



FIRMWARE 5.0

SHORT MANUAL

No. 2020-02

Release 0.4

I. Version history

Edition	Date	Author	Changes
0.1	01.04.2019	John Deutinger	First edition, firmware version 4.2-385-2069
0.2	15.05.2019	John Deutinger	Firmware Version 4.2-1.2137
0.3	12.08.2019	John Deutinger	RC1 from version 4.2-1.2308 and first 5.0/64 bits
0.4	18.02.2020	Chris Helbing	Firmware 5.0 only, preparation for release

Table 1: Version history

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1. Introduction

Configuring an OfficeMaster Gate as the link between ISDN and SIP is relatively simple and easy to understand. However, pure SIP connections are increasingly coming into play and complex scenarios with several different SIP remote stations are emerging. The configuration of such solutions is becoming more and more complicated and sometimes reaches its limits. OfficeMaster Gate firmware version 5.x addresses these problems without experienced users having to learn everything from scratch.

The aim is to meet the following requirements:

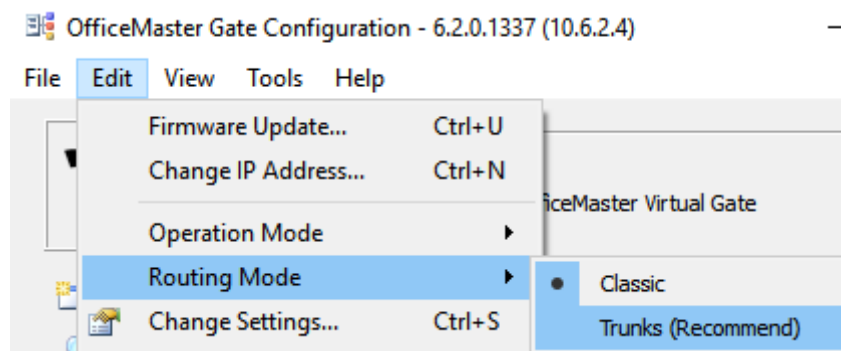
- As far as possible, you want to map everything that is possible at present (exception: fax via OMCUMS component is no longer supported → OfficeMaster DirectSIP Fax as successor)
- Parallel use of multiple SIP trunks with registration
- Simplification of the set of rules: A single rule leads from the source directly to the target - corresponding input and output rules are no longer needed!
- Flexible definition of SIP endpoints/nodes (internal/external SIP trunks with/without registration, registered terminals, foreign SIP destinations/devices)
- Each of these endpoints is configured with properties - independent of the rule set - and serves as the source or target of rules
- Adding an endpoint is supported by a wizard; useful default settings are made for known remote stations
- The firmware is based on a 64 bit operating system and is therefore future-proof for future hardware generations
- Multiple targets support either load-balancing or failover mode
- Note: The connection of real hardware interfaces (ISDN and analog) has not been tested sufficiently so far and is therefore currently not supported

Hint:

Older preview versions are still based on the 32 bit platform - the numbering is used for differentiation: 4.2.x is 32 bit preview, 5.x is the 64 bit version which requires a complete reinstallation of the operating system. The advantage of the 32 Bit Preview is the possibility to switch at any time between this version and the previous 4.1.X version and also to continue to implement ISDN connections.

Hint:

A new version of "OfficeMaster Gate Configuration" is also required for the preview! With this tool, the "Trunk-Mode" must be activated first. To do this, connect to the SBC and make the appropriate settings via **Edit Routing Mode.** →



1.1. Restrictions in the current version

All essential features for operation as a Session Border Controller are already implemented. However, compared to the 4.1-X firmware, some features in the Web interface and PowerShell cmdlets are no longer fully implemented. Updates on this will follow.

Hint:

All menus and dialogs of the Configuration Tool are shown in this document in the recommended setting "English" (switchable via Help menu).

1.2. Use of this version and further planning

1.2.1. Where can/should the current version be used?

The version described here is primarily intended for internal **evaluation by partners** and technically experienced end customers. From 28.02.2020, **productive use by end customers** will also be supported.

The licensing of the 5-version is done via a basic license as well as additional SIP licenses (identical to the existing licenses for 4.1), which are used as "concurrent calls" (simultaneous calls) in contrast to the previous usage. Thus, any number of SIP trunks with any number of channels (limits are still being defined) can be configured. The total number of possible parallel calls depends only on the number of licensed SIP channels.

Added value compared to version 4:

- Team support, numerous additional codecs are supported
- multiple SIP trunks with registration
- for multiple targets you can choose between load balancing and failover
- Server platforms that only allow 64 bit operating systems

2. Trunk configuration

All trunk types (except ISDN connections) are created and preconfigured with the help of a wizard. Further settings or changes can be made at any time afterwards.

