Manual
Workshops (Excerpt)

Media Gateway Workshops

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Chapter 1 Media Gateway - Connecting ISDN / SIP clients to the SIP provider with bandwidth management

1.1 Introduction

Media gateway serves as a translation instance between different telecommunications networks, e.g between the plain old phone network and the next generation networks (IP networks). With the bintec Media Gateway, a company equipped with an automatic PBX on a wired telephone network can be connected to a SIP Trunking Service Provider on the Internet in order to use IP telephony. The bintec Media Gateway supports the binding of several SIP Provider Accounts.

The following chapter describes how to connect an SIP telephone and an ISDN PBX to the media gateway. The media gateway is connected to an SIP provider and to an external ISDN point-to-multipoint connection simultaneously.

The media gateway is connected to the Internet over an ADSL connection. Consequently, this section also discusses the functions **Application Level Gateway** (NAT Proxy), bandwidth management through the functions **Quality of Service** and **Real Time Jitter Control** (Jitter reduction).

In this example scenario the following call assignment is set up:

Call Assignment

External number	External medium	Internal number	Internal device
0911/2557435	ISDN	10	IP telephone
0911/2558296	ISDN	20	ISDN PBX
0911/30839681	SIP	20	ISDN PBX

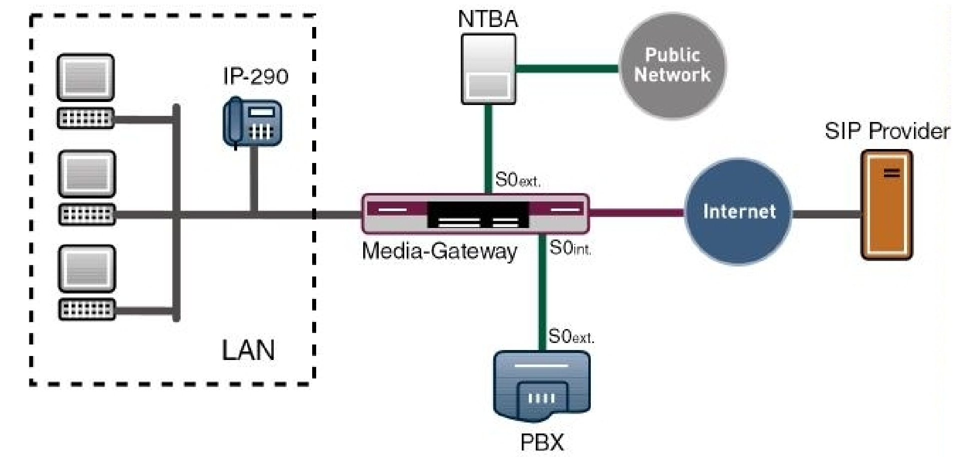

Fig. 1: Example scenario

Requirements

The following are required for the configuration:

- Boot image from version 7.8.2
- A bintec Media Gateway
- The optional DSP module must be set up in the gateway
- The optional licence for the second ISDN interface is required
- The ADSL connection to the gateway must be set up in advance

Configuration in this scenario is carried out using the **GUI** (Graphical User Interface).

1.2 Configuration

1.2.1 Configuring the external ISDN interface

You can use the ISDN BRI interface of your device for both dialup and leased lines over ISDN.

The external ISDN interface is connected directly with an NTBA (Network Termination Basis Connection). On this connection the extensions 2557435 and 2558296 are switched.

Go to the following menu to configure the ISDN interface for your device:

(1) Go to **Physical Interfaces** ->**ISDN Ports** -> **<bri2-0 (TE)>**.

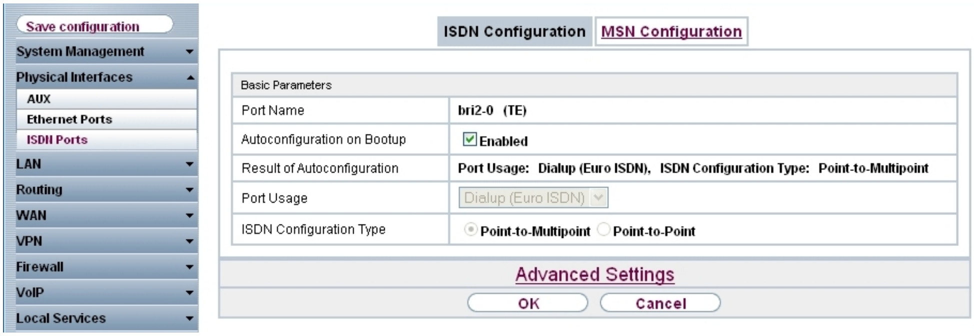

Fig. 2: **Physical Interfaces** -> **ISDN Ports** -> **<bri2-0 (TE)>** .

Relevant fields in the ISDN Configuration menu

Field	Meaning
Port Name	Shows the name of the ISDN port.
Autoconfiguration on Bootup	Here, select whether the ISDN switch type should be automatically recognised.
Result of Autoconfiguration	The status of the ISDN autoconfiguration is displayed here. Automatic D-channel recognition runs until a setting is found. This field cannot be edited.
Port Usage	If the ISDN protocol is not automatically recognised, you must select the port here manually. For this, you must first disable **Automatic Configuration at Start**. Select `Dialup (Euro-ISDN)`.
ISDN Configuration Type	Here, select the ISDN access configuration `Point-to-Multipoint`.

MSN Configuration

If no entry is specified (in the ex works state no MSN number is entered), every incoming ISDN call is accepted by the ISDN Login service. To avoid this, you should make the necessary entries here. As soon as an entry exists, the incoming calls not assigned to any entry are forwarded to the CAPI service.

(1) Go to **Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**.

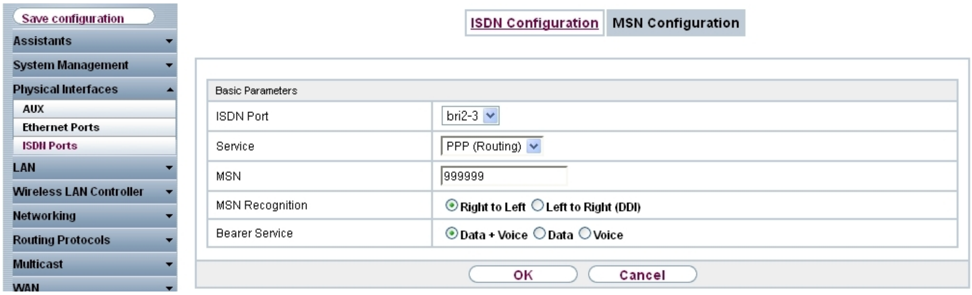

Fig. 3: **Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**

Relevant fields in the MSN Configuration menu

Field	Meaning
ISDN Port	Select the ISDN port for which the MSN is to be configured.
Service	Select the service to which a call is to be assigned on the number below.
MSN	Enter the call number.
MSN Recognition	Select the mode your device is to use for the number comparison for MSN with the `called party number` of the incoming call. By default, the value is set on `right to left`. Select the value `left to right (DDI)` if your device is connected to a point-to-point connection.
Service attribute	Select the type of incoming call (service detection). `Data + Voice` specifies that both data and voice calls (default value) are executed.

1.2.2 Configuring the internal ISDN interface

The optional 1-BRI licence is required to be able to use the second ISDN interface. The second ISDN port is operated as a point-to-multipoint connection in NT mode so that the external ISDN line of the PBX (point-to-multipoint connection; TE mode) can be operated.

To be able to operate the 2. ISDN interface in NT mode, several link plugs (jumpers) in the device must be modified:

For further information on setting the ISDN interfaces see Release Notes 7.5.1 (Chapter: 2.2 Variable switching for ISDN S0 interfaces).

(1) Go to **Physical Interfaces** ->**ISDN Ports**->**<bri2-3 (NT)>**.

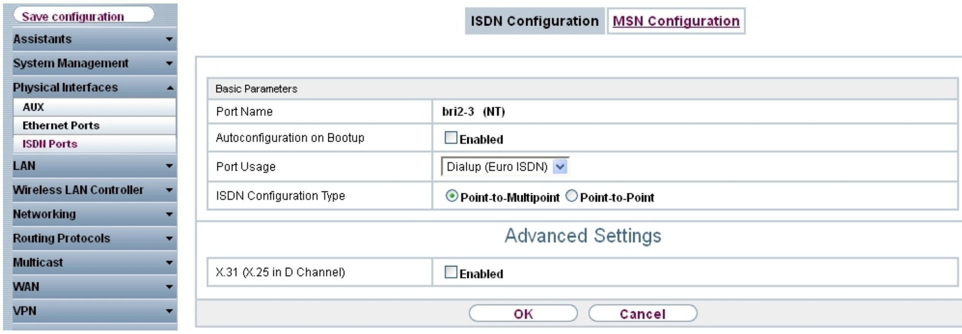

Fig. 4: **Physical Interfaces** ->**ISDN Ports**->**<bri2-3 (NT)>**.

Relevant fields in the ISDN Configuration menu

Field	Meaning
Port Name	Shows the name of the ISDN port.
Port Usage	Here, select *Dialup (Euro ISDN)*.
ISDN Configuration Type	Select the ISDN access configuration *Point-to-Multipoint*.

MSN Configuration

As on the external ISDN port, a dummy subscriber number must also be saved for the internal ISDN port.

(1) Go to **Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**.

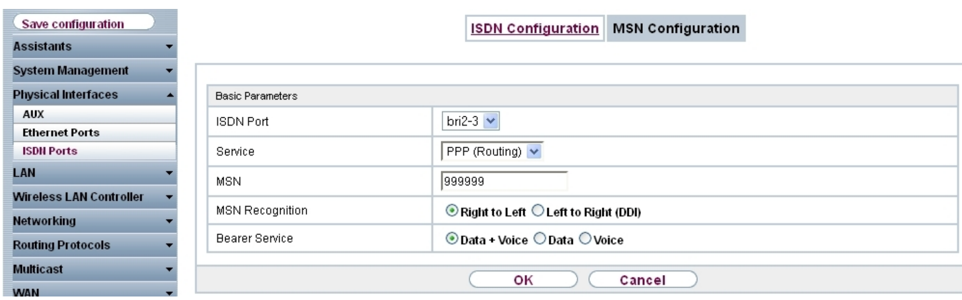

Fig. 5: **Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**

Relevant fields in the MSN Configuration menu

Field	Meaning
ISDN Port	Select the ISDN port for which the MSN is to be configured.
Service	Select the service to which a call is to be assigned on the number below.

Field	Meaning
MSN	Enter the call number.
MSN Recognition	Select the mode your device is to use for the number comparison for MSN with the `called party number` of the incoming call. By default, the value is set on `right to left`.
Service attribute	Select the type of incoming call (service detection). `Data + Voice` specifies that both data and voice calls (default value) are executed.

1.2.3 Enabling the Application Level Gateway for dynamic monitoring of the NAT and firewall

To enable IP telephones to connect by SIP to a VoIP Provider your device has an **Application Level Gateway** (ALG), i.e. an appropriate proxy that implements the necessary NAPT and firewall releases.

Note

The Application Level Gateway must always be used if NAT is enabled on the interface that makes the connection to the Internet.

In the **VoIP** -> **Application Level Gateway** menu, a list is shown of all configured Application Level Gateway entries. These entries enable the ALG. Each entry defines a particular TCP or UDP destination port that is to be supervised by the ALG. In the ex works state, there are two entries configured for the SIP Ports TCP 5060 and UDP 5060 in accordance with the IANA definition.

The router is connected to the internet via an ADSL line. The NAT firewall is active for this connection. No portforwarding is required to open the NAT firewall for the VoIP data (SIP and RTP), instead the **Application Level Gateway** must be enabled. The **Low Latency Transmission** option highlights the VoIP data for the high priority queue relating to QoS.

Go to the following menu to configure an Application Level Gateway entry:

(1) Go to **VoIP** -> **Application Level Gateway** -> **SIP-Proxys** ->**<SIP UDP 5060>** .

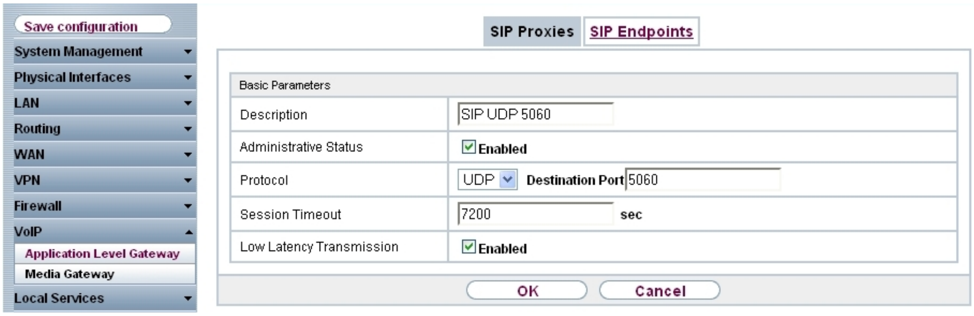

Fig. 6: **VoIP** -> **Application Level Gateway** -> **SIP-Proxys** -><**SIP UDP 5060**> .

Relevant fields in the SIP Proxies menu

Field	Meaning
Description	Displays the name of the Application Level Gateway entry.
Administrative Status	Determines whether or not the proxy should be active.
Protocol	Defines the protocol to be used.
Destination Port	Here you enter the port to be supervised by the proxy.
Session Timeout	Shows the time in seconds for which a session stays up if no data packets are sent or received.
Low Latency Transmission	Mechanism to minimise the transit time of VoIP data packets between two subscribers. This guarantees good voice quality with high line load. Note that **Low Latency Transmission** does not have to be switched on if the media gateway supervises the VoIP connection. The voice quality is optimised by choosing `Enabled`.

1.2.4 Registering the router with VoIP provider sipgate.de

If your want your device is to connect to other SIP servers (e.g. servers of Internet SIP Service providers), you can configure the necessary entries here.

Note

In no case should you use this menu to configure extensions, i.e. for SIP clients or PSTN clients such as SIP telephones, terminal adapters or ISDN telephones Extensions can be configured in the menu **VoIP**-> **Media Gateway** -> **Call transformation** -> **New**.

After installing the DSP module, you can save the login data for the extension.

(1) For this, go to **VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**.

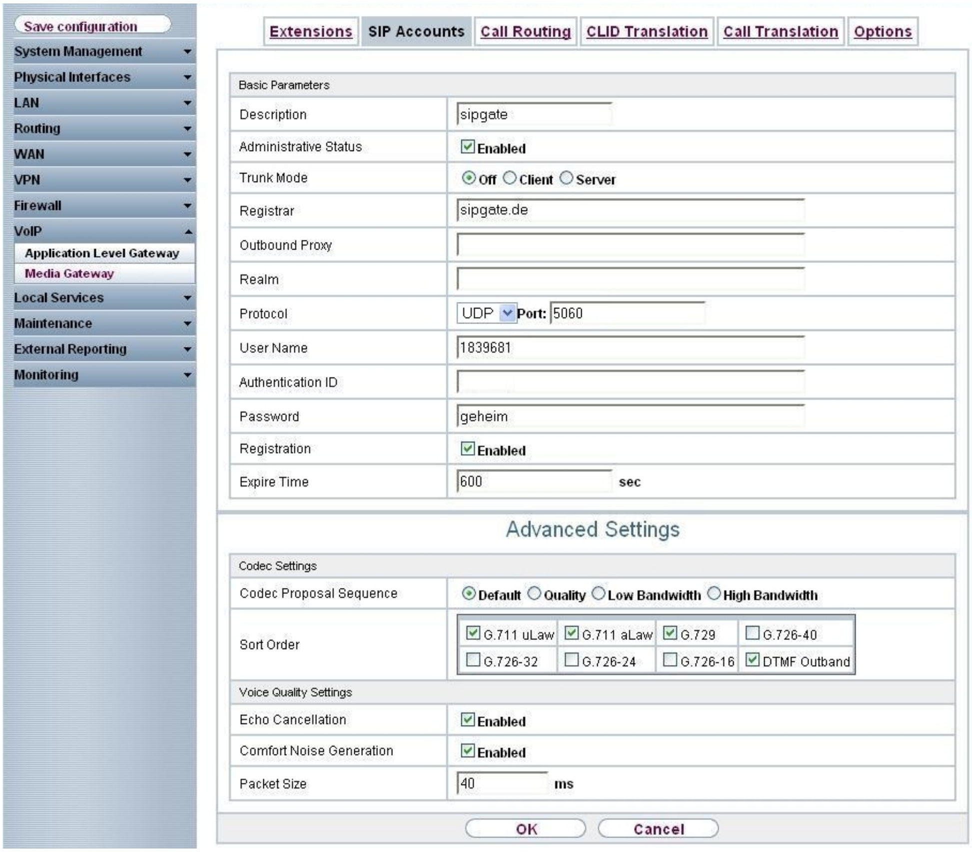

Fig. 7: **VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**

Relevant fields in the SIP Accounts menu

Field	Meaning
Description	Here, assign a name to the account. Maximum number of characters: 40.
Administrative Status	Enable the administrative status of the account.
Trunk Mode	Select the trunk mode to be used. If you select `Off`, the trunk mode is not used.
Registrar	Enter the IP address of the remote SIP terminal (client or server) here. Maximum number of characters: 40.
Protocol	Select the protocol to be used for the connection to the server or proxy.

Field	Meaning
Port	Number of the TCP or UDP port to be used for the connection to the server or proxy.
User Name	Here, enter the username for authentication if your VoIP provider has assigned one to you.
Authentication ID	Enter a name that is to be used for authentication. If you do not enter a name, the name in the **User Name** field is used.
Password	The VoIP provider gives you a PIN or password for authentication. You must enter this value here. Maximum number of characters: 40.
Registration	Enables or disables the SIP REGISTER registration mechanism.
Expire Time	Shows the time in seconds after which the current registration becomes invalid and a new registration request is therefore sent.

In the **Advanced settings** menu, you can define which codecs are used for the chosen account.

Note

The codecs actually used are the intersect of the codecs defined here and those signalled by the provider. For outgoing calls, any remaining codecs are dropped from the list that would require more than the available bandwidth.

Relevant fields in the menu Advanced Settings

Field	Meaning
Codec Proposal Sequence	Determine the order in which the codecs are offered for use by the media gateway. If the first codec cannot be applied, an attempt is made to use the second codec, and so on. Set **Codec Proposal Sequence** to *default*. The codec in the first position will be used. You can sort the codecs according to quality or bandwidth.
Sort Order	Select the codecs to be proposed for the connection. The codecs chosen here are proposed in a certain order, depending on the setting in the Codec Proposal Sequence field.
Echo cancellation	Enable or disable echo cancellation. If *Enabled* is selected, echo feedback is suppressed.

Field	Meaning
Comfort Noise Generation (CNG)	Specify whether Comfort Noise Generation should be used. The slight comfort noise generation prevents subscribers from thinking that the connection is lost during pauses.
Packet Size	The transmission time of an RTP data packet in milliseconds. Possible values: 10 ... 60.

If registration with the VoIP provider is successful, the status in the provider menu shows . The status of the VoIP connection is changed by pressing the button or button in the **Action** column.

(1) Go to **VoIP** -> **Media Gateway** -> **SIP Accounts**.

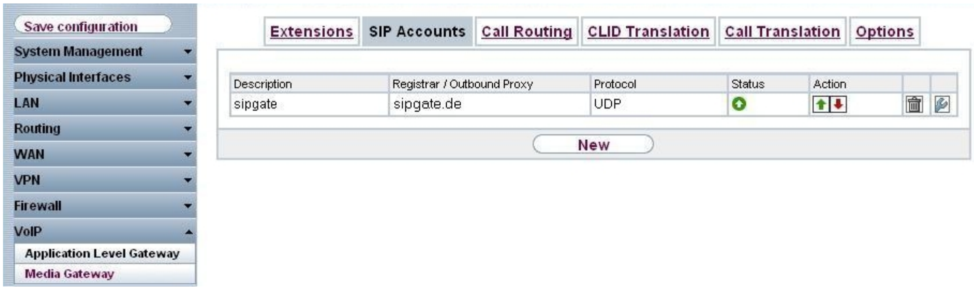

Fig. 8: **VoIP** -> **Media Gateway** -> **SIP Accounts**

1.2.5 Configuring the internal extension

Here you can configure the subscriber numbers for the terminals that are connected to the media gateway.

