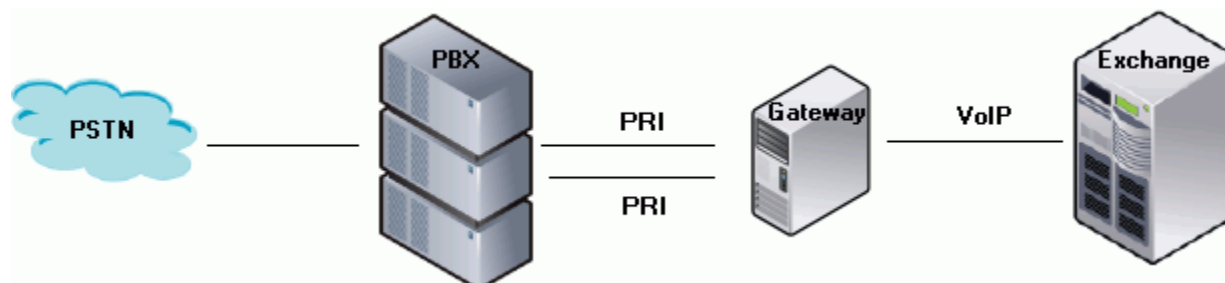
- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select the controllers of the Diva Media Board as sources.
- **Destination:** Select the configured SIP peer as a Master destination.

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

6. Click **Activate Configuration** on the main configuration page to save the settings and activate the changes.

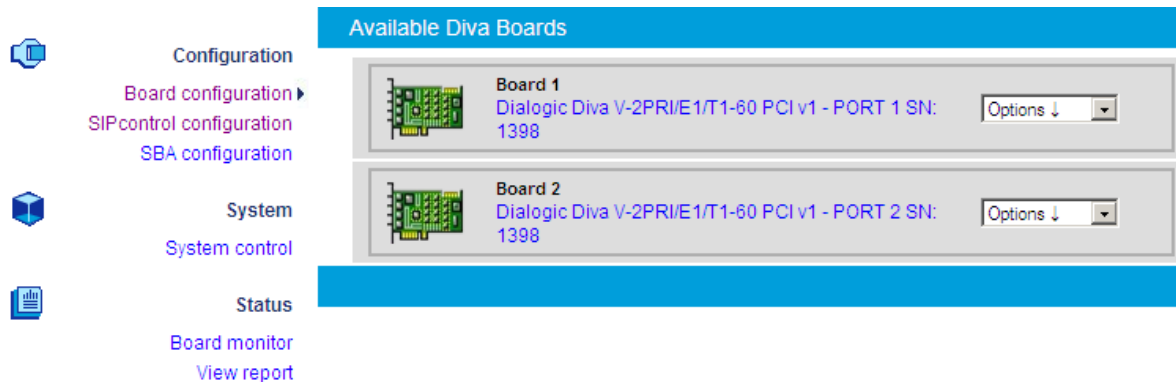
Use Case for Microsoft® Exchange Server 2007

This configuration scenario describes the necessary steps for configuring the gateway computer between the PBX and the Microsoft® Exchange Server 2007, as shown below.



For this configuration scenario, the PSTN interface, the network interface, one SIP peer, and one route need to be configured.

1. Activate the fax license as described in [License Activation](#).
2. Open the Diva SIPcontrol web interface to configure the required parameters. To do so, click **Start > Programs > Dialogic Diva > SIPcontrol Configuration**.
3. In the Diva SIPcontrol web interface, click **Board Configuration** on the left hand side to open the **Available Diva Boards** section.



Click either the board icon or the name of the Diva Media Board to access the board configuration options.

4. Click either the board icon or the name of the Diva Media Board to access the board configuration options.
5. Configure the **D-Channel Protocol** of the PBX. In this example, **PBX. QSIG E1-(QSIG)** is selected:

Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 1, SN:1398

Parameter	Value
D-Channel Protocol:	PBX, QSIG E1 - (QSIG)
Interface Mode/Resource Board:	TE - mode
Direct Dial In (NT2):	Yes
DDI Number Length:	3
DDI Collect Timeout:	0
DDI Special Number:	
Layer 1 Framing:	National default (default)
Layer 2 Connect Mode:	Permanent
Voice Companding:	Force A-Law
View Extended Configuration	No

Save Cancel

6. Click **Save**.

7. Repeat Steps 4 through 6 for the other PRI line.
8. In the Diva SIPcontrol web interface, click **SIPcontrol configuration** on the left hand side to open the **SIPcontrol Configuration** page.
9. In the **Network Interfaces** section, set the listen ports of your Ethernet adapter to **5060**, and enable the listen port by checking the associated check box:

Network Interfaces					
Name	Device	IP address	UDP listen port	TCP listen port	TLS lis
Intel(R) PRO1000 GT Desktop	Intel(R) PRO1000 GT Desktop Adapter - Packet Scheduler Miniport	192.168.213.38	5060 <input checked="" type="checkbox"/>	5060 <input checked="" type="checkbox"/>	5061
Local Loopback Interface	Local Loopback Interface	127.0.0.1	5060 <input type="checkbox"/>	5060 <input type="checkbox"/>	0

10. Configure the SIP peer settings. To do so, open the **SIP Peers** section, click **Add new peer**, and configure the following parameters:

General	
Name:	MS Exchange
Peer type:	MS Exchange 2007 ▼
Host:	IP address of UM server
Port:	5060
IP protocol:	TCP ▼
URI scheme:	SIP (default) ▼
Domain:	

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

In the **Enhanced** Section, enable **Default peer for received SIP calls**:

Enhanced

Default peer for received SIP calls:	<input checked="" type="checkbox"/>
Display name to:	<input type="text"/>
Display name from:	<input type="text"/>
User name to:	<input type="text"/>
User name from:	<input type="text"/>
Gateway prefix:	<input type="text"/>
Reply-To expression:	<input type="text"/>
Reply-To format:	<input type="text"/>
Alive check:	<input type="checkbox"/> <input type="text" value="0"/> seconds (0=auto)
Disconnect tone support:	<input type="checkbox"/>
Support MWI (Message waiting):	<input checked="" type="checkbox"/>
Cause code mapping inbound:	peer default <input type="button" value="v"/>
Cause code mapping outbound:	peer default <input type="button" value="v"/>
Codec profile:	default <input type="button" value="v"/>
Maximum channels:	<input type="text" value="480"/>
Early media support:	<input checked="" type="checkbox"/>
Reliable provisional response:	Optional <input type="button" value="v"/>

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

11. Configure one PSTN to SIP route and one SIP to the PSTN route.

For the PSTN to SIP route, Click **Routing**, click **Add**, and configure the following parameters:

General		
Name:	<input type="text" value="PSTN-SIP-Exchange"/>	
Type: Name	Source	Destin
PSTN: Controller1	<input checked="" type="checkbox"/>	<input type="text" value="-"/>
PSTN: Controller2	<input checked="" type="checkbox"/>	<input type="text" value="-"/>
SIP: MS-Exchange	<input type="checkbox"/>	<input type="text" value="Master"/>
Max. call attempts for this route in a failover scenario:	<input type="text" value="0"/> (0 = try all selected destinations)	

- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select both controllers as sources.
- **Destination:** Select the SIP peer configured above as a Master destination.

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

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

General		
Name:	SIP-PTN-Exchange	
Type: Name	Source	Destin
PSTN: Controller1	<input type="checkbox"/>	Master
PSTN: Controller2	<input type="checkbox"/>	Master
SIP: MS-Exchange	<input checked="" type="checkbox"/>	-
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)	

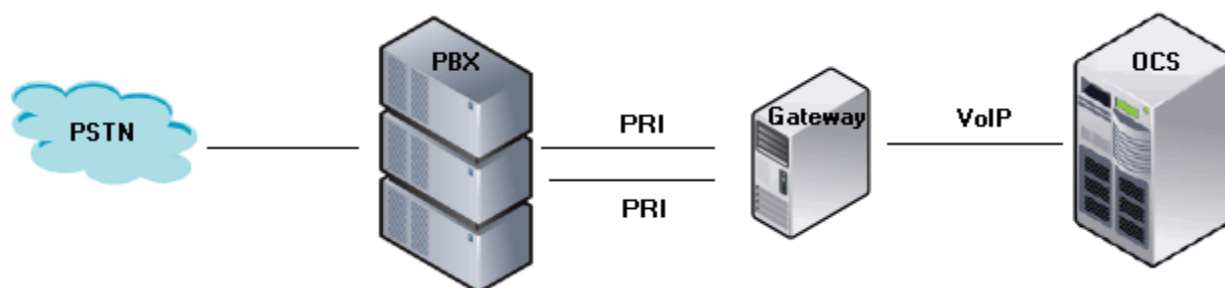
- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select the SIP peer configured above as a source.
- **Destination:** Select both controllers as Master destinations.

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

- Click **Activate Configuration** in the main configuration page to save the settings and activate the changes.

Using the Gateway Computer between the PBX and Microsoft® Office Communications Server 2007

This configuration scenario describes the necessary steps for configuring a gateway computer that has both lines line connected to the PBX. In this scenario, the gateway computer is connected to Microsoft Office Communications Server 2007 via VoIP:



For this configuration scenario, one dialplan, the PSTN interface, the network interface, one SIP peer, and two routes need to be configured.

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

HOME Board Configuration

Configuration

- Board configuration ▶
- SIPcontrol configuration
- SBA configuration

System

- System control

Status

- Board monitor
- View report

Licensing

- License management

Available Diva Boards

Board 1
Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 1 SN: 1033

Board 2
Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 2 SN: 1033

- Click either the board icon or the name of the Diva Media Board to open the **Board Configuration - Detail** page.
- Configure the **D-Channel Protocol** of the PBX. In the example, **PBX.Q.SIG E1-(QSIG)** is selected.

Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 1, SN:1398

Parameter	Value
D-Channel Protocol:	PBX, QSIG E1 - (QSIG)
Interface Mode/Resource Board:	TE - mode
Direct Dial In (NT2):	Yes
DDI Number Length:	3
DDI Collect Timeout:	0
DDI Special Number:	
Layer 1 Framing:	National default (default)
Layer 2 Connect Mode:	Permanent
Voice Companding:	Force A-Law
View Extended Configuration	No

Save Cancel

- Click **Save**.
- Repeat Steps 3 through 5 for the other PRI line.
- In the Diva SIPcontrol web interface, click **SIPcontrol configuration** on the left hand side to open the **SIPcontrol Configuration** page.

8. Under **Dialplans**, click **Add**, and enter the following parameters:

General	
Name:	Dialplan-PBX
Country code:	49
North-American numbering plan:	<input type="checkbox"/>
Area code:	7159 With national prefix
Other local areas:	<input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/>
Base number:	4066
Maximum extension digits:	4
International prefix:	00
National prefix:	0
Access code:	0
PSTN access code provided by SIP caller:	<input checked="" type="checkbox"/>
Incoming PSTN access code provided by PBX:	<input checked="" type="checkbox"/>
Keep SIP URI Domain and Parameters:	<input type="checkbox"/>
<input type="button" value="OK"/> <input type="button" value="Cancel"/>	

- **Name:** Enter a unique name to easily identify the dialplan.
- **Country code:** Enter the country code of the country in which the gateway computer is located.
- **Area code:** Enter the area code of the region in which the gateway computer 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 gateway computer is located.
- **National prefix:** Enter the national prefix that needs to be dialed for long distance calls within the country in which the gateway computer 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 close the window.

- Under **PSTN Interfaces**, configure the first Diva Media Board line. To do so, click **Details** at the right and configure the following parameters:

General	
Hardware description:	Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 1 SN:
PSTN interface number:	1
Name:	Controller1
Address map inbound:	none
Address map outbound:	none
Enhanced	
Address Normalization	
Dialplan:	Dialplan-PBX
Type of number (outbound):	Extension
Encoding (outbound):	Use prefixes
ISDN numbering plan - Default:	unknown
Presentation indicator - Default:	Allowed
Internal interface:	<input type="checkbox"/>

- **Dialplan:** Select the configured dialplan.
- **Type of number (outbound):** Select Extension.
- **Encoding (outbound):** Select Use prefixes.
- Leave the remaining parameters at their default values.

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

- Configure the second line with the same settings as the first line.
- Under **Network Interfaces**, enable your Ethernet adapter, and set the SIP listen ports to **9803**.

Network Interfaces					
Name	Device	IP address	UDP listen port	TCP listen port	TLS listen port
intel(R) PRO1000 EB Network	Intel(R) PRO/1000 EB Network Connection with I/O Acceleration #2	172.16.22.144	9803 <input checked="" type="checkbox"/>	9803 <input checked="" type="checkbox"/>	
intel(R) PRO1000 EB Network1	Intel(R) PRO/1000 EB Network Connection with I/O Acceleration	10.2/2.202.134	<input type="checkbox"/>	<input type="checkbox"/>	
Local Loopback Interface	Local Loopback Interface	127.0.0.1	<input type="checkbox"/>	<input type="checkbox"/>	
RTP start port:	30000				
RTP end port:	39999				

12. Under **SIP Peers**, click **Add new peer**, and configure the following parameters:

General	
Name:	OCS-Mediation-Server
Peer type:	MS OCS 2007/2007 R2 - Mediation Server
Host:	172.16.22.144
Port:	5060
IP protocol:	TCP
URI scheme:	SIP (default)
Domain:	default routing domain of OCS

- **Name:** Enter a unique name to easily identify the SIP peer.
- **Peer type:** Select **MS OCS 2007/ R2 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 close the window.

13. Create one PSTN to SIP route and one SIP to PSTN route. To do so, click **Routing** and click **Add** to open the routing options.

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

General

Name:	PSTN-to-SIP	
Type: Name	Source	Destination
PSTN: Controller1	<input checked="" type="checkbox"/>	.
PSTN: Controller2	<input checked="" type="checkbox"/>	.
SIP: OCS-Mediation-Server	<input type="checkbox"/>	Master
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)	

Address Normalization For Condition Processing (Using Source Dialplan)

Disable inbound and outbound dialplan:	<input type="checkbox"/>
Number format:	International number
Encoding:	Use type flag

- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select both controllers of the Diva Media Board as sources.
- **Destination:** Select the above configured SIP peer as a Master destination.

Under **Address Normalization for Condition Processing (Using Source Dialplan)**, configure the following parameters:

- **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 close the window.

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

General

Name:	SIP-to-PSTN	
Type: Name	Source	Destination
PSTN: Controller1	<input type="checkbox"/>	Master ▼
PSTN: Controller2	<input type="checkbox"/>	Master ▼
SIP: OCS-Mediation-Server	<input checked="" type="checkbox"/>	- ▼
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)	

Address Normalization For Condition Processing (Using Source Dialplan)

Disable inbound and outbound dialplan:	<input type="checkbox"/>
Number format:	Unchanged ▼
Encoding:	Use prefixes ▼

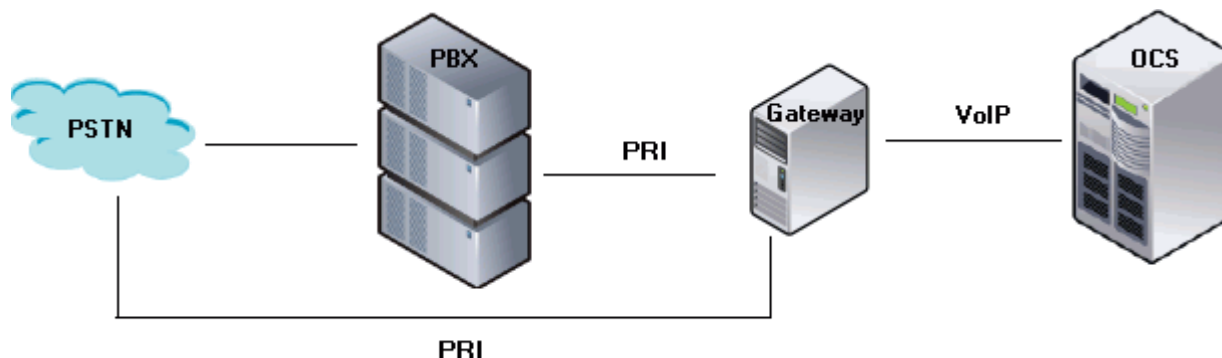
- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select the configured SIP peer as a source.
- **Destination:** Select both controllers of the Diva Media Board as Master destinations.