2.1. ISDN trunks (currently not recommended for productive use)

Existing ISDN and analog interfaces appear automatically in the configuration - depending on hardware and licenses. Most of the settings are made in the "Settings" tab as in older firmware versions. In addition, it is possible to add rules for number manipulation for outgoing calls ("Outbound Number Manipulation") - see Section 2.6

Advanced Configuration

BRI 1 BRI 2 BRI 3 BRI 4 Analog 1 Analog 2 PRI 1 PRI 2 PCM 1 PCM 2 Add

Routing Rules (ISDN) Settings

Number of Channels to Use

Outbound 2 of Total 2

Inbound 2

ISDN Connection

Type Point-to-Point (DID) Point-to-Multipoint (MSN) QSIG Onboard Termination

ISDN Type of Number, Mapping from/to E.164

Type of number	from ISDN	to ISDN
International	+	+
National	+49	+49
Subscriber		

Apply to

<input checked="" type="checkbox"/> Called Party Number	<input checked="" type="checkbox"/> Called Party Number
<input checked="" type="checkbox"/> Calling Party Number	<input checked="" type="checkbox"/> Calling Party Number
<input type="checkbox"/> Redirecting Number	<input type="checkbox"/> Redirecting Number

Outbound Number Manipulation

Add Edit Remove


Name	Rule	Substitution


Add. Parameters

More >> OK Cancel

The configuration of incoming calls is identical for all trunks - it is described under 2.5

2.2. SIP Trunks

The **Add** button  starts the wizard for adding a trunk.



Welcome to the Configuration Wizard

This wizard will help you add and configure a new interface to your OfficeMaster Gate.

Please select one of the options below

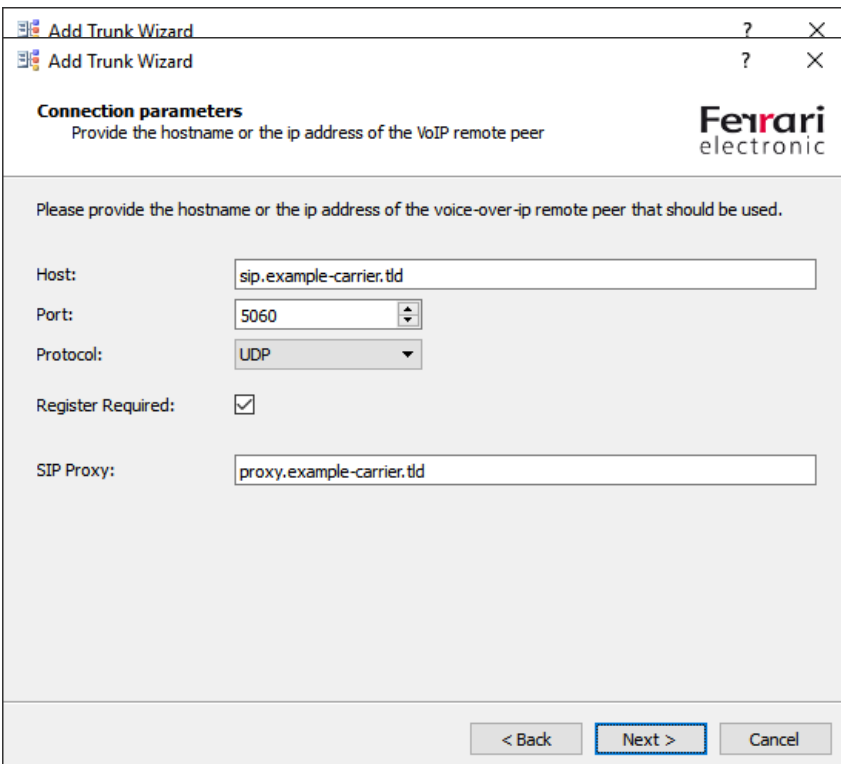
- SIP Trunk
Establish connection to PBX, VoIP system or SIP provider
- SIP Registrar Service
Configure SIP registration server message handling
- Device Group (Phones and other destination)
Direct connect to sip phones or various endpoints

To continue, click Next.

< Back Next > Cancel

Select "SIP Trunk" from the three available types and click "Next >". Then you must select whether a SIP connection to a provider/carrier or to an IP PBX or UC system is to be established:

Here you can select one of the preset profiles. If no suitable template is available, the "< Generic Profile>" is



Connection parameters
Provide the hostname or the ip address of the VoIP remote peer

Please provide the hostname or the ip address of the voice-over-ip remote peer that should be used.

Host:

Port:

Protocol:

Register Required:

SIP Proxy:

< Back Next > Cancel

used. Under Host, specify the destination or SIP domain in the next step. Destination port, protocol, whether registration is required and a different proxy address complete the connection settings. All information can be changed later at any time.

If "Register Required" is selected, the next step is to enter the login data.

Add Trunk Wizard ? X

User information
Provide SIP user information

The remote device requires an user to authenticate herself. The following information are required to establish a connection and are used to authorize the user at the registrar or proxy.

Add Trunk Wizard ? X

Common settings
Select the number of lines to use

Please select the number of lines should be available for services. Please note that the effective number of available channels depends on the installed license.