In this example two internal extensions are used. Extension 10 for the IP telephone and extension 20 for the PBX connected to the internal ISDN port.

Configuring extension 10 - IP telephone

(1) Go to **VoIP** -> **Media Gateway** -> **Extensions** -> **New**.

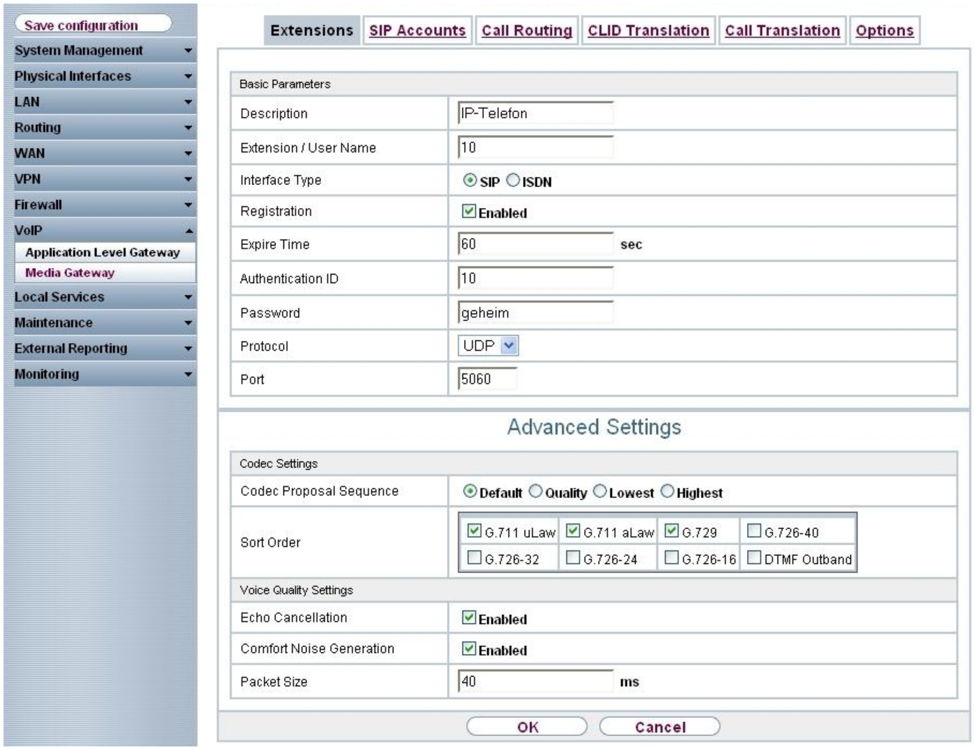

Fig. 9: **VoIP** -> **Media Gateway** -> **Subscriber** -> **New**

Relevant fields in the Extensions menu

Field	Meaning
Description	Enter the name of the terminal.
Extension / User Name	Enter the call number. A maximum of 40 characters can be entered.
Interface Type	Select the interface type to be used.
Registration	Enables or disables the SIP REGISTER registration mechanism.
Expire Time	Enter the time in seconds after which the current registration becomes invalid and a new registration request is therefore sent.
Authentication ID	Here you can enter a name that is to be used for authentication. The name given here must also be entered on the SIP telephone. Maximum number of characters: 20.
Password	Enter a password here. The password given here must also be

Field	Meaning
	entered on the SIP telephone. Maximum number of characters: 20.
Protocol	Select the protocol to be used for data transmission.
Port	Select the port to be used for data transmission.

In the **Advanced Settings** menu you can select the possible codecs for the account.

Relevant fields in the menu Advanced Settings

Field	Meaning
Codec Proposal Sequence	Determine the order in which the codecs are offered for use by the media gateway. If the first codec cannot be applied, an attempt is made to use the second codec, and so on. Set **Codec Proposal Sequence** to *default*. The codec in the first position will be used. You can sort the codecs according to quality or bandwidth.
Sort Order	Select the codecs to be proposed for the connection. The codecs chosen here are proposed in a certain order, depending on the setting in the Codec Proposal Sequence field.
Echo cancellation	Enable or disable echo cancellation. If *Enabled* is selected, echo feedback is suppressed.
Comfort Noise Generation (CNG)	Specify whether Comfort Noise Generation should be used. The slight comfort noise generation prevents subscribers from thinking that the connection is lost during pauses.
Packet Size	The transmission time of an RTP data packet in milliseconds. Possible values: 10 ... 60.

Configuring extension 20 - internal PBX

(1) Go to **VoIP** -> **Media Gateway** -> **Extensions** -> **New**.

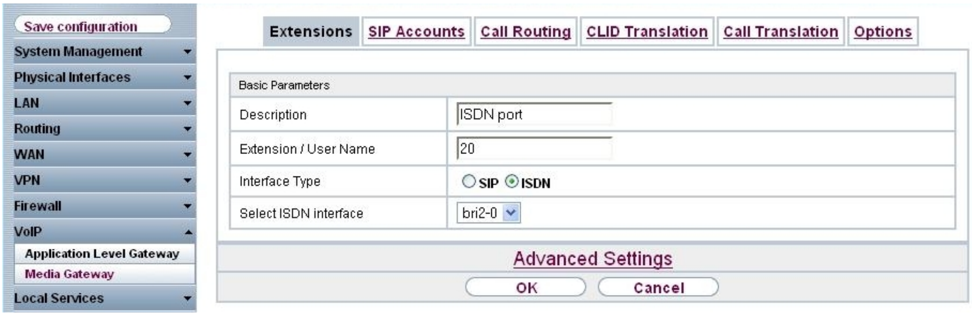

Fig. 10: **VoIP** -> **Media Gateway** -> **Subscriber** -> **New**

Relevant fields in the Extensions menu

Field	Meaning
Description	Enter the name of the terminal.
Extension / User Name	Enter the call number. A maximum of 40 characters can be entered.
Interface Type	Terminal type, an internal PBX is used for the call. The `ISDN` setting can only be selected if ISDN interfaces with the **ISDN Configuration Type** = `Dialup (Euro ISDN) point to multipoint (TE Mode)` is set.
Select ISDN interface	Select an ISDN interface. The ISDN interfaces you can select depends on the device used.

1.2.6 Call assignment - Call Routing - CLID Translation

You can configure the PBX functions for the media gateway in the menu **VoIP** -> **Media Gateway** -> **Options**.

In addition to the call assignment, here you can configure the delay between entering the subscriber number and starting to dial (**Dialling**).

(1) Go to **VoIP** -> **Media Gateway** -> **Options**.

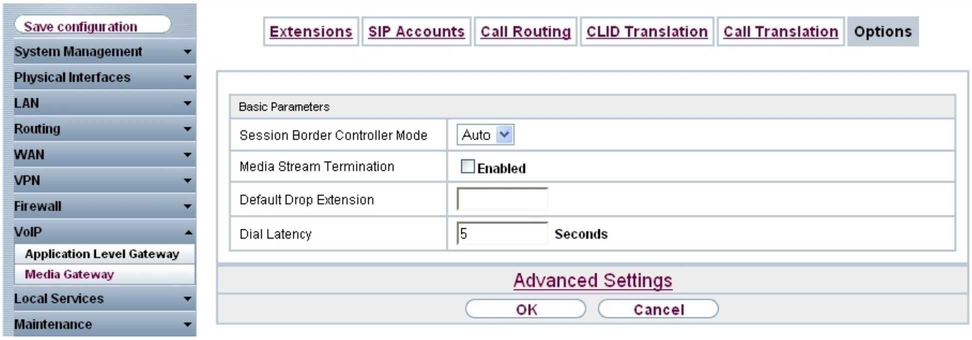

Fig. 11: **VoIP** -> **Media Gateway** -> **Options**

Relevant fields in the Options menu

Field	Meaning
Session Border Controller Mode	Determines the behaviour of the media gateway in combination with a session border controller. • *Auto*: for all extensions that exactly agree with an existing account, the call routing is handled by the session border controller, i.e. all SIP messages configured for the corresponding account are forwarded to the session border controller. For all other extensions, call routing is handled by the media gateway in accordance with the configured call routing entries. Note that the call routing is handled by the media gateway if the provider is not available (backup). • *Off*: call routing is handles exclusively by the media gateway in accordance with the configured call routing and the local extensions. For calls that are to be routed via a particular provider (account), you must configure a corresponding call routing entry. Internal calls (from internal extension to internal extension) that are only to be routed internally do not require an additional call routing entry. • *<SIP Trunk>*: A SIP trunk account is configured and selected for the session border controller under **VoIP** -> **Media Gateway** -> **SIP Accounts**. In this case, the call routing for all extensions is handled by the session border controller, all SIP messages configured for the selected account are forwarded to the session border controller. Note that the call routing is handled by the media gateway if the provider is not available (backup).
Media Stream Termination	Determines how RTP sessions are controlled by the system.

Field	Meaning
	• *Enabled*: RTP sessions are terminated on the media gateway, i.e. all RTP streams are controlled by the media gateway and routed via the media gateway. The participating terminal devices (e.g. SIP telephones) are not connected directly with one another. Note that, for VoIP to VoIP connections, there is no code translation for different VoIP terminal codecs. This is why the codecs from media gateway and VoIP terminals must match; the RTP sessions are not terminated on the media gateway, i.e. all RTP streams are routed from the media gateway without termination. The RTP data packets can be routed in complex networks and thus also via other gateways. • *Disabled* (default value): RTP sessions are not terminated on the media gateway, i.e. all RTP streams are routed by the media gateway without termination. The RTP data packets can be routed in complex networks and thus also via other gateways.
Default Drop Extension	Here you can nominate an extension to receive calls that cannot be routed because there is no valid routing entry for them.
Dialling break	Maximum delay time before the system assumes the telephone number entered is complete and starts the SIP dialling process (sends the SIP INVITE message). This timeout is reset each time that a button is pressed. If you terminate the number entered with #, dialling is immediate.

Call Routing

In the **Call Routing** menu, you can define the conditions for the routing of calls.

In this example, an outgoing call made to a number starting with a 0 is routed with the same subscriber number to the external ISDN connection. If the destination number starts with a 9, the 9 (which is used as a trunk prefix) is replaced by 0049 and is sent over the VoIP connection to the provider sipgate.

Since the internal ISDN connection has been configured as a point-to-multipoint connection, the **type** *external* is used instead of a *trunk*. Proceed as follows to configure the VoIP connection to the provider sipgate.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**.

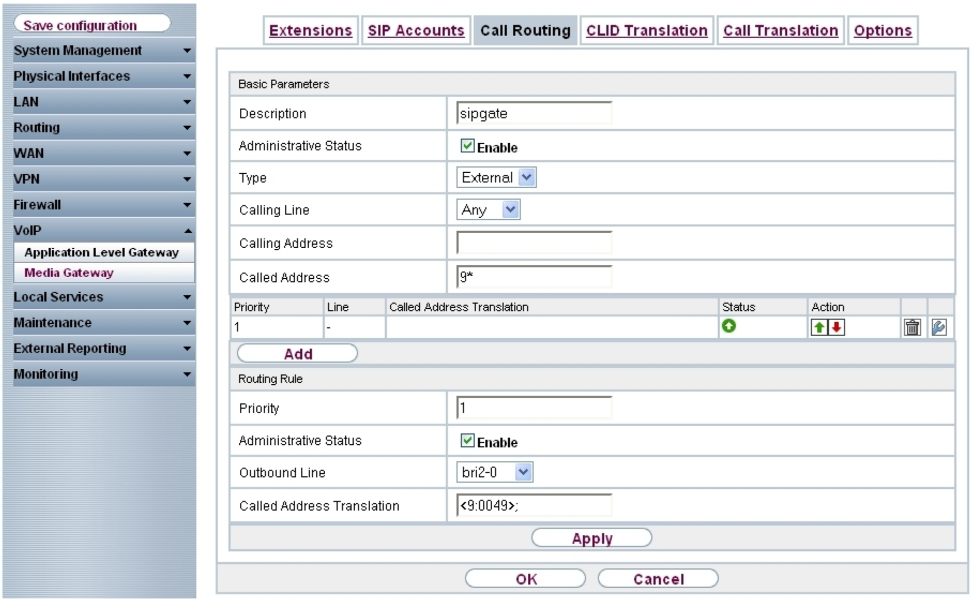

Fig. 12: **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**

Relevant fields in the Call Routing menu

Field	Meaning
Description	Here, enter the name of the call routing entry.
Administrative Status	The entry is used with `enabled`.
Type	Select `External` for calls that are to be routed as outgoing, external calls.
Calling Line	Here you can restrict the routing entry to the line on which the call comes in.
Calling Address	Here you can restrict the routing entry to a particular caller. To do this, you must specify the subscriber number exactly (no wildcards).
Called Address	Here, you can enter an address (call number) that is compared with the dialled address. You can use wildcards here. For example `9*` means that at the end of a character string an arbitrary number of any characters can follow.

You can now create a list with rules that are assigned to the currently selected routing entry, and that serve to manipulate the signalled destination number. You can also delete routing entries.

Use **Add** to create entries.

Relevant fields in the Routing Rule menu

Field	Meaning
Priority	Determines the order of the filter rules, starting with *1* in increasing numerical order.
Administrative Status	The entry is used with *Enable*.
Outbound Line	Defines the PSTN line (PRI, BRI, FXO) or the SIP account used for an outgoing call.
Called Address Translation	The rule shown <9:0049> indicates how the destination number is manipulated. Before it is used for a call, the *9* used as the trunk prefix is replaced by the number *0049*.

CLID Translation

In the **CLID Translation** menu, you can create a list for the translation of subscriber numbers, i.e. this list associates internal and external numbers.

The **CLID Translation** is crucial in this example for incoming calls. For incoming ISDN calls to extension 2557435, this number is replaced by extension 10. The connection is routed to local extension 10 (SIP telephone). Calls on the SIP account and on the ISDN number 2558296 will be routed to the PBX connection.

(1) Go to **VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**.

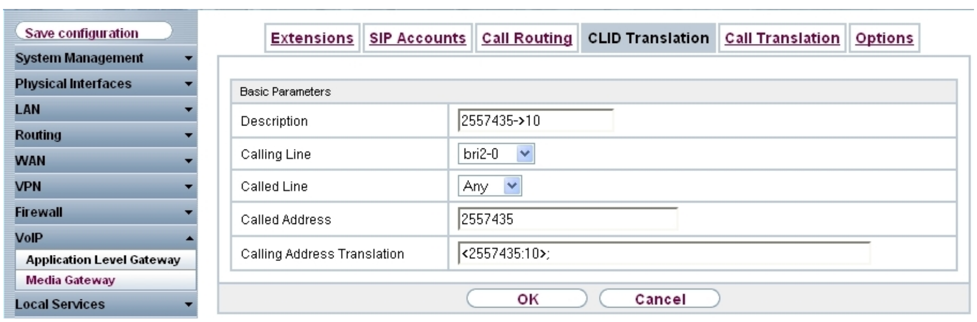

Fig. 13: **VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**

Relevant fields in the CLID Translation menu

Field	Meaning
Description	Here, enter the name of the CLID translation entry.
Call number	Select the line or SIP account via which the calls are to be routed.
Called Line	Here you enter the direction to which the entry is to apply.

Field	Meaning
	Select *Any* for incoming and outgoing calls (bidirectional).
Called Address	Here you have the option of entering the destination address of the call.
Calling Address Translation	Enter the transformation rule applied to the call numbers.

Proceed as with configuration of numbers 1839681 -> 20 and 2558296 -> 20. The final configuration looks like this:

(1) Go to **VoIP** -> **Media Gateway** -> **CLID Translation** .

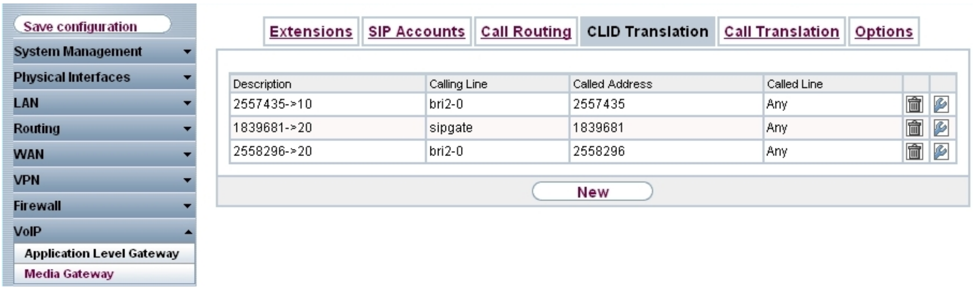

Fig. 14: **VoIP** -> **Media Gateway** -> **CLID Translation**

Real Time Jitter Control

For telephone calls over the Internet, VoIP packets normally have the highest priority. Nevertheless, if the upstream bandwidth is low, noticeable delays in voice transmission can occur when other packets are routed at the same time. The **Real Time Jitter Control** function in the VoIP implementation solves this problem. So as not to block the "line" for VoIP packets for too long, the size of other data packets is reduced if need be during a telephone call.

(1) Go to **WAN**->**Real Time Jitter Control**->**Controlled Interfaces**->**New**.

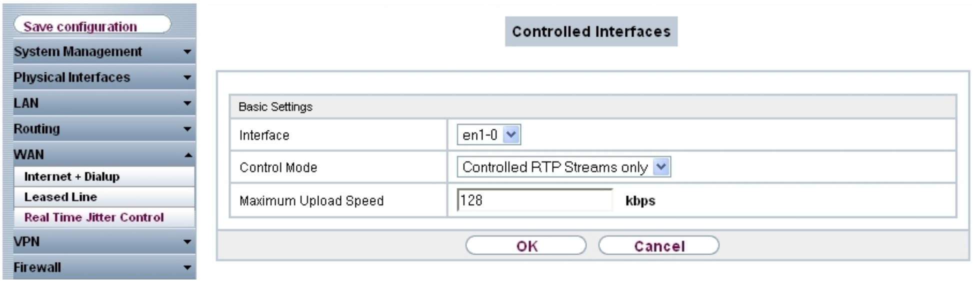

Fig. 15: **WAN**->**Real Time Jitter Control**->**Controlled Interfaces**-> **New**

Relevant fields in the Controlled Interfaces menu

Field	Meaning
Interface	Here you select the Interface on which the voice transmission is to be optimised.
Control Mode	Select the mode for the optimisation. Select *Controlled RTP Streams only*: By means of the data routed through the media gateway, the system recognises VoIP data traffic and optimises the voice transmission. This setting should always be used together with the media gateway.
Maximum Upload Speed (kbit/s)	If you're using an external DSL modem, enter the bandwidth in the upload directions in kbit/s for the selected interface.

Policies

If the Internet connection of the router is used for normal internet traffic or VPN connections, for example, in addition to VoIP data traffic, **QoS** must be enabled.

When setting up the **Application Level Gateway** option, enabling **Low Latency Transmission** automatically performed QoS classification. All VoIP data (SIP and RTP) are highlighted for the *High Priority Queue*. To improve QoS monitoring, a *high priority Queue* without bandwidth restriction should be stored in the QoS configuration alongside the *default Queue*.

(1) Go to **Firewall** -> **Policies** -> **Filter Rules** ->**New**.

