



Dialogic® Diva® SIPcontrol™ Software v1.6

Reference Guide

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License

Under the terms and conditions of this Agreement:

- You may install and use one copy of the Program on a single-user computer, file server, or on a workstation of a local area network.
- Some or all functions of the Program may be available solely if the Program is used with one or more legally acquired Dialogic Activation Key(s).
- To obtain an Activation Key You must first purchase a Proof of Purchase Code (PPC). A PPC may be included in Your software or hardware package or You may have to purchase it separately.
- You will receive Your Activation Key upon registering the Proof of Purchase Code as directed in the PPC document.
- It may be possible to install multiple Activation Keys into the Program; in such a case, the total functionality provided by the Program will be the sum of the licensed functionalities controlled by the installed Activation Keys as long as the maximum capabilities of the Program are not exceeded and the functionalities are compatible.
- Your Activation Key(s) will restrict Your use of the Program. At least one of the following restriction schemes will be available to You when You register each PPC and request an Activation Key.
 - The Activation Key may be associated with a specific Dialogic hardware device. In this case, the licensed functionality controlled by the Activation Key will be available solely if the same Dialogic hardware is present in the computer. You can move the Program to another computer solely if You move the specified Dialogic hardware to the new computer.
 - The Activation Key may be associated with a specific Dialogic-supplied software protection device ('dongle'). In this case, the licensed functionality controlled by the Activation Key will be available solely if the same dongle is present in the computer. You can move the Program to another computer solely if You also move the dongle to the new computer.
 - The Activation Key may be associated with Your specific computer hardware platform. In this case, the licensed functionality controlled by the Activation Key will be available solely if no significant change is made to the hardware installed in the computer. Replacement Activation Keys may be issued at the discretion of Dialogic solely if Dialogic can determine that You have not moved the Program to another computer. Sufficient information must be provided to Dialogic to allow it to make that determination.
- In addition to the above restrictions, each Activation Key may have a specific term of use commencing from the date of PPC registration. In this case, the licensed functionality controlled by the Activation Key will not be available after the Activation Key has expired.
- The Activation process requires that You enter the following information into the web-based system to obtain an Activation Key:
 - PPC
 - The Device ID provided to You by the "Activation" function in the Program
 - Your email address so that the Activation Key can be delivered to You by email
- Dialogic will retain the information above for the following purposes:
 - Validation of future requests from You for replacement Activation Keys
 - Sending renewal reminders to You in the case of limited time licenses.
- If the Dialogic hardware device that Activation Keys are associated with is judged to be defective by Dialogic following its standard practices, Dialogic's Support department will issue to You replacement Activation Keys associated with the replacement device upon receipt of the faulty device by Dialogic. Replacement of the faulty device is subject to the terms

of Dialogic's standard Hardware product Warranty in effect at the time You purchased the hardware product concerned ("Hardware Warranty"). If a valid Advance Replacement Insurance policy contract is in place for the Dialogic hardware product concerned, Dialogic will endeavor to expedite provision of Activation Keys associated with the replacement device.

- If the Dialogic-supplied software protection device (a "dongle" or "USB-stick") that Activation Keys are associated with is judged to be defective by Dialogic following its standard practices, Dialogic Support will issue to You replacement Activation Keys associated with the replacement device upon receipt by Dialogic of the failed device. Replacement of the faulty device is subject to the terms of Dialogic's Hardware Warranty.
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- You may make one copy of Dialogic's documentation provided that all copyright notices contained within the documentation are retained;
- You may transfer the Program, documentation and the license to another eligible party within Your Company if the other party agrees to accept the terms and conditions of this Agreement. If You transfer the Program and documentation You must at the same time either transfer all copies whether in printed or machine readable form to the same party or destroy any copies not transferred;
- You may not rent or lease the Program. You may not reverse engineer, decompile or disassemble the Program. You may not use, copy, modify or transfer the Program and documentation, or any copy, modification or merged portion, in whole or in part, except as expressly provided for in this Agreement;
- You may not modify the Program in order to circumvent or subvert the protection mechanisms inherent in the program or attempt to use a time-limited Activation Key after it has expired;
- If You transfer possession of any copy, modification or merged portion of the Program or documentation to another party in any way other than as expressly permitted in this Agreement, this license is automatically terminated.

Term

The license is effective until terminated. You may terminate it at any time by destroying the Program and documentation together with all copies, modifications and merged portions in any form.

It will also terminate upon conditions set forth elsewhere in this Agreement or if You fail to comply with any terms or conditions of this Agreement at any time. You agree upon such termination to destroy the Program and documentation together with all copies, modifications and merged portions in any form.

Limited Warranty

The only warranty Dialogic makes, beyond the replacement of Activation Keys under the terms set out above, is that the medium on which the Program is recorded will be replaced without charge if Dialogic, in good faith, determines that it was defective in materials or workmanship and if returned to Your supplier with a copy of Your receipt within ninety (90) days from the date You received it. Dialogic offers no warranty for Your reproduction of the Program. This Limited Warranty is void if failure of the Program has resulted from accident, misuse, abuse or misapplication.

Customer Remedies

Dialogic's entire liability and Your and Your Authorized Users exclusive remedy shall be, at Dialogic's option, either (a) return of the price paid or (b) repair or replacement of the Program that does not meet the above Limited Warranty. Any replacement Program will be warranted for the remainder of the original Warranty period.

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Dialogic disclaims all other warranties, either express or implied, including but not limited to implied warranties or merchantability and fitness for a particular purpose and the warranty against latent defects, with respect to the Program and the accompanying documentation. This limited warranty gives You specific legal rights. You may have others, which may vary from jurisdiction to jurisdiction.

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Limit of Liability

Dialogic's entire aggregate liability under any provision of this Agreement shall be limited to the amount actually paid by You for the affected Program.

Right to Audit

If this Program is licensed for use in a Company, Your Company agrees to keep all usual and proper records and books of accounts and all usual proper entries relating to each installation of the Program during the term of this Agreement and for a period of three (3) years thereafter. During this period, Dialogic may cause an audit to be made of the applicable records and of the installations of the Program in order to verify Your compliance with this Agreement and prompt adjustment shall be made to compensate for any errors or omissions disclosed by such audit. Any such audit shall be conducted by an independent certified public accountant selected by Dialogic and shall be conducted during the regular business hours at Your offices and in such a manner as not to interfere with Your normal business activities. Any such audit shall be paid for by Dialogic unless material discrepancies are disclosed.

For such purposes, 'material discrepancies' shall mean the Company exceeding by three percent (3%) or more the number of licensed channels for any function of the Program or the Company exceeding the licensed number of Authorized Users by three percent (3%) or more. If material discrepancies are disclosed, Your Company agrees to pay Dialogic for the costs associated with the audit as well as the license fees for the additional licensed channels or additional Authorized Users. In no event shall audits be made more frequently than semi-annually unless the immediately preceding audit disclosed a material discrepancy.

Supplementary Software

Any Supplementary Software provided with the Dialogic Program referred to in this License Agreement is provided "as is" with no warranty of any kind.

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This Agreement shall be construed and controlled by the laws in force in the Province of Quebec, Canada with the exception of its conflict of laws rules. The United Nations Convention on Contracts for the Sale of Goods does not apply to this Agreement.

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Contractor/ manufacturer is:

DIALOGIC CORPORATION.

9800 Cavendish Blvd., Montreal, Quebec, Canada H4M 2V9

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

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CHAPTER 1

About This Online Guide

How to use this online guide

- To view a section, click the corresponding bookmark located on the left.
- To view a topic that contains further information, click the corresponding blue underlined phrase.
- You may wish to print out the pages required for installing the drivers.