Channels Total:

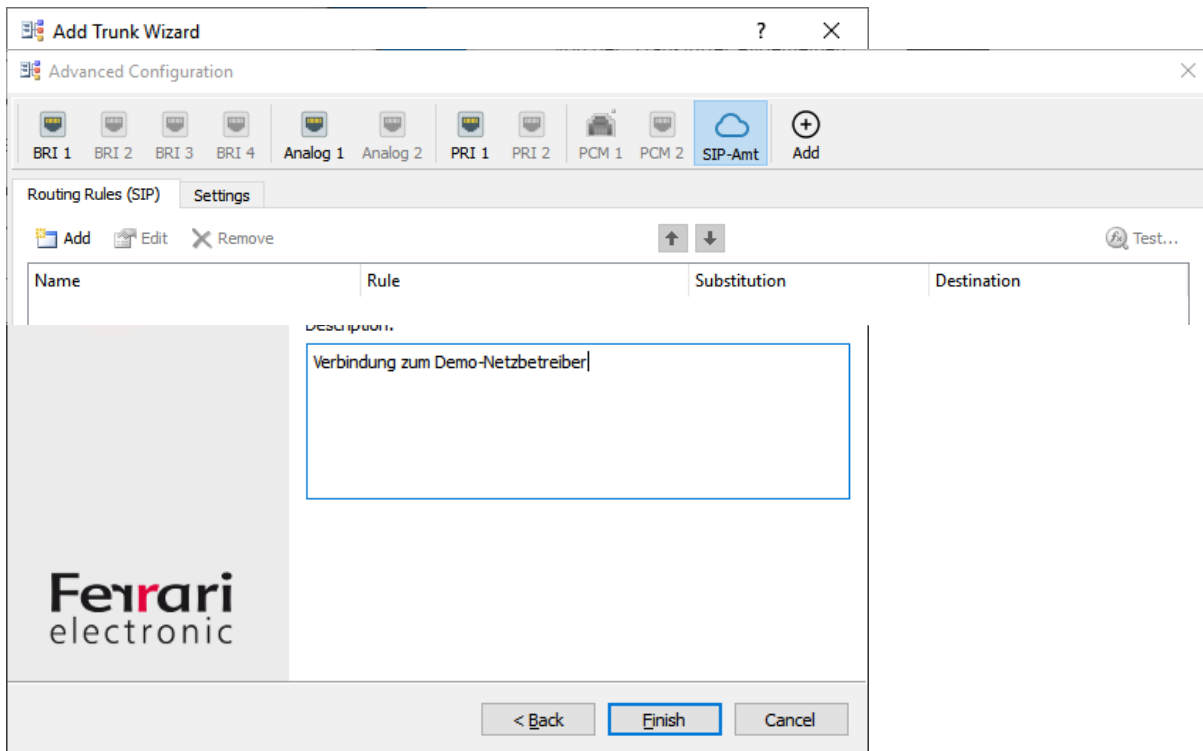
Send:

Receive:

< Back Next > Cancel

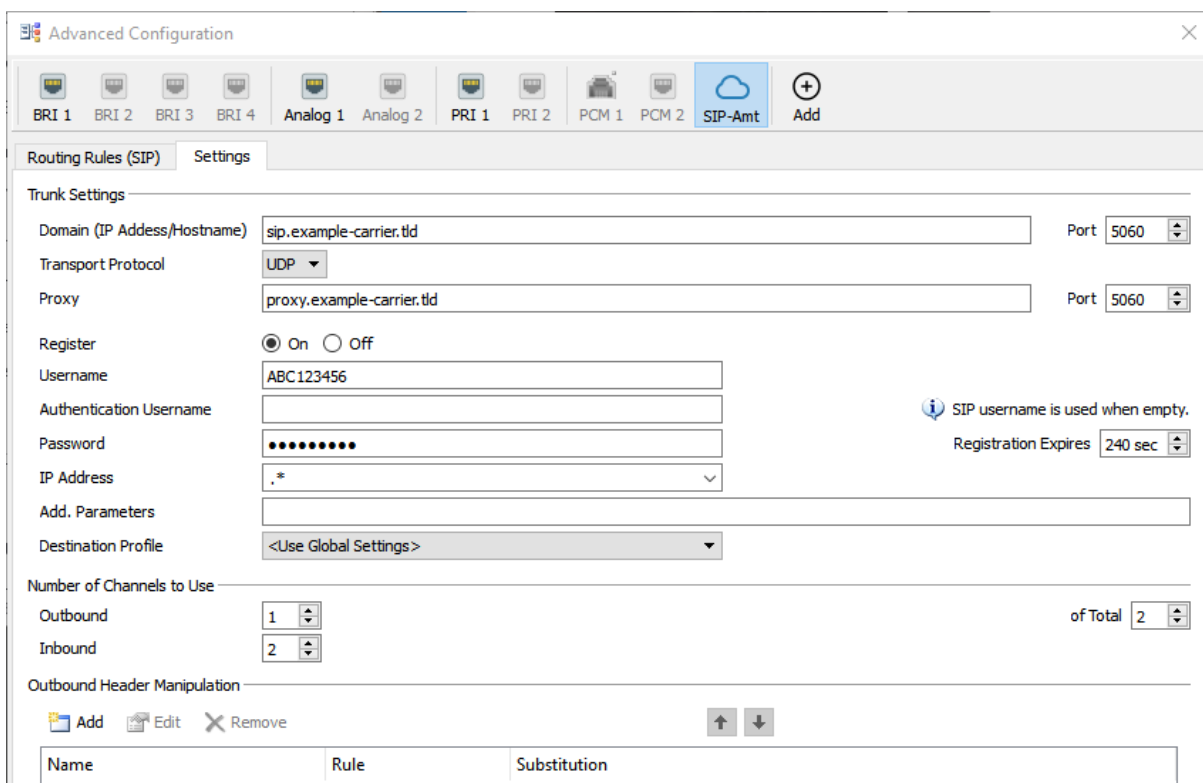
"User (Authorization)" must only be entered if it is different from "User". You can obtain this information from your network operator. The period of validity of the registration may also be specified by the provider. When operating from the internal network via NAT using the UDP protocol, the duration must not be too long, otherwise the incoming path for incoming calls will not remain open in the firewall ("UDP hole punching"). The next step is to configure the number of parallel calls.