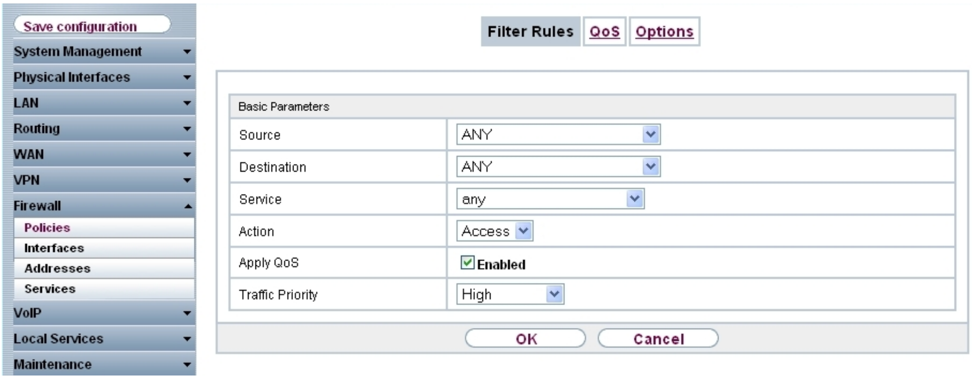

Fig. 16: **Firewall**->**Policies**->**Filter Rules**->**New**

Relevant fields in the Filter Rules menu

Field	Meaning
Source Location	Select one of the preconfigured aliases for the source of the packet. The value `ANY` means that neither the source interface nor the source address is checked.
Destination	Select one of the preconfigured aliases for the destination of the packet. The value `Any` means that neither the destination interface nor the destination address is checked.
Service	Select one of the preconfigured services to which the packet to be filtered must be assigned.
Action	Select the action to be applied to a filtered packet. If you select the `Access` option, packets are passed according to the instruction.
Apply QoS	Select whether you want to enable QoS for this policy with the priority selected in **Data Traffic Priority**
Traffic Priority	Select the priority with which the data specified by the policy is handled on the send side.

Monitoring

The **Monitoring** menu contains submenus that enable you to locate problems in your network and monitor activities, e.g. for bandwidth management.

(1) Go to **Monitoring** -> **Interfaces** -> **Statistics**.

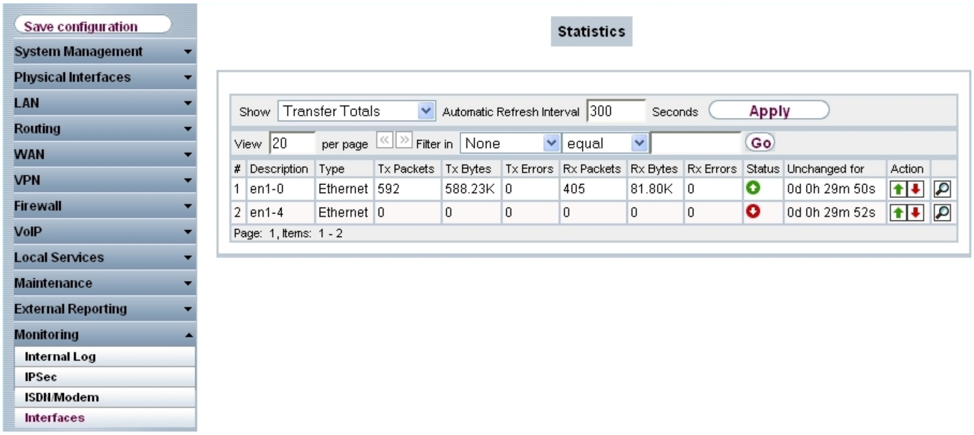

Fig. 17: **Monitoring** -> **Interfaces** -> **Statistics**

You change the state of the interface by pressing the button or button in the Action column. Press the button to display the statistical data for the individual interfaces in detail.

1.3 Overview of configuration steps

Configuring the external ISDN interface

Field	Menu	Value
Autoconfiguration on Bootup	**Physical Interfaces -> ISDN Ports -> ISDN Configuration -> <bri2-0 (TE)>**	`Aktiviert`
Result of Autoconfiguration	**Physical Interfaces -> ISDN Ports -> ISDN Configuration -> <bri2-0 (TE)>**	`Port Usage: Dialup (Euro ISDN), ISDN Configuration Type: Point-to-multipoint`

MSN Configuration

Field	Menu	Value
ISDN Port	**Physical Interfaces -> ISDN Ports -> MSN Configuration -> New**	`bri2-0`
Service	**Physical Interfaces -> ISDN Ports -> MSN Configuration -> New**	e.g. `ISDN Login`
MSN	**Physical Interfaces -> ISDN**	e.g. `999999`

Field	Menu	Value
	Ports -> **MSN Configuration** -> **New**	
MSN Recognition	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*Right to Left*
Service attribute	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*Data + Voice*

Configuring the internal ISDN interface

Field	Menu	Value
Port Usage	**Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-3 (NT)**	*Dialup (Euro ISDN)*
ISDN Configuration Type	**Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-3 (NT)**	*Point-to-multipoint*

MSN Configuration

Field	Menu	Value
ISDN Port	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*bri2-3*
Service	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	e.g. *PPP (routing)*
MSN	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	e.g. *999999*
MSN Recognition	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*Right to Left*
Service attribute	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*Data + Voice*

Application Level Gateway

Field	Menu	Value
Description	**VoIP** -> **Application Level**	*SIP UDP 5060*

Field	Menu	Value
	Gateway -> **SIP Proxies**	
Administrative Status	**VoIP** -> **Application Level Gateway** -> **SIP Proxies**	*Aktiviert*
Protocol	**VoIP** -> **Application Level Gateway** -> **SIP Proxies**	*UDP*
Destination Port	**VoIP** -> **Application Level Gateway** -> **SIP Proxies**	*5060*
Session Timeout	**VoIP** -> **Application Level Gateway** -> **SIP Proxies**	*7200*
Low Latency Transmission	**VoIP** -> **Application Level Gateway** -> **SIP Proxies**	*Aktiviert*

Configuration of SIP accounts

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *sipgate*
Administrative Status	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Aktiviert*
Trunk Mode	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Off*
Registrar	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *sipgate.de*
Protocol	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*UDP*
Port	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*5060*
User Name	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *1839681*
Password	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *secret*
Registration	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Aktiviert*
Expire Time	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*600*
Codec Proposal Sequence	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**-> **Ad-**	*Default*

Field	Menu	Value
	vanced Settings	
Sort Order	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**-> **Advanced Settings**	*G.711 uLaw*, *G.711 aLaw*, *G.729*, *DTMF Outband*
Echo cancellation	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**-> **Advanced Settings**	*Aktiviert*
Comfort Noise Generation (CNG)	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**-> **Advanced Settings**	*Aktiviert*
Packet Size	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**-> **Advanced Settings**	e.g. *40*

Configuring the internal extension

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	e.g. *IP Telephone*
Extension / User Name	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	e.g. *10*
Interface Type	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	e.g. *SIP*
Registration	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	*Aktiviert*
Expire Time	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	*60* Sec
Authentication ID	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	e.g. *10*
Password	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	e.g. *secret*
Protocol	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	e.g. *UDP*
Port	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	*5060*
Codec Proposal Sequence	**VoIP** -> **Media Gateway** -> **Extensions** -> **New**-> **Advanced Settings**	*Default*
Sort Order	**VoIP** -> **Media Gateway** ->	*G.711 uLaw*, *G.711 aLaw*,

Field	Menu	Value
	Extensions -> **New**-> **Advanced Settings**	*G.729*, *DTMF Outband*
Echo cancellation	**VoIP** -> **Media Gateway** -> **Extensions** -> **New**-> **Advanced Settings**	*Aktiviert*
Comfort Noise Generation (CNG)	**VoIP** -> **Media Gateway** -> **Extensions** -> **New**-> **Advanced Settings**	*Aktiviert*
Packet Size	**VoIP** -> **Media Gateway** -> **Extensions** -> **New**-> **Advanced Settings**	e.g. *40*

Configuring the internal PBX

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	e.g. *ISDN port*
Extension / User Name	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	e.g. *20*
Interface Type	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	e.g. *ISDN*
Select ISDN interface	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	e.g. *bri2-3*

Configuration of PBX functions

Field	Menu	Value
Session Border Controller Mode	**VoIP** -> **Media Gateway** -> **Options**	*Auto*
Media Stream Termination	**VoIP** -> **Media Gateway** -> **Options**	Disabled
Dialling break	**VoIP** -> **Media Gateway** -> **Options**	*5* seconds

Call Routing

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *sipgate*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Enable*

Field	Menu	Value
Type	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*External*
Calling Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Any*
Called Address	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *9**
Priority	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*1*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*Enable*
Outbound Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	e.g. *bri2-0*
Called Address Translation	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	e.g. *<9:0049>;*

CLID Translation

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *2557435-> 10*.
Call number	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *bri2-0*
Called Line	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	*Any*
Called Address	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *2557435*
Calling Address Translation	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *<2557435:10>;*
Description	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *1839681-> 20*.
Call number	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *sipgate*
Called Line	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	*Any*
Called Address	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *1839681*
Calling Address Translation	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *<1839681:20>;*.

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *2558296-> 20*.
Call number	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *bri2-0*
Called Line	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	*Any*
Called Address	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *2558296*
Calling Address Translation	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *<2558296:20>;*.

Controlled Interfaces

Field	Menu	Value
Interface	**WAN**->**Real Time Jitter Control**->**Controlled Interfaces**->**New**	e.g. *en1-0*
Control Mode	**WAN**->**Real Time Jitter Control**->**Controlled Interfaces**->**New**	*Controlled RTP only*
Maximum Upload Speed	**WAN**->**Real Time Jitter Control**->**Controlled Interfaces**->**New**	e.g. *128 kbit/s*.

Filter Rules

Field	Menu	Value
Source Location	**Firewall**->**Policies**->**Filter Rules**->**New**	e.g. *ANY*
Destination	**Firewall**->**Policies**->**Filter Rules**->**New**	e.g. *ANY*
Service	**Firewall**->**Policies**->**Filter Rules**->**New**	e.g. *any*
Action	**Firewall**->**Policies**->**Filter Rules**->**New**	*Access*
Apply QoS	**Firewall**->**Policies**->**Filter Rules**->**New**	*Aktiviert*
Traffic Priority	**Firewall**->**Policies**->**Filter Rules**->**New**	*High*

Chapter 2 Media Gateway - Connection of an Asterisk IP PBX to a point-to-point ISDN access

2.1 Introduction

The following chapter describes how to configure the **bintec R4100** as a media gateway to connect an Asterisk IP PBX to a point-to-point ISDN access.

A few extracts are shown from the Asterisk IP PBX configuration to ensure that registration is successful, and that the subscriber number is transmitted correctly. At the exchange, an ISDN party line is used (consisting of two point-to-point ISDN accesses with four B-channels) with the subscriber number 0911/7660069(0-9).

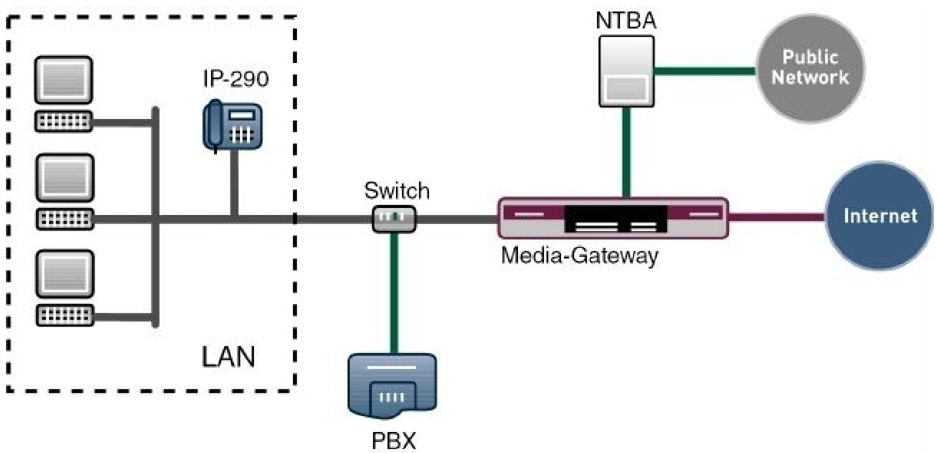

Fig. 18: Example scenario

Requirements

The following are required for the configuration:

- Boot image from version 7.8.4
- A bintec Media Gateway

Configuration in this scenario is carried out using the **GUI** (Graphical User Interface).

2.2 Configuration

2.2.1 Configuring the bintec R4100 media gateway

ISDN interface configuration

The ISDN interfaces ISDN-0 and ISDN-1 are used to connect the media gateway to the point-to-point ISDN access. The **ISDN Configuration Type** must be set to `Dialup (Euro ISDN) point-to-point (TE Mode)` based on the point-to-point ISDN access for the two interfaces **BRI2-0** and **BRI2-1**.

Go to the following menu to configure the ISDN interface for your device:

(1) Go to **Physical interfaces** -> **ISDN-Ports** -> **<bri2-0 (TE)>** .

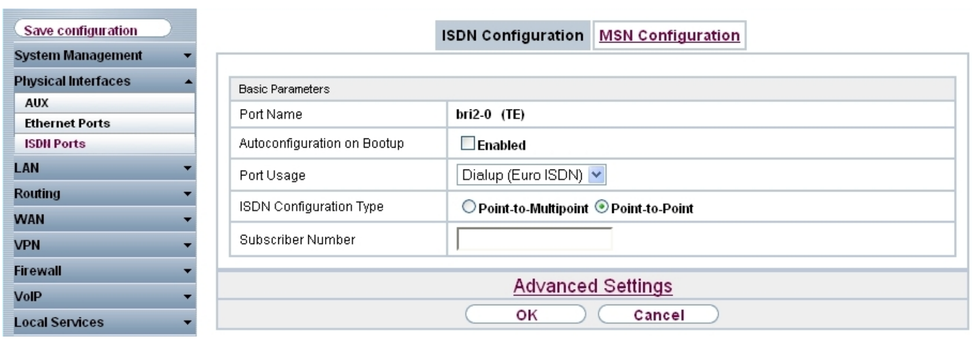

Fig. 19: **Physical Interfaces ->ISDN Ports-><bri2-0 (TE)>** .

Relevant fields in the ISDN Configuration menu

Field	Meaning
Port Name	Shows the name of the ISDN port.
Autoconfiguration on Bootup	Here, select whether the ISDN switch type should be automatically recognised.
Result of Autoconfiguration	The status of the ISDN autoconfiguration is displayed here. Automatic D-channel recognition runs until a setting is found. This field cannot be edited.
Port Usage	If the ISDN protocol is not automatically recognised, you must select the port here manually. For this, you must first disable **Automatic Configuration at Start**.

Field	Meaning
	Select *Dialup (Euro-ISDN)*.
ISDN Configuration Type	Here, select ISDN access configuration *Point-to-Point*.

ISDN Trunks

The **ISDN Trunks** menu appears only if your device has at least one ISDN point-to-point connection (BRI or PRI) and this connection is configured in NT mode.

In this example, an ISDN party line (consisting of two point-to-point ISDN accesses) is used at the exchange. Both ISDN ports must be joined together so that both ISDN connections can be used as a bundle. For this, go to the following menu:

(1) Go to **VoIP** -> **Media Gateway** -> **ISDN Trunks**-> **New**.

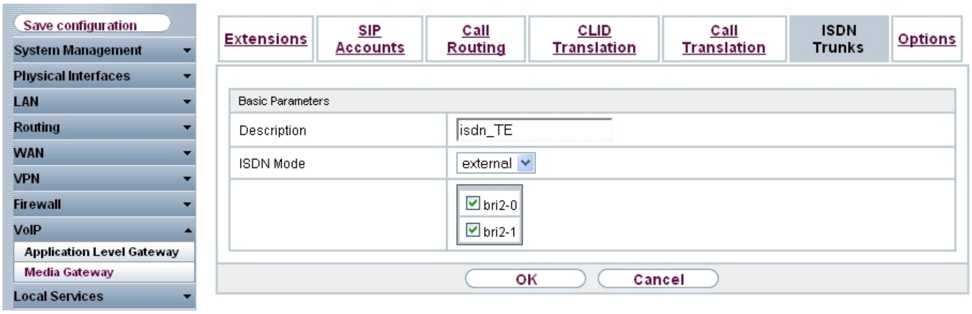

Fig. 20: **VoIP** ->**Media Gateway**-> **ISDN Trunks**-> **New**

Relevant fields in the ISDN Trunks menu

Field	Meaning
Description	Here you give the party line a name. Maximum number of characters: 20.
ISDN Mode	Displays the mode in which the party line is to be operated. *External*: Point-to-point TE connection (to connect to the point-to-point ISDN access). Enable the ISDN connections to be used for the party line.

Connecting the Asterisk to the bintec R4100

An account must be created in which the media gateway is run as a SIP Server to register the Asterisk IP PBX on **bintec R4100**.

In the **Trunk Settings** submenu you can define the settings for direct dial-in. An incoming call can be routed to just one terminal device (direct dial-in). For an outgoing call, the caller can be indicated to the called party.

Go to the following menu to configure the required accounts:

(1) Go to **VoIP** -> **Media Gateway** -> **SIP Accounts**-> **New**.

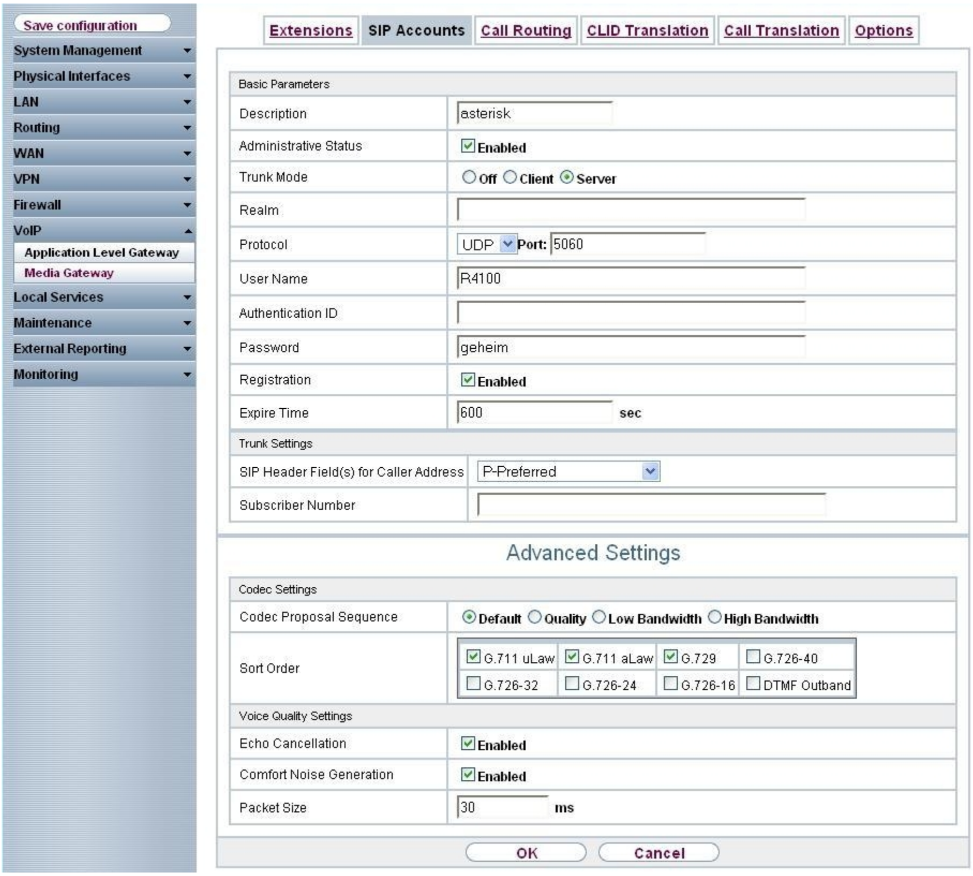

Fig. 21: **VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**

Relevant fields in the SIP Accounts menu

Field	Meaning
Description	Here, assign a name to the account. Maximum number of characters: 40.
Administrative Status	Enable or disable the administrative status of the account.