Structure of this guide

This guide is structured as follows:

Section	Contents
About the Dialogic® Diva® SIPcontrol™ Software	General information about Diva SIPcontrol software, general features, supported Dialogic® Diva® Media Boards, and supported operating systems
Software Installation	Installation of the Diva SIPcontrol software with the setup wizard
License Activation	Activation of the license on the Dialogic activation web site
Configuration	Overview of configuration parameters
How Calls Are Processed	Description of the processing of calls
How Call Addresses Are Processed	Description of the processing of call addresses
How Address Maps Are Processed	Description of the processing of address maps
How Numbers Are Processed	Description of the number normalization with the Diva SIPcontrol software
Software Uninstallation	Uninstallation of the Diva SIPcontrol software
Customer Service	Information on how to get technical support for Dialogic® Diva® products
Appendix	Description of Cause Code mappings, Error codes, and Routing Examples

CHAPTER 2

About the Dialogic® Diva® SIPcontrol™ Software

The Dialogic Diva SIPcontrol software is a gateway that translates call control information from the PSTN into SIP messages and vice versa. The Diva SIPcontrol software is installed on top of a Dialogic® Diva® Media Board, allowing the Diva board to be used as a SIP gateway within the computer or server that hosts the media server platform. The Diva SIPcontrol software is delivered with a default license for the simultaneous use of eight channels.

New features in the Dialogic® Diva® SIPcontrol™ software version 1.6

Enhanced routing

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

Enhanced address manipulation

- define the number manipulation on three different stages of the call routing (inbound, route selection, outbound)
- reduced complexity of the regular expressions introduced in version 1.5.1 of the Dialogic® Diva® SIPcontrol™ software with the introduction of address maps
- different dialplans can be entered for each controller and each SIP peer, which can ease the deployment in a multiple location environment

Improved support for North-American numbering plan

- support for configuration of multiple area codes to be handled as local; therefore, the Dialogic® Diva® SIPcontrol™ dialplan engine is able to automatically format dialed numbers according to local phone provider requirements without any additional regular expressions

New configuration frontend

- standard web browsers can be used for configuration
- remote configuration of the Diva SIPcontrol software from any computer in the network

Configuration changes during runtime

- modify most parameters of the Diva SIPcontrol software without the need to restart the service; active calls are not affected by configuration updates and continue undisturbed

New SIP feature

- support for SIP session timer (RFC4028)

Configurable cause codes

- configurable translation of ISDN cause code to SIP response code and vice versa; consequently, the Diva SIPcontrol software can adapt to the specific behavior of used the PSTN PBX and/or SIP peer

Feature Overview

General feature

- Configuration via web interface

Call handling

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

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

Media processing

- G.711 A-Law and u-Law
- RTP dynamic payload audio/telephony event
- RTP profile name RTP/AVP
- DTMF via RTP payload (RFC 2833)
- PSTN-side fax tone detection via RTP event (RFC 2833)
- Echo Canceller with 128 ms echo tail used with RTP
- T.38 Fax up to V.34 (SuperG3 Fax)

Reliability

- Load Balancing and Failover on PSTN side
- Load Balancing and Failover on SIP side (uses OPTIONS for Keep-Alive check)
- Alive check for active calls on SIP side via SIP session timer

Supported RFCs

- RFC3261 - Session Initiation Protocol
- RFC2833 - RTP Payload for DTMF Digits, Telephony Tones and Telephony Signals
- RFC3264 - An Offer/Answer Model with Session Description Protocol
- RFC3265 - SIP-specific Event Notification

- RFC3389 - RTP Payload for Comfort Noise
- RFC3398 - ISDN to SIP mapping
- RFC3515 - REFER method
- RFC3550 - Realtime Transport Protocol (RTP)
- RFC3842 - Message Waiting Indication for SIP
- RFC3891 - SIP "Replaces" header
- RFC3892 - SIP Referred - By Mechanism
- RFC4028 - Session Timers in SIP
- RFC4497 - Interworking between SIP and QSIG
- RFC4566 - Session Description Protocol (SDP)
- Draft: Diversion Indication in SIP (draft-levy-sip-diversion-08)

Supported hardware

The Dialogic® Diva® SIPcontrol™ software supports the following Dialogic® Diva® Media Boards (up to 120 channels are supported):

Dialogic® Diva® ISDN BRI Media Boards

- Diva BRI-2M 2.0
- Diva V-BRI-2
- Diva 4BRI-8M 2.0
- Diva V-4BRI-8

Dialogic® Diva® ISDN PRI Media Boards**Diva PRI 3.0:**

- Diva PRI/E1/T1-8
- Diva PRI/T1-24
- Diva PRI/E1-30

Diva V-PRI:

- Diva V-PRI/T1-24
- Diva V-PRI/E1-30

Diva Multiport PRI:

- Diva V-2PRI/E1/T1
- Diva V-4PRI/E1/T1

Diva PRI PCIe:

- Diva PRI/T1-24 PCIe
- Diva PRI/E1-30 PCIe

Diva V-PRI PCIe:

- Diva PRI/T1-24 PCIe
- Diva PRI/E1-30 PCIe

Dialogic® Diva® Analog Media Boards

- Diva Analog-2
- Diva V-Analog-4
- Diva Analog-4
- Diva V-Analog-8
- Diva Analog-8

Supported software

The Dialogic® Diva® SIPcontrol™ software supports Dialogic® Diva® for Windows® software version 8.3 or higher.

Supported operating systems

The Dialogic® Diva® SIPcontrol™ software supports the following operating systems:

- Windows® XP 32-bit
- Windows Server® 2003 32-bit and 64-bit

CHAPTER 3

Software Installation

To install the Dialogic® Diva® SIPcontrol™ software, use the Dialogic® Diva® SIPcontrol™ Setup Wizard as described below:

Note: If you want to upgrade from Diva SIPcontrol software version 1.5.1, DO NOT uninstall the software before you install the Diva SIPcontrol software version 1.6, otherwise you might lose some settings, including your regular expressions.

1. Insert your Dialogic® Diva® Media Board into the computer as described in the installation guide that came with your Diva board.
2. Install the Dialogic® Diva® for Windows® software as described in the "Dialogic® Diva® for Windows® Reference Guide". We recommend to install version 8.3.2, but the Diva SIPcontrol software also runs on 8.3 of the Diva for Windows software. The Reference Guide is available on your Dialogic® Diva® Software Suite CD-ROM, or on the Dialogic web site under: www.dialogic.com.

Note: If the correct version of the Diva software (8.3 or higher) is not detected during the installation, an error message is displayed and the installation is aborted.

3. Go to the directory in which the Windows® installer package "DSSIPControl.msi" is located and double-click it.
4. In the welcome dialog box, click **Next**.
5. The **End-User License Agreement** box appears. Accept the license agreement to start the installation.
6. If you are upgrading from a former version of the Diva SIPcontrol software and you kept the configuration files, the **Existing configuration found** box appears. Select whether you want to reuse the existing configuration files and click **Next**.
7. The **Ready to Install the Program** box appears. Click **Install** to install the Diva SIPcontrol software.
8. If the installation terminates prematurely, verify that:
 - the installed Diva Media Board is supported; see [Supported hardware](#) on page 11 for more information,
 - the correct Diva for Windows software is installed; see [Supported software](#) on page 12 for more information,
 - the operating system is supported by the Diva SIPcontrol software; see [Supported operating systems](#) on page 12 for more information,
 - the CAPI driver is installed correctly, inserted in the Dialogic® Diva® Configuration Manager and connected to the Diva Media Board.

If the installation still cannot be completed despite of that, contact Dialogic Customer Support under www.dialogic.com/support.

9. After the installation is complete, the **Completing the Diva SIPcontrol Wizard** box appears. Click **Finish** to exit the installation.
10. Now, you can configure the settings. To do so, click **Start > Programs > Diva for Windows > SIPcontrol configuration**. If you need help during the configuration, click the parameter to display its online help text.

CHAPTER 4

License Activation

Note: The Dialogic® Diva® SIPcontrol™ software includes a default license for eight (8) channels. If you require more than eight channels simultaneously, then you will need to purchase another license. After purchasing the license, you will need to generate and activate it to unlock functionality in the product. As soon as you have activated the purchased license, the channels of the default license cannot be used any longer.

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 the 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 Dialogic® Diva® SIPcontrol™ software configuration web interface. For more information, see [To activate the license file](#).

Device Unique ID (DUID)

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

To get the DUID:

1. Click **Start > Programs > Diva for Windows > SIPcontrol configuration** to open the Dialogic® Diva® SIPcontrol™ software configuration web interface.
2. Click **License Management** on the left side of the Diva SIPcontrol software configuration web interface to open the **License Status** dialog.
3. In the **License Status** dialog, copy the DUID number of the Dialogic® 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 the Dialogic® Diva® SIPcontrol™ software 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**.