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

- Click **Activate Configuration** in the main configuration page to save the settings and to activate the changes.
- Configure Microsoft Office Communications Server 2007

Using the Gateway Computer Between the PBX/PSTN and Microsoft® Office Communications Server 2007

This configuration scenario describes the necessary steps for configuring a gateway computer that has one line connected to the PBX and the other line connected directly to the PSTN. In this scenario, the gateway computer is connected to the Microsoft® Office Communications Server 2007 via VoIP:



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

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



3. Click either the board icon or the name of the first Diva Media Board to access the board configuration options.
4. Configure the **D-Channel Protocol** of port 1. In the example, **PBX. QSIG E1 - (Q.SIG)** is selected:

Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 1, SN:1398	
Parameter	Value
D-Channel Protocol:	PBX, QSIG E1 - (QSIG)
Interface Mode/Resource Board:	TE - mode
Direct Dial In (NT2):	Yes
DDI Number Length:	3
DDI Collect Timeout:	0
DDI Special Number:	
Layer 1 Framing:	National default (default)
Layer 2 Connect Mode:	Permanent
Voice Companding:	Force A-Law
View Extended Configuration	No

5. Click **Save**.
6. Click **Board configuration** again, and select port 2 of your Diva Media Board to access the board configuration options.
7. Configure the **D-Channel Protocol** of the PRI line connected to the PSTN. In the example, **Europe/other countries, Euro-ISDN (ETSI-DSS1)-(ETSI)** is selected:

Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 2, SN:1398

Parameter	Value
D-Channel Protocol:	Europe/other countries, Euro-ISDN (ETSI-DSS1) - (ETSI)
Interface Mode/Resource Board:	TE - mode
Direct Dial In (NT2):	Yes
DDI Number Length:	3
DDI Collect Timeout:	0
DDI Special Number:	
Layer 1 Framing:	National default (default)
Layer 2 Connect Mode:	Permanent
Voice Companding:	Force A-Law
View Extended Configuration	No

Save Cancel

8. Click **Save**.
9. In the Diva SIPcontrol web interface, click **SIPcontrol configuration** on the left hand side to open the **SIPcontrol Configuration** page.
10. Configure two dialplans; one for the line connected to the PBX and one for the line connected directly to the PSTN.

To create the dialplan for the line connected to the PBX, open the **Dialplans** section, click **Add**, and configure the following parameters:

General	
Name:	Dialplan-at-PBX
Country code:	49
North-American numbering plan:	<input type="checkbox"/>
Area code:	7159 With national prefix
Other local areas:	<input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/>
Base number:	4066
Maximum extension digits:	4
International prefix:	00
National prefix:	0
Access code:	0
PSTN access code provided by SIP caller:	<input checked="" type="checkbox"/>
Incoming PSTN access code provided by PBX:	<input checked="" type="checkbox"/>
Keep SIP URI Domain and Parameters:	<input type="checkbox"/>
<input type="button" value="OK"/> <input type="button" value="Cancel"/>	

- **Name:** Enter a unique name to easily identify the dialplan.
- **Country code:** Enter the country code of the country in which the gateway computer is located.
- **Area code:** Enter the area code of the region in which the gateway computer 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 gateway computer is located.

- **National prefix:** Enter the national prefix that needs to be dialed for long distance calls within the country in which the gateway computer 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 close the window.

To configure the dialplan for the line connected directly to the PSTN, click **Add**, and configure the following parameters. I changed the screenshot, because it had an “nn” in the International prefix field, which is not allowed.

General	
Name:	Dialplan-at-PSTN
Country code:	49
North-American numbering plan:	<input type="checkbox"/>
Area code:	7159 With national prefix ▼
Other local areas:	<input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/>
Base number:	4066
Maximum extension digits:	4 ▼
International prefix:	00
National prefix:	0
Access code:	0
PSTN access code provided by SIP caller:	<input type="checkbox"/>
Incoming PSTN access code provided by PBX:	<input type="checkbox"/>
Keep SIP URI Domain and Parameters:	<input type="checkbox"/>
<input type="button" value="OK"/> <input type="button" value="Cancel"/>	

- **Name:** Enter a unique name to easily identify the dialplan.
- **Country code:** Enter the country code of the country in which the gateway computer is located.
- **Area code:** Enter the area code of the region in which the gateway computer 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 gateway computer is located.
- **National prefix:** Enter the national prefix that needs to be dialed for long distance calls within the country in which the gateway computer is located.

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

11. Under **PSTN Interfaces**, 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 on which you configured the switch type of your PBX. Similarly, the controller number that you configure with the PSTN-specific settings needs to correspond to the line number in the Diva Configuration Manager on which you configured the switch type of your PSTN line.

To configure the PBX-specific parameters, click **Details** at the right of the controller connected to the PBX. and configure the following parameters:

General	
Hardware description:	Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 1 SN: 1
PSTN interface number:	1
Name:	Controller-to-PBX
Address map inbound:	none
Address map outbound:	none

Enhanced

Address Normalization	
Dialplan:	Dialplan-at-PBX
Type of number (outbound):	Extension
Encoding (outbound):	Use prefixes
ISDN numbering plan - Default:	unknown
Presentation indicator - Default:	Allowed
Internal interface:	<input type="checkbox"/>

PSTN Call Transfer Settings

Message Waiting Indication (MWI)

OK Cancel

- **Dialplan:** Select the dialplan you configured for the PBX.
- **Type of number (outbound):** Select **Extension**.
- **Encoding (outbound):** Select **Use prefixes**.

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

To configure the PSTN-specific parameters, click **Details** at the right of the controller connected to the PSTN line, and configure the following parameters:

General	
Hardware description:	Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 2 SN: 1
PSTN interface number:	2
Name:	Controller-to-PSTN
Address map inbound:	none
Address map outbound:	none
Enhanced	
Address Normalization	
Dialplan:	Dialplan-at-PSTN
Type of number (outbound):	National number
Encoding (outbound):	Use prefixes
ISDN numbering plan - Default:	unknown
Presentation indicator - Default:	Allowed
Internal interface:	<input type="checkbox"/>
PSTN Call Transfer Settings	
Message Waiting Indication (MWI)	
<input type="button" value="OK"/> <input type="button" value="Cancel"/>	

- **Dialplan:** Select the dialplan you configured for the PSTN
- **Type of number (outbound):** Select **National number**.
- **Encoding (outbound):** Select **Use prefixes**.

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

- Under **Network Interfaces**, enable your Ethernet adapter, and set the SIP listen ports to **9803**:

Network Interfaces					
Name	Device	IP address	UDP listen port	TCP listen port	TLS listen port
intel(R) PRO1000 EB Network	Intel(R) PRO/1000 EB Network Connection with I/O Acceleration #2	172.16.22.144	9803 <input checked="" type="checkbox"/>	9803 <input checked="" type="checkbox"/>	
intel(R) PRO1000 EB Network1	Intel(R) PRO/1000 EB Network Connection with I/O Acceleration	10.2/2.202.134	<input type="text"/> <input type="checkbox"/>	<input type="text"/> <input type="checkbox"/>	
Local Loopback Interface	Local Loopback Interface	127.0.0.1	<input type="text"/> <input type="checkbox"/>	<input type="text"/> <input type="checkbox"/>	
RTP start port:	<input type="text" value="30000"/>				
RTP end port:	<input type="text" value="30000"/>				

13. Under **SIP Peers**, click **Add new peer**, and configure the following parameters:

General	
Name:	<input type="text" value="OCS-Mediation-Server"/>
Peer type:	<input type="text" value="MS OCS 2007/2007 R2 - Mediation Server"/>
Host:	<input type="text" value="172.16.22.144"/>
Port:	<input type="text" value="5060"/>
IP protocol:	<input type="text" value="TCP"/>
URI scheme:	<input type="text" value="SIP (default)"/>
Domain:	<input type="text" value="default routing domain of OCS"/>
Enhanced	
Security	
Session Timer	
Address Normalization	
Dialplan:	<input type="text" value="Dialplan-at-PBX"/>
Number format (outbound):	<input type="text" value="Unchanged"/>
Encoding (outbound):	<input type="text" value="Use prefixes"/>
Address map inbound:	<input type="text" value="none"/>
Address map outbound:	<input type="text" value="none"/>
Authentication	
<input type="button" value="OK"/> <input type="button" value="Cancel"/>	

- **Name:** Enter a unique name to easily identify the SIP peer.
- **Peer type:** Select **MS OCS 2007/2007 R2 - 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**, select the dialplan you configured for the controller connected to the PBX.

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

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

General		
Address map name:	<input type="text" value="SIPto-PSTN-Address-Map"/>	
Rule name:	<input type="text" value="Remove Outside Access Digit"/>	
Stop on match:	<input type="checkbox"/>	
Enhanced configuration:	<input checked="" type="checkbox"/>	

Called address rules		
	Condition	Result
Address:	<input type="text" value="^0"/>	<input type="text"/>
Name:	<input type="text"/>	<input type="text"/>
Number type:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> International <input checked="" type="checkbox"/> National <input checked="" type="checkbox"/> Network specific <input checked="" type="checkbox"/> Subscriber <input checked="" type="checkbox"/> Abbreviated	<input type="text" value="No change"/>
Numbering plan:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> ISDN/telephony E.164 <input checked="" type="checkbox"/> National standard <input checked="" type="checkbox"/> Private	<input type="text" value="No change"/>

Calling address rules
Redirect address rules

<input type="button" value="OK"/> <input type="button" value="Cancel"/>

- **Address map name:** Enter a descriptive name for the address map.
- **Rule name:** Enter a name that explains the address map rule.
- **Address Condition** in the **Called address rules** section: Enter the expression **^0** to remove the outside access digit.

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

15. Create three routes:

- Route from Microsoft® Office Communications Server 2007 directly to the PSTN
- Route from Microsoft® Office Communications Server 2007 via the PBX to the PSTN
- Route from the PSTN/PBX via the gateway computer to Microsoft® Office Communications Server 2007

See [Creating the Routes for This Scenario](#) for instructions.

In this scenario, the order of the routes is important, because only one route will be configured with a condition. To change the order of the routes 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:

Routing						
Name	Sources	Destinations	Address map	Enabled		
SIP-to-PSTN	OCS-Mediation-Server	Controller-to-PSTN (Master)	SIP-to-PSTN-Address-Map	<input checked="" type="checkbox"/>	Up	Down Details
SIP-to-PBX	OCS-Mediation-Server	Controller-to-PBX (Master)	none	<input checked="" type="checkbox"/>	Up	Down Details
PSTN-and-PBX-to-SIP	Controller-to-PBX, Controller-to-PSTN	OCS-Mediation-Server (Master)	none	<input checked="" type="checkbox"/>	Up	Down Details
Add						

16. Click **Activate Configuration** in the main configuration page to save the settings and activate the changes.

17. Configure Microsoft® Office Communications Server 2007.

Creating the Routes for This Scenario

Create the routes in the following order:

1. Create the route from Microsoft® Office Communications Server 2007 to the PSTN first. To do so, open the **Routing** section, click **Add** and configure the following parameters:

General					
Name:	SIPto-PSTN				
Type: Name	Source	Destination			
PSTN: Controller-to-PBX	<input type="checkbox"/>	-			
PSTN: Controller-to-PSTN	<input type="checkbox"/>	Master			
SIP: OCS-Mediation-Server	<input checked="" type="checkbox"/>	-			
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)				

Address Normalization For Condition Processing (Using Source Dialplan)	
Disable inbound and outbound dialplan:	<input type="checkbox"/>
Number format:	Extension
Encoding:	Use prefixes

Conditions					
Extended Condition	Item	Called	Calling	Redirect	Action
<input type="checkbox"/>	Address	^0			delete
Add new condition					

Address Manipulation	
Address map:	SIPto-PSTN-Address-Map

- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select the SIP peer configured above as a Master destination.
- **Destination:** Select the controller you configured for the PSTN.

Under Address Normalization for Condition Processing (Using Source Dialplan), configure the following parameters:

- **Number format:** Select **Extension** from the dropdown menu.
- **Encoding:** Select **Use prefixes** from the dropdown menu.

Under **Conditions**, click **Add new condition**, and enter **^0** in the **Called number** field to omit the outside access digit.

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

2. Configure the route from Microsoft® Office Communications Server 2007 to the PBX. To do so, click **Add** and configure the following parameters:

General		
Name:	SIP-to-PBX	
Type: Name	Source	Destination
PSTN: Controller-to-PBX	<input type="checkbox"/>	Master
PSTN: Controller-to-PSTN	<input type="checkbox"/>	.
SIP: OCS-Mediation-Server	<input checked="" type="checkbox"/>	.
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)	
Address Normalization For Condition Processing (Using Source Dialplan)		
Disable inbound and outbound dialplan:	<input type="checkbox"/>	
Number format:	Unchanged	
Encoding:	Use type flag	
Conditions		
Address Manipulation		
<input type="button" value="OK"/> <input type="button" value="Cancel"/>		

- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select the controller you configured for the PBX.
- **Destination:** Select the configured SIP peer as a Master destination.

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

3. Create the route from the PSTN/PBX to Microsoft® Office Communications Server 2007. To do so, click **Add**, and Configure the following parameters:

General					
Name:		PSTN-and-PBXto-SIP			
Type: Name	Source	Destination			
PSTN: Controller-to-PBX	<input checked="" type="checkbox"/>	-			
PSTN: Controller-to-PSTN	<input checked="" type="checkbox"/>	-			
SIP: OCS-Mediation-Server	<input type="checkbox"/>	Master			
Max. call attempts for this route in a failover scenario:		0 (0 = try all selected destinations)			

Address Normalization For Condition Processing (Using Source Dialplan)	
Disable inbound and outbound dialplan:	<input type="checkbox"/>
Number format:	International number
Encoding:	Use type flag

Conditions					
Extended Condition	Item	Called	Calling	Redirect	Action
<input type="checkbox"/>	Address				delete
Add new condition					

Address Manipulation	
Address map:	none

- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select both controllers of the Diva Media Board.
- **Destination:** Select the SIP peer configured above as a Master destination.

Under **Address Normalization for Condition Processing (Using Source Dialplan)**, configure the following parameters:

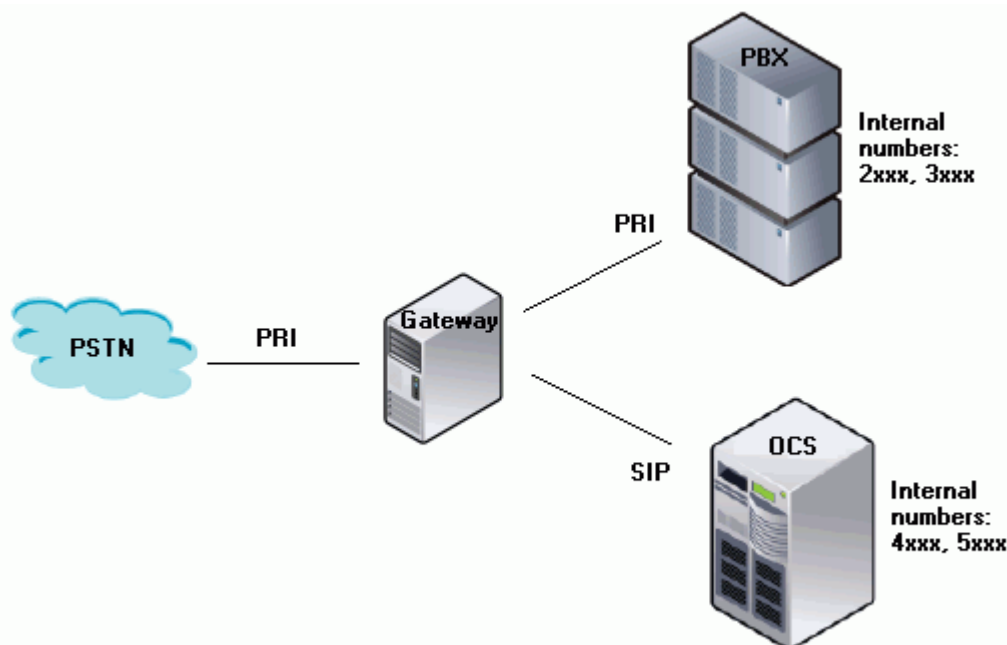
- **Number format:** Select **International number** from the dropdown menu.
- **Encoding:** Select **Use type flag** from the dropdown menu.