Field	Meaning
Trunk Mode	Select the trunk mode to be used. If you select `Server`, the media gateway is run as a SIP Server.
Realm	Here you can enter a further domain name of the SIP Proxy server. Enter a name only if this is explicitly specified by the provider. The field can also be used to identify the authorised user.
Protocol	Select the protocol to be used for the connection to the server or proxy.
Port	Number of the TCP or UDP port to be used for the connection to the server or proxy.
User Name	Here, enter the username for authentication if your VoIP provider has assigned one to you.
Authentication ID	Enter a name that is to be used for authentication. If you do not enter a name, the name in the **User Name** field is used.
Password	The VoIP provider gives you a PIN or password for authentication. You must enter this value here. Maximum number of characters: 40.
Registration	Enables or disables the SIP REGISTER registration mechanism.
Expire Time	Shows the time in seconds after which the current registration becomes invalid and a new registration request is therefore sent.
SIP Header Field(s) for Caller Address	This option defines where and how the DDI sender (caller) address is sent for outgoing calls. Select `P-Preferred`. The so-called "p-preferred-identity" field is added to the SIP header and contains the sender address.
Call number	Here you can set a number that is added as a prefix for outgoing calls and is removed from the sender address for incoming calls.

In the **Advanced Settings** menu, perform the settings for the SIP protocol and other specific settings.

In the **Codec Settings** submenu you can define which codecs are used for the chosen account.

The codec settings for the RTP streams can be applied without changes.

Some fields are optional and only have to be set if required for the corresponding account.

Relevant fields in the menu Advanced Settings

Field	Meaning
Codec Proposal Sequence	Determine the order in which the codecs are offered for use by the media gateway. If the first codec cannot be applied, an attempt is made to use the second codec, and so on. Set **Codec Proposal Sequence** to *default*. The codec in the first position will be used. You can sort the codecs according to quality or bandwidth.
Sort Order	Select the codecs to be proposed for the connection. The codecs chosen here are proposed in a certain order, depending on the setting in the Codec Proposal Sequence field.
Echo cancellation	Enable or disable echo cancellation. If *Enabled* is selected, echo feedback is suppressed.
Comfort Noise Generation (CNG)	Specify whether Comfort Noise Generation should be used. The slight comfort noise generation prevents subscribers from thinking that the connection is lost during pauses.
Packet Size	The transmission time of an RTP data packet in milliseconds. Possible values: 10 ... 60.

Media Gateway Configuration

You can configure the PBX functions for the media gateway in the menu **VoIP** -> **Media Gateway** -> **Options**.

Incoming and outgoing calls are managed and terminated by the **bintec R4100** using the **Session Border Controller** and **Media Stream Termination** settings shown.

(1) Go to **VoIP** -> **Media Gateway** -> **Options**.

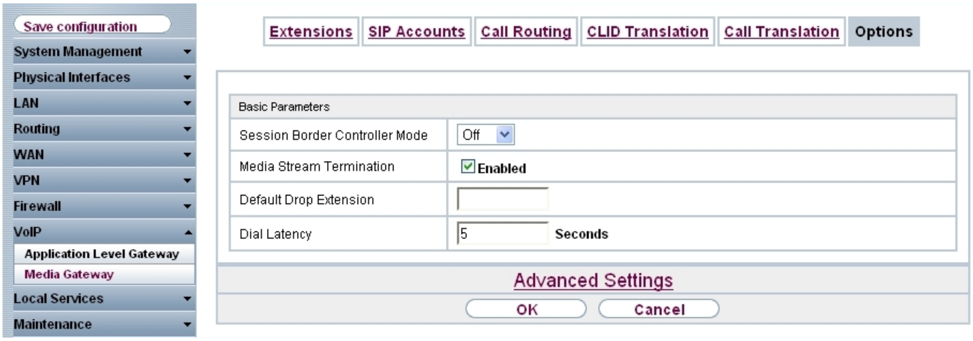

Fig. 22: **VoIP** -> **Media Gateway** -> **Options**

Relevant fields in the Options menu

Field	Meaning
Session Border Controller Mode	Determines the behaviour of the media gateway in combination with a session border controller. • *Auto*: for all extensions that exactly agree with an existing account, the call routing is handled by the session border controller, i.e. all SIP messages configured for the corresponding account are forwarded to the session border controller. For all other extensions, call routing is handled by the media gateway in accordance with the configured call routing entries. Note that the call routing is handled by the media gateway if the provider is not available (backup). • *Off*: call routing is handles exclusively by the media gateway in accordance with the configured call routing and the local extensions. For calls that are to be routed via a particular provider (account), you must configure a corresponding call routing entry. Internal calls (from internal extension to internal extension) that are only to be routed internally do not require an additional call routing entry.
Media Stream Termination	Determines how RTP sessions are controlled by the system. • *Enabled*: RTP sessions are terminated on the media gateway, i.e. all RTP streams are controlled by the media gateway and routed via the media gateway. The participating terminal devices (e.g. SIP telephones) are not connected directly with one another. Note that, for VoIP to VoIP connections, there is no code translation for different VoIP terminal codecs. The codecs of

Field	Meaning
	media gateway and VoIP terminals must therefore agree. RTP sessions are not terminated on the media gateway, i.e. all RTP streams are routed by the media gateway without termination. The RTP data packets can be routed in complex networks and thus also via other gateways. • `Disabled` (default value): RTP sessions are not terminated on the media gateway, i.e. all RTP streams are routed by the media gateway without termination. The RTP data packets can be routed in complex networks and thus also via other gateways.
Default Drop Extension	Here you can nominate an extension to receive calls that cannot be routed because there is no valid routing entry for them.
Dialling break	Maximum delay time before the system assumes the telephone number entered is complete and starts the SIP dialling process (sends the SIP INVITE message). This timeout is reset each time that a button is pressed. If you terminate the number entered with #, dialling is immediate.

Call routing / Extension translation

In the **Call Routing** menu, you can define the conditions for the routing of calls.

In this example, 10 terminals are connected to the Asterisk IP PBX. The numbers 100 to 109 are used for the extension numbers. Based on the following call routing all calls with a destination number starting with 10 are routed to the Asterisk IP PBX. This setting is necessary to allow incoming calls (from ISDN to Asterisk).

(1) Go to **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**.

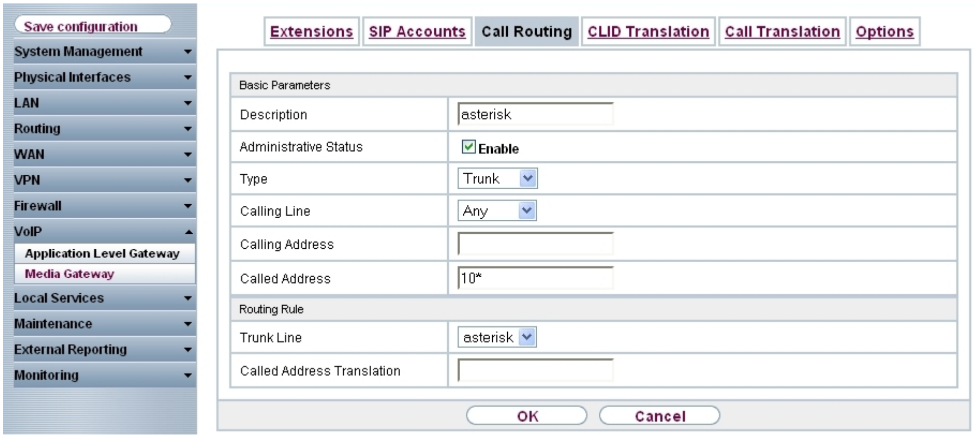

Fig. 23: **VoIP** -> **Media Gateway** -> **Call Routing**-> **New**

Relevant fields in the Call Routing menu

Field	Meaning
Description	Here, enter the name of the call routing entry.
Administrative Status	The entry is used with *enabled*.
Type	Select *Trunk* for calls that are routed to a PBX behind the media gateway.
Calling Line	Here you can restrict the routing entry to the line on which the call comes in.
Calling Address	Here you can restrict the routing entry to a particular caller. To do this, you must specify the subscriber number exactly (no wildcards).
Called Address	Here, you can enter an address (call number) that is compared with the dialled address. You can use wildcards here. For example *10** means that at the end of a character string an arbitrary number of any characters can follow.
Trunk Line	Defines the line (PRI, BRI, FXO) or the SIP account used for an incoming connection.

An additional entry is required for outgoing connections (from Asterisk to ISDN). If wildcards * are used in the **Called Address** option, all other calls are routed via the two point-to-point ISDN accesses. The following setting ensures that all outgoing calls are routed via the ISDN party line.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**.

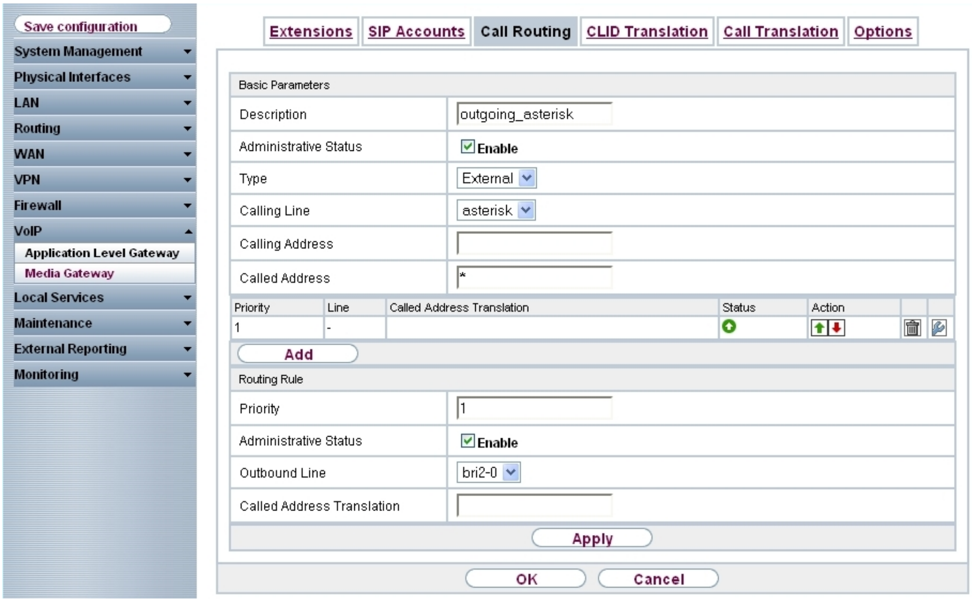

Fig. 24: **VoIP** -> **Media Gateway** -> **Call Routing**-> **New**

Relevant fields in the Call Routing menu

Field	Meaning
Description	Here you give the entry a name.
Administrative Status	The entry is used with *enabled*.
Type	Select *External* for calls that are to be routed as outgoing external calls. This can be done using standard SIP accounts or SIP trunking accounts in DDI client mode.
Calling Line	Here you can restrict the routing entry to the line on which the call comes in.
Called Address	Here you can enter an address numerically (e.g. a subscriber number) or alphanumerically that is to be compared with a dialled address. You can use wildcards. * means that at the end of a character string any number of characters may follow, If the configured address agrees with the signalled address, the routing entry is used.

You can now create a list with connections over which outgoing calls can be sent. If the line (SIP provider or ISDN line) cannot be used with Order 1, the line with the next highest order will be used to establish the connection.

Use **Add** to create entries.

Relevant fields in the Routing Rule menu

Field	Meaning
Priority	Determines the order of the filter rules, starting with `1` in increasing numerical order.
Admin Status	The entry is used with `Enable`.
Outbound Line	Defines the PSTN line (PRI, BRI, FXO) or the SIP account used for an outgoing call.

Call Translation

In the **Call Translation** menu, you can create a list for the translation of subscriber numbers, i.e. this list associates internal and external numbers.

Since the local extension numbers (numbers 100 to 109) must differ from the external numbers (0911/7660069(0-9)), the subscriber numbers have to be manipulated. In the **Call Translation** menu, with outgoing connections the last digit of the outgoing subscriber number is kept while the previous digit of the subscriber number changes, e.g.

Local number = 100 ; External number = 091176600690 or

Local number = 101 ; External number = 091176600691 and so on

(1) Go to **VoIP** -> **Media Gateway** -> **Call Translation** -> **New**.

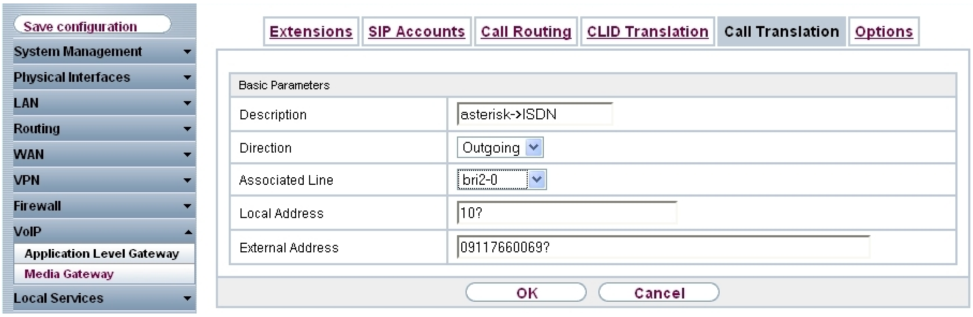

Fig. 25: **VoIP** -> **Media Gateway** -> **Call Translation** -> **New**

Relevant fields in the Call Translation menu

Field	Meaning
Description	Give the number translation a name.
Direction	Here you enter the direction to which the entry is to apply.

Field	Meaning
	Select *Outgoing* for outgoing calls.
Associated Line	Determines the line or SIP account via which the calls are to be routed.
Local Address	Here you enter the internal number (e.g. extension or PBX number). For outgoing calls, the signalled Calling Party Number (corresponds in the menu to the **Local Address** field) is translated to the **External Address**. Numerical and alphanumerical characters are permissible. ? is a placeholder for an arbitrary digit. Note **Local Address** and **External Address** must contain the same number of wildcards.
External Address	Enter the external number here. For outgoing calls, the signalled called party number (corresponding in the menu to the **Local Address** field) is translated to the **External Address**.

In our example the destination number is transmitted without a dialling code for incoming connections (from ISDN to Asterisk). Consequently, a further **Call Translation** rule is required. For example, an incoming call to the subscriber number 76600695 is changed to destination number 105. **Call Routing** is then initiated and the call is routed to the Asterisk IP PBX via the SIP trunk.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Translation** -> **New**.

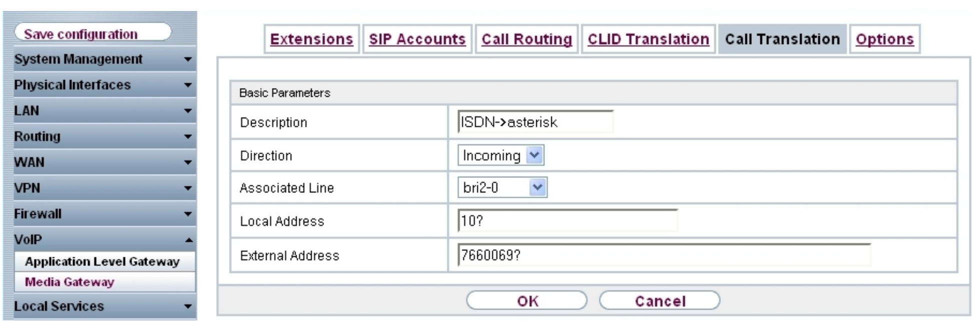

Fig. 26: **VoIP** -> **Media Gateway** -> **Call Translation** -> **New**

Relevant fields in the Call Translation menu

Field	Meaning
Description	Give the number translation a name.
Direction	Here you enter the direction to which the entry is to apply. Select *Incoming* for incoming calls.
Associated Line	Determines the line or SIP account via which the calls are to be routed.
Local Address	Here you enter the internal number (e.g. extension or PBX number). For outgoing calls, the signalled Calling Party Number (corresponds in the menu to the **Local Address** field) is translated to the **External Address**. Numerical and alphanumerical characters are permissible. ? is a placeholder for an arbitrary digit. Note **Local Address** and **External Address** must contain the same number of wildcards.
External Address	Enter the external number here. For incoming calls, the signalled Called Party Number (corresponds in the menu to the **External Address** field) is translated to the **Local Address**.

2.2.2 Configuring Asterisk IP PBX

The two configuration files `sip.conf` and `extensions.conf` are adjusted to the scenario described above. The Asterisk configuration file `sip.conf` is used to store general settings along with the registration of the Asterisk IP PBX on **bintec R4100** and the registration of the IP telephone on Asterisk IP PBX. Call routing for the IP PBX is defined in the Asterisk configuration file `extensions.conf`. Both Asterisk configuration files are represented below.

`sip.conf`

```
[general]
port=5060                ; UDP Port to bind to (SIP standard port is 5060)
bindaddr=0.0.0.0         ; IP address to bind to (0.0.0.0 binds to all)
context=default-context
maxexpirey=300           ; Max length of incoming registration we allow
defaultexpirey=60        ; Default length of incoming/outoing registration
disallow=all             ; First disallow all codecs
allow=alaw
allow=ulaw               ; Allow codecs in order of preference
allow=g729
musicclass=default       ; Sets the default music on hold class for all SIP calls
                         ; This may also be set for individual users/peers
language=en              ; Default language setting for all users/peers
                         ; This may also be set for individual users/peers
rtptimeout=60            ; Terminate call if 60 seconds of no RTP activity
                         ; when we're not on hold
rtpholdtimeout=300       ; Terminate call if 300 seconds of no RTP activity
                         ; when we're on hold (must be > rtptimeout)
useragent=Asterisk       ; Allows you to change the user agent string
nat=no                   ; NAT settings
                         ; yes = Always ignore info and assume NAT
                         ; no = Use NAT mode only according to RFC3581
                         ; never = Never attempt NAT mode or RFC3581 support
                         ; route = Assume NAT, don't send rport (work around more UNIDEN bugs)

; R4100 registration
register => R4100:asterisk@192.168.0.254/R4100 ; Register R4100

[R4100]
host=192.168.0.254
context=R4100-in
type=peer
dtmfmode=rfc2833
allow=alaw
allow=ulaw
allow=g729
insecure=very
username=R4100
fromuser=R4100
secret=asterisk
canreinvite=no
```

```
; R4100 registration
register => R4100:asterisk@192.168.0.254/R4100 ; Register R4100
[R4100]
host=192.168.0.254
context=R4100-in
type=peer
dtmfmode=rfc2833
allow=alaw
allow=ulaw
allow=g729
insecure=very
username=R4100
fromuser=R4100
secret=asterisk
canreinvite=no

; registration of IP Clients
[100]
type=friend
context=default-context
secret=pwd
host=dynamic
canreinvite=no
[101]
type=friend
context=default-context
secret=pwd
host=dynamic
canreinvite=no
[102]
type=friend
context=default-context
secret=pwd
host=dynamic
canreinvite=no
[103]
type=friend
context=default-context
secret=pwd
host=dynamic
canreinvite=no
[104]
type=friend
context=default-context
secret=pwd
host=dynamic
canreinvite=no
[105]
type=friend
context=default-context
secret=pwd
host=dynamic
canreinvite=no
[106]
type=friend
context=default-context
secret=pwd
host=dynamic
canreinvite=no
[107]
type=friend
context=default-context
secret=pwd
host=dynamic
canreinvite=no
[108]
type=friend
context=default-context
secret=pwd
host=dynamic
canreinvite=no
[109]
type=friend
context=default-context
secret=pwd
host=dynamic
canreinvite=no
```