Diva Activation

http://www.dialogic.com/activate/akiweb_req.asp

Date: Bearbeiten Ansicht Favoriten Extras ?

★ ★ ★ Diva Activation

Dialogic WORLDWIDE | CONTACT | DEVELOPER RESOURCE CENTER | PARTNER RESOURCE CENTER | SITEMAP GO

HOME | PRODUCTS | PURCHASE | PARTNERS | SERVICES & SUPPORT | NEWS & EVENTS | ABOUT US

Dialogic Diva Activation

PPC

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

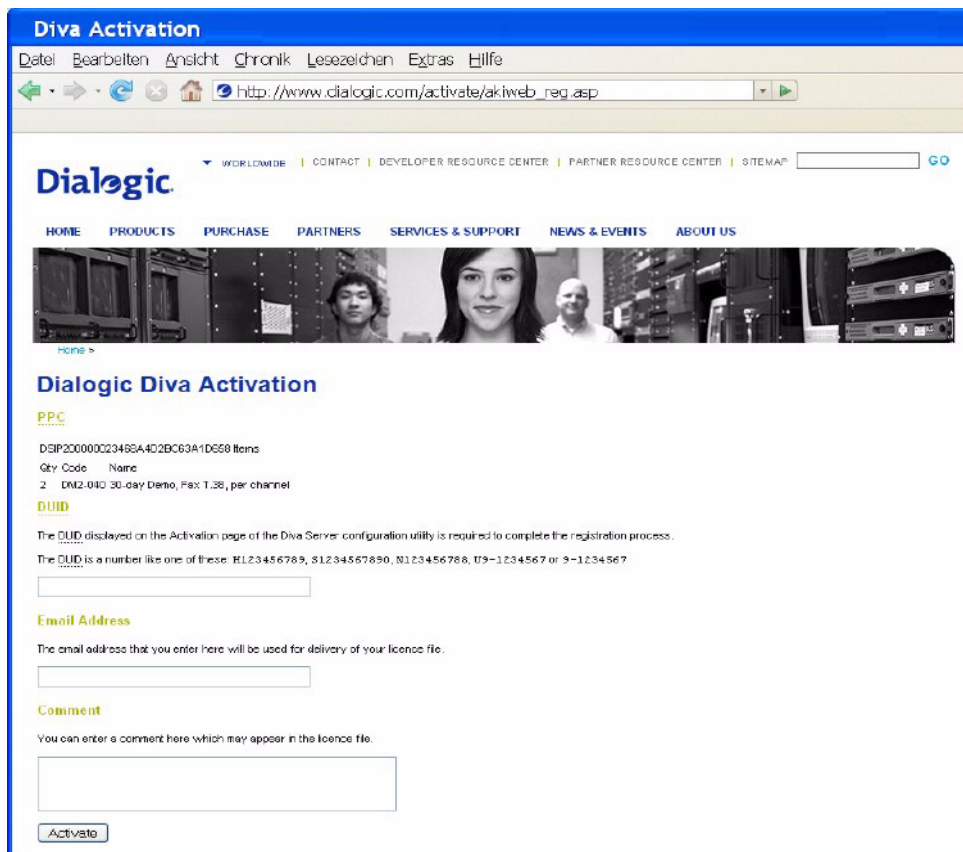
The PPC is a string of letters and digits similar to this: DSIF10000101A1G0F86GF624D0D9C0

Check

Seeing Beyond Tomorrow

HOME | PRODUCTS | PURCHASE | PARTNERS | DEVELOPERS | SERVICES & SUPPORT | DOWNLOADS | CONTACT | ABOUT US
LEGAL | PRIVACY POLICY | SITEMAP | **RSS** | CAREERS | NEWSLETTER SIGNUP

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



Diva Activation

Dialogic

HOME PRODUCTS PURCHASE PARTNERS SERVICES & SUPPORT NEWS & EVENTS ABOUT US

Dialogic Diva Activation

PPC

DSP20000023465A402BC63A1D658 Items

Qty Code Name

2 DM2-040 30-day Demo, Fax T.38, per channel

DUID

The DUID displayed on the Activation page of the Diva Server configuration utility is required to complete the registration process.

The DUID is a number like one of these: 8123456789, 81234567890, 8123456788, 09-1234567 or 9-1234567

Email Address

The email address that you enter here will be used for delivery of your license file.

Comment

You can enter a comment here which may appear in the license file.

Paste your Device Unique ID (DUID) that you saved earlier, and enter your email address to which the license file should be sent.

4. Click **Activate** to generate the license file that will be sent to the email address you have entered.
5. Save the license file and activate it. For more information, see [To activate the license file](#) below.

To activate the license file

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

1. Click **License Management** on the left side of the Dialogic® Diva® SIPcontrol™ software configuration web interface to open the **License Status** dialog.
2. In the **License Status** dialog, click **Browse**, go to the directory in which you saved the license file, and click **Open**.
3. Click **Upload** to activate the license file.

CHAPTER 5

Configuration

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

To open the Diva SIPcontrol software web interface:

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

To do so:

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

To do so:

- Click **Start > Settings > Control Panel > Administrative Tools**.
- In the **Administrative Tools** window, select **Services**.
- In the **Services** window, right-click the **Dialogic Diva WebConfig** service and select **Restart**.

Now, you can access the Diva SIPcontrol software web interface on any of the IP addresses of the computer where SIPcontrol is installed and configure the setting according to your needs. The following parameters can be configured:

- [PSTN Interface Configuration](#) on page 18
- [Network Interface Configuration](#) on page 21
- [SIP Peer Configuration](#) on page 22
- [Routing Configuration](#) on page 25
- [Dialplan Configuration](#) on page 27
- [Address Map Configuration](#) on page 29
- [Cause Code Mapping Configuration](#) on page 30
- [System Settings](#) on page 31

Mandatory configurations are:

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

Configuration Tips and Hints

- All 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.
- In the configuration the term "inbound" refers to settings that need to be valid for calls coming from the PSTN or SIP to the computer on which the Dialogic® Diva® SIPcontrol™ software is installed. Respectively, "outbound" refers to settings valid for calls from SIPcontrol to the PSTN or SIP.
- Changes to the configuration will only take effect after you clicked **Save** at the bottom of the main configuration page.
- Some changes require a restart of the Diva SIPcontrol software. Such changes include changing the IP address of the operating system, changing the default gateway, activating/deactivating the network adapter.
Note: The restart will terminate active connections.
- The configuration session times out after 30 minutes of inactivity and a new login is required to access the session again. If the new login screen appears when you try to save the configuration, login again and click the "Back" button of the browser. The configuration session opens with the settings before the time out and you can save the configuration.
- To remove the password login page, logout from the web interface and restart the Dialogic Diva WebConfig service as described in [Restart the Dialogic Diva WebConfig service](#), on page 17. Then, open the Windows Explorer, go to **Program Files > Diva Server > httpd > login**, and delete the login file.
- To open the online help for a specific parameter, click the parameter and a window with the help text pops up.

PSTN Interface Configuration

The PSTN configuration allows for configuring different settings per CAPI controller. The controllers are read in when the configuration page is loaded. Currently, the number and order of the controllers can be changed only with local console access to the Dialogic® Diva® SIPcontrol™ software computer via Dialogic® Diva® Configuration Manager. In the Diva Configuration Manager, delete the bindings between the CAPI service and the Dialogic® Diva® Media Board and reconnect them in the required order.

At least one controller must be enabled; otherwise, it is ignored both for inbound and outbound calls.

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

The following menus are available for configuration:

- [General Configuration](#) below
- [Address Normalization](#) on page 19
- [Call Transfer](#) on page 20
- [Message Waiting Indication](#) on page 21

General Configuration

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

General Configuration	
Hardware description:	Dialogic Diva PRI/E1/T1-8 PCI v3 SN: 1056
PSTN interface number:	2
Name:	<input type="text" value="Controller2"/>
Address map inbound:	<input type="button" value="none"/>
Address map outbound:	<input type="button" value="none"/>

Hardware description:	Displays the installed Dialogic® Diva® Media Board. This entry is predefined by the system and cannot be changed.
PSTN interface number:	Displays the number of the CAPI controller. The number is set automatically by the system.