Finally, short names for the trunk and optionally a more detailed description are configured.



The trunk now appears in addition to the previous ones.

In the "Settings" tab, all settings can now be checked, corrected or supplemented.

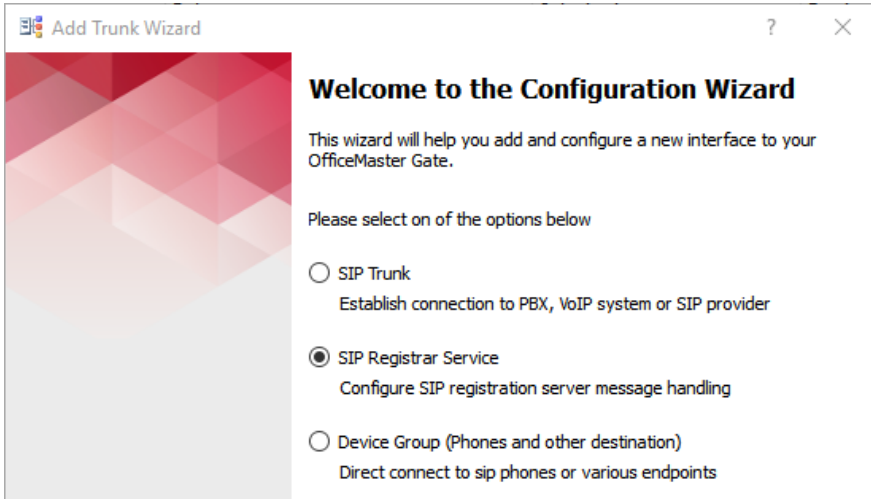


Additional settings ("Additional Parameters", "Destination Profile") are discussed in section 5. Important is the field

"IP Address": If an incoming INVITE cannot be assigned to this trunk otherwise by the field "Domain", the sender IP address of the remote terminal assigned to this trunk should be entered here (Attention: Syntax as regular expression!). More on this in section 4.1.

2.3. SIP Registrar Service

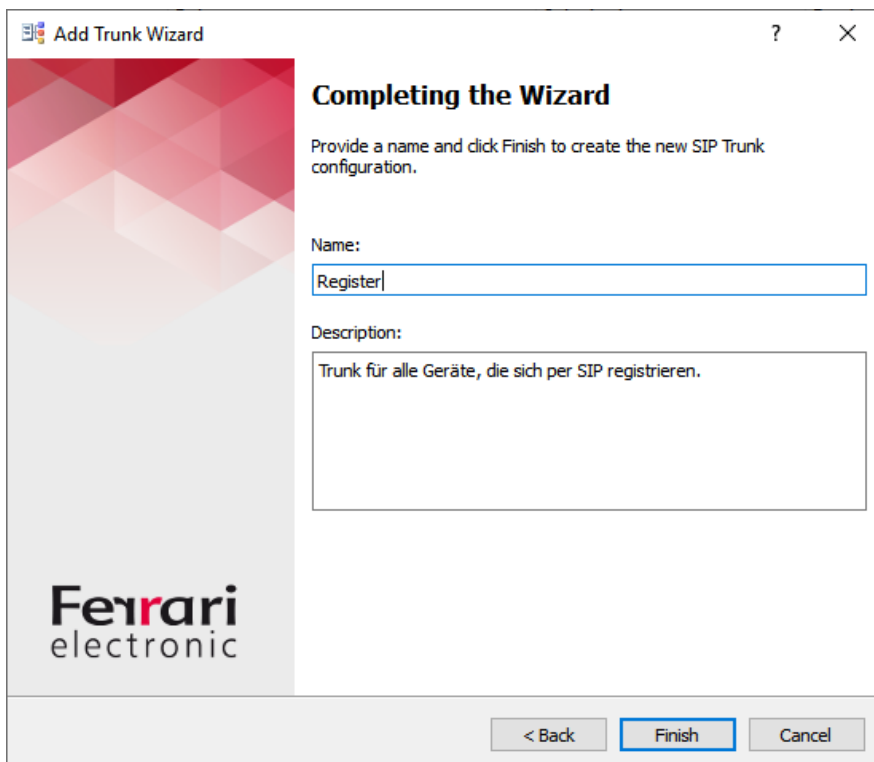
A special trunk that can only be created once is the "SIP Registrar Service". It contains the settings for the registration of devices and the set of rules that is applied to registered devices.