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

Using the Gateway Computer Between the PSTN and PBX/Microsoft® Office Communications Server 2007

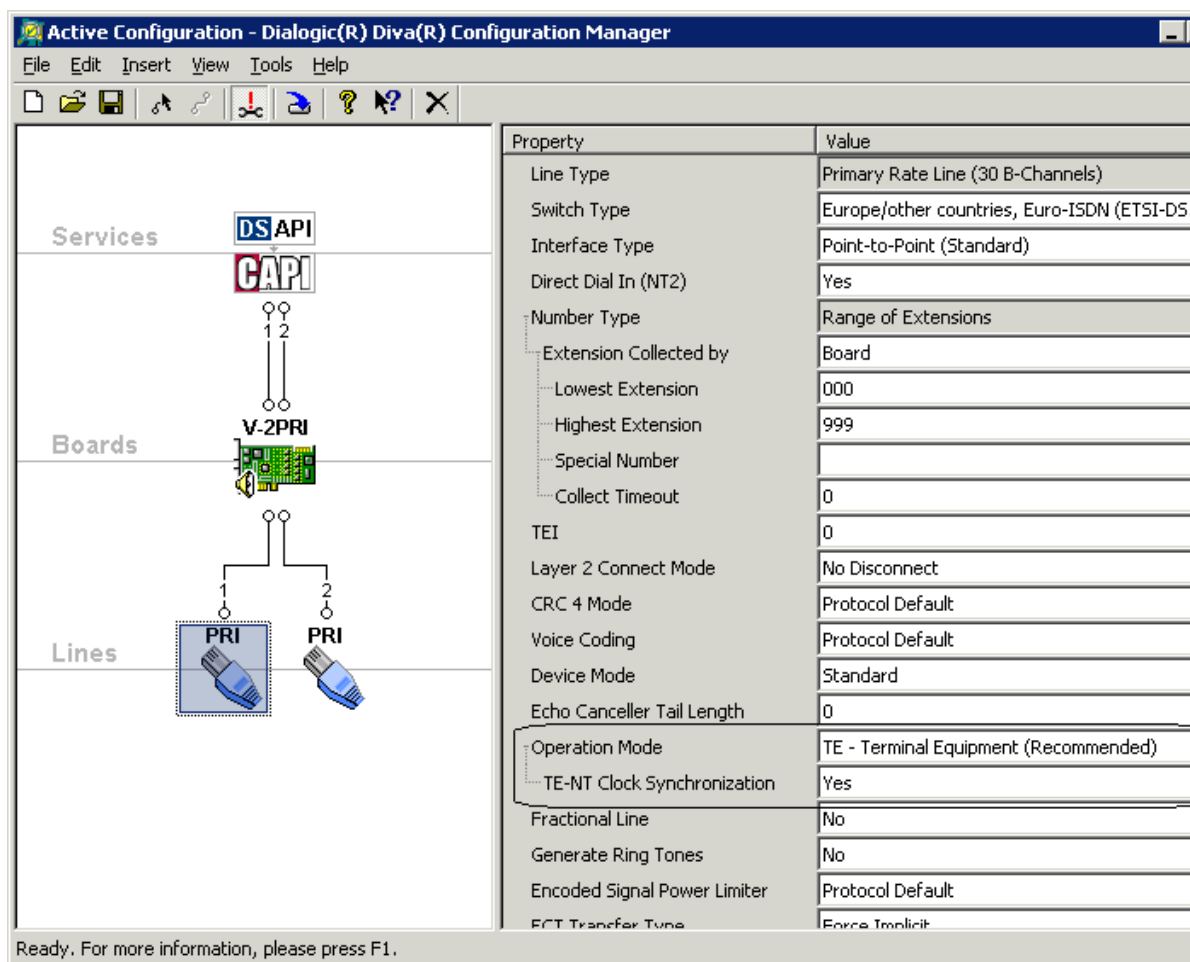
This configuration scenario describes the necessary steps if the gateway computer is connected between the PSTN and PBX/Microsoft® Office Communications Server 2007. This way, Diva SIPcontrol can also route calls from the PBX to the PSTN, and vice versa. One PRI line is connected to the PBX and one PRI line is connected directly to the PSTN. This scenario also assumes that the PBX was previously connected to the PSTN directly and the PBX has not been reconfigured at all to cope with any changes introduced by the gateway or Microsoft® Office Communications Server 2007.

The Microsoft® Mediation Server is installed on the gateway computer. The PBX is configured for the extensions starting with 2 or 3, and the Microsoft® Office Communications Server 2007 is configured for the extensions starting with 4 or 5. Diva SIPcontrol expects a PSTN access code in calls from Microsoft® Office Communications Server 2007 to the PSTN or from the PBX to the PSTN; in the following example, it is an additional 0 (zero). Diva SIPcontrol is configured to remove the access code before forwarding the call to the PSTN. For convenience, Diva SIPcontrol is also configured to add the outside access code to the calling number in calls coming from the PSTN. Since the PBX is not aware of the presence of the gateway and Microsoft® Office Communications Server 2007, it does not send or expect to receive any outside access code. Therefore, Diva SIPcontrol also removes the outside access code in calls to the PBX and adds it in calls from PBX.



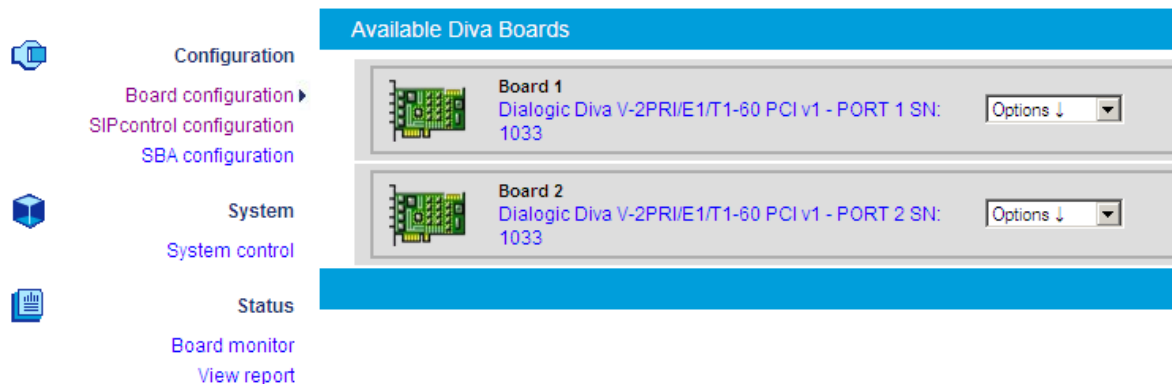
For this configuration scenario, one dialplan, five address maps, the two PSTN interfaces, the network interface, two SIP peers, and five routes need to be configured.

1. Open the Diva Configuration Manager. To do so, click **Start > Programs > Dialogic Diva > Configuration Manager**.
2. Click the line icon for port one and under **Operation Mode** set **TE-NT Clock Synchronization** to **Yes**:



Open the Diva web interface to configure the required parameters. To do so, click **Start > Programs > Dialogic Diva > SIPcontrol Configuration**.

- Open the Diva web interface to configure the required parameters. To do so, click **Start > Programs > Dialogic Diva > SIPcontrol Configuration**.
- In the Diva 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 Diva Media Board line (the PRI line connected to the PSTN) to open the **Board Configuration - Detail** page, and then configure the following parameters:

Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 1, SN:1398

Parameter	Value
D-Channel Protocol:	Europe/other countries, Euro-ISDN (ETSI-DSS1) - (ETSI)
Interface Mode/Resource Board:	TE - mode
Direct Dial In (NT2):	Yes
DDI Number Length:	3
DDI Collect Timeout:	0
DDI Special Number:	
Layer 1 Framing:	National default (default)
Layer 2 Connect Mode:	No disconnect
Voice Companding:	Protocol default
View Extended Configuration	No

Save Cancel

- D-Channel Protocol:** Select **Europe/other countries, Euro-ISDN (ETSI-DSS1) - (ETSI)** as the D-channel protocol.
- DDI Number Length:** Set the value to **3**.

Click **Save** to close the window.

- Click **Board configuration** again and select Port 2 of your Diva Media Board to open the **Board Configuration - Detail** page for this board and configure the following parameters:

Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 2, SN:1398	
Parameter	Value
D-Channel Protocol:	Europe/other countries, Euro-ISDN (ETSI-DSS1) - (ETSI)
Interface Mode/Resource Board:	NT - mode
Direct Dial In (NT2):	Yes
DDI Number Length:	20
DDI Collect Timeout:	3
DDI Special Number:	
Layer 1 Framing:	National default (default)
Layer 2 Connect Mode:	No disconnect
Voice Companding:	Protocol default
View Extended Configuration	No

- **D-Channel Protocol:** Select **Europe/other countries, Euro-ISDN (ETSI-DSS1) - (ETSI)** as the D-channel protocol.
- **Interface Mode/Resource Board:** Select **NT-mode**.
- **DDI Number Length:** Select **20** to cover the possible length of the called number in an outgoing call.
- **DDI Collect Timeout:** Select a timeout in seconds after which the Diva Board stops collecting the digits of the calling number and passes the number to the application. In the example, the timeout value is **3** seconds.

Click **Save** to close the window.

7. In the Diva web interface, click **SIPcontrol configuration** on the left hand side to open the **SIPcontrol Configuration** page.
8. Open the **Dialplans** section, click **Add**, and configure the following parameters for the dialplan:

General	
Name:	Dialplan-PSTN
Country code:	49
North-American numbering plan:	<input type="checkbox"/>
Area code:	7159 With national prefix
Other local areas:	<input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/>
Base number:	4066
Maximum extension digits:	4
International prefix:	00
National prefix:	0
Access code:	0
PSTN access code provided by SIP caller:	<input checked="" type="checkbox"/>
Incoming PSTN access code provided by PBX:	<input checked="" type="checkbox"/>
Keep SIP URI Domain and Parameters:	<input type="checkbox"/>
<input type="button" value="OK"/> <input type="button" value="Cancel"/>	

- **Name:** Enter a unique name to easily identify the dialplan.
- **Country code:** Enter the country code of the country in which the gateway computer is located.
- **Area code:** Enter the area code of the region in which the gateway computer 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 gateway computer is located.

- **National prefix:** Enter the national prefix that needs to be dialed for long distance calls within the country in which the gateway computer 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 close the window.

9. Create five address maps: two for incoming calls, two for outgoing calls, and one for the special number, which is normally the reception's number. The first four address maps are necessary for either adding or omitting the OAD (Outside Access Digit). See [Creating Address Maps for This Scenario](#) for instructions.
10. Under **PSTN Interfaces**, configure Controller 1 with PSTN-specific settings and Controller 2 with PBX-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 on which you configured the switch type of your PBX. Similarly, the controller number that you configure with the PSTN-specific settings needs to correspond to the line number in the Diva Configuration Manager on which you configured the switch type of your PSTN line.

To configure PSTN-specific parameters, click **Details** at the right of the controller connected to the PSTN line, and configure the following parameters:

General	
Hardware description:	Dialogic Diva V-2PRI/E1/T1 - PORT 1 SN: 1033
PSTN interface number:	1
Name:	Controller-to-PSTN
Address map inbound:	PSTN-Access-Inbound
Address map outbound:	PSTN-Access-Outbound

Enhanced	
Address Normalization	
Dialplan:	Dialplan-PSTN
Number format (outbound):	National number
Encoding (outbound):	Use prefixes
Default numbering plan:	unknown
Default presentation indicator:	Allowed
Internal interface:	<input type="checkbox"/>

PSTN Call Transfer Settings	
Message Waiting Indication (MWI)	

OK Cancel

- **Name:** Enter a unique name for this controller.
- **Address map inbound:** Select the inbound address map that you configured for access from the PSTN.
- **Address map outbound:** Select the outbound address map that you configured for access to the PSTN.

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

- **Dialplan:** Select the dialplan you configured for the PSTN.
- **Number format (outbound):** Select **National number** from the dropdown menu.
- **Encoding (outbound):** Select **Use prefixes** from the dropdown menu.

Leave the remaining parameters at their default values.

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

To configure the controller that is connected to the PBX later in the configuration, click **Details** at the right of the respective controller, and configure the following parameters:

General	
Hardware description:	Dialogic Diva V-2PRI/E1/T1 - PORT 2 SN: 1033
PSTN interface number:	2
Name:	Controller-to-PBX
Address map inbound:	From-PBX
Address map outbound:	To-PBX

Enhanced	
Address Normalization	
Dialplan:	Dialplan-PSTN
Number format (outbound):	National number
Encoding (outbound):	Use prefixes
Default numbering plan:	unknown
Default presentation indicator:	Allowed
Internal interface:	<input checked="" type="checkbox"/>

PSTN Call Transfer Settings	
Message Waiting Indication (MWI)	

- **Name:** Enter a unique name for this controller.
- **Address map inbound:** Select the inbound address map that you configured for access from the PBX.
- **Address map outbound:** Select the outbound address map that you configured for access to the PBX.

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

- **Dialplan:** Select the dialplan you configured for the PSTN.
- **Number format (outbound):** Select **National number** from the dropdown menu.
- **Encoding (outbound):** Select **Use prefixes** from the dropdown menu.
- **Internal Interface:** This option must be activated, because this interface connects to internal devices only. (It does not directly connected to the PSTN.)

Leave the remaining parameters at their default values.

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

11. Under **Network Interfaces**, enable your Ethernet adapter, and set the SIP listen ports to **9803**:

Network Interfaces						
Name	Device	IP address	UDP listen port	TCP listen port	TLS list	
Intel(R) PRO1000 GT Desktop	Intel(R) PRO1000 GT Desktop Adapter - Packet Scheduler Miniport	192.168.213.38	9803 <input checked="" type="checkbox"/>	9803 <input checked="" type="checkbox"/>	5061	
Local Loopback Interface	Local Loopback Interface	127.0.0.1	5060 <input type="checkbox"/>	5060 <input type="checkbox"/>	0	

12. Create a SIP peer for the Microsoft® Mediation Server installed on the gateway computer. To configure this SIP peer, open the **SIP Peers** section, click **Add**, and enter the following parameters:

General

Name:

OCS-Mediation-Server

Peer type:

MS OCS 2007/2007 R2 - Mediation Server

Host:

192.168.212.136

Port:

5060

IP protocol:

TCP

URI scheme:

SIP (default)

Domain:

ocs-name.ad-domain.tld

Enhanced

Security

Session Timer

Address Normalization

Dialplan:

Dialplan-PSTN

Number format (outbound):

International number

Encoding (outbound):

Use type flag

Address map inbound:

none

Address map outbound:

none

Authentication

OK

Cancel

- **Name:** Enter a unique name to easily identify the SIP peer.
- **Peer type:** Select **MS OCS 2007/2007 R2 - 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**, configure the following parameters:

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

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

13. Create the following routes:

- PSTN to Microsoft® Office Communications Server 2007
- Microsoft® Office Communications Server 2007 to the PBX
- Microsoft® Office Communications Server 2007 to the PSTN
- A route for calls to the reception

See [Creating Routes for This Scenario](#) for instructions.

In this scenario, the order of the routes is important because of the used conditions.

Creating Address Maps for This Scenario

1. To create the first address map for incoming calls from the PSTN, open the **Address Maps** section, click **Add**, and configure the following parameters:

General

Address map name:	<input type="text" value="PSTN-Access-Inbound"/>
Rule name:	<input type="text" value="Add OAD in Calling Number"/>
Stop on match:	<input type="checkbox"/>
Enhanced configuration:	<input checked="" type="checkbox"/>

Called address rules

Calling address rules

	Condition	Result
Address:	<input type="text" value="^[0-9].*"/>	<input type="text" value="0\$&"/>
Name:	<input type="text"/>	<input type="text"/>
Number type:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> International <input checked="" type="checkbox"/> National <input checked="" type="checkbox"/> Network specific <input checked="" type="checkbox"/> Subscriber <input checked="" type="checkbox"/> Abbreviated	<input type="text" value="No change"/>
Numbering plan:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> ISDN/telephony E.164 <input checked="" type="checkbox"/> National standard <input checked="" type="checkbox"/> Private	<input type="text" value="No change"/>
Presentation:	<input checked="" type="checkbox"/> Allowed <input checked="" type="checkbox"/> Restricted <input checked="" type="checkbox"/> Number not available <input checked="" type="checkbox"/> Undefined	<input type="text" value="No change"/>

Redirect address rules

- **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.