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

Address map inbound: Select the name of a regular expression list to be applied on incoming calls on this interface. See [Address Map Configuration](#) on page 29 for more information about setting up a regular expression list. If you upgraded from Dialogic® Diva® SIPcontrol software version 1.5 or 1.5.1, an Address Map is automatically generated here to enforce the same number processing behaviour in Diva SIPcontrol software version 1.6 as in former Diva SIPcontrol software versions. If you used regular expressions in Diva SIPcontrol software version 1.5.1, they will be included in this address map as well, unless they cannot be converted to the new scheme. In this case, the entry **<Use Windows Registry values>** is available. The Diva SIPcontrol software will then use the regular expressions defined in the registry keys that were used by Diva SIPcontrol software 1.5.1.

Regular expressions may be used to add or remove dial prefixes required by a PBX or to rewrite public phone numbers of different number ranges into a common format. See the [Examples](#) on page 38 for more information.

Address map outbound: Select the name of a regular expression list to be applied on outgoing calls on this interface. See [Address Map Configuration](#) on page 29 for more information about setting up a regular expression list. If you upgraded from Dialogic® Diva® SIPcontrol software version 1.5 or 1.5.1, an Address Map is automatically generated here to enforce the same number processing behaviour in Diva SIPcontrol software version 1.6 as in former Diva SIPcontrol software versions. If you used regular expressions in Diva SIPcontrol software version 1.5.1, they will be included in this address map as well, unless they cannot be converted to the new scheme. In this case, the entry **<Use Windows Registry values>** is available. The Diva SIPcontrol software will then use the regular expressions defined in the registry keys that were used by Diva SIPcontrol software 1.5.1.

Regular expressions may be used to add or remove dial prefixes required by a PBX or to rewrite public phone numbers of different number ranges into a common format. See the [Examples](#) on page 38 for more information.

Address Normalization

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

Address Normalization	
Dialplan:	none ▼
Number format (outbound):	Unchanged ▼
Encoding (outbound):	Use type flag ▼
Default numbering plan:	unknown ▼

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

In most cases, all 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.

You need to configure the local dialplan as described under [Dialplan Configuration](#) on page 27 before you can select it here.

Number format (outbound): This parameter determines the shortest format allowed that is sent out in outbound calls. 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 if no country code is given or if the area code matches the location settings.

Extension: The number is converted to the extension only if no additional information is given or if the country / area code and basic phone number match the location settings.

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

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

Call Transfer

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

PSTN Call Transfer Settings	
The Call Transfer settings depend on the capabilities of the communication platform (PBX, switch).	
Blind Call Transfer (A- and C-Party on PSTN side)	
Transfer type:	With consultation call (Explicit Call Transfer) ▼
Invoke Call Transfer in state:	Proceeding ▼
Use same channel for consultation call:	<input type="checkbox"/>
Primary call on Hold before transfer:	<input type="checkbox"/>
Use tromboning if transfer fails (needs two bearer channels):	<input checked="" type="checkbox"/>
Supervised Call Transfer (A- and C-Party on PSTN side)	
Transfer type:	With consultation call (Explicit Call Transfer) ▼
Use tromboning if transfer fails (needs two bearer channels):	<input checked="" type="checkbox"/>

- Transfer type:** The following options are available:
- With consultation call (Call Deflection):** The call is transmitted automatically.
 - With consultation call (Explicit Call Transfer):** After the transfer to the destination party, the channel is freed. The transfer may be announced or unannounced.
 - With consultation call via tromboning:** The call transfer is emulated. Two B-channels are blocked during the call transfer.
- Complete transfer in state:** The blind call transfer is typically handled via an implicit call to the transfer destination. Once this call reaches the state specified via the option **Invoke Call Transfer in state**, the call transfer is completed. Default setting is **Connected**. If the calling party should hear the ring back tone from the transfer destination, this parameter must be set to **Proceeding** or **Alerting**.
- Use same channel for implicit call:** The B-channel used for the primary call is used for the consultation call as well. This requires that the option **Hold primary call before transfer** is enabled. For Dialogic® Diva® Analog Media Boards and protocols using inband signaling, this option must be enabled.
- Primary call on Hold before transfer:** Select this option to place the primary call on hold before a call to the transfer destination is initiated.
- Use tromboning if transfer fails (needs two bearer channels):** Select this option if the Call Transfer should be emulated in case it could not be transferred with **Call Deflection** or **Explicit Call Transfer**.

Message Waiting Indication

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

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 for updating of the message waiting indication. This authentication is done by a **Controlling user number**. The administrator of the PBX can provide this number.

Controlling user provided number:

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

Network Interface Configuration

The network interface configuration allows for configuring of the global network parameters of the Dialogic® Diva® SIPcontrol™ software, such as the IP addresses and the ports the Diva SIPcontrol software will be listening on. Diva SIPcontrol software v1.6 supports only one single IP address and port number.

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

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

Network Interface Configuration					
Name	Device	IP Address	Protocol	SIP Listen Port	Enabled
3Com 3C920 Integrated Fast Ether	3Com 3C920 Integrated Fast Ethernet Controller (3C905C-TX Compatible) - Packet Scheduler Miniport	192.168.212.72	all	9803	<input checked="" type="checkbox"/>
Local Loopback Interface	Local Loopback Interface	127.0.0.1	all	5060	<input type="checkbox"/>
RTP Start Port:	<input type="text" value="30000"/>				
RTP End Port:	<input type="text" value="39999"/>				
Jitterbuffer Size Min [ms]:	<input type="text" value="0"/>				

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 the Dialogic® Diva® SIPcontrol™ software is installed.

Note: If you have several network adapters pointing to the same network and you enabled one of these adapters to work with the Diva SIPcontrol software, you need to change the IP address of the adapter that is not enabled. Otherwise, calls cannot be processed correctly.

Protocol

From the dropdown menu, select the IP protocol supported in calls from SIP: either TCP, UDP or both.

SIP Listen Port

Port for incoming SIP calls. The standard port 5060 can be used if no other SIP application is running on the same computer as the Dialogic® Diva® SIPcontrol™ software.

If you disable this option, the Diva SIPcontrol software will not accept calls from SIP on this port.

Enabled

Enable the network interface to use for the configuration. Note that you can only enable one network interface.

RTP Start Port	Defines the lowest port of the range in which the Dialogic® Diva® SIPcontrol™ software receives RTP streams. Change this value only if problems occur.
RTP End Port	Defines the highest port of the range in which the Dialogic® Diva® SIPcontrol™ software receives RTP streams. Change this value only if problems occur.
Jitterbuffer Size Min (ms)	Specifies the minimum time in milliseconds used by the Dialogic® Diva® SIPcontrol™ software to buffer RTP data before streaming it to the ISDN. Normally, the value is adjusted automatically according to network quality and network usage, but in some cases it may need to be increased to prevent buffer underruns. However, increasing the value also increases latency and therefore it should be left at the default of 0 milliseconds.

SIP Peer Configuration

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

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

- [Edit SIP Peer Configuration](#)
- [Session Timer](#) on page 24
- [Address normalization](#) on page 24

Edit SIP Peer Configuration

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

Edit SIP Peer Configuration	
Name:	peer1
Peer type:	Default
Default SIP to PSTN peer:	<input checked="" type="checkbox"/>
Host:	
Port:	5060
IP protocol:	TCP
Domain:	
Display name to:	
Display name from:	
User name to:	
User name from:	
Force T.38 reinvite:	<input type="checkbox"/>
Alive check:	<input type="checkbox"/>
Address map inbound:	none
Address map outbound:	none
Cause code mapping inbound:	peer default
Cause code mapping outbound:	peer default
Maximum channels:	120