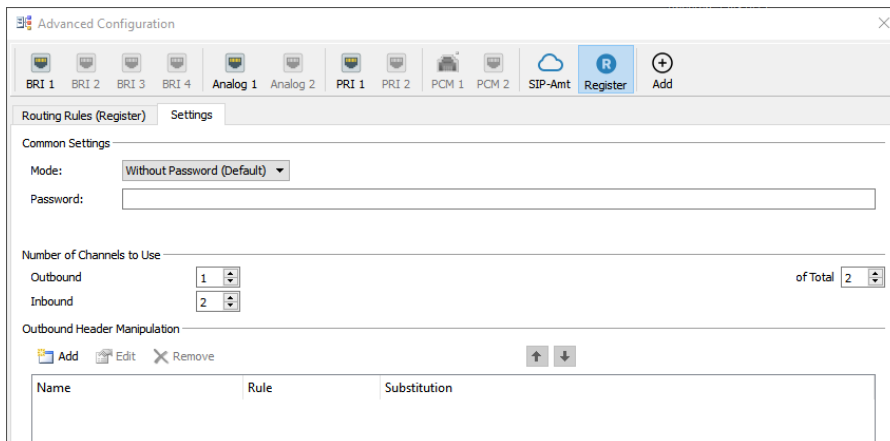
The next step is to select how SIP devices can register:

- Without Password (Default) - Login without password allowed
- Global Password - Login with global password possible
- Disabled - registration not allowed

This setting can be changed at any time. Finally, the name and description of the trunk is defined.



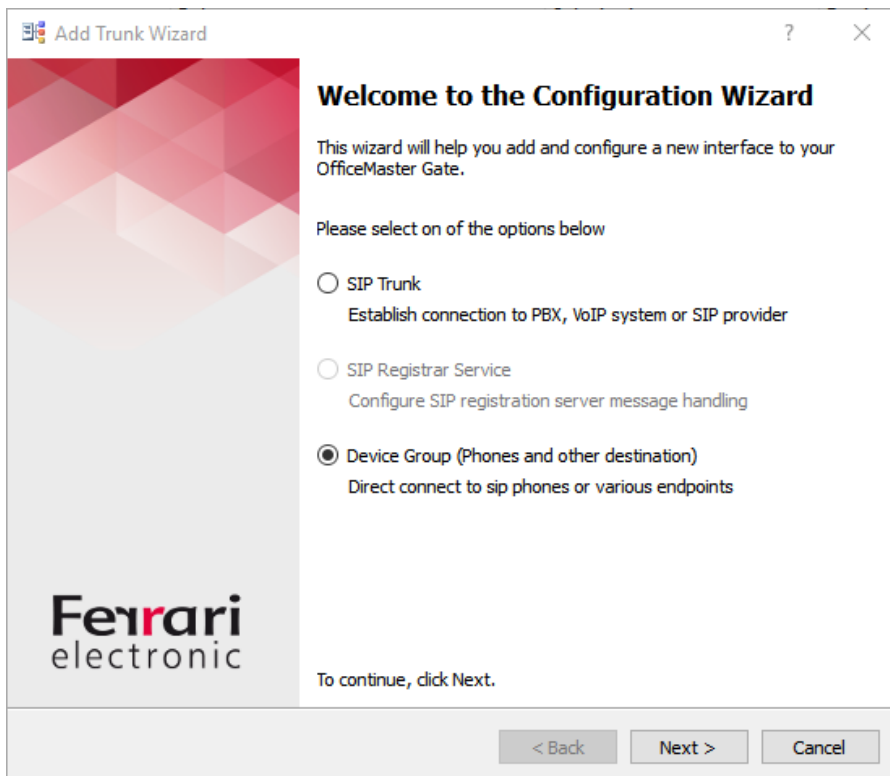
In the "Settings" tab, the settings can be changed subsequently:



2.4. device groups

Besides registered SIP devices, it is also possible to use devices without registration. Analogue telephones and fax machines connected to SIP ATAs can be used with or without registration - the same applies to SIP devices. Without registration, such devices can be grouped e.g. by location or by usage (telephones, fax machines, etc.) to use separate settings and sets of rules for each group.

Such groups are created as trunks as usual using the wizard:



A short name and a detailed description is defined for the device group:

The screenshot shows the 'Add Trunk Wizard' dialog box, titled 'Completing the Wizard'. It contains the following fields and controls:

- Name:** A text input field containing the text 'Faxer'.
- Description:** A text area containing the text 'Device Group für die Steuerung von Faxgeräten über SIP-ATAs'.
- Buttons:** '< Back', 'Finish', and 'Cancel' buttons are located at the bottom right.
- Logo:** The Ferrari electronic logo is visible in the bottom left corner.

Members of this group are added in "Settings" ("Add"):

The screenshot shows the 'Add Device Group Member' dialog box with the following configuration:

- Name:** Faxgerät Empfang
- Phone Number:** 100
- Address:** 192.168.5.100
- Port:** 5060
- Protocol:** UDP
- Add. Parameters:** (Empty text field)
- Active:** Active
- Buttons:** 'OK' and 'Cancel' buttons are located at the bottom right.

Adding multiple times creates a list of group members to which common settings ("OutBound Header Manipulation") and voting rules apply.

This trunk type can be created multiple times and can be provided with members.

2.5. Routing rules

Each trunk has its own set of rules for handling incoming calls. A rule refers directly to one or more destinations - nothing more is needed to handle a call.