extensions.conf

```
[general]
static=yes            ;For now only the option yes is implemented, (so setting it to no won't have any effect)
writeprotect=no       ;Then you can save dialplan from the CLI command 'save dialplan'
autofallthrough=yes   ;If this option is set, after finishing with things to do, Asterisk will hang up the call.
                       If not set, Asterisk will wait
                      ;for another extension to be dialed. It is highly recommended this option to be set to yes.
[default-context]     ;entry point for local extensions (initial context)
include => local      ;context for local calls - behind asterisk PBX
include => R4100-out  ;context for outgoing calls - Asterisk to mediagatway
include => R4100-in   ;context for incoming calls - mediagateway to Asterisk

[R4100-out]                                                  ;context for outgoing calls
exten => _0X.,1,SIPAddHeader(P-Preferred-Identity:
<tel:${CALLERID(num)}>)                                      ;SIP-Header(invite) will be enlarged by
                                                             "P-Prefered-Identity" and set to Caller-address
exten => _0X.,2,Dial(SIP/${EXTEN}@R4100,60,tr)               ;Dial command initiates a new call to the dialed
                                                             number (=exten), 60sec timeout
exten => _0X.,3,Playback(invalid)                            ;If priority 1 and 2 fails "invalid" will be played
exten => _N.,1,Hangup                                        disconnect

[R4100-in]                                                   ;context for incoming calls
exten => _R4100,1,SET(SRC_ADDRESS=${SIP_HEADER
(P-Preferred-Identity):5})                                   ;SRC_ADDRESS is filled with P-Prefered-Identity and
                                                             5colums at the beginning will be removed
exten => _R4100,n,SET(DEST_ADDRESS=${SIP_HEADER(TO)})        ;DEST_ADDRESS is filled with SIP-Header-TO field
exten => _R4100,n,SET(DEST_ADDRESS=${CUT(DEST_ADDRESS,:,2)});content of DEST_ADDRESS is removed up to the ":"
exten => _R4100,n,SET(DEST_ADDRESS=${CUT(DEST_ADDRESS,@,1)});content DEST_ADDRESS which begins with a "@" is removed
exten => _R4100,n,SET(SRC_ADDRESS=${CUT(SRC_ADDRESS,@,1)})  ;content of SRC_ADDRESS which begins with "@" will be removed
exten => _R4100,n,SET(CALLERID(num)=${SRC_ADDRESS})         ;Asterisk option CALLERID is set to SRC_ADDRESS
exten => _R4100,n,Dial(SIP/${DEST_ADDRESS},60)               ;Dial command initiates a new call with destination
                                                             = DEST_ADDRESS, 60sec timeout

[local]                                                      ;context for local calls - behind asterisk PBX
exten => _10X,1,SET(DEST_ADDRESS=${SIP_HEADER(TO)})          ;DEST_ADDRESS is set to the content of the SIP header "TO"
exten => _10X,n,SET(DEST_ADDRESS=${CUT(DEST_ADDRESS,:,2)})   ;content of DEST_ADDRESS is removed up to the ":"
exten => _10X,n,SET(DEST_ADDRESS=${CUT(DEST_ADDRESS,@,1)})   ;content of SRC_ADDRESS which begins with "@" will be removed
exten => _10X,n,Dial(SIP/${DEST_ADDRESS},60,t)               ;Dial command initiates an call to DEST_ADDRESS
exten => _10X,2,Playback(invalid)                            ;If priority 1 and 2 fails "invalid" will be played
exten => _10X,3,Hangup                                       ;disconnect
```

2.3 Overview of configuration steps

Configuring the external ISDN interface

Field	Menu	Value
Port Usage	**Physical Interfaces** -> **ISDN Ports** -> **<bri2-0 (TE)>**	`Dialup (Euro ISDN)`
ISDN Configuration Type	**Physical Interfaces** -> **ISDN Ports** -> **<bri2-0 (TE)>**	`Point-to-point`
Port Usage	**Physical Interfaces** -> **ISDN Ports** -> **<bri2-1 (TE)>**	`Dialup (Euro ISDN)`
ISDN Configuration Type	**Physical Interfaces** -> **ISDN Ports** -> **<bri2-1 (TE)>**	`Point-to-point`

Compile ISDN Trunks

Field	Menu	Value
Description	**VoIP** ->**Media Gateway**-> **ISDN Trunks**-> **New**	isdn_TE
ISDN Mode	**VoIP** ->**Media Gateway**-> **ISDN Trunks**-> **New**	*External* Enable *bri2-0* and *bri2-1*

Configuration of SIP accounts

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *asterisk*
Administrative Status	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Aktiviert*
Trunk Mode	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	Server
Protocol	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *UDP*
Port	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*5060*
User Name	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *R4100*
Password	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *secret*
Registration	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Aktiviert*
Expire Time	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*600* Sec
SIP Header Field(s) for Caller Address	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *P-Preferred*
Codec Proposal Sequence	**VoIP** -> **Media Gateway** -> **SIP Accounts** ->**New Advanced Settings**	e.g. *Standard*
Echo cancellation	**VoIP** -> **Media Gateway** -> **SIP Accounts** ->**New Advanced Settings**	*Aktiviert*
Comfort Noise Generation	**VoIP** -> **Media Gateway** -> **SIP Accounts** ->**New Advanced Settings**	*Aktiviert*

Field	Menu	Value
Packet Size	**VoIP** -> **Media Gateway** -> **SIP Accounts** ->**New Advanced Settings**	e.g. *30* ms

Call assignment for incoming calls

Field	Menu	Value
Session Border Controller Mode	**VoIP** -> **Media Gateway** -> **Options**	*Off*
Media Stream Termination	**VoIP** -> **Media Gateway** -> **Options**	*Aktiviert*
Dialling break	**VoIP** -> **Media Gateway** -> **Options**	e.g. *5* seconds

Call Routing

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *asterisk*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Aktiviert*
Type	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Trunk*
Calling Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *Any*
Called Address	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *10**
Trunk Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *asterisk*

Call routing for outgoing calls

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *outgoing_asterisk*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Enable*
Type	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*External*
Calling Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *asterisk*

Field	Menu	Value
Called Address	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. ***
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*Enable*
Outbound Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	e.g. *bri2-0*

Call Translation

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *asterisk->ISDN*
Direction	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	*Outgoing*
Associated Line	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *bri2-0*
Local Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *10?*
External Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *09117660069?*
Description	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *ISDN->asterisk*
Direction	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	*Incoming*
Associated Line	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *bri2-0*
Local Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *10?*
External Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *7660069?*

Chapter 3 Media Gateway - Configuring the connection of an ISDN PABX to a SIP trunking account with provider QSC

3.1 Introduction

The following chapters describe how to configure a **bintec R4100** as a media gateway to connect an ISDN PBX to a QSC SIP trunking account. The ISDN PBX has been set up for operation on a point-to-point ISDN access.

In our example, the main number for the point-to-point ISDN access 9673 and the direct dialling range numbers of the extensions is a two-digit number. The PABX is connected to the media gateway over an ISDN port. The second ISDN port of the media gateway is connected with an exchange-based point-to-point ISDN access and serves as an ISDN backup line.

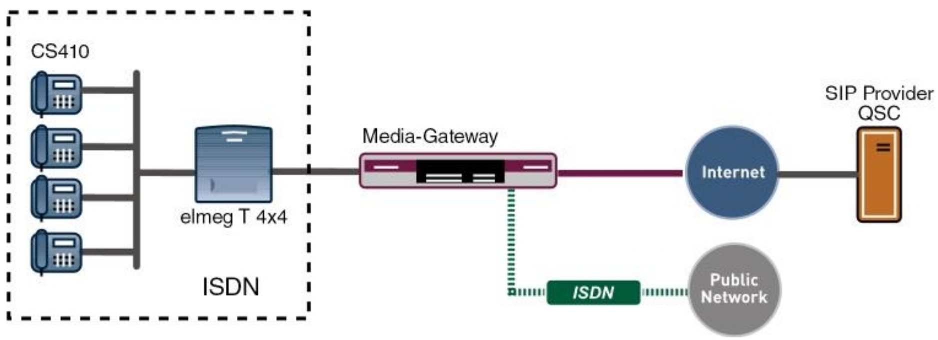

Fig. 27: Example scenario

Requirements

The following are required for the configuration:

- A bintec media gateway with system software 7.6.6.
- A DSP module (4-way) must be installed
- An Internet connection with sufficient bandwidth (recommended uplink bandwidth >= 256 kbp/s)

Configuration in this scenario is carried out using the **GUI** (Graphical User Interface).

3.2 Configuration

3.2.1 Configuring the ISDN interfaces

ISDN port ISDN-0 on **bintec R4100** is connected to the NTBA (Network Termination Basis Connection) of the ISDN backup line. The ISDN ports of the media gateway are already enabled in ISDN TE mode in the ex works state and the ISDN switch type is recognised automatically when starting the media gateway. As a result, no changes have to be made for this ISDN port.

(1) Go to **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-0 (TE)>**

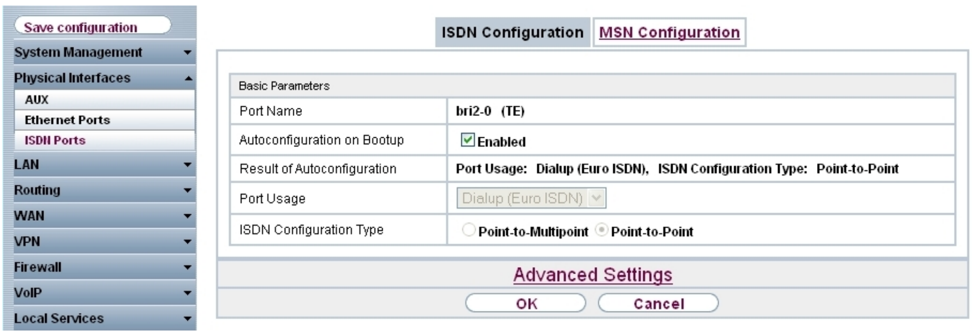

Fig. 28: **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-0 (TE)>**

ISDN mode must be changed to *NT Mode* before the ISDN PBX can connect to the media gateway at port *ISDN-1*. To do this, the housing on the media gateway must be opened. The link plugs for the ISDN-0 and ISDN-1 interfaces can be found on all devices on the main PCB behind the ISDN terminal block.

Insert the link plugs for interfaces ISDN-0 and ISDN-1 as follows:

Internal/external switching	J0M	External (factory default setting)
Internal/external switching	J1M	internal
Power supply	J0P	Off (factory default setting)
Power supply	J1P	On
Terminator	J0T	Off (factory default setting)
Terminator	J1T	On

For further information on setting the ISDN interfaces see Release Notes 7.5.1 (Chapter: 2.2 Variable switching for ISDN S0 interfaces).

Once the `ISDN-1` interface has been set by changing the link plugs to ISDN NT mode, you can configure the port for connecting the PBX. As the PBX has been configured for operation on a point-to-point ISDN access, you must set the **ISDN Configuration Type** to `Dialup (Euro ISDN) point-to-point (NT Mode)` on the media gateway.

(1) Go to **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-1 (NT)**

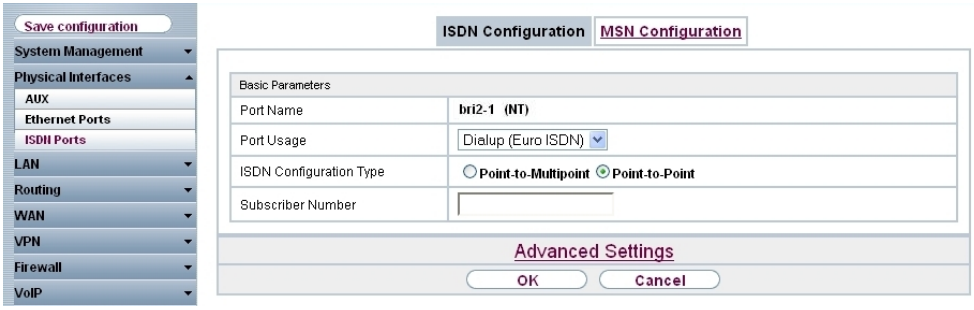

Fig. 29: **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-1 (NT)**

Relevant fields in the ISDN Configuration menu

Field	Meaning
Port Name	Shows the name of the ISDN port.
Port Usage	Select the protocol that you wish to use for the ISDN port, in this case `Dialup (Euro-ISDN)`.
ISDN Configuration Type	Here, select ISDN access configuration `Point-to-Point`.

3.2.2 Configuring the QSC SIP trunking accounts

The login data for registering the SIP trunking accounts with provider QSC are entered in the **SIP Accounts** menu.

In the **Trunk Settings** submenu, you can define the settings for direct dial-in. An incoming call can be routed to just one terminal device (direct dial-in). For an outgoing call, the caller can be indicated to the called party.

The following settings ensures that your own subscriber number is transmitted correctly with outgoing calls. With a QSC SIP trunking account, your own subscriber number (with attached direct dialling range number) is indicated in the SIP header field for caller address `Display and User Name`.

To create the account, add a new entry and configure the account as indicated below.

(1) Go to **VoIP** -> **Media Gateway** -> **SIP Accounts**-> **New**.

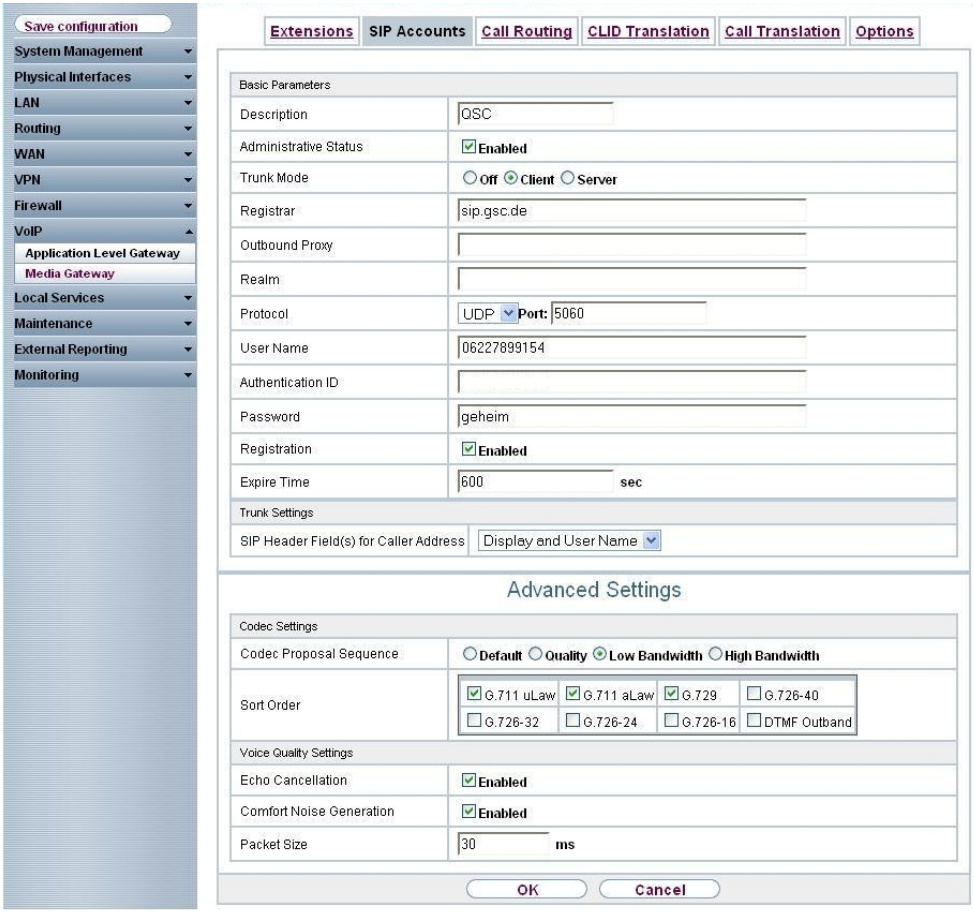

Fig. 30: **VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**

Relevant fields in the SIP Accounts menu

Field	Meaning
Description	Here, assign a name to the account. Maximum number of characters: 40.
Administrative Status	Enable the administrative status of the account.
Trunk Mode	Select the trunk mode to be used. If you select `Client`, the media gateway is run as a SIP client.
Registrar	Here, enter the IP address of the SIP registrar or of the SIP proxy server. Maximum number of characters: 40.

Field	Meaning
Protocol	Select the protocol to be used for data transport.
Port	Number of the TCP or UDP port to be used for the connection to the server or proxy.
User Name	Here, enter the username for authentication if your VoIP provider has assigned one to you.
Authentication ID	Enter a name that is to be used for authentication. If you do not enter a name, the name in the **User Name** field is used.
Password	The VoIP provider gives you a PIN or password for authentication. You must enter this value here. Maximum number of characters: 40.
Registration	Enables or disables the SIP REGISTER registration mechanism.
Expire Time	Shows the time in seconds after which the current registration becomes invalid and a new registration request is therefore sent.
SIP Header Field(s) for Caller Address	This option defines where and how the DDI sender (caller) address is sent for outgoing calls. Select `Display and User Name`. The sender address is transferred to the SIP header in the **Display**field and in the **User** field.

In the **Advanced Settings** menu, perform the settings for the SIP protocol and other specific settings.

In the **Codec Settings** submenu you can define which codecs are used for the selected account.

Note

The codecs actually used are the intersect of the codecs defined here and those signalled by the provider. For outgoing calls, any remaining codecs are dropped from the list that would require more than the available bandwidth.

Some fields are optional and only have to be set if required for the corresponding account.

Relevant fields in the menu Advanced Settings

Field	Meaning
Codec Proposal Sequence	Determine the order in which the codecs are offered for use by

Field	Meaning
	the media gateway. If the first codec cannot be used, the second is tried and so on. Select `Low Bandwidth`. As a result, the bintec media gateway gives preference to compressing codecs in order to occupy as little bandwidth as possible for the VoIP connections (RTP streams).
Sort Order	Select the codecs to be proposed for the connection.
Echo cancellation	Enable or disable echo cancellation. If `Enabled` is selected, echo feedback is suppressed.
Comfort Noise Generation (CNG)	Specify whether Comfort Noise Generation should be used. The slight comfort noise generation prevents subscribers from thinking that the connection is lost during pauses.
Packet Size	The transmission time of an RTP data packet in milliseconds. Possible values: 10 ... 60.

If registration with the VoIP provider is successful, the status in the provider menu shows . The status of the VoIP connection is changed by pressing the button or button in the **Action** column.

(1) Go to **VoIP** -> **Media Gateway** -> **SIP Accounts**.

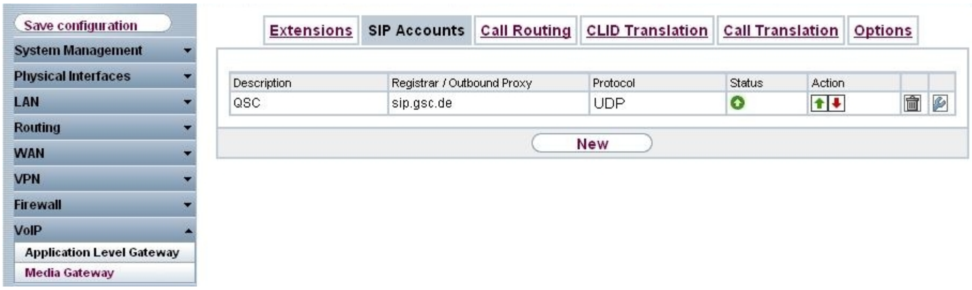

Fig. 31: **VoIP** -> **Media Gateway** -> **SIP Accounts**

3.2.3 Extension Assignment / Translation / Call Routing

In the **Call Routing** menu, the destination number determines which line is used to route incoming and outgoing calls.

Since the external numbers of the ISDN PABX differ from the extensions used for the external QSC SIP trunking account and the ISDN backup line, the extensions must first be translated.