Name:	Enter a name for the SIP peer. A SIP peer is a specific endpoint to and from which the Dialogic® Diva® SIPcontrol™ software establishes the calls.
Peer type:	Some SIP peers need a specific peer, such as a Microsoft® Exchange Server, to work faultlessly with the Dialogic® Diva® SIPcontrol™ software. If this is the case for your configuration, select the specific SIP peer. If not, select Default .
Default SIP to PSTN peer:	Enable this option if the selected peer type should be used as default peer. Calls from unconfigured SIP peers will be assigned to this peer, and therefore are handled with these settings. If several peers are configured as default, the Dialogic® Diva® SIPcontrol™ software takes the first to transmit the call.
Host:	Enter the host name or IP address of the peer. The name must be resolvable by local name resolution. During the establishment of a call, the host name is sent by this peer exactly as entered here, unless an address map applies that converts the host name in a different format. For more information about name resolution, see the Windows® documentation.
Port:	Displays the SIP port on which the remote peer is listening. The default is 5060, which is the standard port for SIP.
IP protocol:	Select the IP protocol to be used for calls to this peer. Calls from this peer are accepted with all protocols and on all ports/addresses configured in Network Interface Configuration as described on page 21.
Domain:	Enter the domain name, e.g., dialogic.com, or the IP address. The domain name must comply with the DNS rules.
Display name to:	Enter the name that is to be sent in the "To" header of the INVITE message on calls from the PSTN to SIP.
Display name from:	Enter the name that is to be sent in the "From" header of the INVITE message on calls from the PSTN to SIP. To sent the calling party number include an asterisk (*) in the display name. For instance, if the display name is "Dialogic *" and the calling number is 123, then the remote side receives "Dialogic 123". To include an asterisk in the display name, enter "*" to include a backslash enter "\\".
User name to:	You 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 Dialogic® Diva® Analog Media Boards. If a call from SIP does not contain a user name, the name entered here is transmitted as calling party number to the PSTN.
User name from:	Enter the user name that is added to the SIP address when a number from the PSTN is suppressed. You can also enter the complete SIP address consisting of <username>@<local-IP/hostname>. If a call from SIP does not contain a user name, the name entered here is transmitted as called party number to the PSTN.
Force T.38 reinvoke:	Some peers do not switch the media channel to T.38 if they receive a fax call, e.g., if they do not evaluate the fax calling tone. If you select this option, the Dialogic® Diva® SIPcontrol™ software tries to initiate the media channel switch.
Alive check:	If you select this option, the Dialogic® Diva® SIPcontrol™ software will send "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.
Address map inbound:	Name of the regular expressions list applied to the addresses received on calls from this SIP peer. See Address Map Configuration on page 29 for more information about setting up a regular expression list. Regular expressions may be used to add or remove dial prefixes required by a PBX or to rewrite public phone numbers of different number ranges into a common format. See the Examples on page 38 for more information.

Address map outbound:	<p>Select the name of a regular expression list to be applied on outgoing calls on this interface. See Address Map Configuration on page 29 for more information about setting up a regular expression list.</p> <p>Regular expressions may be used to add or remove dial prefixes required by a PBX or to rewrite public phone numbers of different number ranges into a common format. See the Examples on page 38 for more information.</p>
Cause code mapping inbound:	Select the cause code mapping for incoming calls that you configured under Cause Code Mapping Configuration as described on page 30.
Cause code mapping outbound:	Select the cause code mapping for outgoing calls that you configured under Cause Code Mapping Configuration as described on page 30.
Maximum channels:	Specifies the number of channels that this SIP peer is able to handle at the same time. This setting is used by the Dialogic® Diva® SIPcontrol™ software to distribute calls in a load-balancing scenario.

Session Timer

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

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

Use session timer:	Activates session monitoring via SIP session timers, with the timeout values given here. Refer to RFC4028 for details.
Interval:	If Use session timer is enabled, you can set a timeout in seconds until a call is considered to be aborted. Normally, refreshes are performed after the first half of the interval has been 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 the Dialogic® Diva® SIPcontrol™ software will accept. The minimum value is 90 seconds.

Address normalization

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

Address Normalization	
Dialplan:	<input type="text" value="none"/>
Number format (outbound):	<input type="text" value="Unchanged"/>
Encoding (outbound):	<input type="text" value="Use type flag"/>

Dialplan:	<p>Select the local dialplan to be used by the dialplan module of the Dialogic® Diva® SIPcontrol™ software. You need to configure the local dialplan under Dialplan Configuration as described on page 27 before you can select it here.</p> <p>The dialplan selected here applies only to outgoing calls.</p>
Number format (outbound):	<p>This parameter determines the shortest format allowed that is sent out in outbound calls. 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 signaled in the SIP message 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 if no country code is given or if the area code matches the location settings.</p>

Extension:	The number is converted to the extension only if no additional information is given or if the country / area code and basic phone number match the location settings.
Encoding (outbound):	Determines if numbers in outbound calls should either be encoded as unknown number with national or international prefix digits or as national or international call with type flags.

Routing Configuration

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

- called, calling and redirected number of a call, for which the redirected number is only available for calls originating in the PSTN,
- the source where a call originated, i.e. a PSTN interface name or a specific SIP peer,
- the current channel allocation across a set of several possible destinations in a load-balancing environment, and
- the current status of a destination. See [How Calls Are Processed](#) on page 32 for more information.

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

For more information about possible routing configurations, see [Routing examples](#) on page 51.

The following menus are available for configuration:

- [Edit Routing Configuration](#) on page 25
- [Address Normalization For Condition Processing \(Using Source Dialplan\)](#) on page 26
- [Conditions](#) on page 26
- [Address Manipulation](#) on page 27

Edit Routing Configuration

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

Edit Routing Configuration		
Name:	<input type="text" value="Routing1"/>	
Direction:	<input type="button" value="PSTN to SIP"/>	
Select sources		
Controller1	<input checked="" type="checkbox"/>	
Select destinations	Loadbalancing / Failover	
	Master	Slave
testpeer	<input checked="" type="checkbox"/>	<input type="checkbox"/>
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".

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

Select Sources: Depending on the selected direction, this part either lists all configured PSTN interfaces or SIP peers. The route will only be considered for a call if the call originated from a selected source.
Note: A source may be selected even if it is currently disabled. In this case, however, the call will already have been rejected before the route is queried. At least one source interface is required for the route.

- Select Destinations:** You can select the possible destinations for the route, i.e., the set of CAPI controllers or SIP peers to which the call may be routed. The master or slave setting allows for configuring priorities. The Dialogic® Diva® SIPcontrol™ software will always try to establish a call to one of the masters first and considers the slaves only if all masters have failed or could not accept calls due to their call load.
- Max. call attempts for this route in a failover scenario:** Enter the number of times, the Dialogic® Diva® SIPcontrol™ software should try to call the recipient in a failover environment. If you enter 0 (zero), the Diva SIPcontrol software tries all selected destinations. A value of 1 disables the failover functionality and tries only the first destination of a route.

Address Normalization For Condition Processing (Using Source Dialplan)

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

Address Normalization For Condition Processing (Using Source Dialplan)	
Number format:	Unchanged ▼
Encoding:	Use prefixes ▼

- Number format:** This parameter determines the shortest format allowed that is sent out on outgoing calls. 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 if no country code is given or if the area code matches the location settings.
 - Extension:** The number is converted to the extension only if no additional information is given or if the country / area code and basic phone number match the location settings.
- Encoding:** Determines if outgoing numbers should either be encoded as unknown number with national or international prefix digits or as national or international call with type flags.

Conditions

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

Conditions			
Called number	Calling number	Redirect number	
<input type="text"/>	<input type="text"/>	<input type="text"/>	Delete
Add			

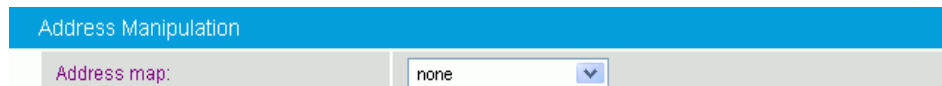
- Called number:** If the routing is supposed to be valid only for specific calls, enter the called party number to which the route should apply to. Then the Dialogic® Diva® SIPcontrol™ software compares the current called party number against the here entered called number. If it does not match, the Diva SIPcontrol software verifies the next routing until it finds a match.
- Note:** A route can only be matched if all three condition parts (called number, calling number, and redirect number) match their call address counterpart in any of the lines. Empty condition entries match always, i.e., a line with all three condition parts left empty will always apply, thus working as a default route.
- Calling number:** If the routing is supposed to be valid only for specific calls, enter the calling party number to which the route should apply to. Then the Dialogic® Diva® SIPcontrol™ software compares the current calling party number against the here entered calling number. If it does not match, the Diva SIPcontrol software verifies the next routing until it finds a match.
- Note:** A route can only be matched if all three condition parts (called number, calling number, and redirect number) match their call address counterpart in any of the lines. Empty condition entries match always, i.e., a line with all three condition parts left empty will always apply, thus working as a default route.