- **Address Condition** in the **Calling address rules** section: Enter **^[0-9].***. This expression matches all calls from the PSTN.
- **Address Result** in the Calling address rules section: Enter the 0 (zero), so that it is added in the output, followed by the **\$&**, which adds the incoming number after the **0**.

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

2. To create the second address map for outgoing calls to the PSTN, click **Add** again, and configure the following parameters:

General		
Address map name:	PSTN-Access-Outbound	
Rule name:	Remove OAD in Called Number	
Stop on match:	<input type="checkbox"/>	
Enhanced configuration:	<input type="checkbox"/>	

Called address rules		
Address:	^0	
Name:		
Number type:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> International <input checked="" type="checkbox"/> National <input checked="" type="checkbox"/> Network specific <input checked="" type="checkbox"/> Subscriber <input checked="" type="checkbox"/> Abbreviated	No change
Numbering plan:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> ISDN/telephony E.164 <input checked="" type="checkbox"/> National standard <input checked="" type="checkbox"/> Private	No change

Calling address rules

Redirect address rules

- **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.
- **Address Condition** in the **Called address rules** section: Enter **^0**. With this expression, the OAD (0) will be removed from the outgoing calls.

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

3. The third address map is only necessary if the PBX is configured as if it is still connected to the PSTN. To create the this address map, click **Add** again, and configure the following parameters:

General

Address map name:	<input type="text" value="From-PBX"/>
Rule name:	<input type="text" value="Add OAD in Called Number"/>
Stop on match:	<input type="checkbox"/>
Enhanced configuration:	<input type="checkbox"/>

Called address rules

	Condition	Result
Address:	<input type="text" value="^[0-9].*"/>	<input type="text" value="0\$&"/>
Name:	<input type="text"/>	<input type="text"/>
Number type:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> International <input checked="" type="checkbox"/> National <input checked="" type="checkbox"/> Network specific <input checked="" type="checkbox"/> Subscriber <input checked="" type="checkbox"/> Abbreviated	<input type="text" value="No change"/>
Numbering plan:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> ISDN/telephony E.164 <input checked="" type="checkbox"/> National standard <input checked="" type="checkbox"/> Private	<input type="text" value="No change"/>

Calling address rules

Redirect address rules

- **Address map name:** Enter a name for the address map for calls from the PBX.
- **Rule name:** Enter a name that describes the address map rule. This name will be displayed on the main configuration page.

Under **Called Address rules**, configure the following parameters:

- **Address Condition** in the **Calling address rules** section: Enter `^[0-9].*`. This expression matches all calls from the PSTN.
- **Address Result** in the **Calling address rules** section: Enter the **0** (zero), so that it is added in the output, followed by the **\$&**, which adds the incoming number after the **0**.

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

The fourth address map is also only necessary if the PBX is configured as if it is still connected to the PSTN. To create this address map, click **Add** again, and configure the following parameters:

General		
Address map name:	<input type="text" value="To-PBX"/>	
Rule name:	<input type="text" value="Remove OAD in Calling Number"/>	
Stop on match:	<input type="checkbox"/>	
Enhanced configuration:	<input type="checkbox"/>	

Called address rules		

Calling address rules		
	Condition	Result
Address:	<input type="text" value="^0"/>	<input type="text"/>
Name:	<input type="text"/>	<input type="text"/>
Number type:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> International <input checked="" type="checkbox"/> National <input checked="" type="checkbox"/> Network specific <input checked="" type="checkbox"/> Subscriber <input checked="" type="checkbox"/> Abbreviated	<input type="text" value="No change"/>
Numbering plan:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> ISDN/telephony E.164 <input checked="" type="checkbox"/> National standard <input checked="" type="checkbox"/> Private	<input type="text" value="No change"/>
Presentation:	<input checked="" type="checkbox"/> Allowed <input checked="" type="checkbox"/> Restricted <input checked="" type="checkbox"/> Number not available <input checked="" type="checkbox"/> Undefined	<input type="text" value="No change"/>

Redirect address rules		

OK	Cancel
----	--------

- **Address map name:** Enter an address map name for calls from the gateway to the PBX.
- **Rule name:** Enter a name that describes the address map rule. This name will be displayed on the main configuration page.
- **Address Condition** in the **Calling address rules** section: Enter **^0**. With this expression, the OAD will be removed from calls to the PBX.

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

The fifth address map is necessary to map the number of unassigned internal numbers to the reception number. In the following example, the reception has the extension 2000. The address map is later used in a route for unassigned internal numbers. All numbers using this route will end up at the reception. To create the this address map, click **Add** again, and configure the following parameters:

General

Address map name:	<input type="text" value="Map-to-Central-Number"/>
Rule name:	<input type="text" value="Map-to-Central-Number-1"/>
Stop on match:	<input type="checkbox"/>
Enhanced configuration:	<input type="checkbox"/>

Called address rules

	Condition	Result
Address:	<input type="text" value="^.*\$"/>	<input type="text" value="+49715940662000"/>
Name:	<input type="text"/>	<input type="text"/>
Number type:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> International <input checked="" type="checkbox"/> National <input checked="" type="checkbox"/> Network specific <input checked="" type="checkbox"/> Subscriber <input checked="" type="checkbox"/> Abbreviated	<input type="text" value="No change"/>
Numbering plan:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> ISDN/telephony E.164 <input checked="" type="checkbox"/> National standard <input checked="" type="checkbox"/> Private	<input type="text" value="No change"/>

Calling address rules

Redirect address rules

- **Address map name:** Enter an address map name for calls to the reception.
- **Rule name:** Enter a name that describes the address map rule. This name will be displayed on the main configuration page.

Under **Called address rules**, configure the following parameters:

- **Address Condition:** Enter **^.*\$**. Because this address map will be associated with a route for internal unassigned numbers, this expression will match all of those numbers.
- **Address Result:** Enter the number of the reception, to which the unassigned internal numbers will be routed. Include the country code, area code, base number, and extension, as shown in the graphic above.

Creating Routes for This Scenario

Create the routes in the following order:

1. Create the route from the PSTN to Microsoft® Office Communications Server 2007 first. To do so, open the **Routing** section, click **Add** and configure the following parameters:

General					
Name:		Route-to-OCS			
Type: Name	Source	Destination			
PSTN: Controller-to-PBX	<input checked="" type="checkbox"/>	-			
PSTN: Controller-to-PSTN	<input checked="" type="checkbox"/>	-			
SIP: OCS-Mediation-Server	<input type="checkbox"/>	Master			
Max. call attempts for this route in a failover scenario:		0 (0 = try all selected destinations)			

Address Normalization For Condition Processing (Using Source Dialplan)	
Disable inbound and outbound dialplan:	<input type="checkbox"/>
Number format:	International number
Encoding:	Use type flag

Conditions					
Extended Condition	Item	Called	Calling	Redirect	Action
<input type="checkbox"/>	Address	^.*\$			delete
Add new condition					

Address Manipulation	
OK Cancel	

- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select both controllers as sources.
- **Destination:** Select the SIP peer you configured for the Microsoft® Mediation Server as a Master destination.
- **Number format:** Select **International number** from the dropdown menu.
- **Encoding:** Select **Use type flag** from the dropdown menu.

- Under **Conditions**, click **Add**, and in **Called number**, enter a regular expression that matches only the numbers of the Microsoft® Office Communications Server 2007 in E.164 format including country code, area code, trunk prefix, and extension number. In this scenario, the extensions at the Office Communications Server start with 4 or 5. Within the regular expression, this is represented by [45].

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

- Create the second route from the Microsoft® Office Communications Server 2007 to the PBX. To do so, click **Add** in the **Routing** section, and configure the following parameters:

General					
Name:	Route-to-PBX				
Type: Name	Source	Destination			
PSTN: Controller-to-PBX	<input type="checkbox"/>	Master			
PSTN: Controller-to-PSTN	<input checked="" type="checkbox"/>	-			
SIP: OCS-Mediation-Server	<input checked="" type="checkbox"/>	-			
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)				
Address Normalization For Condition Processing (Using Source Dialplan)					
Disable inbound and outbound dialplan:	<input type="checkbox"/>				
Number format:	International number				
Encoding:	Use type flag				
Conditions					
Extended Condition	Item	Called	Calling	Redirect	Action
<input type="checkbox"/>	Address	^+4971594066(23)			delete
Add new condition					
Address Manipulation					
OK Cancel					

- Name:** Enter a unique name to easily identify the route.
- Source:** Select the PSTN controller and the SIP peer you configured for the Microsoft® Mediation Server as sources.
- Destination:** Select the PBX controller as a Master destination.
- Number format:** Select **International number** from the dropdown menu.
- Encoding:** Select **Use type flag** from the dropdown menu.
- Under **Conditions**, click **Add**, and in **Called number**, enter a regular expression that matches only the subscriber numbers of the PBX in E.164 format including country code, area code, trunk prefix, and extension number. In this scenario, the extensions at the PBX start with 2 or 3. Within the regular expression, this is represented by [23].

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

3. Create the third route from the Microsoft® Office Communications Server 2007 to the PSTN. To do so, click **Add** in the **Routing** section, and configure the following parameters:

General		
Name:	Route-to-PSTN	
Type: Name	Source	Destination
PSTN: Controller-to-PBX	<input checked="" type="checkbox"/>	-
PSTN: Controller-to-PSTN	<input type="checkbox"/>	Master
SIP: OCS-Mediation-Server	<input checked="" type="checkbox"/>	-
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)	
Address Normalization For Condition Processing (Using Source Dialplan)		
Disable inbound and outbound dialplan:	<input type="checkbox"/>	
Number format:	International number	
Encoding:	Use type flag	
Conditions		
Address Manipulation		
<div>OK Cancel</div>		

- **Source:** Select the PBX controller and the SIP peer configured for the Microsoft® Mediation Server as sources.
- **Destination:** Select the PSTN controller as a Master destination.
- Under **Address Normalization For Condition Processing (Using Source Dialplan)**, configure the following parameters:
 - **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 close the window.

4. Create the fourth route for calls to the reception. To do so, click **Add** in the **Routing** section, and configure the following parameters:

General					
Name:	Route-to-Central-Number				
Type: Name	Source	Destination			
PSTN: Controller-to-PBX	<input type="checkbox"/>	Master			
PSTN: Controller-to-PSTN	<input checked="" type="checkbox"/>	-			
SIP: OCS-Mediation-Server	<input type="checkbox"/>	-			
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)				

Address Normalization For Condition Processing (Using Source Dialplan)	
Disable inbound and outbound dialplan:	<input type="checkbox"/>
Number format:	International number
Encoding:	Use type flag

Conditions					
Extended Condition	Item	Called	Calling	Redirect	Action
<input type="checkbox"/>	Address	^+4971594066(016789)			delete
Add new condition					

Address Manipulation	
Address map:	Map-to-Central-Number

- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select the PSTN controller as a source.
- **Destination:** Select the SIP peer configured for the Microsoft® Mediation Server as a destination. We select this destination, because in this scenario, the reception is connected to the Microsoft® Mediation Server. If the reception is connected to the PBX instead, select the PBX controller as the destination.

Under **Address Normalization For Condition Processing (Using Source Dialplan)**, configure the following parameters:

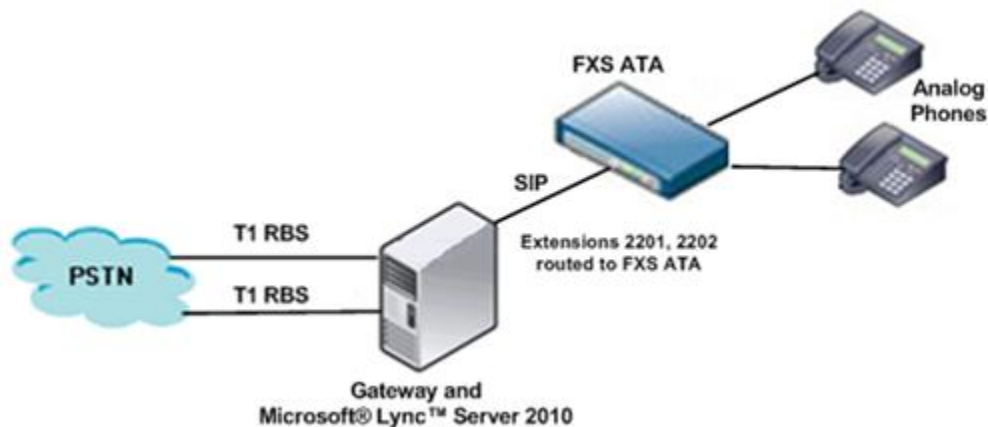
- **Number format:** Select **International number** from the dropdown menu.
- **Encoding:** Select **Use type flag** from the dropdown menu.
- **Conditions:** Click **Add**, and in **Called number**, enter a regular expression that matches all extensions that should be routed to the reception, i.e., all unassigned extensions. In our example, the unassigned extensions do NOT start with 0, 1, 6, 7, 8, or 9, because those extensions are not used by the PBX or Microsoft® Office Communications Server 2007.
- **Address map** in the **Address Manipulation** section: Select the mapping for the reception.

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

Using the Gateway Computer Between the PSTN and Microsoft® Lync™ Server 2010

This configuration scenario describes the necessary steps for configuring a gateway computer that is directly connected to the PSTN via two T1 RBS links and also connected to a Lync SBA server on the SIP side. This scenario has the following characteristics:

- There is no PBX between the gateway and the PSTN.
- The Lync Server is co-located with the gateway. Together, the Lync Server and the gateway create a Survivable Branch Appliance Solution (SBA).
- The gateway is connected to an FXS (Foreign eXchange Subscriber interface) Analog Telephone Adapter (ATA) via SIP. The FXS ATA is connected to analog phones or fax devices.
- Internal extensions have four digits.
- The Lync Server routes calls to extensions 2201 and 2202 to the FXS ATA via the gateway.
- The gateway blocks calls from the PSTN that have origination extensions of 2201 or 2202, because the Lync Server would treat these calls as calls from the FXS ATA rather than calls from the PSTN.



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



3. Click either the board icon or the name of the first Diva Media Board line to open the **Board Configuration - Detail** page.
4. Configure the first board by configuring the following parameters:

Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 1, SN:1033

Parameter	Value
D-Channel Protocol:	USA, RBS T1 (Robbed Bit Signaling) - (RBSCAS)
Interface Mode/Resource Board:	TE - mode
Direct Dial In (NT2):	Yes
DDI Number Length:	4
DDI Collect Timeout:	2 (default)
DDI Special Number:	
Layer 1 Framing:	National default (default)
Layer 2 Connect Mode:	Permanent
Voice Companding:	Protocol default
View Extended Configuration	No

Save Cancel

- **D-Channel Protocol:** Select a protocol with T1 Robbed Bit Signaling (RBS).
- **DDI Number Length:** Select **4**.

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

5. Configure the second board with the same parameters as the first one.
6. In the Diva SIPcontrol web interface, click **SIPcontrol configuration** on the left hand side to open the **SIPcontrol configuration** page.