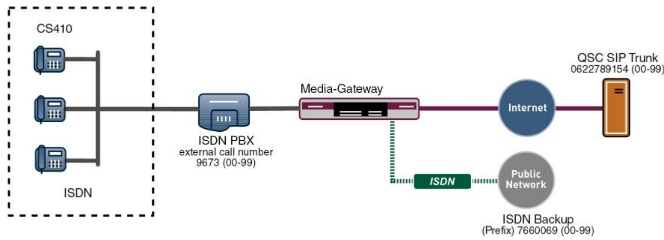

Fig. 32: Extension numbers

PBX Configuration

You can configure the PBX functions for the media gateway in the menu **VoIP** -> **Media Gateway** -> **Options**.

Incoming and outgoing calls are managed and terminated by the **bintec R4100** using the **Session Border Controller** and **Media Stream Termination** settings shown.

(1) Go to **VoIP** -> **Media Gateway** -> **Options**.

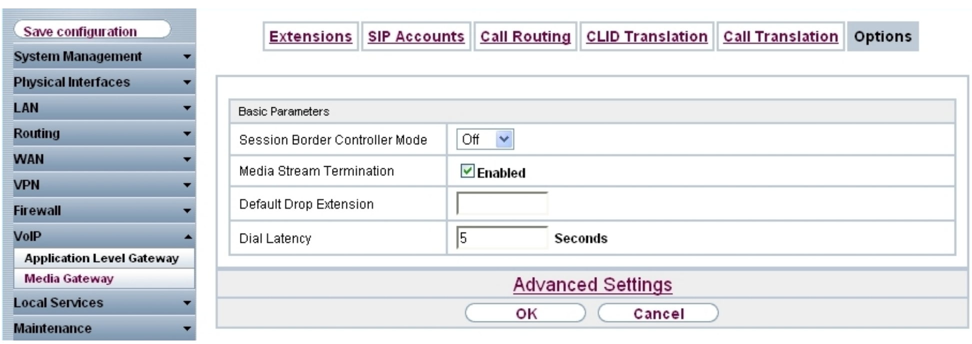

Fig. 33: **VoIP** -> **Media Gateway** -> **Options**

Relevant fields in the Options menu

Field	Meaning
Session Border Controller Mode	Determines the behaviour of the media gateway in combination with a session border controller. • `Auto`: for all extensions that exactly agree with an existing account, the call routing is handled by the session border controller, i.e. all SIP messages configured for the corresponding account are forwarded to the session border controller. For all

Field	Meaning
	other extensions, call routing is handled by the media gateway in accordance with the configured call routing entries. Note that the call routing is handled by the media gateway if the provider is not available (backup). • `Off`: call routing is handles exclusively by the media gateway in accordance with the configured call routing and the local extensions. For calls that are to be routed via a particular provider (account), you must configure a corresponding call routing entry. Internal calls (from internal extension to internal extension) that are only to be routed internally do not require an additional call routing entry.
Media Stream Termination	Determines how RTP sessions are controlled by the system. • `Enabled`: RTP sessions are terminated on the media gateway, i.e. all RTP streams are controlled by the media gateway and routed via the media gateway. The participating terminal devices (e.g. SIP telephones) are not connected directly with one another. Note that, for VoIP to VoIP connections, there is no code translation for different VoIP terminal codecs. This is why the codecs from media gateway and VoIP terminals must match; the RTP sessions are not terminated on the media gateway, i.e. all RTP streams are routed from the media gateway without termination. The RTP data packets can be routed in complex networks and thus also via other gateways. • `Disabled` (default value): RTP sessions are not terminated on the media gateway, i.e. all RTP streams are routed by the media gateway without termination. The RTP data packets can be routed in complex networks and thus also via other gateways.
Dialling break	Shows the maximum delay time before the system assumes the telephone number entered is complete and starts the SIP dialling process (sends the SIP INVITE message). This timeout is reset each time that a button is pressed. Default value: `5`.

Call Translation

The ISDN PBX uses the master subscriber number 9673 on the external connection and a two-digit extension block (00-99), which indicates the respective extension.

In this example, an incoming call via the QSC SIP trunk is signalled to the media gateway with the called party number (destination number) 06227899154 and the two-digit direct dialling range number (00-99). For the incoming call to be transferred successfully, the media gateway must change this called party number from 06227899154 [extension number] to 9673 [extension number].

With an outgoing call, the ISDN PBX signals to the media gateway the calling party number (subscriber number of the caller) 9673 with an attached extension number. The media gateway then initiates an outgoing call over the QSC SIP trunk and uses the number 06227899154 [extension] as the calling party number.

You can configure how called party numbers for incoming calls and calling party numbers for outgoing calls are translated in the **Call Translation** menu.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Translation** -> **New**.

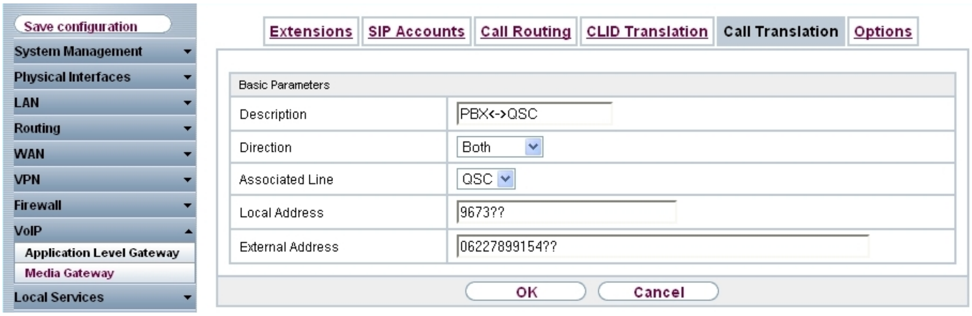

Fig. 34: **VoIP** -> **Media Gateway** -> **Call Translation** -> **New**

Relevant fields in the Call Translation menu

Field	Meaning
Description	Give the number translation a name.
Direction	Here you enter the direction to which the entry is to apply. Select `Both` for incoming and outgoing calls (bidirectional).
Associated Line	Determines the line or SIP account via which the calls are to be routed.
Local Address	Here you enter the internal number (e.g. extension or PBX number). For outgoing calls, the signalled Calling Party Number (corresponds in the menu to the **Local Address** field) is translated to the **External Address**.

Field	Meaning
	Numerical and alphanumerical characters are permissible. ? is a placeholder for an arbitrary digit. Note **Local Address** and **External Address** must contain the same number of wildcards.
External Address	Enter the external number here. For outgoing calls, the signalled called party number (corresponding in the menu to the **Local Address** field) is translated to the **External Address**.

Call translation between ISDN PBX and the ISDN backup line function according to the same principle. For example, with an incoming call over the ISDN backup line, called party number 7660069-20 is translated to called party number 9673-20 and then signalled to the ISDN PBX by the call routing. For example, with an outgoing call calling party, number 9673-20 is translated to calling party number 7660069-20 and then signalled over the ISDN backup line using call routing.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Translation** -> **New**.

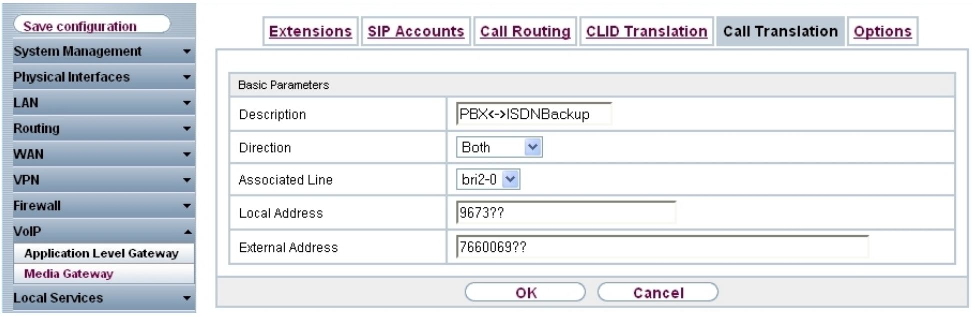

Fig. 35: **VoIP** -> **Media Gateway** -> **Call Translation** -> **New**

3.2.4 Translating the calling party number (CLID)

In the **CLID Translation** menu, you define the processing of the (calling party number) for incoming calls. You can, for example, add a prefix to a received call number in order to route corresponding outgoing calls via a particular account.

If the ISDN PBX for outgoing connections requires a specific dialling code for the trunk prefix (e.g. a leading 0), the calling party number must also be changed for incoming calls. The calling party number must be changed to enable a callback directly from the caller list of a telephone. If an incoming call is signalled from extension 091196730, for example, the calling party number of the caller must be displayed on the telephone with a leading zero (e.g. 0091196730). This change to the calling party number (for incoming calls) can be configured in the **CLID Translation** menu.

The following chapter describes how to change the calling party number for calls signalled over the QSC SIP trunking account.

(1) Go to **VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**.

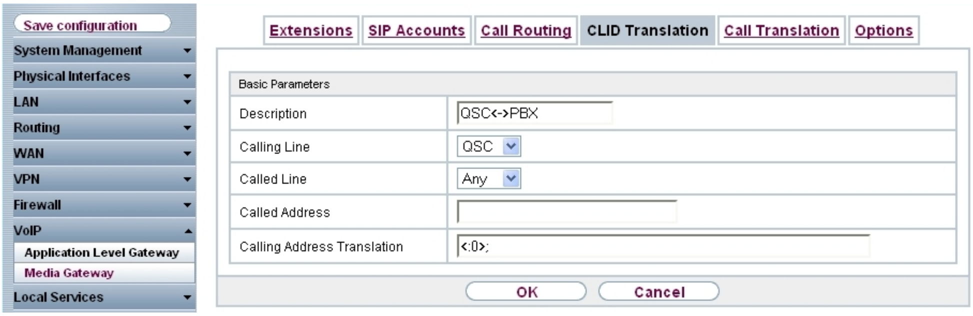

Fig. 36: **VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**

This section describes how to change the calling party number for calls signalled over the ISDN backup line.

(1) Go to **VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**.

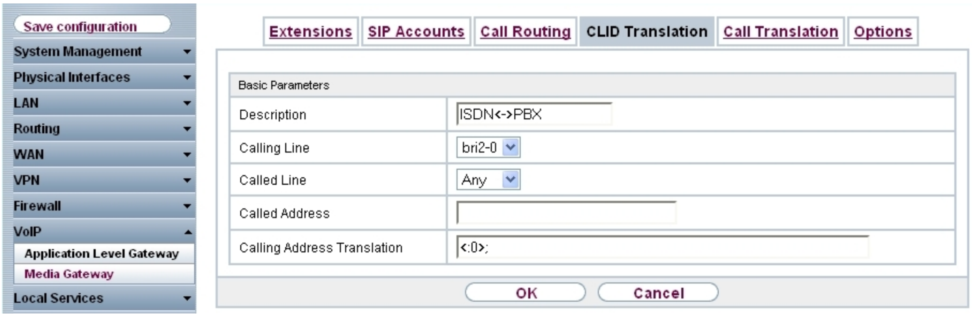

Fig. 37: **VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**

Relevant fields in the CLID Translation menu

Field	Meaning
Description	Give the entry a name.
Call number	Select the line or SIP account via which the calls are to be routed.
Called Line	Here you enter the direction to which the entry is to apply. Select `Any` for incoming and outgoing calls (bidirectional).

Field	Meaning
Calling Address Translation	Transformation rule to be used on the subscriber number The calling party number transmitted by the provider is preceded with a leading zero according to the rule mechanism.

Configuration of call routing

In the **Call Routing** menu, there occurs a definition of which SIP account or ISDN line is used when establishing a new call. Two entries are required to convert the extension numbers shown above.

Note

In principle, care must be taken when configuring call routing that the rules for call translation take priority over call routing. This means that the numbers converted after the call translation must be taken into account in the call routing menu.

With incoming calls, the called party number is changed using call translation to 9673 (main number of the ISDN PBX) with the attached extension number (e.g. 9673-20). The following call routing entry routes all calls with a destination number starting with 9673 to ISDN Port bri2-3 and therefore to the ISDN PBX.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**.

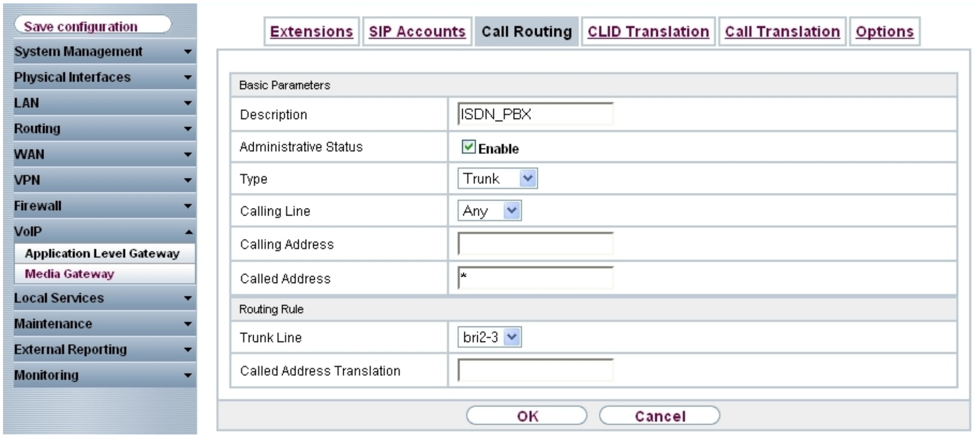

Fig. 38: **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**

Relevant fields in the Call Routing menu

Field	Meaning
Description	Here you give the entry a name.
Administrative Status	The entry is used with *Enable*.
Type	Select *Trunk* for calls that are routed to a PBX behind the media gateway.
Calling Line	Here you can restrict the routing entry to the line on which the call comes in.
Called Address	Here you can enter an address numerically (e.g. a subscriber number) or alphanumerically (e.g. for a trunk) that is to be compared with a dialled address. You can use wildcards. * means that at the end of a character string any number of characters may follow, If the configured address agrees with the signalled address, the routing entry is used.
Trunk Line	Defines the ISDN port for a call routed to the ISDN PBX.

An additional entry is required for outgoing connections. If wildcards "*" are used in the **Called Address** option, all outgoing calls are routed via the SIP accounts/ISDN lines listed in the table. In the following configuration the media gateway mainly initiates outgoing connections over the QSC SIP trunk (Order 1). If the QSC SIP trunk fails (e.g. due to a failed SIP registration) the outgoing calls are routed over the ISDN backup line (Order 2).

The following section shows the call routing entries that are required for outgoing connections.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**.

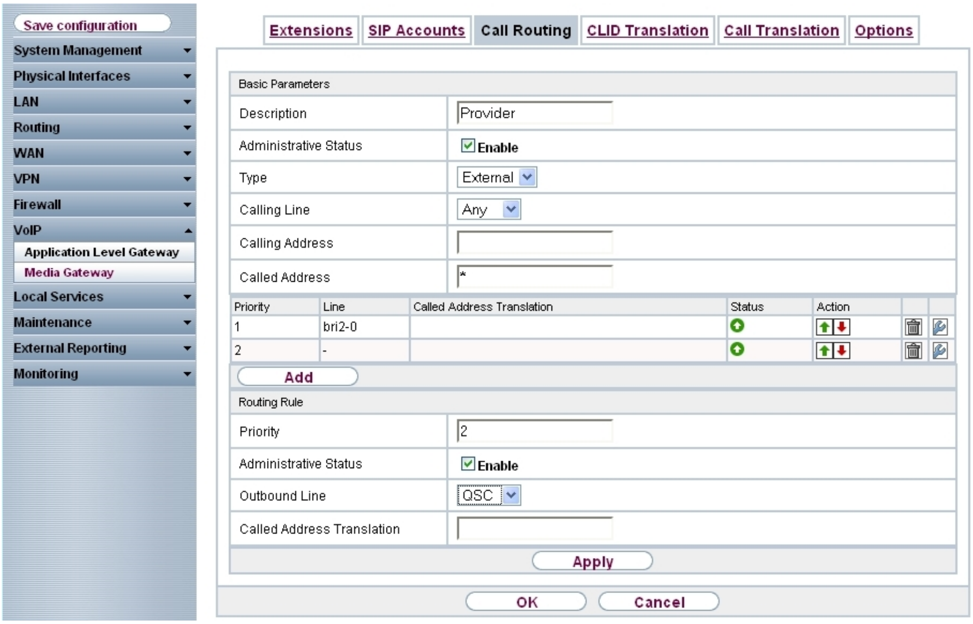

Fig. 39: **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**

Relevant fields in the Call Routing menu

Field	Meaning
Description	Here you give the entry a name.
Administrative Status	The entry is used with `Enable`.
Type	Select `External` for calls that are to be routed as outgoing external calls. This can be done using standard SIP accounts or SIP trunking accounts in DDI client mode.
Calling Line	Here you can restrict the routing entry to the line on which the call comes in.
Called Address	Here you can enter an address that is compared with the dialled address. You can use wildcards. If wildcards * are used in the **Called Address** option, all calls that cannot be handled by another call routing are routed via the SIP accounts/ISDN lines listed in the table.

You can now create a list with connections over which outgoing calls can be sent. If the line (SIP provider or ISDN line) cannot be used with Order 1, the line with the next highest order will be used to establish the connection.

Use **Add** to create entries.

Relevant fields in the Routing Rule menu

Field	Meaning
Priority	Determines the order of the filter rules, starting with *1* in increasing numerical order.
Administrative Status	The entry is used with *Enable*.
Outbound Line	Defines the PSTN line (PRI, BRI, FXO) or the SIP account used for an outgoing call.

3.2.5 Enabling the Application Level Gateway for dynamic monitoring of the NAT and firewall instance

To enable IP telephones to connect by SIP to a VoIP Provider your device has an **Application Level Gateway** (ALG), i.e. an appropriate proxy that implements the necessary NAPT and firewall releases.

In our example the media gateway is connected to the internet over an ADSL path. For security reasons **Network Address Translation** and the **Stateful Inspection Firewall** have been enabled. The **Application Level Gateway** must be enabled to prevent any negative interference by the firewall to VoIP calls (e.g. blocking the RTP stream). During a VoIP call the **Application Level Gateway** dynamically authorises access to the internet for the required SIP and RTP connections.

In the ex works state two proxy entries are predefined for the SIP **Application Level Gateway**.

(1) Go to **VoIP** -> **Application Level Gateway** ->**SIP Proxies**.

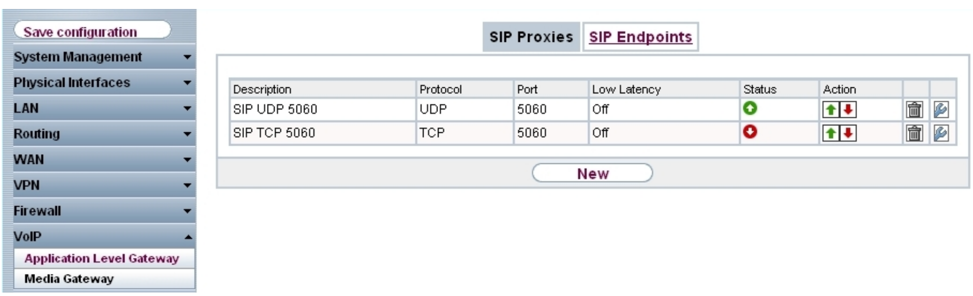

Fig. 40: **VoIP** -> **Application Level Gateway** -> **SIP Proxies**

In our example the *SIP UDP 5060* proxy entry is enabled.

(1) Go to **VoIP** -> **Application Level Gateway** -> **<SIP UDP 5060>** .