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

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

Address Manipulation

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



Address Map: If a route matches, the address manipulation setting allows for modifying the call addresses according to your needs. Example: If all calls with the called party number starting with "9" should be directed to a specific peer, it might be desirable to remove this digit. This can be done with a special address map configured. Note that you need to configure the address map as described under [Address Map Configuration](#) on page 29 before you can select it here.

Dialplan Configuration

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

The dialplan module supports the following features:

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

Important information about the outside access digit configuration

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

- The Diva SIPcontrol software's number normalization function does not remove outside access digits as a PBX can do for external calls. If the Diva SIPcontrol software needs to behave like a PBX with an outside access digit for external calls, you need to use the Address Map functionality in combination with a Routing module.

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

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

Edit Dialplan Configuration									
Name:	<input type="text" value="Dialplan1"/>								
Country code:	<input type="text" value="1"/>								
North-American numbering plan:	<input type="checkbox"/>								
Area code:	<input type="text"/> With national prefix ▼								
Other local areas:	<table border="1"> <tr><td><input type="text"/></td></tr> <tr><td><input type="text"/></td></tr> <tr><td><input type="text"/></td></tr> <tr><td><input type="text"/></td></tr> <tr><td><input type="text"/></td></tr> <tr><td><input type="text"/></td></tr> <tr><td><input type="text"/></td></tr> <tr><td><input type="text"/></td></tr> </table>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>	<input type="text"/>
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<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 type="checkbox"/>								
Incoming PSTN access code provided by PBX:	<input type="checkbox"/>								

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

Country code: Enter the country where computer with the installed Dialogic® Diva® SIPcontrol™ software is located.

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

Area code: If you do not use the North-American numbering plan (NANP), enter the area code without the leading zero here. If the NANP is needed for your configuration, enter the code for the home area here and enter the codes for the other local areas in **Other local areas**.
 If you need to use NANP, you can choose between transmitting the number:
With national prefix: The long-distance code is added to the number.
Local: The number is transmitted without any area code.
Without national prefix: The number is transmitted without the long-distance prefix.

Other local areas: You 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 country and area code. If you use MSNs, leave this field empty and enter the length of the MSNs in Maximum extension digits .
Maximum extension digits:	Specify the maximum number of extension digits. Use the "arrow up" and "arrow down" buttons to do so.
International prefix:	Enter the international prefix for your country, e.g., 00.
National prefix:	Enter the digits of the national prefix, e.g., 0 in Germany.
Access code:	Enter the digits that are needed to get access to the public network, e.g., 9.
PSTN access code provided by the SIP caller:	Select this option, if the SIP caller has to provide the access code. If the length of the called number is not sufficient to identify it as an internal number, you have to activate this option to avoid ambiguous numbers. This is usually the case, if you are not using the North-American Numbering Plan (NANP).
Incoming PSTN access code provided by the PBX:	Select this option, if the PBX adds the access code to the calling number for incoming external calls.

Address Map Configuration

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

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

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

Edit Address Map Configuration	
Address map name:	<input type="text" value="AddressMap1"/>
Rule name:	<input type="text" value="AddressMap1.1"/>
Called address expression:	<input type="text"/>
Called address format:	<input type="text"/>
Calling address expression:	<input type="text"/>
Calling address format:	<input type="text"/>
Redirect address expression:	<input type="text"/>
Redirect address format:	<input type="text"/>
Stop on match:	<input 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 dialogs where an address map can be selected.

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

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

Called address expression: If the regular expression entered here matches a called address, the format string is applied to the result. See the reference guide for more information on regular expressions.

Called address format:	If the address format entered here matches a called address, the format string is applied to the result. See the reference guide for more information on regular expressions and formats.
Calling address expression:	If the regular expression entered here matches a calling address, the format string is applied to the result. See the reference guide for more information on regular expressions.
Calling address format:	If the address format entered here matches a calling address, the format string is applied to the result. See the reference guide for more information on regular expressions and formats.
Redirect address expression:	If the regular expression entered here matches a redirected address, the format string is applied to the result. See the reference guide for more information on regular expressions.
Redirect address format:	If the address format entered here matches a redirected address, the format string is applied to the result. See the reference guide for more information on regular expressions and formats.
Stop on match:	If all expressions match all addresses of a call, this flag determines if the Dialogic® Diva® SIPcontrol™ software should continue to search for matching rules. If set, the address matching is aborted.

The expressions match from the beginning, but they do not need to match the complete number. The part of the number that does not match, is added to the number, for example:

Number: 1234567

Expression: 123

Format: 4567

Result: 45674567

If the complete number needs to be matched, you need to add "\$" to the expression, for example:

Number: 1234

Expression: 1234\$

Format: 4567

Result: 4567

Note: In calls from SIP, an expression followed by "\$" does not match if the SIP address consists of <number>@<hostname>, e.g., 1234\$ does not match the address 1234@sipcontrol.dialogic.com.

Cause Code Mapping Configuration

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

If the internal default mapping table provided by the Dialogic® Diva® SIPcontrol™ software does not fulfill your needs, e.g., because your local PBX uses non-standard cause codes, you can configure your own cause code mapping table, which will be then inquired before the default table is being checked. See [Cause code mapping](#) on page 44 for the cause/response code mapping table.

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

Edit Cause Code Mapping		
Name:	<input type="text" value="CauseMapping1"/>	
Direction:	<input type="text" value="PSTN to SIP"/>	
PSTN cause code	SIP response code	
<input type="text"/>	<input type="text"/>	<input type="button" value="Delete"/>
<input type="button" value="Add"/>		
Default:	<input type="text"/>	

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. The values are only valid in the range from 1 to 127.

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

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

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

System Settings

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

System Settings	
Event Log Level:	<input type="text" value="Errors, Warnings, Informational Messages"/>
Debug Level:	<input type="text" value="Full"/>

Event Log Level: A computer with the Dialogic® Diva® SIPcontrol™ software installed, may write different types of events into the System Event Log. The details for each event log are described in the Appendix under [Event logging](#) on page 49.

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

CHAPTER 6

How Calls Are Processed

The Dialogic® Diva® SIPcontrol™ software uses an endpoint-based approach to process calls, that means that every PSTN interface and every configured SIP peer is considered as a single endpoint.

Each call originates at a specific endpoint (on the SIP side after assigning the SIP call request to one of the configured peers) and needs a route to find its designated endpoint (the destination). Thus, the most simple configuration needs one PSTN endpoint, one SIP peer, and one route.

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

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

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. Verify if at least one call address condition of this route matches the called, calling, and redirected addresses of the call request; if not, find another route.
4. If any route condition matched, verify in the list of configured destinations which one is the most preferred. This is done based on settings. See [Some important things to know about call processing](#) below for more information.
5. Try to establish the call via this destination. If the destination is unavailable or rejects the call, try the next destination of the route. Note that the call will be aborted immediately if a cause code, signaling final failures, e.g., user busy or unallocated number is received.
6. The call is established.

Some important things to know about call processing

- Each route may point to several destinations, between which 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 would also match the call criteria, only the first matching route is ever evaluated; therefore, default routes should be created carefully.
- Load-balancing/fail-over is only performed between the destinations of a single route.
- Routes without any conditions do always match (as long as the source endpoint is listed in route sources).

Emergency calls

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

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

Routing conditions

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

CHAPTER 7

How Call Addresses Are Processed

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

The Diva SIPcontrol software converts addresses automatically, without any intervention from the user. That means, that the Diva SIPcontrol software adds or removes a special prefix to a number with a known number type, e.g. "+" for international numbers, when converting between a number and an address. See [Common formats](#): on page 37 for a list of prefixes.

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

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

Possible scenarios

- At a PSTN interface, a line access digit must be prepended in order to call to the public network, while another PSTN interface is directly connected and does not need an access digit.
Solution: Add a regular expression to outbound address map of the first interface.
- All calls to a number beginning with "9" shall be routed to one specific SIP peer, while removing this digit.
Solution: Manipulate the called number in the route. By this approach, the SIP peer may receive also calls to other numbers (via other routes) without having to deal with different number formats.
- SIP peer "A" needs the dialed numbers to be formatted in E.164 format, while SIP peer "B", which is in load-balancing or fail-over partnership with "A", needs it in an extension-only format.
Solution: Define different number formats in the SIP peer settings.
- SIP peer "A" is located at a different location than SIP peer "B", e.g., London and Stuttgart; therefore, both need different location settings regarding country and area codes, etc.
Solution: Create different dialplans and assign each dialplan to one SIP peer.