Example of a rule from SIP trunk to Skype for Business - 4-digit extensions starting with 1:

The screenshot shows the 'Call Processing' configuration window with the 'Advanced' tab selected. The 'General' section includes a 'Display Name' field containing 'Rufe zu Skype4B' and a 'Type' dropdown set to 'Routing'. The 'Match and Replace' section features a 'Pre-Check Number List' dropdown set to '<Not specified>' and an unchecked checkbox for 'Treat as negative lookup list'. Below this are four rows of fields: 'To', 'From', 'P-Asserted-Id (PAI)', and 'Diversion'. Each row has a 'Pattern' field and a 'Replace with' field. The 'To' pattern is '(\\+493012341...)' and its replacement is '\\1'. The 'From' pattern is '(.*)' and its replacement is '\\2'. The 'PAI' pattern is '(.*)' and its replacement is '\\3'. The 'Diversion' pattern is '(.*)' and its replacement is '\\4'. The 'Destination' list contains one entry, 'S4BTLS'. The 'Selection Mode' dropdown is set to 'Load Balancing'. At the bottom, there is a 'Disable the rule' checkbox, 'OK', and 'Cancel' buttons.

As in older firmware versions, a number list (from OfficeMaster Directory Service) can be specified as an additional condition. In this example, a certain number range is selected by a "Regular Expression" - the plus sign at the beginning must therefore be preceded by a \, so that the comparison takes place with + as a normal sign.

Hint:

If several targets have been selected, you can choose between "Load Balancing" and "Failover" in the "Selection Mode" - this possibility did not exist in previous firmware versions!

Example of a multiple destination with "failover" - the priority is determined by the order of the destinations:

General

Display Name:

Type:

Match and Replace

Pre-Check Number List:

Treat as negative lookup list

Pattern: Replace with:

To:

From:

P-Asserted-Id (PAI):

Diversion:

Destination

- SIP-Amt
- BRI 1
- PRI 1

2.6. Correction rules for outgoing calls

Each trunk can be addressed as a destination by different other trunks. It may happen that different phone number formats are used depending on the source. Some SIP trunks use E.164, while others use different schemes (e.g. 004930... or 030...). To work with mixed sources, correction rules ("Outbound Header Manipulation") are used at the outgoing trunk.

Example of conversion to E.164 for Skype for Business:

Outbound Header Manipulation		
Name	Rule	Substitution
Internationale Nummern	00(.*) (.*) (.*) (.*)	+1, \2, \3, \4
Nationale Nummern	0(.*) (.*) (.*) (.*)	+49 \1, \2, \3, \4

If outgoing SIP trunks do not support E.164 format, the following strategy is recommended: All incoming rules are always converted to E.164 (this may require multiple rules). This is used to generate the format that the respective SIP trunk expects - including the optional specification of the sender as "P-Asserted-Identity" (PAI) header.

3. Example configurations

Here a few typical examples in the environment of OfficeMaster are documented.

3.1. Fax with OfficeMaster DirectSIP and other FoIP remote peers

As already mentioned in section 1 the "old" control with the OMCUMS component is no longer supported. Instead, the new firmware behaves as a fax proxy, which means that any signaling regarding T.38 or G.711 is passed through to the other side. The fax behavior (T.38 Send / Accept / Reject Re-Invite) is thus controlled by the remote stations involved. If both sides agree on T.38, T.38 UDPTL packets are passed through transparently, otherwise faxes are sent via G.711 pass-through.

3.2. Use with SIP2Lync

The SIP2Lync server is initially set up as a separate trunk.

Example:

The screenshot shows the 'Advanced Configuration' window for the 'SIP2Lync' trunk. The interface includes a navigation bar with various trunk types (BRI, Analog, PRI, PCM, SIP-Amt, OM G.711, OM T.38, SIP2Lync) and tabs for 'Routing Rules (SIP)' and 'Settings'. The 'Settings' tab is active, displaying the following configuration:

- Trunk Settings:**
 - Domain (IP Address/Hostname): Port:
 - Transport Protocol:
 - Proxy: Port:
 - Register: On Off
 - Username:
 - Authentication Username: SIP username is used when empty.
 - Password:
 - IP Address:
 - Add. Parameters:
 - Destination Profile:
- Number of Channels to Use:**
 - Outbound: of Total
 - Inbound:

Furthermore, a "register" trunk is used for the SIP devices. Now only one rule has to be added between the two trunks.

In the SIP2Lync Trunk:

The screenshot shows the 'Call Processing' configuration window with the 'Advanced' tab selected. The configuration is as follows:

- General:**
 - Display Name: Rufe von SIP2Lync zu SIP-Geräten
 - Type: Routing
- Match and Replace:**
 - Pre-Check Number List: <Not specified>
 - Treat as negative lookup list
 - Pattern: Replace with:
 - To: (.*) \1
 - From: (.*) \2
 - P-Asserted-Id (PAI): (.*) \3
 - Diversion: (.*) \4
- Destination:** Register

In the "Register" Trunk:

The screenshot shows the 'Call Processing' configuration window with the 'Advanced' tab selected. The configuration is as follows:

- General:**
 - Display Name: Rufe von SIP-Geräten zu SIP2Lync
 - Type: Routing
- Match and Replace:**
 - Pre-Check Number List: <Not specified>
 - Treat as negative lookup list
 - Pattern: Replace with:
 - Called Party Number: (.*) \1
 - Calling Party Number: (.*) \2
 - Calling Party Number 2: (.*) \3
 - Redirecting Number: (.*) \4
- Destination:** SIP2Lync

As before, the global SIP2Lync setting must be configured - nothing more is required.