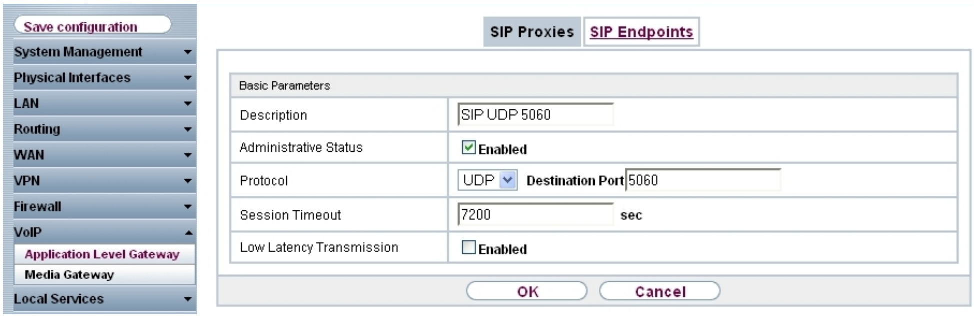

Fig. 41: **VoIP** -> **Application Level Gateway** -> **<SIP UDP 5060>**

Relevant fields in the SIP Proxy menu

Field	Meaning
Description	Name of the proxy entry.
Administrative Status	Set **Administrative Status** to `Active`.
Protocol	Defines the protocol to be used.
Destination Port	Here you enter the port to be supervised by the proxy.
Session Timeout	Shows the time in seconds for which a session stays up if no data packets are sent or received.
Low Latency Transmission	Mechanism to minimise the transit time of VoIP data packets between two subscribers. This guarantees good voice quality with high line load. Note that **Low Latency Transmission** does not have to be switched on if the media gateway supervises the VoIP connection. If `Enabled`, the voice quality is optimised, if `Disabled`, the voice quality is not optimised.

Real Time Jitter Control

For telephone calls over the Internet, VoIP packets normally have the highest priority. Nevertheless, if the upstream bandwidth is low, noticeable delays in voice transmission can occur when other packets are routed at the same time. The **Real Time Jitter Control** function in the VoIP implementation solves this problem. So as not to block the "line" for VoIP packets for too long, the size of other data packets is reduced if need be during a telephone call.

Note

When using the media gateway, **Control Mode** should always be set on `Controlled RTP only`.

(1) Go to **WAN** -> **Real Time Jitter Control** -> **Controlled Interfaces** -> **New**.

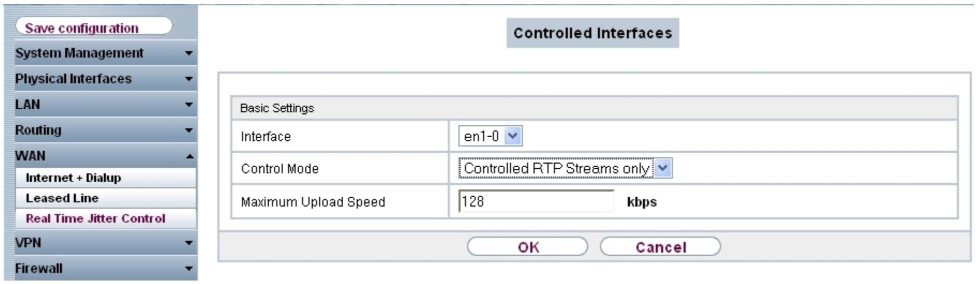

Fig. 42: **WAN** -> **Real Time Jitter Control** -> **Controlled Interfaces** -> **New**

Relevant fields in the Controlled Interfaces menu

Field	Meaning
Interface	Here you select the connection on which the voice transmission is to be optimised.
Control Mode	Select the mode for the optimisation. Select `Controlled RTP Streams only`: By means of the data routed through the media gateway, the system recognises VoIP data traffic and optimises the voice transmission. This setting should always be used together with the media gateway.
Maximum Upload Speed	If you use an external DSL modem, you must enter the bandwidth.

3.3 Overview of configuration steps

Configuring the external ISDN interface

Field	Menu	Value
Port Name	**Physical Interfaces** -> **ISDN Ports** -> **<bri2-0 (TE)>**	*bri2-0 (TE)*
Autoconfiguration on Bootup	**Physical Interfaces** -> **ISDN Ports** -> **<bri2-0 (TE)>**	*Aktiviert*
Result of Autoconfiguration	**Physical Interfaces** -> **ISDN Ports** -> **<bri2-0 (TE)>**	*Port Usage: Dialup (Euro-ISDN), ISDN Configuration Type; Point-to-point*
Port Name	**Physical interfaces** -> **ISDN Ports** -> **<bri2-1 (NT)>**	*bri2-1 (NT)*
Port Usage	**Physical interfaces** -> **ISDN Ports** -> **<bri2-1 (NT)>**	*Dialup (Euro ISDN)*
ISDN Configuration Type	**Physical interfaces** -> **ISDN Ports** -> **<bri2-1 (NT)>**	*Point-to-point*

Configuration of SIP accounts

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *QSC*
Administrative Status	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Aktiviert*
Trunk Mode	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Client*
Registrar	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *sip.qsc.de*
Protocol	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*UDP*
Port	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*5060*
User Name	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *06227899154*
Password	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *secret*

Field	Menu	Value
Registration	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Aktiviert*
Expire Time	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *600*sec
SIP Header Field(s) for Caller Address	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Display and User Name*
Codec Proposal Sequence	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**-> **Advanced Settings**	*Low Bandwidth*
Echo cancellation	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**-> **Advanced Settings**	*Aktiviert*
Comfort Noise Generation	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**-> **Advanced Settings**	*Aktiviert*
Packet Size	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**-> **Advanced Settings**	*30* ms

Call Assignment

Field	Menu	Value
Session Border Controller Mode	**VoIP** -> **Media Gateway** -> **Options**	*Off*
Media Stream Termination	**VoIP** -> **Media Gateway** -> **Options**	*Aktiviert*
Dialling break	**VoIP** -> **Media Gateway** -> **Options**	e.g. *5* seconds

Call Translation

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *PBX<->QSC*
Direction	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *Both*
Associated Line	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *QSC*
Local Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *9673??*

Field	Menu	Value
External Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. `0622789154??`
Description	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. `PBX<->ISDNBackup`
Direction	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. `Both`
Associated Line	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. `bri2-0`
Local Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. `9673??`
External Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. `7660069??`

Configuration of CLID translation

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. `QSC<->PBX`
Call number	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	`QSC`
Called Line	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	`Any`
Calling Address Translation	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. `<:0>;`
Description	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. `ISDN<->PBX`
Call number	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. `bri2-0`
Called Line	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	`Any`
Calling Address Translation	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. `<:0>;`

Configuration of call routing

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. `ISDN_PBX`
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	`Enable`

Field	Menu	Value
Type	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Trunk*
Calling Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Any*
Called Address	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. ***
Trunk Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *bri2-3*
Description	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *Provider*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Enable*
Type	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*External*
Calling Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Any*
Called Address	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. ***
Priority	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*1*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*Enable*
Outbound Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	e.g. *bri2-0*
Priority	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*2*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*Enable*
Outbound Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	e.g. *QSC*

Application Level Gateway

Field	Menu	Value
Description	**VoIP** -> **Application Level Gateway** -> **<SIP UDP 5060>**	e.g. *SIP UDP 5060*
Administrative Status	**VoIP** -> **Application Level**	*Aktiviert*

Field	Menu	Value
	Gateway -> **<SIP UDP 5060>**	
Protocol	**VoIP** -> **Application Level Gateway** -> **<SIP UDP 5060>**	*UDP*
Destination Port	**VoIP** -> **Application Level Gateway** -> **<SIP UDP 5060>**	*5060*
Session Timeout	**VoIP** -> **Application Level Gateway** -> **<SIP UDP 5060>**	*7200*
Low Latency Transmission	**VoIP** -> **Application Level Gateway** -> **<SIP UDP 5060>**	*Disabled*

Real Time Jitter Control

Field	Menu	Value
Interface	**WAN** -> **Real Time Jitter Control** -> **Controlled Interfaces** -> **New**	e.g. *en1-0*
Control Mode	**WAN** -> **Real Time Jitter Control** -> **Controlled Interfaces** -> **New**	*Controlled RTP only*
Maximum Upload Speed	**WAN** -> **Real Time Jitter Control** -> **Controlled Interfaces** -> **New**	e.g. *128* kbit/s

Chapter 4 Media Gateway - Configuring the connection of an ISDN PABX to a SIP trunking account with provider Toplink

4.1 Introduction

The following chapters describe how to configure a **bintec R4100** as a media gateway to connect an ISDN PBX to a Toplink SIP trunking account. The ISDN PBX has been set up for operation on a point-to-point ISDN access.

In our example, the main number for the point-to-point ISDN access 9673 and the direct dialling range numbers of the extensions is a two-digit number. The PABX is connected to the media gateway over an ISDN port. The second ISDN port of the media gateway is connected with an exchange-based point-to-point ISDN access and serves as an ISDN backup line.

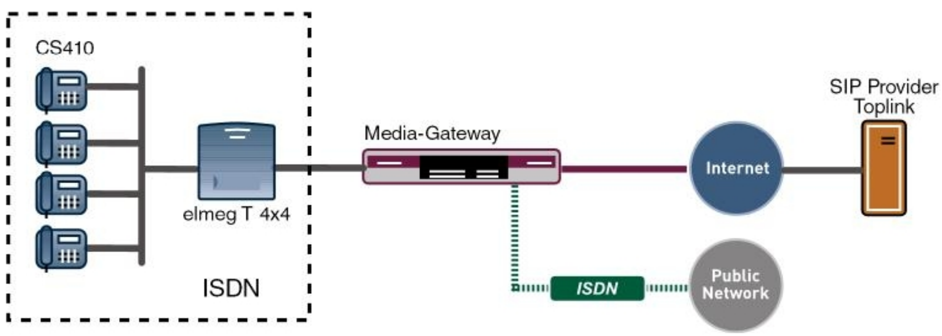

Fig. 43: Example scenario

Requirements

The following are required for the configuration:

- A bintec media gateway with system software 7.6.6.
- A DSP module (4-way) must be installed
- An Internet connection with sufficient bandwidth (recommended uplink bandwidth >= 256 kbp/s)

Configuration in this scenario is carried out using the **GUI** (Graphical User Interface) .

4.2 Configuration

4.2.1 Configuring the ISDN interfaces

ISDN port ISDN-0 on **bintec R4100** is connected to the NTBA (Network Termination Basis Connection) of the ISDN backup line. The ISDN ports of the media gateway are already enabled in ISDN TE mode in the ex works state and the ISDN switch type is recognised automatically when starting the media gateway. As a result, no changes have to be made for this ISDN port.

(1) Go to **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-0 (TE)**

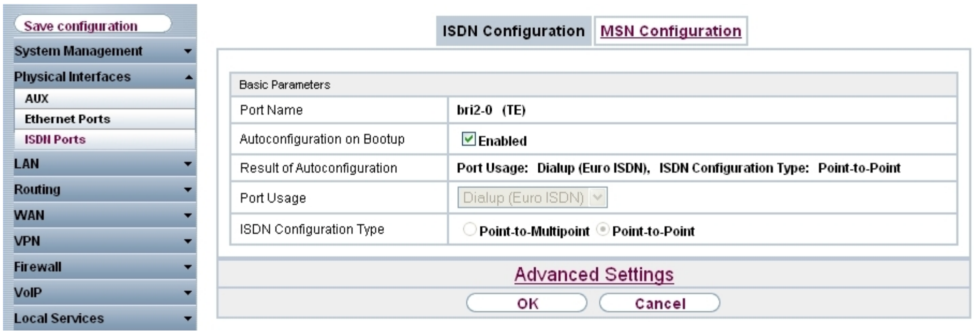

Fig. 44: **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-0 (TE)**

ISDN mode must be changed to `NT Mode` before the ISDN PBX can connect to the media gateway at port `ISDN-1`. To do this, the housing on the media gateway must be opened. The link plugs for the ISDN-0 and ISDN-1 interfaces can be found on all devices on the main PCB behind the ISDN terminal block.

Insert the link plugs for interfaces ISDN-0 and ISDN-1 as follows:

Internal/external switching	J0M	External (factory default setting)
Internal/external switching	J1M	internal
Power supply	J0P	Off (factory default setting)
Power supply	J1P	On
Terminator	J0T	Off (factory default setting)
Terminator	J1T	On

For further information on setting the ISDN interfaces see Release Notes 7.5.1 (Chapter: 2.2 Variable switching for ISDN S0 interfaces).

Once the *ISDN-1* interface has been set by changing the link plugs to ISDN NT mode, you can configure the port for connecting the PBX. As the PBX has been configured for operation on a point-to-point ISDN access, you must set the **ISDN Configuration Type** to *Dialup (Euro ISDN) point-to-point (NT Mode)* on the media gateway.

(1) Go to **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-1 (NT)>**

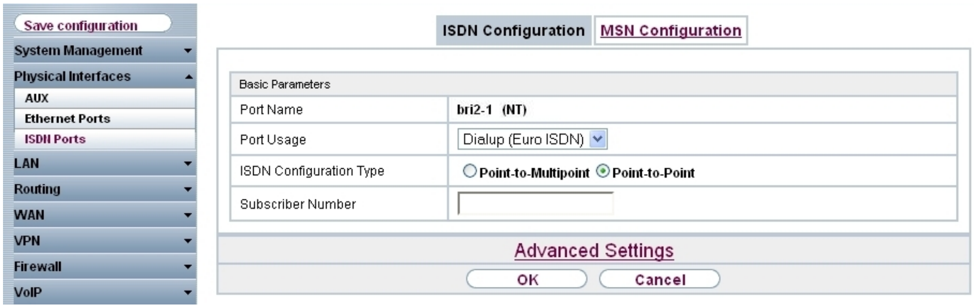

Fig. 45: **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-1 (NT)>**

Relevant fields in the ISDN Configuration menu

Field	Meaning
Port Name	Shows the name of the ISDN port.
Port Usage	Select the protocol that you want to use for the ISDN port.
ISDN Configuration Type	Here, select ISDN access configuration *Point-to-Point*.

4.2.2 Configuring the Toplink SIP trunking accounts

The login data for registering the SIP trunking accounts with provider Toplink are entered in the **SIP Accounts** menu. To create the account, add a new entry and configure the account as indicated below.

In the **Trunk Settings** submenu, you can define the settings for direct dial-in. An incoming call can be routed to just one terminal device (direct dial-in). For an outgoing call, the caller can be indicated to the called party.

The following settings ensures that your own subscriber number is transmitted correctly with outgoing calls. With a Toplink SIP trunking account, your own subscriber number (with attached direct dialling range number) is indicated in the SIP header field for caller address *P-preferred*.

(1) Go to **VoIP** -> **Media Gateway** -> **SIP Accounts**-> **New**.

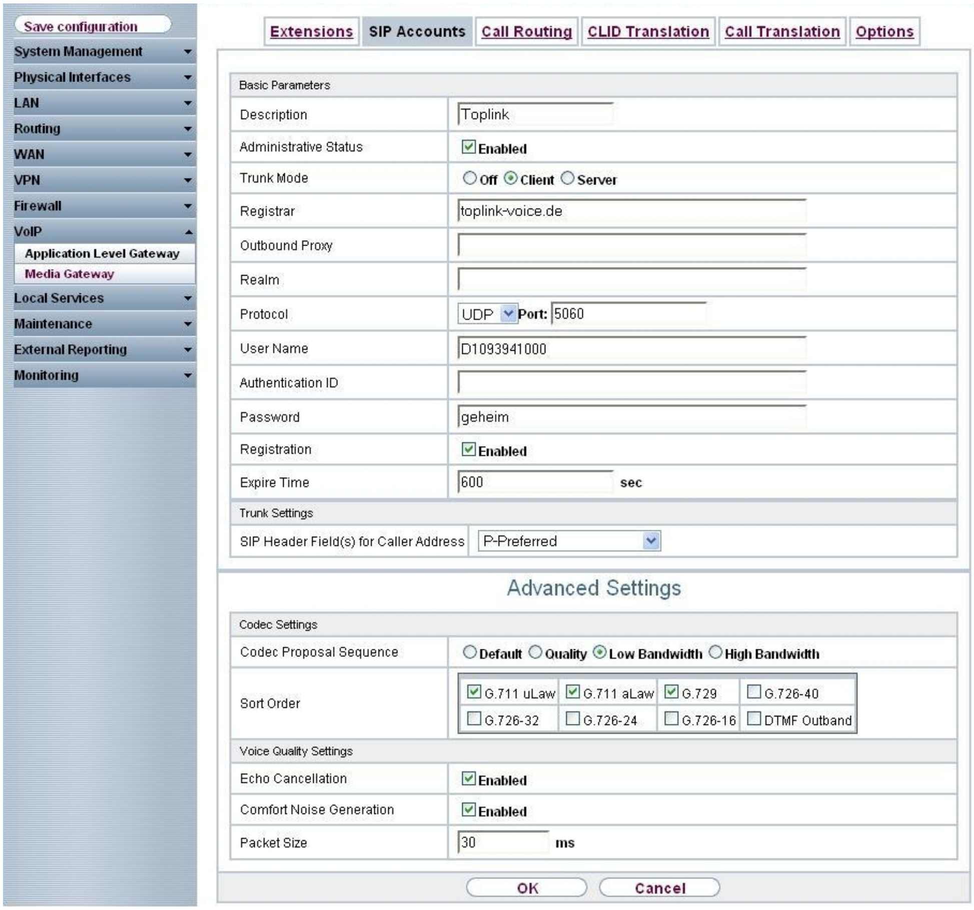

Fig. 46: **VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**

Relevant fields in the SIP Accounts menu

Field	Meaning
Description	Here, assign a name to the account. Maximum number of characters: 40.
Administrative Status	Enable the administrative status of the account.
Trunk Mode	Select the trunk mode to be used. If you select `Client`, the media gateway is run as a SIP client.
Registrar	Here, enter the IP address of the SIP registrar or of the SIP proxy server. Maximum number of characters: 40.
Protocol	Select the protocol to be used for data transport.
Port	Number of the TCP or UDP port to be used for the connection to the server or proxy.

Field	Meaning
User Name	Here, enter the username for authentication if your VoIP provider has assigned one to you.
Authentication ID	Enter a name that is to be used for authentication. If you do not enter a name, the name in the **User Name** field is used.
Password	The VoIP provider gives you a PIN or password for authentication. You must enter this value here. Maximum number of characters: 40.
Registration	Enables or disables the SIP REGISTER registration mechanism.
Expire Time	Shows the time in seconds after which the current registration becomes invalid and a new registration request is therefore sent.
SIP Header Field(s) for Caller Address	This option defines where and how the DDI sender (caller) address is sent for outgoing calls. Select *P-Preferred*. The so-called "p-preferred-identity" field is added to the SIP header and contains the sender address.

In the **Advanced Settings** menu, perform the settings for the SIP protocol and other specific settings.

In the **Codec Settings** submenu you can define which codecs are used for the selected account.

Note

The codecs actually used are the intersect of the codecs defined here and those signalled by the provider. For outgoing calls, any remaining codecs are dropped from the list that would require more than the available bandwidth.

Some fields are optional and only have to be set if required for the corresponding account.

Relevant fields in the menu Advanced Settings

Field	Meaning
Codec Proposal Sequence	Determine the order in which the codecs are offered for use by the media gateway. If the first codec cannot be used, the second is tried and so on. Select *Low Bandwidth*. As a result, the bintec media gateway gives preference to compressing codecs in order to occupy as

Field	Meaning
	little bandwidth as possible for the VoIP connections (RTP streams).
Sort Order	Select the codecs to be proposed for the connection.
Echo cancellation	Enable or disable echo cancellation. If `Enabled` is selected, echo feedback is suppressed.
Comfort Noise Generation (CNG)	Specify whether Comfort Noise Generation should be used. The slight comfort noise generation prevents subscribers from thinking that the connection is lost during pauses.
Packet Size	The transmission time of an RTP data packet in milliseconds. Possible values: 10 ... 60.

If registration with the VoIP provider is successful, the status in the provider menu shows . The status of the VoIP connection is changed by pressing the button or button in the **Action** column.

(1) Go to **VoIP** -> **Media Gateway** -> **SIP Accounts**.