7. Create a certificate request file and private key file with a third party program. Send the certificate request file to the CA server used for providing the Lync Server certificates. Retrieve signed certificate and the CA certificate from CA server. For information about generating certificate files, key files, and certificate request files with OpenSSL and Microsoft® Active Directory Certificate Services, see *Deploying a Dialogic® 4000 Media Gateway as a Survivable Branch Appliance for Microsoft® Lync™ Server 2010*.
8. Upload the CA certificate file, the signed certificate, and key file to Diva SIPcontrol, and select the authentication mode:
 1. Click **Security Profiles** (about halfway down the page), and then click **Details** to open the Security Profiles options:

Upload Certificate and Key Files

Certificate authority file: Not available	<input style="width: 80%;" type="text"/> <input type="button" value="Browse..."/> <input type="button" value="Upload"/>
Certificate file: Not available	<input style="width: 80%;" type="text"/> <input type="button" value="Browse..."/> <input type="button" value="Upload"/>
Key file: Not available	<input style="width: 80%;" type="text"/> <input type="button" value="Browse..."/> <input type="button" value="Upload"/>

Global Security Parameters

Host name:	<input style="width: 80%;" type="text"/> must match 'CommonName' of certificate
Supported cipher levels:	High: <input checked="" type="checkbox"/> Medium: <input checked="" type="checkbox"/> Low: <input type="checkbox"/>
Authentication mode:	<input type="text" value="Standard TLS Authentication"/>
Certificate date verification:	<input type="checkbox"/>

2. In the Certificate authority file field, use **Browse** to locate the Certificate Authority file (*certAuth.cer*).
 3. Click **Upload** to upload the *certAuth.cer* file to Diva SIPcontrol
- A message box appears.
4. Click **OK** to upload the file.
 5. Repeat Steps 2 – 5 for the Certificate file (*.cer) and Key file (*.csr).

6. In the Global Security Parameters section, fill in the following fields:
 - **Host name:** Enter the common name used in the certificate to identify the Diva SIPcontrol host machine.
 - **Authentication mode:** Select Mutual authentication.
7. Click **OK** at the bottom of the page to close the Security Profiles window.
9. At the bottom of the SIPcontrol main page, click **Activate Configuration** to save the configuration.
10. **Restart Diva SIPcontrol** to use the new configuration. Click **System Control** on the left side of the web interface, and then click **Restart** in the SIPcontrol field. When Diva SIPcontrol restarts, click **SIPcontrol configuration** on the left side of the web interface.
11. Create two codec profiles: One for the Lync Server and the other for FXS ATA 3.
To create the a codec profile for the Lync Server, open the **Codec Profiles** section, click **Add**, and configure the following parameters:

General	
Name:	Lync-Codecs
Audio Codecs	
Available Codecs G.729 G.726 16 kbps G.726 24 kbps G.726 32 kbps G.726 40 kbps	Selected Codecs G.711 A-Law G.711 u-Law
Use Codec --> <-- Remove Codec	
Up Down	
G.711 A-Law Codec Settings Packet interval default: 20 Voice activity detection: Comfort Noise Generation: <input checked="" type="checkbox"/>	
Audio Quality	
Support comfort noise payload: <input checked="" type="checkbox"/> Echo canceller: <input checked="" type="checkbox"/>	Noise suppressor:
DTMF Codec	
Transmit as RTP event: <input checked="" type="checkbox"/> Automatic payload type: <input checked="" type="checkbox"/> Disable CNG event: <input type="checkbox"/>	Manual payload type value:
OK Cancel	

- Enter a unique name to easily identify the codec.
- In the **Audio Codecs** section, enable **Comfort Noise Generation** and **Voice activity detection**.

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

To create a codec profile for the FSX ATA, open the **Codec Profiles** section, click **Add**, and configure the following parameters:

General

Name: ATACodecs

Audio Codecs

Available Codecs

- G.729
- G.726 16 kbps
- G.726 24 kbps
- G.726 32 kbps
- G.726 40 kbps

Selected Codecs

- G.711 u-Law

Use Codec -->

<-- Remove Codec

Up Down

G.711 u-Law Codec Settings

Packet interval default: 20

Voice activity detection: ☐

Comfort Noise Generation: ☐

Audio Quality

Support comfort noise payload: ☐

Noise suppressor: ☐

Echo canceller: ☒

DTMF Codec

Transmit as RTP event: ☒

Automatic payload type: ☐

Manual payload type value: 10

Disable CNG event: ☒

OK Cancel

- Enter a unique name to easily identify the codec.
- Remove **G.711 A-Law** from the list of selected codecs.
- In the **Audio Quality** section, disable **Support comfort noise payload**.
- Click **OK** to save the settings and close the window.

12. Create a dialplan that reflects the PSTN dialing rules for the T1 trunks connected to the gateway. To do so, open the **Dialplans** section, click **Add**, and configure the following parameters:

General	
Name:	US Dialplan
Country code:	1
North-American numbering plan:	<input checked="" type="checkbox"/>
Area code:	716 With national prefix ▾
Other local areas:	<input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/> <input type="text"/>
Base number:	555
Maximum extension digits:	4 ▾
International prefix:	011
National prefix:	1
Access code:	<input type="text"/>
PSTN access code provided by SIP caller:	<input checked="" type="checkbox"/>
Incoming PSTN access code provided by PBX:	<input checked="" type="checkbox"/>
Keep SIP URI Domain and Parameters:	<input type="checkbox"/>
<input type="button" value="OK"/> <input type="button" value="Cancel"/>	

- **Name:** Enter a unique name to easily identify the dialplan.
- **North-American number plan:** Enable **North-American numbering plan** if the dialplan is for North American numbers (country code 1).
- **Country code:** Enter the country code of the country in which the gateway computer is located.
- **Area code:** Enter the area code of the region in which the gateway computer 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 gateway computer is located.

- **National prefix:** Enter the national prefix that needs to be dialed for long distance calls within the country in which the gateway computer is located.
- **Access code:** Enter the digit that is necessary to get access to the public network, if any.

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

13. Create an address map for calls from the FXS ATA to the Lync Server, to make them appear like internal Lync Server calls. To do this, open the **Address Maps** section, click **Add**, and configure the following parameters:

General		
Address map name:	<input type="text" value="FromATA"/>	
Rule name:	<input type="text" value="FromATA.1"/>	
Stop on match:	<input checked="" type="checkbox"/>	
Enhanced configuration:	<input type="checkbox"/>	

Called address rules		
Address:	<input type="text" value="^(.*)@192\168\212\138"/>	<input type="text" value="\$1@sba.domain.example.com"/>
Name:	<input type="text"/>	<input type="text"/>
Number type:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> International <input checked="" type="checkbox"/> National <input checked="" type="checkbox"/> Network specific <input checked="" type="checkbox"/> Subscriber <input checked="" type="checkbox"/> Abbreviated	<input type="text" value="No change"/>
Numbering plan:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> ISDN/telephony E.164 <input checked="" type="checkbox"/> National standard <input checked="" type="checkbox"/> Private	<input type="text" value="No change"/>

Calling address rules		
Address:	<input type="text" value="^(.*)@192\168\212\138"/>	<input type="text" value="\$1@sba.domain.example.com"/>
Name:	<input type="text"/>	<input type="text"/>
Number type:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> International <input checked="" type="checkbox"/> National <input checked="" type="checkbox"/> Network specific <input checked="" type="checkbox"/> Subscriber <input checked="" type="checkbox"/> Abbreviated	<input type="text" value="No change"/>
Numbering plan:	<input checked="" type="checkbox"/> Unknown <input checked="" type="checkbox"/> ISDN/telephony E.164 <input checked="" type="checkbox"/> National standard <input checked="" type="checkbox"/> Private	<input type="text" value="No change"/>
Presentation:	<input checked="" type="checkbox"/> Allowed <input checked="" type="checkbox"/> Restricted <input checked="" type="checkbox"/> Number not available <input checked="" type="checkbox"/> Undefined	<input type="text" value="No change"/>

Redirect address rules		
<input type="button" value="OK"/> <input type="button" value="Cancel"/>		

- **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.
- Optionally enable **Stop on match** to have the gateway stop searching for matching rules when all specified address conditions match all addresses of a call.

Under **Called address rules**, configure the following parameters:

- **Address Condition:** Enter the called address condition as shown in the graphic above. This condition matches all calls to 192.168.212.138, which is the IP address of the FSX ATA.
- **Address Result:** Enter the called address result as shown in the graphic above, with the FDQN of the SBA after the @ sign. This address result converts the matched called addresses to a form accepted by the Lync Server.
- Under **Calling address rules**, configure the following parameters:
- **Address Condition:** Enter the calling address condition as shown in the graphic above. This condition matches all calls from 192.168.212.138, which is the IP address of the FXS ATA.
- **Address Result:** Enter the calling address result as shown in the graphic above, with the FDQN of the SBA after the @ sign. This address result converts the matched calling addresses to a form accepted by the Lync Server.

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

14. Under **PSTN Interfaces**, configure both Controller 1 and Controller 2 with PSTN-specific settings. The controller numbers that you configure with PSTN-specific settings needs to correspond to the line numbers in the Diva Configuration Manager on which you configured the switch type of your PSTN lines.

To configure PSTN-specific parameters for Controller 1, click **Details** to the right of the respective controller, and configure the following parameters:

General	
Hardware description:	Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 1 SN
PSTN interface number:	1
Name:	Controller1
Address map inbound:	none
Address map outbound:	none

Enhanced	
Address Normalization	
Dialplan:	US Dialplan
Type of number (outbound):	Unchanged
Encoding (outbound):	Use type flag
ISDN numbering plan - Default:	unknown
Presentation indicator - Default:	Allowed
Internal interface:	<input type="checkbox"/>

PSTN Call Transfer Settings	
Message Waiting Indication (MWI)	

- **Name:** Enter a unique name for this controller.
- **Dialplan:** Select the configured dialplan.

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

Configure Controller 2 by using the same settings you specified for Controller 1, except for the value of the Name parameter:

General	
Hardware description:	Dialogic Diva V-2PRI/E1/T1-60 PCI v1 - PORT 2 SN:
PSTN interface number:	2
Name:	Controller2
Address map inbound:	none
Address map outbound:	none

Enhanced	
Address Normalization	
Dialplan:	US Dialplan
Type of number (outbound):	Unchanged
Encoding (outbound):	Use type flag
ISDN numbering plan - Default:	unknown
Presentation indicator - Default:	Allowed
Internal interface:	<input type="checkbox"/>

PSTN Call Transfer Settings	
Message Waiting Indication (MWI)	

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

- Under **Network Interfaces**, enable your Ethernet adapter, and set the TCP listen port to **5081** and the TLS listen port to **5082**:

Network Interfaces					
Name	Device	IP address	UDP listen port	TCP listen port	TLS listen
Intel(R) PRO1000 EB Network	Intel(R) PRO/1000 EB Network Connection with I/O Acceleration #2	192.168.212.136	5060 <input type="checkbox"/>	5081 <input checked="" type="checkbox"/>	5082
Local Loopback Interface	Local Loopback Interface	127.0.0.1	0 <input type="checkbox"/>	0 <input type="checkbox"/>	0

RTP start port:	30000
RTP end port:	39999

- Create the following SIP peers, as described in [Creating the SIP Peers for This Scenario](#):
 - A SIP peer for the Lync Server
 - Two SIP peers to correspond to each ATA extension.

- A dummy SIP peer that is used for rejecting PSTN calls made to the same extensions as the FXS ATA (extensions 2201 and 2202).
17. Once you have created the SIP Peers, you need to disable the dummy SIP peer, which is the fourth one. To do this, access the SIP Peers section of the main SIP Configuration screen, and uncheck the **Enabled** parameter for this peer.

The SIP Peers section of the main configuration screen should now look like this:

SIP Peers							
Name	Host	Port	IP protocol	Codec Profile	Dialplan	Enabled	
Lync	sba.domain.example.com	5067	TLS	Lync-Codecs	US Dialplan	<input checked="" type="checkbox"/>	Details De
ATA2201	192.168.212.138	5068	TCP	ATACodecs	US Dialplan	<input checked="" type="checkbox"/>	Details De
ATA2202	192.168.212.138	5070	TCP	ATACodecs	US Dialplan	<input checked="" type="checkbox"/>	Details De
Forbidden Call	127.0.0.1	5060	TCP	default	none	<input type="checkbox"/>	Details De
Default peer for received SIP calls: Lync							Add new peer

18. Create the following routes in the order specified:
- Route to block numbers from the PSTN that have the same extension as the ATA
 - Route from the PSTN to the Lync Server
 - Route for call transfers in a Lync Server environment
 - Two routes from the Lync Server to the FXS ATA extensions
 - Route from the Lync Server to the PSTN
 - Route from FXS ATA to the Lync Server

In this scenario, the order of the routes is important because of the conditions used. For instructions on creating routes, see [Creating the Routes for This Scenario](#).

Creating the SIP Peers for This Scenario

Create the SIP Peers in this order:

1. Create the first SIP peer for the Lync Server. To do this, open the **SIP Peers** section, click **Add**, and enter the following parameters:

General	
Name:	Lync
Peer type:	MS Lync 2010 - Mediation Server
Host:	sba.domain.example.com
Port:	5067
IP protocol:	TLS
URI scheme:	SIP (default)
Domain:	domain.example.com

Enhanced	
Default peer for received SIP calls:	<input checked="" type="checkbox"/>
Display name to:	
Display name from:	
User name to:	
User name from:	
Gateway prefix:	
Reply-To expression:	
Reply-To format:	
Alive check:	<input checked="" type="checkbox"/> 0 seconds (0=auto)
Disconnect tone support:	<input type="checkbox"/>
Support MWI (Message waiting):	<input checked="" type="checkbox"/>
Cause code mapping inbound:	peer default
Cause code mapping outbound:	peer default
Codec profile:	Lync-Codecs
Maximum channels:	480
Early media support:	<input checked="" type="checkbox"/>
Reliable provisional response:	Optional

Security	
Signaling accept level:	Accept unencrypted and encrypted calls
Media security level:	Require SRTP for all calls

Session Timer	
---------------	--

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

Authentication	
----------------	--

OK Cancel

- **Name:** Enter a unique name to easily identify the SIP peer.
- **Peer type:** Select **MS Lync 2010 - Mediation Server** from the dropdown menu.
- **Host:** Enter the FQDN of the Lync Server to which you are connected, which in this scenario, is also the FQDN of the gateway.

Under **Enhanced**, configure the following parameters:

- **Default peer for received SIP calls:** Enable this parameter.
- **Codec profile:** Select the codec you defined for the Lync Server.

Under **Security**, configure the following parameter:

- **Media security level:** Select **Require SRTP for all calls** from the dropdown menu.

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

- **Dialplan:** Select the configured dialplan.
- **Number format (outbound):** Select **International number** from the dropdown menu.
- **Encoding (outbound):** Select **Use type flag** from the dropdown menu.

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

2. Create the second SIP peer for the first ATA extension. To do this, open the **SIP Peers** section, click **Add**, and enter the following parameters:

General	
Name:	ATA2201
Peer type:	Grandstream HT-502
Host:	192.168.212.138
Port:	5068
IP protocol:	TCP
URI scheme:	SIP (default)
Domain:	

Enhanced	
Default peer for received SIP calls:	<input type="checkbox"/>
Display name to:	
Display name from:	
User name to:	
User name from:	
Gateway prefix:	
Reply-To expression:	
Reply-To format:	
Alive check:	<input type="checkbox"/> 1800 seconds (0=auto)
Disconnect tone support:	<input type="checkbox"/>
Support MWI (Message waiting):	<input checked="" type="checkbox"/>
Cause code mapping inbound:	peer default
Cause code mapping outbound:	peer default
Codec profile:	ATACodecs
Maximum channels:	480
Early media support:	<input type="checkbox"/>
Reliable provisional response:	Disabled

Security	

Session Timer	

Address Normalization	
Dialplan:	US Dialplan
Number format (outbound):	Unchanged
Encoding (outbound):	Use type flag
Address map inbound:	none
Address map outbound:	none

Authentication	

- **Name:** Enter a unique name to easily identify the SIP peer.
- **Peer type:** Select **Grandstream HT-502** from the dropdown menu.
- **Host:** Enter the host name for the SIP peer.

Under **Enhanced**, select the codec you defined for the ATA as the **Codec profile**, and disable **Reliable provisional response**.

Under **Address Normalization**, select the configured dialplan.