4. Troubleshooting

In this section you will find help for diagnosis if something does not work as intended. Syslog files are an essential basis for this - logging should therefore always be activated! For the analysis of log files the "Syslog-Analyzer" (syslogwin.exe) is used. It is recommended to activate the setting Edit "→Treat all as Regular Expression" in this tool - then search terms will always be interpreted as Regular Expressions (alternatively a search string can start with a plus sign, then the rest will also be processed as Regular Expression). The following examples assume that the setting described here is active.

4.1. Assignment of incoming SIP calls

If calls are not processed as expected, it is often because the incoming INVITE is not assigned to the correct trunk and therefore the wrong set of rules is used. For the assignment, IP addresses and domain names from SIP headers are compared with the trunk configurations. In addition, a trunk found in this way must contain at least one rule whose conditions are met. This makes it possible to use several trunks with the same domains, which cover different number ranges. These number ranges must be selected accordingly in the call rules. If an assignment is not possible, the sender address or an address range in the "IP Address" field can also be specified as a regular expression (default ".*", i.e. any address). Example: `10\.\10\.` * for addresses in the range 10.10.0.0/16

To check the assignment to trunks in the syslog, the incoming INVITE must first be found. Entering "Rx: INV" is sufficient to list all incoming calls. By clicking on the call to be examined, the INVITE is displayed in the upper window. The processing of the configured trunks (from left to right) and their rules can be seen immediately before the INVITE and can be found there by scrolling back.

Example: Call via Telekom SIP trunk that exists twice for different phone numbers (shortened)

The first two trunks are not active:

ROUTE: check D-Channel 1

ROUTE: check D-Channel 2

Trunk 3 is active, but does not use the Telekom domain:

ROUTE: check D-Channel 3

ROUTE: check D-Channel 3 is active

ROUTE: trunk based destination route: 4d trunk fromHost test if "sip-trunk.telekom.de" is contained in "10.6.1.33" (10.6.1.33)

Trunk 4 is active and configured for Telekom, but there is no rule matching the phone number

ROUTE: check D-Channel 4

ROUTE: check D-Channel 4 is active

ROUTE: trunk based destination route: 4d trunk fromHost test if "sip-trunk.telekom.de" is contained in "sip-trunk.telekom.de" (sip-trunk.telekom.de)

ROUTE: trunk based destination route: 4e trunk fromHost found sip-trunk.telekom.de

ROUTE: RouteIsdnInitiated("+49301234567,+491729876543,,") startAfter='nil' ConfigId=4

ROUTE: test 1 To Skype4B Hans "+49301234567,+491729876543,,", "^(.*) (.*) (.*) (.*)"

ROUTE: test 2 DirectSIP T.38 pass-through "+49301234567,+491729876543,,", "^(.*) (.*) (.*) (.*)"

Trunk 5 is active and configured for Telekom, there is also a suitable rule

ROUTE: check D-Channel 5

ROUTE: check D-Channel 5 is active

```

ROUTE: trunk based destination route: 4d trunk fromHost test if "sip-trunk.telekom.de" is contained
in "sip-trunk.telekom.de" (sip-trunk.telekom.de)
ROUTE: trunk based destination route: 4e trunk fromHost found sip-trunk.telekom.de
ROUTE: RouteIsdnInitiated("+49301234567,+491729876543,,") startAfter='nil' ConfigId=5
ROUTE: test 1 To Skype4B Hans "+49301234567,+491729876543,,", "^(. *7),(. *),(. *),(. *)"
ROUTE: found 1 To Skype4B Hans "+4930455946", "+491729876543", "", "", "
    string_replacedCallingPartyNumber2=\3
string_replacedCalledPartyNumber=+4930455946
string_Destination=trunk
    string_RegularExpressionForIpAddressMatch=.*
    string_replacedRedirectedPartyNumber=\4
bool_InvertNumberList=false
string_RegularExpression=^(.*1),(. *),(. *),(. *)
string_disabled=no
struct_DestinationList
    1
string_Id=2004.a
string_NumberList=
    string_replacedCallingPartyNumber1=\2
string_Description=To Skype4B Hans
ushort_DestinationSelectionMode=0
string_AdditionalParameters=
string_action=stop
struct_DestinationId
    1
destination=7
ROUTE: check D-Channel 6
...

```

At the end the list of found target trunks is given:
D-Channel list is seven,

4.2. Syslog Call Analysis

To evaluate individual SIP calls, they must first be found in the syslog. A participating telephone number can be used as a search term; search terms such as "Tx: INVITE" for outgoing calls or "Rx: INVITE" for incoming calls are also possible. In both cases it is helpful to evaluate the time of the call. Right-clicking within a SIP message displays a context menu. The selection "Filter Call SIP Commands" automatically generates a suitable search term to filter out the entire conversation of this call. By clicking on individual lines in the upper window, the corresponding position in the syslog is positioned.

5. Specialsettings

In certain situations it is necessary to set a special behavior different from the default, e.g. to use certain optional SIP headers, to use other processes (e.g. fax with or without T.38) and so on. There are two different possibilities for this: Parameter profiles and additional parameters.

5.1. Parameter Profiles

These already exist in earlier firmware versions and are used to group several parameters. Such a parameter set can be selected by its name directly in the configuration program.