How addresses are manipulated

Note: Each step is optional.

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

CHAPTER 8

How Address Maps Are Processed

Address maps are processed as follows:

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

CHAPTER 9

How Numbers Are Processed

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

1. [Number normalization based on a dialplan](#) as described below.
2. [Number modification using regular expressions](#) as described on page 37.

Number normalization based on a dialplan

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

The Diva SIPcontrol software version 1.6 also supports dialplans using the North-American numbering plan (NANP). See [North-American numbering plan](#) on page 28 for more information.

The number normalization is done in two steps:

1. The received called, calling and redirected numbers are analyzed based on the dialplan configured for the PSTN Interface or SIP Peer.
2. The number is converted into the configured target format. Available are six target formats (two target formats are new in Diva SIPcontrol software version 1.6):
 - **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 number 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, if required. Exception: Numbers with different country code will be converted to an international number with prefix for international calls and outside access digit, if required.
 - **National number with number type (new):** If possible, all numbers are converted to a national number with the number type flag set to "national". Exception: Numbers with different country code will be converted to an international number with number type set to "international". **Note:** This target format should not be used for calls to SIP networks.
 - **Extension only with prefixes:** All numbers are reduced as much as possible, only the required prefixes are prepended.
 - **Extension only with number type (new):** All numbers are reduced as much as possible. Instead of prefixes the appropriate number type is set. **Note:** This target format should not be used for calls to SIP networks.

Some important information about the outside access digit configuration

- Configure the outside access digit only, if there is a PBX between the PSTN and the Dialogic® Diva® SIPcontrol™ software, and if this PBX requires the outside access digit for external calls. If you need to configure the outside access digit, configure also the related options:
 - **Incoming PSTN access code provided by the PBX:** This option defines if the Diva SIPcontrol software expects the outside access digit in the calling number in external calls from the PBX. Usually, the PBX prepends the outside access digit to the calling number in incoming external calls in order to enable callback functionality at internal phones. If this is the case, enable this option.
 - **PSTN access code provided by the SIP caller:** This option defines, if Dialogic Diva SIPcontrol expects the outside access digit in the called number in external calls from the SIP side to the PSTN. Usually, it is required to prepend the outside access digit to call an external number from an internal phone (In this case, these are phones on the SIP side.). However, in some configurations this is not required,

especially in a configuration that is part of the North-American numbering plan (NANP), where an internal number can be identified based on its length. If it is possible to identify an internal call purely by the length of the called number, this option can be disabled. In all other configurations with outside access digit this option has to be enabled.

It is recommended to have this option enabled in dialplans with outside access digit.

- The Diva SIPcontrol software's number normalization function does not remove outside access digits as a PBX can do for external calls. If the Diva SIPcontrol software needs to behave like a PBX with an outside access digit for external calls, you need to use the Address Map functionality in combination with a Routing module.

Number modification using regular expressions

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

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

The Dialogic® Diva® SIPcontrol™ software uses the style of regular expressions used by Perl; therefore, most tutorials and how-to's covering Perl regular expressions should apply to the Diva SIPcontrol software.

Common expressions

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

Common formats:

Character	Meaning
0-9, +	Inserts the respective character into the output
(?n(digits))	Inserts the digits given only if the n th sub-expression of the expression matched
\$&	Outputs what matched the whole expression
\$n	Outputs the n th matched sub-expression
+	Indicates an international number type, if it is the first character in the string
N	Indicates a national number type, if it is the first character in the string
S	Indicates a subscriber number type, if it is the first character in the string

Examples

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

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

Omit the prefix digits

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

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

Expression entry: ^33

Format entry: (none)

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

Add the prefix digits

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

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

Expression entry: .*

Format entry: 9\$&

Remove the prefix only for local area

Task: If a call to a local number is initiated, no long distance code should be dialed. In this example the local area code is 972 and the long distance code is 01.

Example: 01-972-444-5555 should be dialed as 972-444-5555.

Expression entry: ^(01972)

Format entry: 972

Add the prefix for non-local calls

Task: If a call should be placed to a non-local party, 011 should be added before the dialed number. In this example the local area codes are 111, 222, and 333.

Example: 111-444-5555 should be dialed as is, while 472-666-777 should be dialed as 011-472-666-7777.

First expression entry: ^(111|222|333)

First format entry: \$&

Stop on Match: true

Second expression entry: .*

Second format entry: 011\$&

Stop on Match: true

Replace the international number type by prefix

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

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

Expression entry: ^\+

Format entry: 01

Replace the international dial prefix by number type

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

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

Expression entry: ^01

Format entry: +

Replace an extension by another

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

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

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

First format entry: 2222

Stop on Match: true

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

Second format entry: 4444

Stop on Match: true

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

Modify the number to local dialplan requirements

Task: Modify the number so the Dialogic® Diva® SIPcontrol™ software dials out correctly according to the dialed type of number.

Example: Consider the following dialing requirements. There are three local area codes (406, 214, 972) that must be dialed without long distance code.

9-1-406->	9	local calls without area code
9-1-972-473-45->	45	internal calls 4-digit dialing for 4500-4599
9-1-214->	9-214	for local calls with area code
9-1-972->	9-972	for local calls with area code

Expression entry 1: ^91406

Format entry 1: 9

Stop on Match: true

Expression entry 2: ^9197247345

Format entry 2: 45

Stop on Match: true

Expression entry 3: ^91214

Format entry 3: 9214

Stop on Match: true

Expression entry 4: ^91972

Format entry 4: 9972

Stop on Match: true

Note: The order of the expression parts is important here, because the internal calls (9-1-972-473-45) need to be matched before the local calls (9-1-972).

Replace the "N" in a national number

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

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

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

Expression: ^N

Format entry: 0

CHAPTER 10

Software Uninstallation

Note: If you want to upgrade from Dialogic® Diva® SIPcontrol™ software version 1.5.1 to Diva SIPcontrol software version 1.6, DO NOT uninstall the software before you install the Diva SIPcontrol software version 1.6, otherwise you might lose some settings, including your regular expressions.

To uninstall the Diva SIPcontrol software:

1. Click **Start > Settings > Control Panel**.
2. Double-click **Add or Remove Programs**.
3. In the **Add or Remove Programs** box, select "Diva® SIPcontrol™ - version 1.5.1" and click **Change**.
4. In the welcome dialog box, click **Next**.
5. In the **Remove the Program** box, click **Remove** to uninstall Diva SIPcontrol software.
6. In the **Setup Wizard Completed** box, click **Finish**. Diva SIPcontrol software is now uninstalled.
7. If you want to uninstall Dialogic® Diva® for Windows® see the "Dialogic® Diva® for Windows® Reference Guide. The Reference Guide is available on your Dialogic® Diva® Software Suite CD-ROM or on the Dialogic web site under: www.dialogic.com.

CHAPTER 11

Customer Service

Dialogic provides various options and arrangements for obtaining technical support for your Dialogic® product. We recommend that you use the Dialogic® Diva® Support Tools first before contacting your Dialogic Corporation supplier. Also we suggest that you visit our help web with its detailed information about a variety of topics. In the unusual case that neither your supplier nor the information on the help web cannot adequately address your support issue contact our Customer Support.

For more information see:

- [Dialogic® Diva® Support Tools](#)
- [Dialogic Corporation Help Web](#)
- [Dialogic Corporation Customer Support](#)

Dialogic® Diva® Support Tools

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

- Dialogic® Diva® Line Test: With the Diva Line Test tool you can test your hardware and perform simple phone test calls, call transfers, or basic inbound and outbound calls.
- Dialogic® Diva® Diagnostics: With the Diva Diagnostics tool you can write traces for each adapter or driver into a file.
- Dialogic® Diva® Management tool: With the Diva Server 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 tools see the respective online help file.

If you cannot solve the problem with the tools, contact your Dialogic Corporation supplier.