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

3. Create the third SIP peer for the second ATA extension. Use the same configuration values as you did for the first ATA extension, except for the values of the **Name**, **Host**, and **Port** parameters in the **General** section.
4. Create the fourth SIP peer as a dummy peer that will be used for rejecting calls. (The routing rules will determine which calls will be rejected.) To create this SIP peer, open the **SIP Peers** section, click **Add**, and enter values for the **Name**, **Host**, and **Port** parameters in the **General** section. The values for these parameters do not matter. For example:

General	
Name:	<input type="text" value="Forbidden Call"/>
Peer type:	<input type="text" value="Default"/>
Host:	<input type="text" value="127.0.0.1"/>
Port:	<input type="text" value="5060"/>
IP protocol:	<input type="text" value="TCP"/>
URI scheme:	<input type="text" value="SIP (default)"/>
Domain:	<input type="text"/>

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

Creating Routes for This Scenario

Create the routes in the following order:

1. Create the first route to block numbers from the PSTN that have the same extension as the ATA (extensions 2201 and 2202). To create the first route, open the **Routing** section, click **Add** and configure the following parameters:

General					
Name:		Block ATA numbers from PSTN			
Type: Name	Source	Destination			
PSTN: Controller1	<input checked="" type="checkbox"/>	-			
PSTN: Controller2	<input checked="" type="checkbox"/>	-			
SIP: Lync	<input type="checkbox"/>	-			
SIP: ATA2201	<input type="checkbox"/>	-			
SIP: ATA2202	<input type="checkbox"/>	-			
SIP: Forbidden Call (disabled)	<input type="checkbox"/>	Master			
Max. call attempts for this route in a failover scenario:		0 (0 = try all selected destinations)			
Address Normalization For Condition Processing (Using Source Dialplan)					
Disable inbound and outbound dialplan:		<input type="checkbox"/>			
Number format:		International number			
Encoding:		Use type flag			
Conditions					
Extended Condition	Item	Called	Calling	Redirect	Action
<input type="checkbox"/>	Address		^\+17165552201\$		delete
<input type="checkbox"/>	Address		^\+17165552202\$		delete
Add new condition					
Address Manipulation					
OK Cancel					

- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select both PSTN controllers as sources.
- **Destination:** Select the disabled SIP peer as a Master destination.

Under **Address Normalization for Condition Processing (Using Source Dialplan)**, configure the following parameters:

- **Use Dialplan:** Leave this field enabled (the default) so that the gateway uses the configured Dialplan when selecting this route.
- **Number format:** Select **International number** from the dropdown menu.
- **Encoding:** Select **Use type flag** from the dropdown menu.

Under **Conditions**, click **Add**, and in **Calling number**, enter the regular expressions that match the extensions for the ATA including country code, area code, trunk prefix, and extension number. In this scenario, the expressions `^\+17165552201$` and `^\+17165552202$` match both extensions for the ATA.

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

2. Create the second route from the PSTN to the Lync Server. To do so, click **Add** in the **Routing** section, and configure the following parameters:

General		
Name:	PSTN-to-SIP	
Type: Name	Source	Destination
PSTN: Controller1	<input checked="" type="checkbox"/>	-
PSTN: Controller2	<input checked="" type="checkbox"/>	-
SIP: Lync	<input type="checkbox"/>	Master
SIP: ATA2201	<input type="checkbox"/>	-
SIP: ATA2202	<input type="checkbox"/>	-
SIP: Forbidden Call (disabled)	<input type="checkbox"/>	-
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)	
Address Normalization For Condition Processing (Using Source Dialplan)		
Disable inbound and outbound dialplan:	<input type="checkbox"/>	
Number format:	Unchanged	
Encoding:	Use type flag	
Conditions		
Address Manipulation		
<div>OK Cancel</div>		

- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select both PSTN controllers as sources.
- **Destination:** Select the SIP Peer configured for the Lync Server as a Master destination.

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

3. Create the third route to have all call transfers routed by the Lync Server. To do so, click **Add** in the **Routing** section, and configure the following parameters:

General		
Name:	Call transfer routing by Lync	
Type: Name	Source	Destination
PSTN: Controller1	<input type="checkbox"/>	-
PSTN: Controller2	<input type="checkbox"/>	-
SIP: Lync	<input checked="" type="checkbox"/>	Master
SIP: ATA2201	<input type="checkbox"/>	-
SIP: ATA2202	<input type="checkbox"/>	-
SIP: Forbidden Call (disabled)	<input type="checkbox"/>	-
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)	

Address Normalization For Condition Processing (Using Source Dialplan)	
Disable inbound and outbound dialplan:	<input type="checkbox"/>
Number format:	Unchanged
Encoding:	Use type flag

Conditions				
Extended Condition	Item	Called	Calling	Redirect
<input type="checkbox"/>	Address	^192\168\212\136		
<input type="checkbox"/>	Address	^(?)SBA\domain\example\com		
Add new condition				

Address Manipulation	
OK Cancel	

- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select the SIP peer configured for the Lync Server as both a source and Master destination. This causes all calls matching the specified conditions to be routed back to the Lync Server, so it can handle the routing.

Under **Conditions**, enter the address of the Lync Server as both an FQDN and an IP address. This condition will match all transfer requests from the Lync Server.

Note: If the Gateway is connected to a Lync Front End Server Pool instead of a Lync SBA or single Lync Front End Server, you need to use the FQDN of the pool and IP address of each pool member to match all transfer requests from the Lync Server.

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

4. Create the fourth route to route calls from the Lync Server to one of the ATA extensions. To do so, click **Add** in the **Routing** section, and configure the following parameters:

General				
Name:	To ATA1			
Type: Name	Source	Destination		
PSTN: Controller1	<input type="checkbox"/>	-		
PSTN: Controller2	<input type="checkbox"/>	-		
SIP: Lync	<input checked="" type="checkbox"/>	-		
SIP: ATA2201	<input type="checkbox"/>	Master		
SIP: ATA2202	<input type="checkbox"/>	-		
SIP: Forbidden Call (disabled)	<input type="checkbox"/>	-		
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)			
Address Normalization For Condition Processing (Using Source Dialplan)				
Disable inbound and outbound dialplan:	<input type="checkbox"/>			
Number format:	International number			
Encoding:	Use type flag			
Conditions				
Extended Condition	Item	Called	Calling	Redirect
<input type="checkbox"/>	Address	^+17165552201\$		
Add new condition				
Address Manipulation				
OK Cancel				

- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select the SIP peer configured for the Lync Server as a source.
- **Destination:** Select the SIP peer configured for one of the ATA extensions as a Master destination.

Under **Address Normalization for Condition Processing (Using Source Dialplan)**, configure the following parameters:

- **Use Dialplan:** Leave this field enabled (the default) so that the gateway uses the configured Dialplan when selecting this route.
- **Number format:** Select **International number** from the dropdown menu.
- **Encoding:** Select **Use type flag** from the dropdown menu.

Under **Conditions**, click **Add**, and in **Called number**, enter a regular expression that matches all calls to the ATA extension 2201.

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

5. Create the fifth route to route calls from the Lync Server to the second ATA extension. To do so, use the same settings you specified for the fourth route (in Step 12), except for the value of the **Name** and **Called number** parameters. For **Called number**, enter a regular expression that matches all calls to the ATA extension 2202:

General

Name:	ToATA2	
Type: Name	Source	Destination
PSTN: Controller1	<input type="checkbox"/>	-
PSTN: Controller2	<input type="checkbox"/>	-
SIP: Lync	<input checked="" type="checkbox"/>	-
SIP: ATA2201	<input type="checkbox"/>	-
SIP: ATA2202	<input type="checkbox"/>	Master
SIP: Forbidden Call (disabled)	<input type="checkbox"/>	-
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)	

Address Normalization For Condition Processing (Using Source Dialplan)

Disable inbound and outbound dialplan:	<input type="checkbox"/>
Number format:	International number
Encoding:	Use type flag

Conditions

Extended Condition	Item	Called	Calling	Redirect	Action
<input type="checkbox"/>	Address	^+17165552202\$			delete
Add new condition					

Address Manipulation

OK Cancel

6. Create the sixth route to route calls from the Lync Server to the PSTN. To do so, click **Add** in the **Routing** section, and configure the following parameters:

General		
Name:	Lync-to-PSTN	
Type: Name	Source	Destination
PSTN: Controller1	<input type="checkbox"/>	Master
PSTN: Controller2	<input type="checkbox"/>	Master
SIP: Lync	<input checked="" type="checkbox"/>	-
SIP: ATA2201	<input type="checkbox"/>	-
SIP: ATA2202	<input type="checkbox"/>	-
SIP: Forbidden Call (disabled)	<input type="checkbox"/>	-
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)	
Address Normalization For Condition Processing (Using Source Dialplan)		
Conditions		
Address Manipulation		
<input type="button" value="OK"/> <input type="button" value="Cancel"/>		

- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select the SIP peer configured for the Lync Server as a source.
- **Destination:** Select both controllers as Master destinations.

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

7. Create the seventh route to route calls from the FSX ATA to the Lync Server. To do so, click **Add** in the **Routing** section, and configure the following parameters:

General		
Name:	From ATA	
Type: Name	Source	Destination
PSTN: Controller1	<input type="checkbox"/>	-
PSTN: Controller2	<input type="checkbox"/>	-
SIP: Lync	<input type="checkbox"/>	Master
SIP: ATA2201	<input checked="" type="checkbox"/>	-
SIP: ATA2202	<input checked="" type="checkbox"/>	-
SIP: Forbidden Call (disabled)	<input type="checkbox"/>	-
Max. call attempts for this route in a failover scenario:	0 (0 = try all selected destinations)	
Address Normalization For Condition Processing (Using Source Dialplan)		
Conditions		
Address Manipulation		
Address map:	FromATA	
<input type="button" value="OK"/> <input type="button" value="Cancel"/>		

- **Name:** Enter a unique name to easily identify the route.
- **Source:** Select the SIP peers for the two ATA extensions as sources.
- **Destination:** Select the SIP peer for the Lync Server as a master destination.

Under **Address Manipulation**, select the address map configured for calls from the ATA to the Lync Server. This address map makes the calls appear like internal calls to the Lync Server.

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13. SNMP Support

Activating 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 Diva System Release software. To activate the SNMP service, use the Diva Configuration Manager as described below.

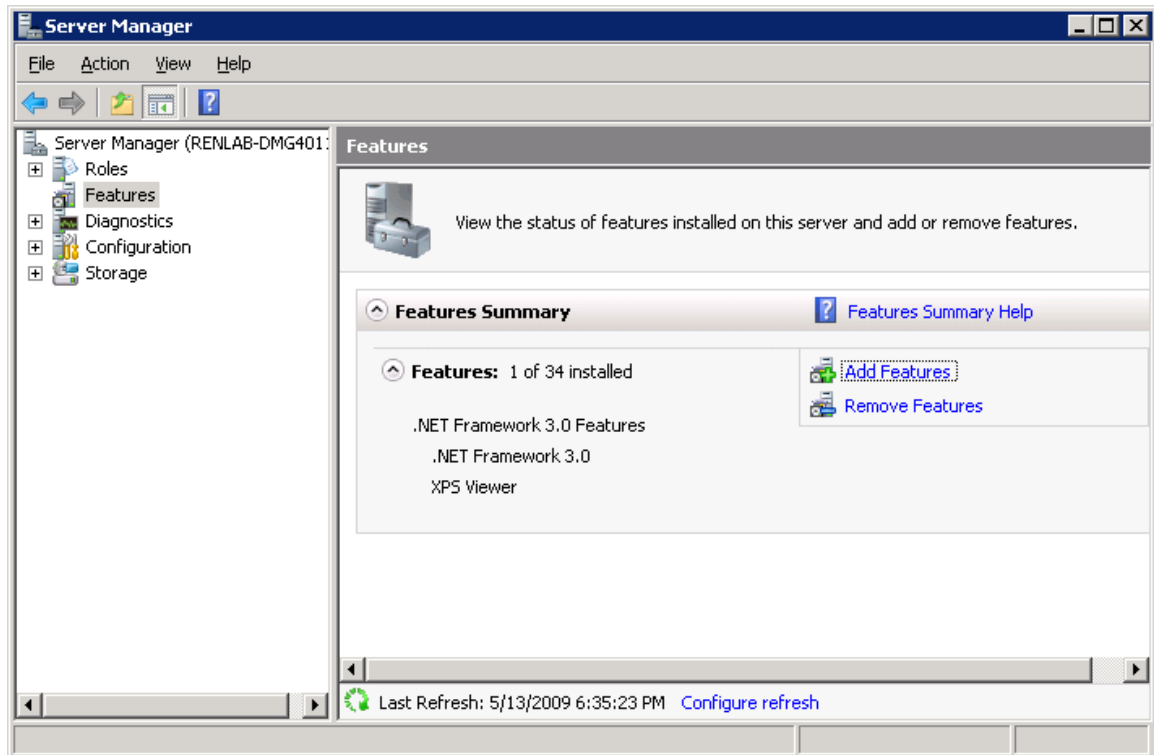
To activate SNMP support for a Diva Media Board, follow these steps:

1. [Install the Windows® SNMP Service.](#)
2. [Add the SNMP Service in the Diva Configuration Manager.](#)
3. Install an SNMP tool, e.g., Net.SNMP (optional, for testing only).
4. Restart your computer.
5. [Verify the SNMP service status.](#)
6. [Verify the function of the SNMP Service.](#)

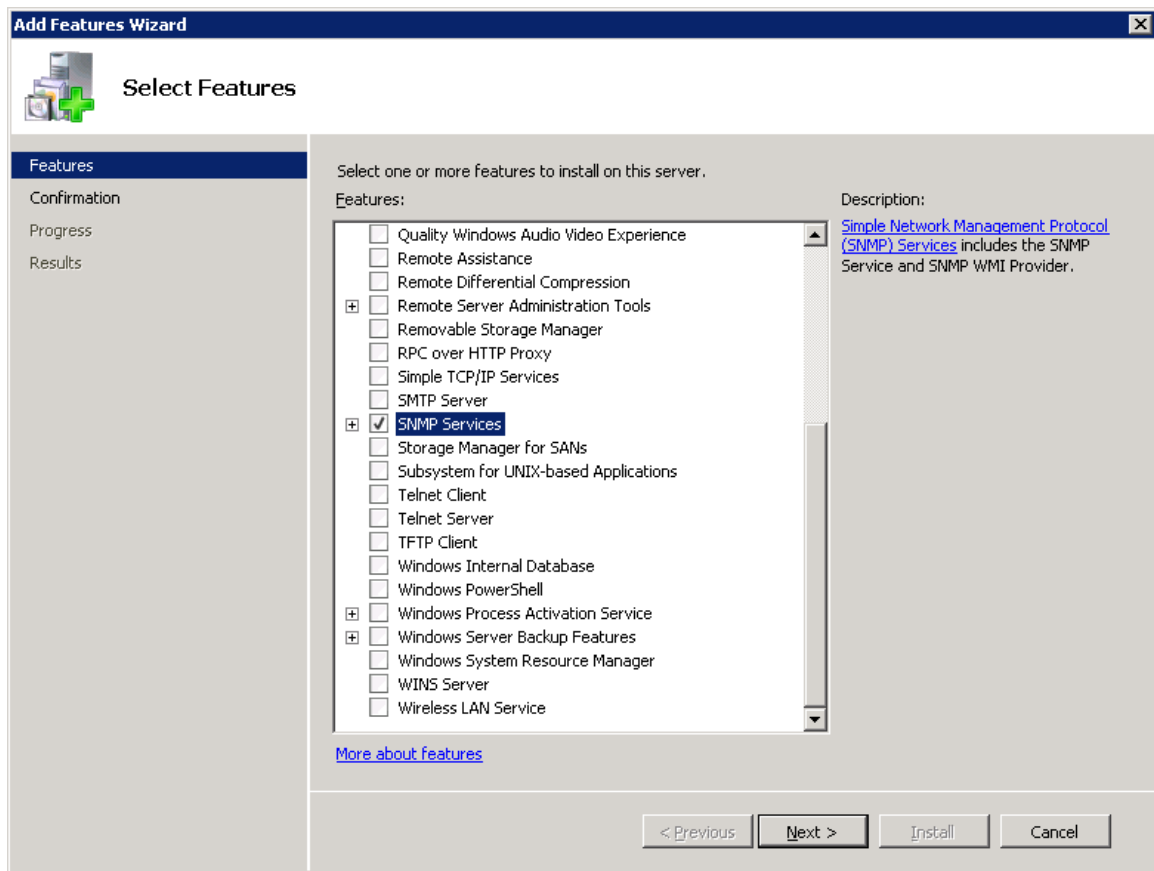
Installing the Windows SNMP Service

To install the Windows® SNMP Service under Microsoft® Windows Server® 2008 and Microsoft® Windows Server® 2008 R2:

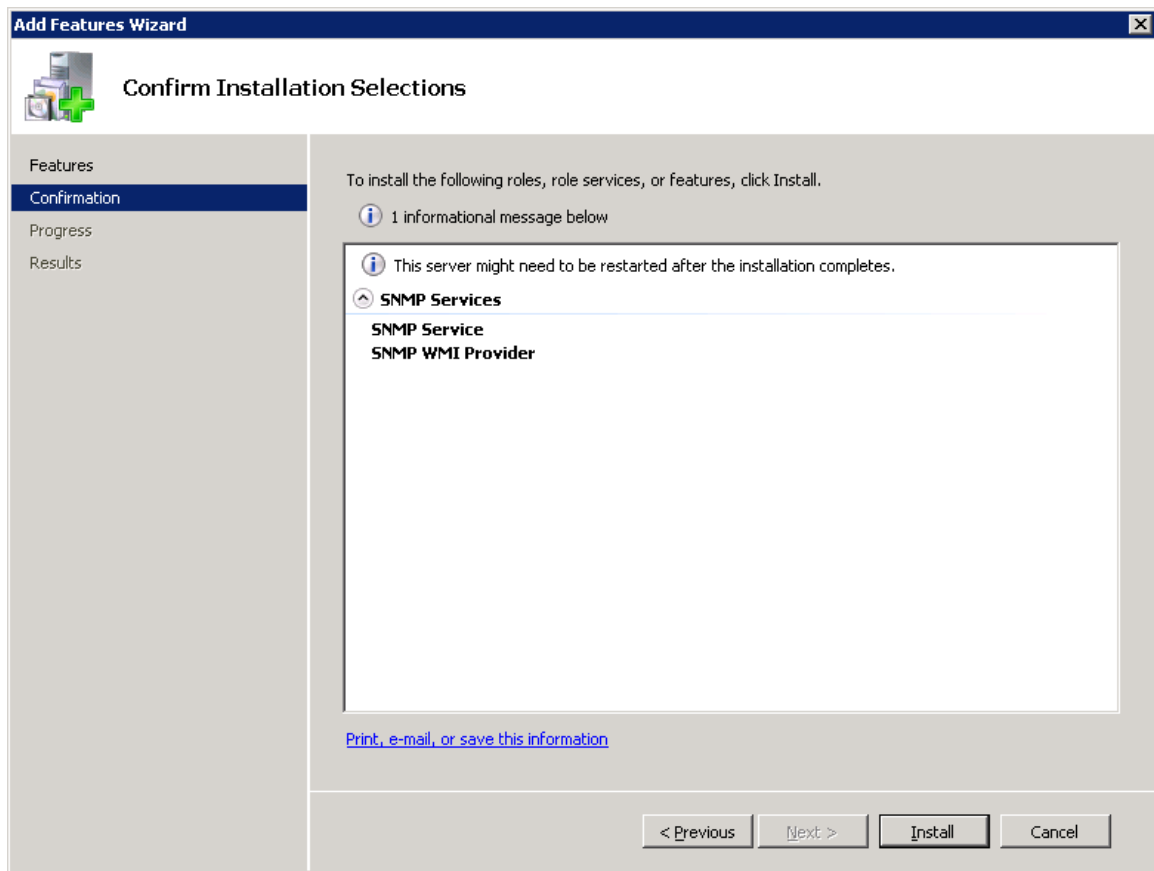
1. Open the Microsoft® Server Manager via **Start > Server Manager**.
2. In the **Server Manager** window, go to **Features** and click the **Add Features** link.



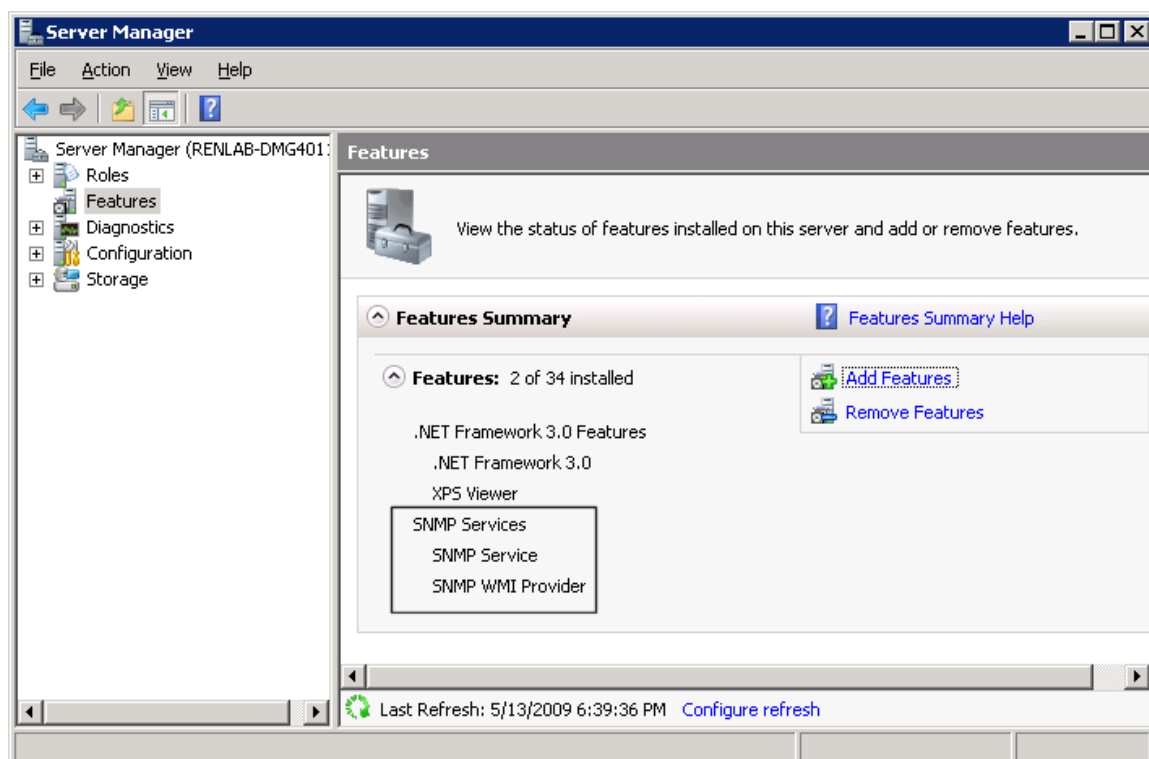
3. In the **Add Features Wizard** window, select **SNMP Services** and click **Next**.



4. Click Install, to install 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 in [Adding the SNMP Service in the Dialogic® Diva® Configuration Manager](#).

Adding the SNMP Service in the Dialogic® Diva® Configuration Manager

To add the SNMP service in the Dialogic® Diva® Configuration Manager, follow these steps:

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.

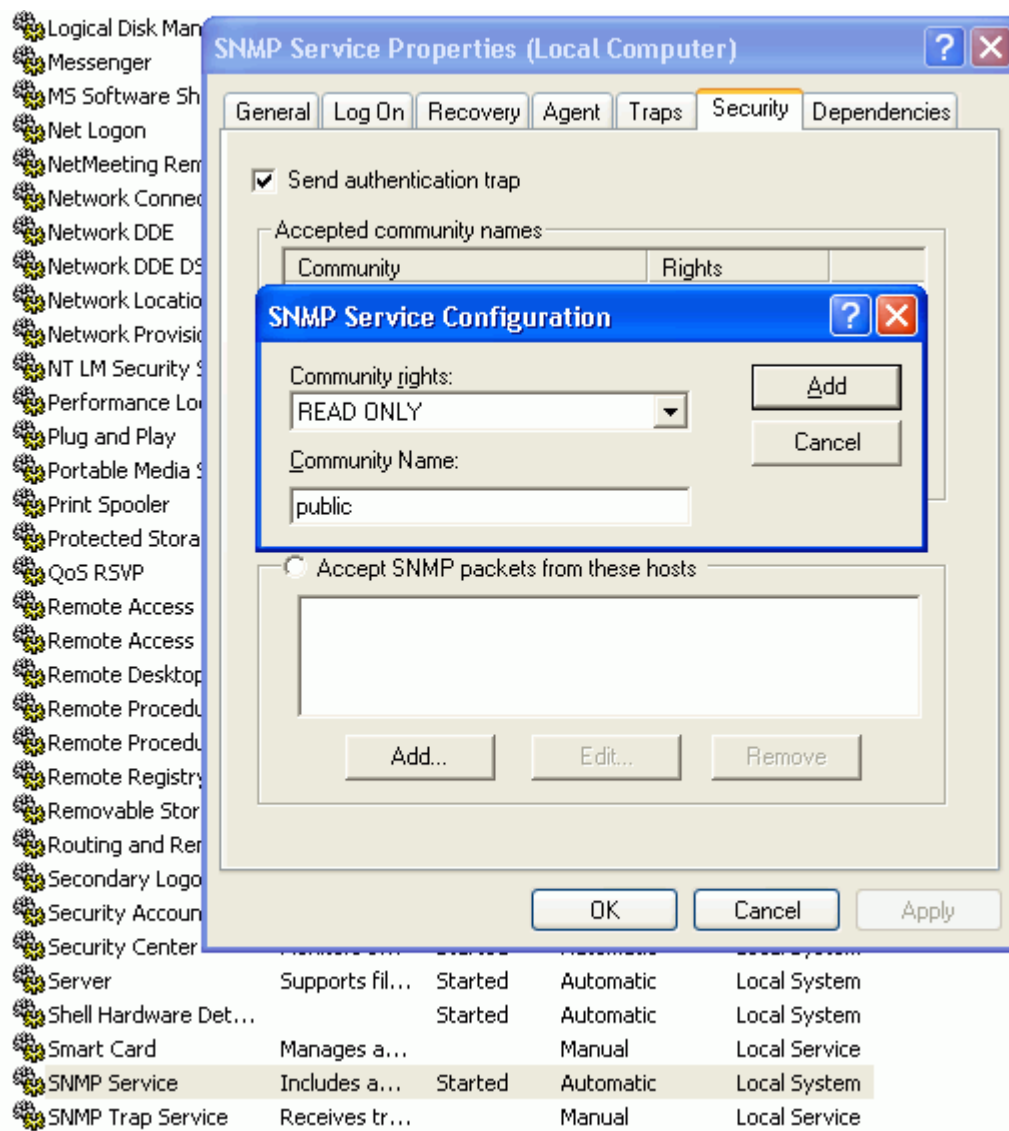
Note: You do not need to connect the SNMP service to any Dialogic® Diva® Media Board. The SNMP is always available for all installed Diva Media Boards.

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

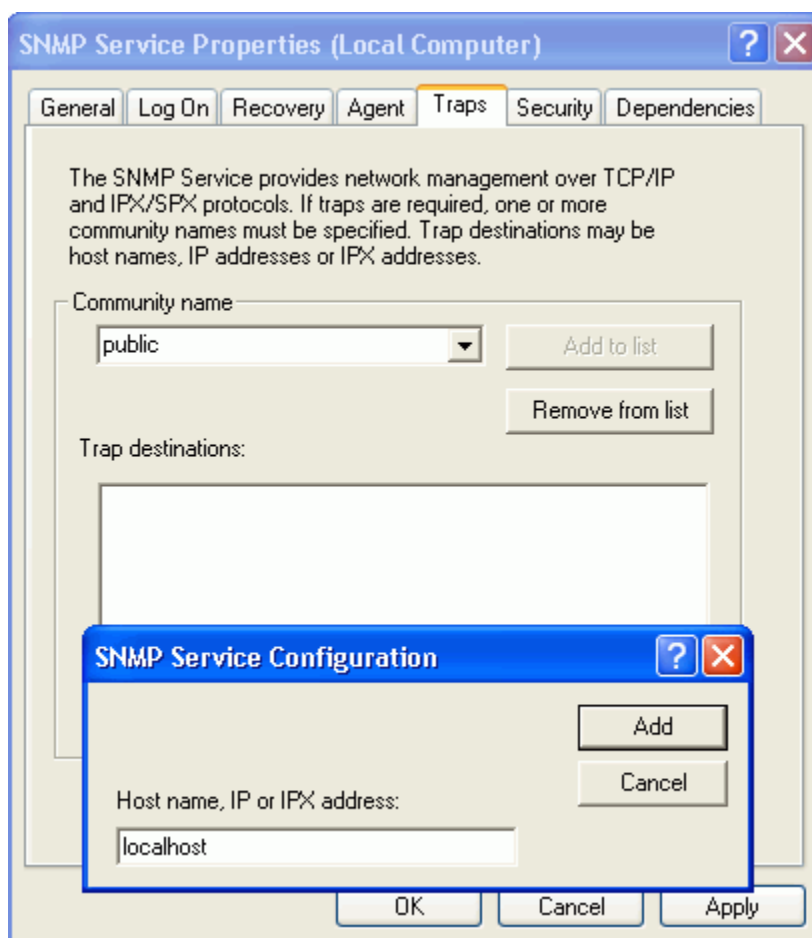
Verifying the SNMP Service Status

To verify the SNMP service status, follow these steps:

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.

Verifying the Function of the SNMP Service

To verify the function of the SNMP service, follow these steps:

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 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 Diva SNMP service and about the relationship between supported OIDs and Diva Media Board management interface variables.

MIB-II (RFC 1213/2233)	Path	Description
MIB-II	interfaces.ifTable.ifEntry.	
	ifIndex	Unique index of Diva interfaces starting with ifIndex-offset + 1 (see option -oN). First, all installed 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 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 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	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 subaddress
	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**.

14. Verifying the Line Configuration with the Dialogic® Diva® Line Test Tool

How to Verify the Line Configuration with the Dialogic® Diva® Line Test Tool

To check the line configuration, use the Diva Line Test tool available under **Start > Programs > Dialogic Diva > Line Test**. You need to login via Remote Desktop or on the local console with local administrative rights to use this tool.

The Diva Line Test tool offers the following tests:

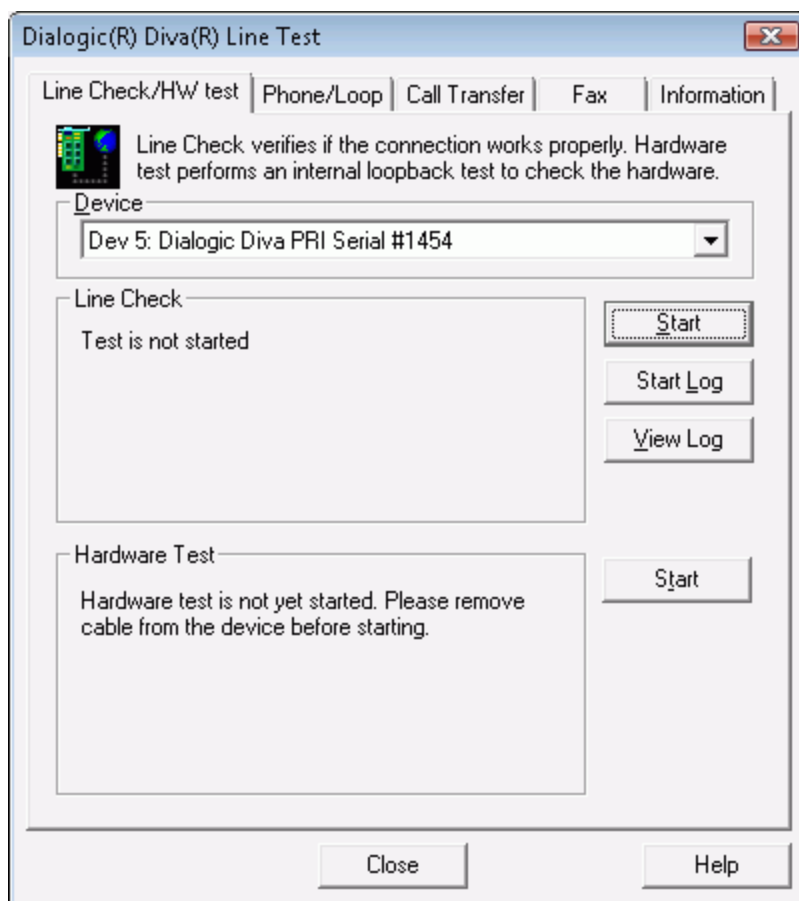
- **Line Check test**: Performs a quick check of your Diva System Release software installation and the physical connection.
- **Hardware test**: Performs a test of the physical controller only.
- **Phone/Loop test**: Performs basic inbound or outbound phone tests to test the connection to other telephones or to itself.
- **Call Transfer test**: 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.