Example (Destination Profile in Trunk Settings):

The screenshot shows the 'Advanced Configuration' window with the 'SIP-Amt' tab selected. Under 'Trunk Settings', the following fields are visible:

- Domain (IP Address/Hostname): sip.example-carrier.tld
- Transport Protocol: UDP
- Proxy: proxy.example-carrier.tld
- Register: On Off
- Username: ABC123456
- Authentication Username: (empty)
- Password: (masked with dots)
- IP Address: *
- Add. Parameters: (empty)
- Destination Profile: OMG Cisco (Fax via T.38)

The list of selectable parameter profiles can be extended if required. A separate documentation for specialists ("OfficeMaster Gate Previously Undocumented Parameters") describes how to do this and which parameters can be used.

5.2. Additional parameters

At various places in the configuration tool, the field "Add. Parameters" (Additional Parameters). Individual settings can be made here, and several such values can also be combined. The meaning of the parameters and syntax are also described in the document mentioned in section 5.1

5.3. Parameters per trunk / per rule

Parameters can be configured in various places:

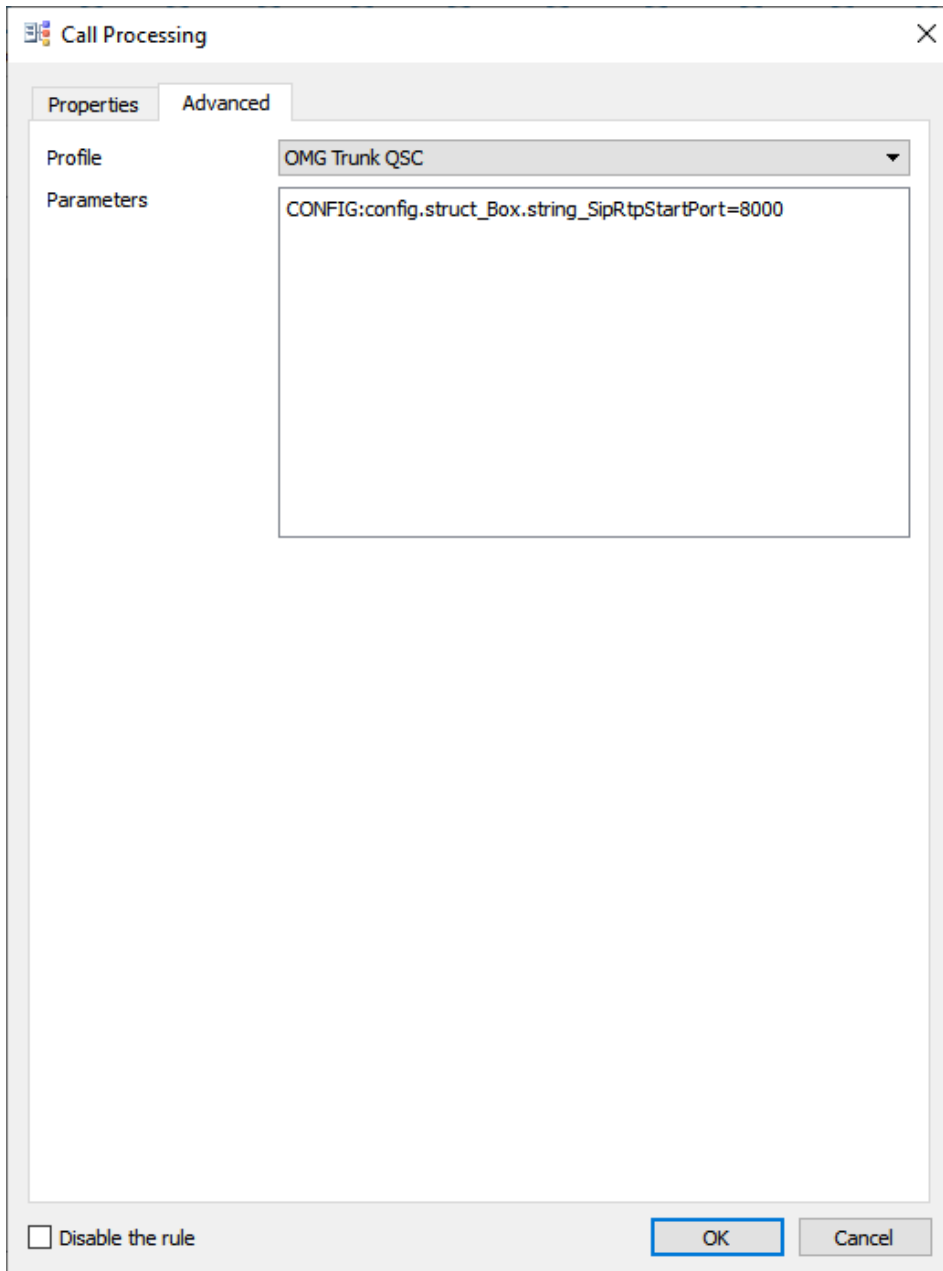
- In the trunk settings
- In a single rule

5.3.1. Parameters in Trunk Settings

Parameter profiles and additional parameters with the prefix "CONFIG:" apply to the respective trunk for both incoming and outgoing calls. If additional parameters are only valid for incoming or only for outgoing calls, the prefix "IN:" or "OUT:" must be used instead of "CONFIG:".

5.3.2. Parameters in a rule

Parameter profiles and/or additional parameters are specified on the "Advanced" tab:



The settings per rule apply to the incoming connection, but not to the outgoing connection!