Dialogic Corporation Help Web

If your supplier is unable to help you to solve your problem, you can visit the Dialogic Help Web. It contains detailed information on such subjects as:

- Installation and upgrade of Dialogic® Diva® drivers, configuration scenarios, and applications.
- Diagnostic and testing utilities.
- Basic issues, error messages and how to resolve them.
- "How to" guides and wizards.
- Online training, which, as of the date of publication of this document, is offered for ISDN, the Dialogic® Diva® SDK, and Dialogic® X.25 products. The training is aimed toward technical support people, but much of the courses are also suitable for a non-technical audience.

For more information, visit our Help Web at www.dialogic.com/support/helpweb.

Dialogic Corporation Customer Support

If the information on the Dialogic Help Web was not sufficient to help you solve your problem, contact Dialogic Customer Support. See www.dialogic.com/support/contact for details on how to contact us.

To provide help, Dialogic Customer Support will likely need from you:

- A debug trace (see the Dialogic® Diva® Diagnostics Online Help file - DivaTrace.chm, and
- A copy of your active configuration (see the Dialogic® Diva® Configuration Manager Online Help file - DSMain.chm).

- A copy of your Dialogic® Diva® SIPcontrol™ software configuration. To save a copy, open a dos box, change to the directory in which the Diva SIPcontrol software files are stored and execute: regfile.exe save <nameofthefile>.txt or click **Show Configuration** at the bottom of the configuration web interface and copy and paste the contents into a separate file and save it as text file.

CHAPTER 12

Appendix

The appendix contains information regarding [Cause code mapping](#) below, [Event logging](#) on page 49, and [Routing examples](#) on page 51.

Cause code mapping

If the Dialogic® Diva® SIPcontrol™ software uses Microsoft® Office Communications Server 2007 as SIP peer, the cause/response code tables are used as specified by Microsoft. See [Default cause code mapping for Microsoft® Office Communications Server \(OCS\) 2007 SIP peers](#) on page 47 for a detailed list of cause/response codes.

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

Default cause code mapping

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

For ISDN to SIP code mappings, see [ISDN cause code to SIP response code](#) below.

For SIP to ISDN code mappings, see [SIP response code to ISDN cause code](#) on page 45.

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

ISDN cause code	Description	SIP response code forwarded to the SIP peer	Description
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	500	Server internal error
127	Interworking, unspecified	500	Server internal error
Any code other than listed above:		500	Server internal error

SIP response code to ISDN cause code

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

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

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

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

For ISDN to SIP code mappings, see [Microsoft® OCS 2007 ISDN cause code to SIP response code](#) on page 47.

For SIP to ISDN code mappings, see [Microsoft® OCS 2007 SIP response code to ISDN cause code](#) on page 48.

Microsoft® OCS 2007 ISDN cause code to SIP response code

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

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

Microsoft® OCS 2007 SIP response code to ISDN cause code

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

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

Event logging

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

- [Errors](#)
- [Warnings](#)
- [Informational messages](#)

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

Errors

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

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

Event ID	Event Text	Event Description
2000	Service could not start. <Reason>	The <Reason> is a text that explains why the service could not start.
2001	Service could not stop. <Reason>	The <Reason> is a 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.

Warnings

A warning is an event that is not necessarily significant. But it might indicate a possible future problem.

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

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

Informational messages

Informational messages refer to the successful operation 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.
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.

Routing examples

This section describes the configuration of four possible routing scenarios:

[Direct routing between one PSTN interface and one SIP peer](#) below

[Connect two SIP peers to two PSTN interface exclusively](#) below

[Connect two SIP peers to same PSTN interface](#) on page 52

[Load-balancing or fail-over between two SIP peers](#) on page 52

Direct routing between one PSTN interface and one SIP peer

If you need 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 needed:

1. Under **PSTN Interface Configuration**, enable and configure all PSTN interfaces connected to a PBX. Confirm each dialog box with **OK**.
2. Under **SIP Peer Configuration**, create a SIP peer with the necessary settings and make sure that the option **Default SIP to PSTN peer** is enabled. Confirm with **OK**.
3. Under **Routing Configuration**, create route no. 1 and do the following:
 - Select **PSTN to SIP** as direction.
 - Enable all required PSTN interfaces.
 - Select the SIP peer configured in step 2 as **Master** destination.
 - Set the parameter **Number format** to **Unchanged**.
 - Confirm with **OK**.
4. Under **Routing Configuration**, create route no. 2 and do the following:
 - Select **SIP to PSTN** as direction.
 - Enable the SIP peer configured in step 2 as source peer.
 - Enable all required PSTN interfaces as **Master** destination.
 - Set the parameter **Number format** to **Unchanged**.
 - Confirm with **OK**.
5. Save the configuration in the main configuration interface.

Connect two SIP peers to two PSTN interface exclusively

If you want to connect two SIP peers to two PSTN interfaces, so that each SIP peer may use one interface exclusively, 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 Interface Configuration**, enable and configure the two PSTN interfaces. Confirm with **OK**.
2. Under **SIP Peer Configuration**, create both SIP peers and make sure the entries in **Host** match exactly the host address sent by the peer for outgoing calls. Confirm with **OK**.
3. Under **Routing Configuration**, create route no. 1 and do the following:
 - Select **PSTN to SIP** as direction.
 - Enable the first PSTN interface as source
 - Enable the first SIP peer configured in step 2 as **Master** destination.
 - Confirm with **OK**.
4. Under **Routing Configuration**, create route no. 2 and repeat step 3 for the second PSTN interface and the second SIP peer.
5. Under **Routing Configuration**, create route no. 3 and do the following:

- Select **SIP to PSTN** as direction.
 - Enable the first SIP peer configured in step 2 as source peer.
 - Enable the first PSTN interfaces as **Master** destination.
 - Confirm with **OK**.
6. Under **Routing Configuration**, create route no. 4 and repeat step 5 for the second PSTN interface and the second SIP peer.
 7. Save the configuration in the main configuration interface.

Connect two SIP peers to same PSTN interface

If you want to connect two SIP peers to the same PSTN interface so that all calls from 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 Interface Configuration**, enable and configure the PSTN interface. Confirm with **OK**.
2. Under **SIP Peer Configuration**, create both SIP peers and make sure the entries in **Host** match exactly the host address sent by the peer for outgoing calls. Confirm with **OK**.
3. Under **Routing Configuration**, create route no. 1 and do the following
 - Select **PSTN to SIP** as direction.
 - Enable the first PSTN interface as source.
 - Enable the first SIP peer configured in step 2 as **Master** destination.
 - Under **Conditions**, click **Add** and set the **Called address** to "1.*".
 - Confirm with **OK**.
4. Under **Routing Configuration**, create route no. 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 Configuration**, create route no. 3 and do the following:
 - Select **SIP to PSTN** as direction.
 - Enable both SIP peers as source peer.
 - Enable the first PSTN interfaces as **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 be also 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 fail-over between two SIP peers

If two SIP servers should be configured as load-balancing or failover, carry out the following configuration.

1. Under **PSTN Interface Configuration**, enable and configure all required PSTN interfaces. Confirm with **OK**.
2. Under **SIP Peer Configuration**, create both SIP peers and make sure the entries in **Host** match exactly the host address sent by the peer for outgoing calls. If you configure a fail-over, SIP peer 1 (the master) should have the option **Alive check** enabled. Confirm with **OK**.
3. Under **Routing Configuration**, create route no. 1 and do the following:

- Select **PSTN to SIP** as direction.
 - Enable the first PSTN interface as source.
 - Enable the first SIP peer configured in step 2 as **Master** destination. For load-balancing configurations, SIP peer no. 2 should be configured as **Master** destination and for failover configurations it should be configured as **Slave** destination.
 - Confirm with **OK**.
4. Under **Routing Configuration**, create route no. 2 and do the following:
 - Select **SIP to PSTN** as direction.
 - Enable both SIP peers as source peer.
 - Enable the first PSTN interfaces as **Master** destination.
 - Confirm with **OK**.
 5. Save the configuration in the main configuration interface.
 6. Under **Routing Configuration**, create route no. 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 Configuration**, create route no. 3 and do the following:
 - Select **SIP to PSTN** as direction.
 - Enable both SIP peers as source peer.
 - Enable the first PSTN interfaces as **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 be also directed to one peer, remove the condition from the respective PSTN to SIP route and move the route to the end of the list.