Performing a Line Check Test

To perform a line check test with the Diva Line Test tool, follow these steps:

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



2. Under **Device** select the line of the 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.
 - Switch type, network type, and ISDN or phone numbers are configured correctly in the Diva Configuration Manager.
 - SPIDs (Service Profile Identifiers) are configured correctly in the Diva Configuration Manager if you use a North American switch type.
 - Diva Media Board is not conflicting with any other hardware.
 - 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.

Performing a 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.

To perform a hardware test, follow these steps:

1. Open the Diva Line Test tool and click the **Line Check/HW Test** tab.
2. Select under **Device** the line of the 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
- Diva Media Board is loaded correctly
- Newest Diva System Release software is installed
- 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](#).

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.

Performing a 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 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](#).

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](#).
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".

Performing a Call Transfer Test

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 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](#).
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](#).
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 Diva Line Test tool is opened.

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

- Call should be transferred without consultation call, or
- Call should be answered and the incoming call should be placed on hold, or
- 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.

Performing a Fax Test

With the fax test you can send or receive a fax document that you can view with a fax viewer. To perform a fax test, follow these steps:

1. Under **Device** select the line of the 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 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 [Setting Up a Test for Incoming Fax Calls](#).

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](#).
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.

Setting Up a Test for Incoming Fax Calls

To set up a test for incoming fax calls, follow these steps:

1. Open two 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. For more information, see [Customer Service](#).

Writing a Message into a Trace File

To write a message into a trace file, follow these steps:

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 Diva Line Test Tool is open. Once you close the tool, the unsaved wave file will be deleted.

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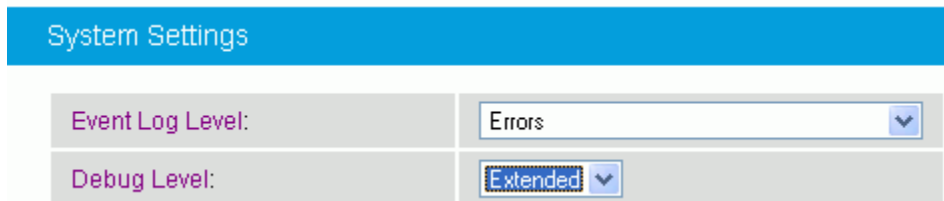
15. Creating a Trace with the Dialogic® Diva® Diagnostics Tool

How to Create a Trace with the Dialogic® Diva® Diagnostics Tool

To create a trace for the DMG4000 Gateway, you need to set the correct debug level in the Diva SIPcontrol web interface first. Then you can create a trace in the Diva Diagnostics tool. You need to login via Remote Desktop or on the local console with local administrative rights to use this tool.

To set the correct debug level in the Diva SIPcontrol 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**.

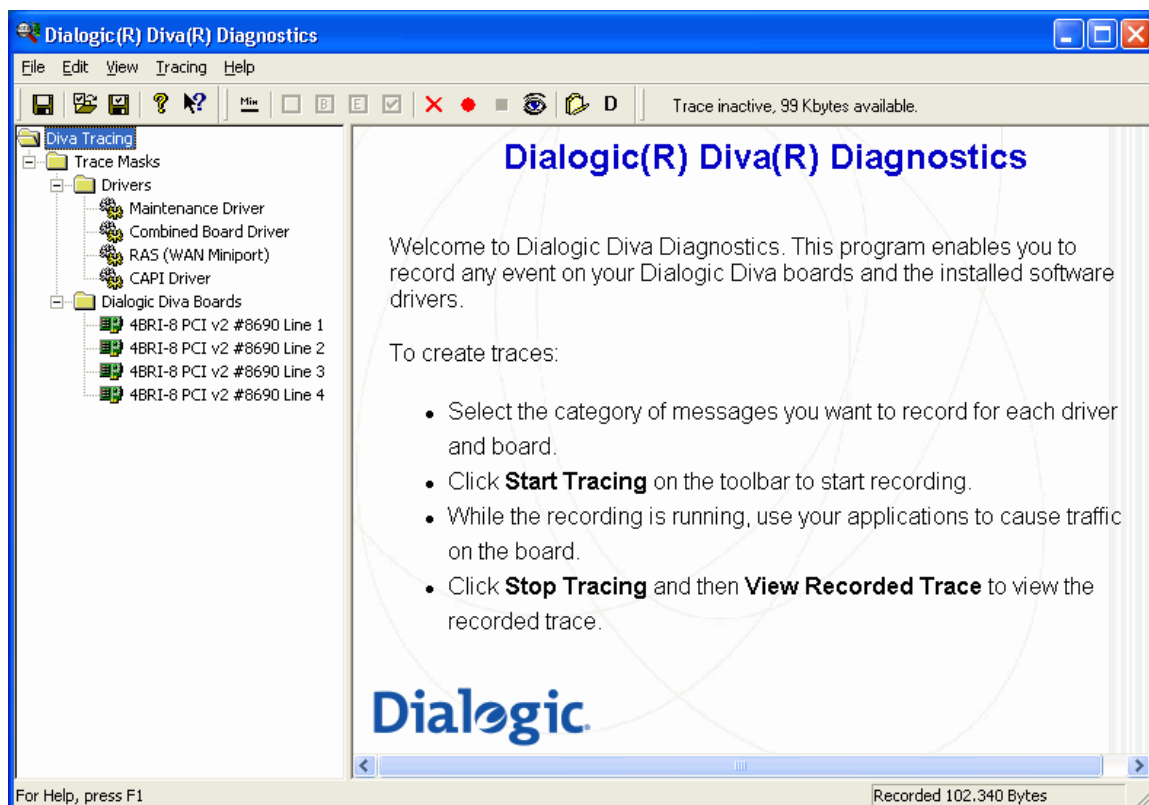


The screenshot shows a web interface titled "System Settings". Below the title, there are two rows of settings. The first row is labeled "Event Log Level:" and has a dropdown menu set to "Errors". The second row is labeled "Debug Level:" and has a dropdown menu set to "Extended".

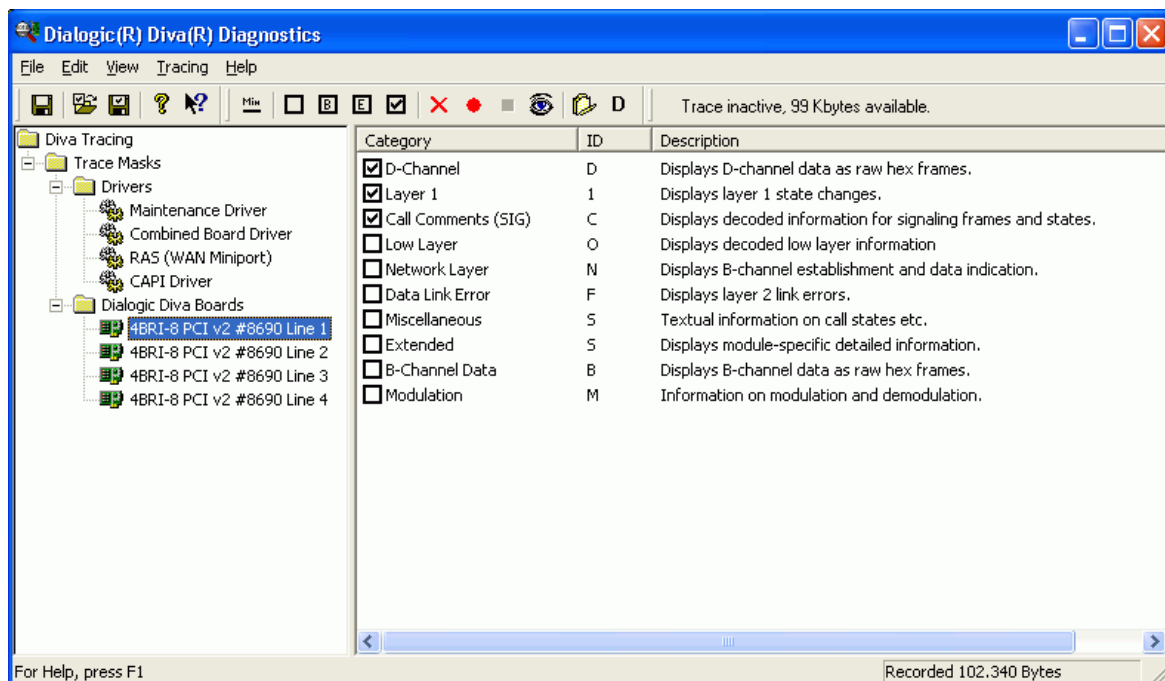
System Settings	
Event Log Level:	Errors
Debug Level:	Extended


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 Diva Media Boards.



2. Click the controller of the 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**.

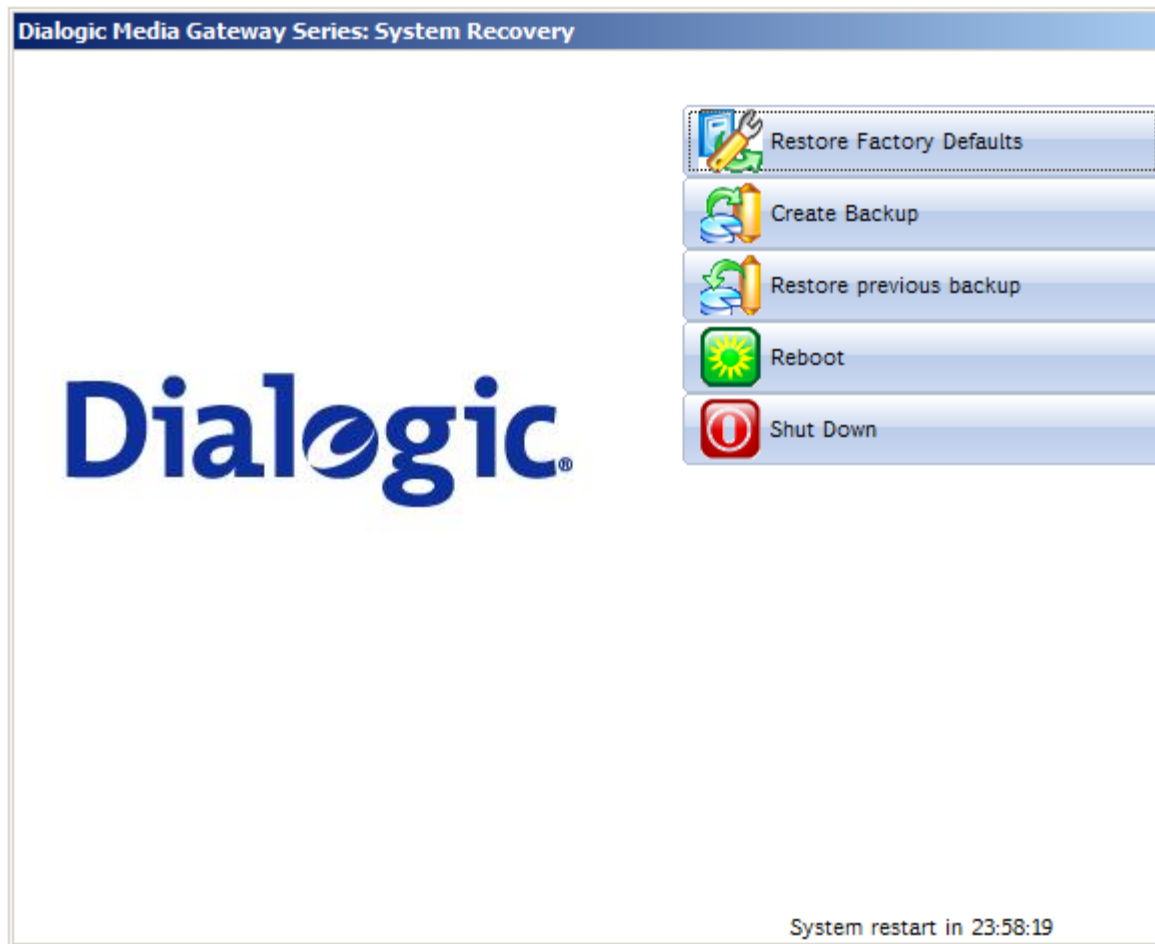
For detailed information about the Diva Diagnostics tool, see the Dialogic® Diva® Diagnostics Online Help under **Help > Help Topics**.

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16. Backing Up and Restoring the Configuration

How to Back Up and Restore the Configuration

The DMG4000 Gateways provide a restore and backup menu. You can access this menu if you press the left ALT key during the boot sequence. The menu provides the following options:



Restore Factory Defaults: The system partition C: is set to default factory state. If you choose this option and you accept to restore the factory defaults, the current system partition will be deleted.

Create Backup: An image of the current DMG4000 Gateway system partition C:, including all settings, is created and stored under D:/backup. The backup is overwritten every time you create a new backup. If you need to keep a backup with certain settings, save it in a different directory or on an external medium.

Restore previous backup: With this option, you can restore a previously saved backup of the system partition, including all DMG4000 Gateway settings. This might be necessary if you need to re-activate the settings of a configuration you saved earlier. If you choose this option, the current system partition C: will be deleted.

Reboot: Restarts the DDMG4000 Gateway.

Shut Down: Shuts down the DMG4000 Gateway.

If you do not choose any of the settings, the DMG4000 Gateway will be restarted after the time displayed at the bottom of the dialog box has elapsed.

17. Customer Service

Customer Service

Dialogic provides various options and arrangements for obtaining technical support for your Dialogic® product. We recommend that you use the Diva Support Tools first before contacting your Dialogic supplier. Also, we suggest that you visit Dialogic Technical Services & Support site, as it includes detailed information about a variety of topics. In the unusual case that neither your supplier nor the information on the Services & Support site is able to 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:

- 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.
- Diva Diagnostics: With the Diva Diagnostics tool, you can write traces for each Diva Media Board or driver into a file.
- 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.

For more information about the Diva Support Tools, see the respective online help files.

If you cannot address the 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 Services & Support web site, where you can find:

- A help web section for Dialogic® products at <http://www.dialogic.com/support/helpweb>
- A download section to install the newest version of your software at <http://www.dialogic.com/support/downind.asp>
- A training section with information about webinars, online, and onsite trainings at <http://www.dialogic.com/training/default.htm>
- A manuals section that provides a complete set of available documentation at <http://www.dialogic.com/manuals/default.htm>
- Technical discussion forums about different developer-specific Q&A at <http://www.dialogic.com/den/groups/developers/default.aspx>
- The Dialogic Customer Support site. For detailed information about how to contact Dialogic Customer Support, see [Dialogic Customer Support](#).

Dialogic Customer Support

If the information on the Services & Support site was not sufficient to help you address your issue, contact Dialogic Customer Support. See www.dialogic.com/support/contact for details.

Please note that when you contact Dialogic Customer Support, you may need to provide or have handy one or more of the following:

- A debug trace (see the Dialogic® Diva® Diagnostics Online Help file - DivaTrace.chm.)
- A copy of your active Diva System Release configuration (see the Dialogic® Diva® Configuration Manager Online Help file - *DSMain.chm*).
- A copy of your Diva SIPcontrol configuration.

To save a copy of your Diva SIPcontrol configuration, follow these steps:

1. Access the Diva SIPcontrol web interface.
2. In the Overview section, click **Save GUI settings**.
A User Prompt dialog appears.
3. Enter a name for the saved profile, and click **OK**. The profile name can contain characters, digits, and the underscore character (_).
4. In the Config.-Profiles field in the Overview section, select the saved profile, and click **Export to file**.
The File Download dialog appears.
5. Click **Save** to save the exported file, and then follow the prompts.
6. Send the saved file to Dialogic Customer Support.