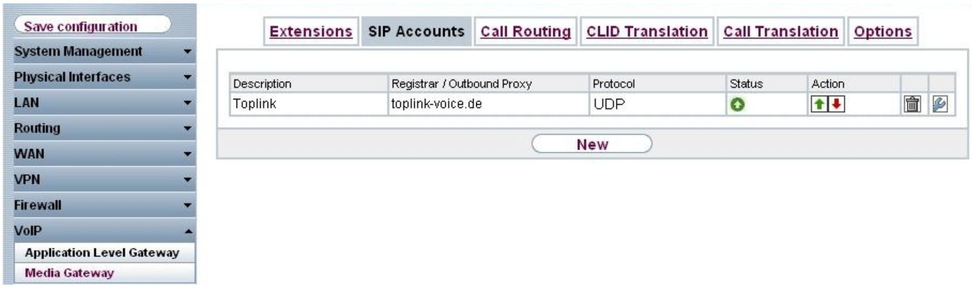

Fig. 47: **VoIP** -> **Media Gateway** -> **SIP Accounts**

4.2.3 Extension Assignment / Translation / Call Routing

In the **Call Routing** menu, the destination number determines which line is used to route incoming and outgoing calls.

Since the external numbers of the ISDN PBX differ from the extensions used for the Toplink SIP trunking account and the ISDN backup line, the extensions must first be translated.

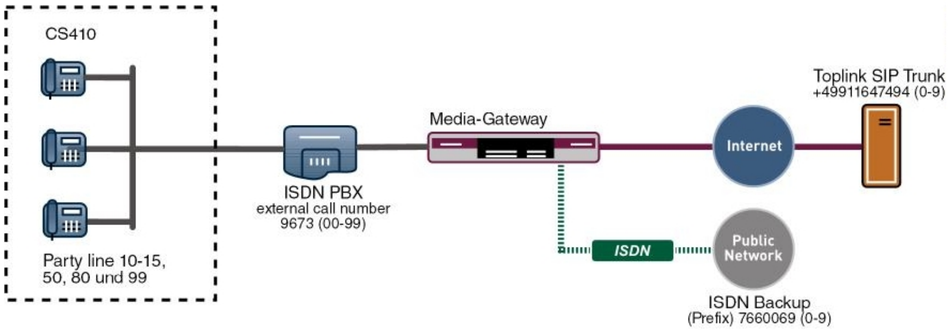

Fig. 48: Extension numbers

Call Translation

The ISDN PBX uses the master subscriber number 9673 on the external ISDN port and a two-digit extension block (00-99), which indicates the respective extension.

The Toplink SIP trunking account used in this example allows up to 10 extensions to be connected with a one-digit direct dialling range number. In this example the following extensions have been connected to the ISDN PBX: 10, 11, 12, 13, 14, 15, 50, 80, 99.

An incoming call via the Toplink SIP trunking account is signalled to the media gateway with the called party number (destination number) +49911647494 and a one-digit direct dialling range number (0-9). For the call to be transferred successfully, the media gateway must change this called party number from +49911647494 [extension number] to 9673 [extension number]. With an outgoing call, the ISDN PBX signals to the media gateway the calling party number (subscriber number of the caller) 9673 with an attached extension number (00-99). The media gateway then initiates an outgoing call over the Toplink SIP trunking account and uses the calling party number +49911647494 with the attached one-digit extension number [0-9].

In this example the ISDN PBX uses a two-digit extension number block (9673[00-99]) to refer to the respective extension for the external subscriber number. The Toplink SIP trunking account, however, only provides a one-digit extension number block (+49911647494[0-9]). Consequently, the number of the respective extension cannot be transferred exactly for incoming and outgoing calls.

The following configuration is required for the extension number translation described:

The translation of the subscriber numbers for extensions 10 to 15 can be configured with a single **Call Translation** entry. This simplified method is achieved by using a placeholder ('?'). The configuration shown in this example translates the called party number +49911647494-1 to extension number 9673-11, for example, for an incoming call. With an outgoing call initiated from extension 11, this entry allows the calling party number to be translated from 9673-11 to +49911647494-1. The call translation function always retains the last digit of the dialled subscriber number for incoming calls, and replaces the previous digits of the subscriber number.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**.

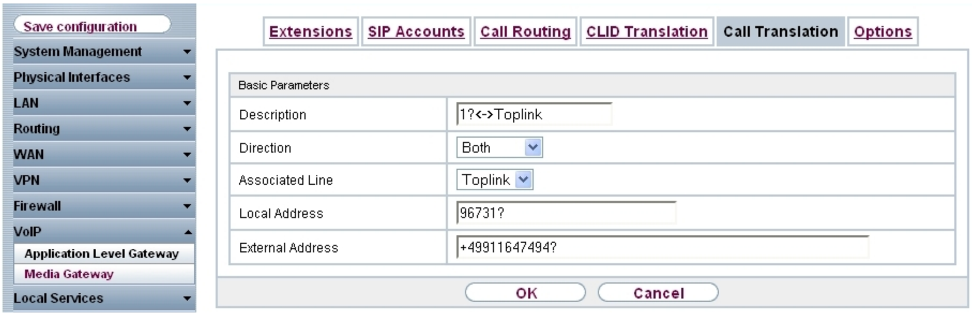

Fig. 49: **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**

Relevant fields in the Call Translation menu

Field	Meaning
Description	Give the number translation a name.
Direction	Here you enter the direction to which the entry is to apply. Select `Both` for incoming and outgoing calls (bidirectional).
Associated Line	Determines the line or SIP account via which the calls are to be routed.
Local Address	Here you enter the internal number (e.g. extension or PBX number). For outgoing calls, the signalled Calling Party Number (corresponds in the menu to the **Local Address** field) is translated to the **External Address**. Numerical and alphanumerical characters are permissible. ? is a placeholder for an arbitrary digit. Note **Local Address** and **External Address** must contain the same number of wildcards.
External Address	Enter the external number here. For outgoing calls, the signalled called party number (corresponding in the menu to the **Local Address** field) is translated to the **External Address**.

A similar entry is required to translate numbers between the ISDN PBX and the ISDN backup line. For example, with an incoming call the called party number 76600691 is translated to 967311 and then signalled to the ISDN PBX by the call translation. With an outgoing call initiated from extension 11, this call translation configuration translates the calling party number from 967311 to 76600691.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**.

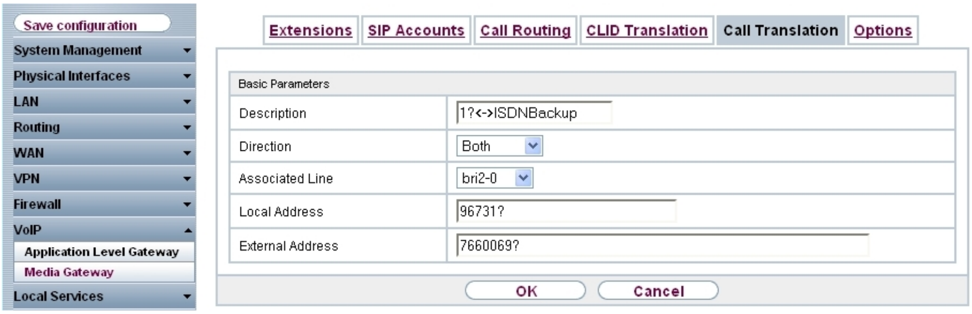

Fig. 50: **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**

Placeholders cannot be used to create the call translation entries for the other direct dialling range numbers used in this example. You must therefore create your own call translation entry for each extension.

The following chapter describes how to configure **Call Translation** for extensions 50, 80 and 99.

The following call translation configuration translates the calling party number 967350 to +499116474946 when making outgoing calls. With incoming calls, the called party number is changed accordingly.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**.

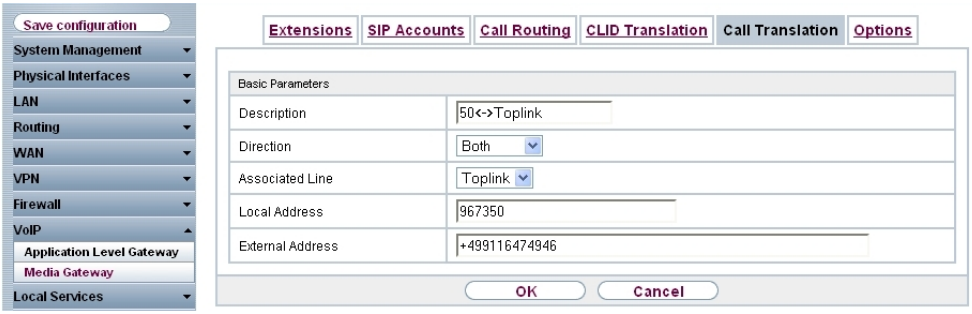

Fig. 51: **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**

The following call translation configuration translates the calling party number 967350 to

76600696 when making outgoing calls. With incoming calls, the called party number is changed accordingly.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**.

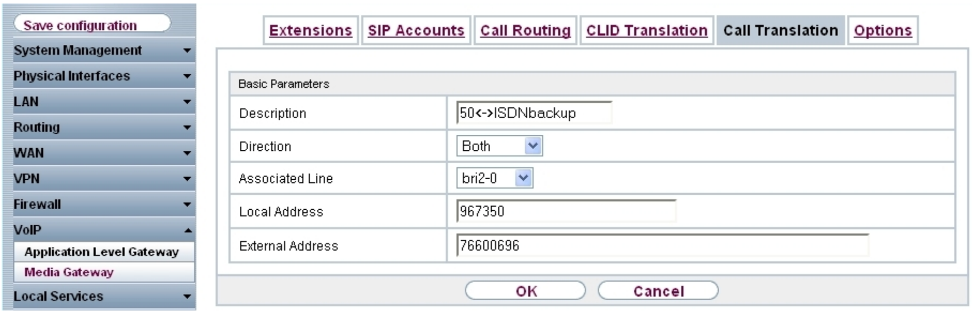

Fig. 52: **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**

The following call translation configuration translates the calling party number 967380 to +499116474947 when making outgoing calls. With incoming calls, the called party number is changed accordingly.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**.

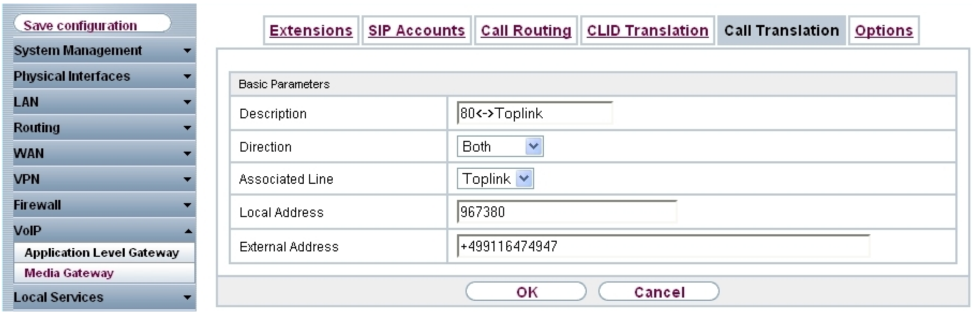

Fig. 53: **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**

The following call translation configuration translates the calling party number 967380 to 76600697 when making outgoing calls. With incoming calls, the called party number is changed accordingly.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**.

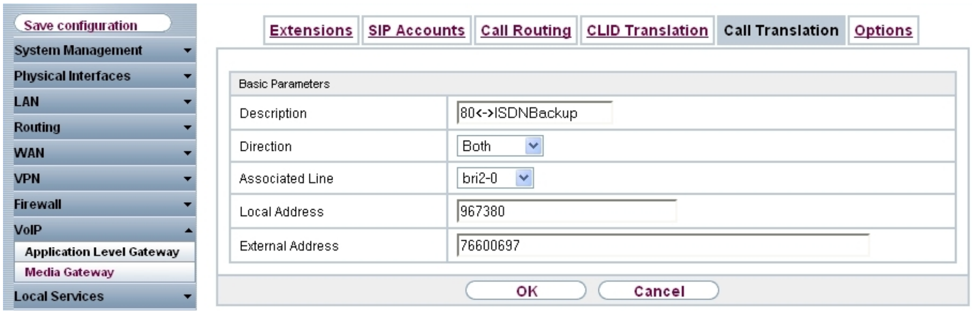

Fig. 54: **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**

The following call translation configuration translates the calling party number 967399 to +499116474948 when making outgoing calls. With incoming calls, the called party number is changed accordingly.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**.

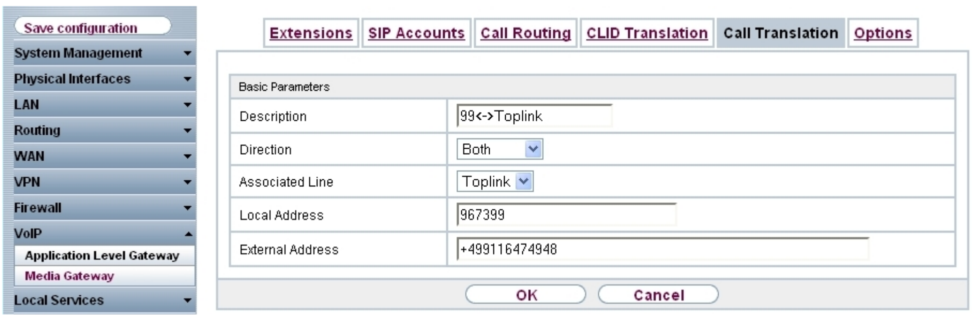

Fig. 55: **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**

The following call translation configuration translates the calling party number 967399 to 76600698 when making outgoing calls. With incoming calls, the called party number is changed accordingly.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**.

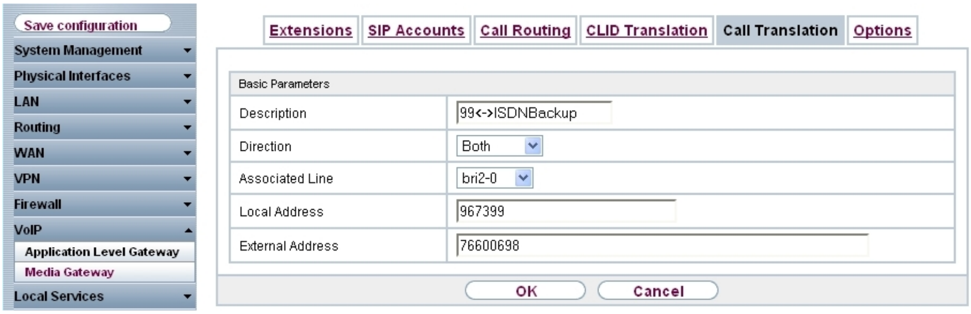

Fig. 56: **VoIP** -> **Media Gateway** -> **Call Translation** ->**New**

The complete configuration then looks like this:

(1) Go to **VoIP** -> **Media Gateway** -> **Call Translation**.

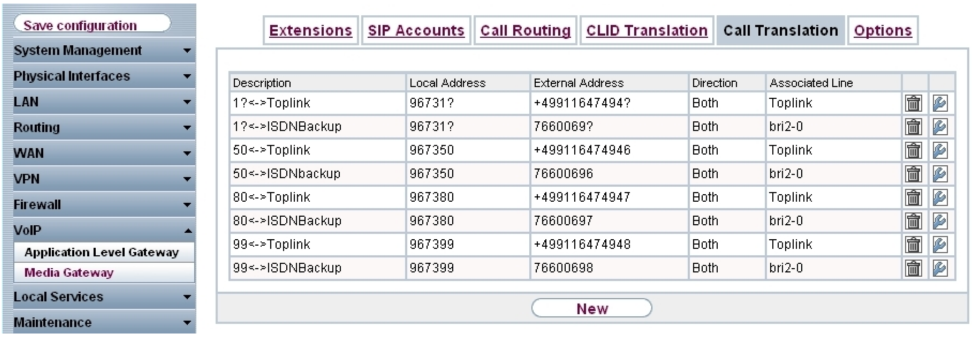

Fig. 57: **VoIP** -> **Media Gateway** -> **Call translation**

4.2.4 Translating the calling party number (CLID)

If the ISDN PBX for outgoing connections requires a specific dialling code for the trunk prefix (e.g. a leading 0), the calling party number must also be changed for incoming calls. The calling party number must be changed to enable a callback directly from the caller list of a telephone.

If an incoming call is signalled from extension 091196730, for example, the calling party number of the caller must be displayed on the telephone with a leading zero (e.g. 0091196730). This change to the calling party number (for incoming calls) can be configured in the **CLID Translation** menu.

The following chapter describes how to change the calling party number for incoming calls over the Toplink SIP trunk. The calling party number (e.g. 091196730) transmitted by the provider is preceded with a leading zero (e.g. 0091196730) according to the rule mechanism.

(1) Go to **VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**.

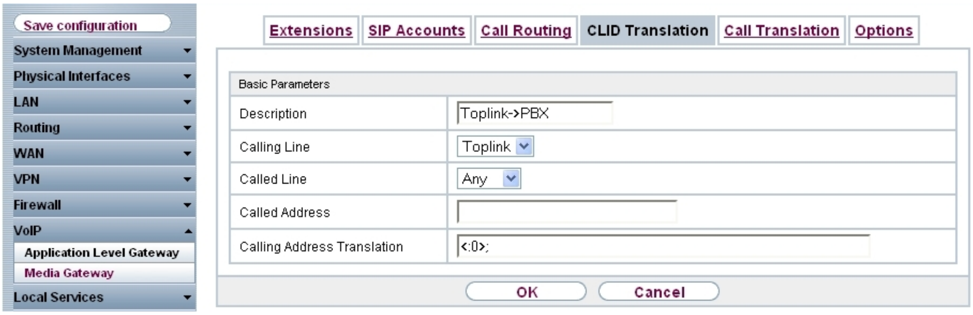

Fig. 58: **VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**

This section describes how to change the calling party number for incoming calls over the ISDN backup line.

(1) Go to **VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**.

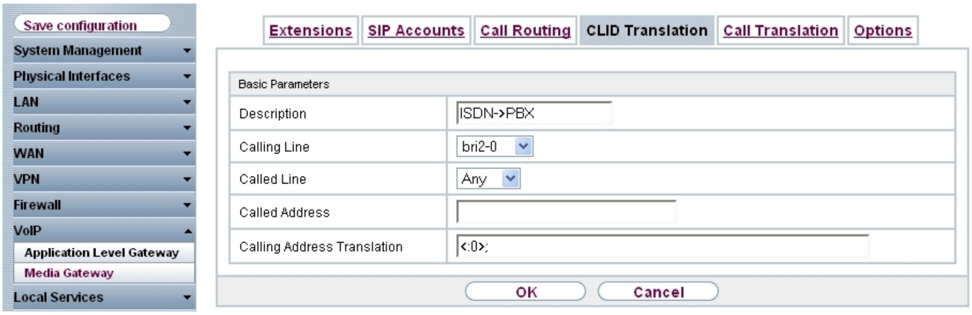

Fig. 59: **VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**

Relevant fields in the CLID Translation menu

Field	Meaning
Description	Here, enter the name of the CLID translation entry.
Call number	Select the line or SIP account via which the calls are to be routed.
Called Line	Here you enter the direction to which the entry is to apply. Select `Any` for incoming and outgoing calls (bidirectional).
Called Address	Here you have the option of entering the destination address of the call.
Calling Address Transla-tion	Enter the transformation rule applied to the call numbers.

Field	Meaning
	The calling party number transmitted by the provider is preceded with a leading zero according to the rule mechanism.

Configuration of call routing

In the **Call Routing** menu, there is a definition of which SIP account or ISDN line is used when establishing a call.

Note

In principle, care must be taken when configuring call routing that the rules for call translation take priority over call routing. This means that the numbers converted after the call translation must be taken into account in the call routing menu.

Two entries are required to convert the extension numbers shown above.

With incoming calls, the called party number is changed using the call translation mechanism to 9673 (master subscriber number of the ISDN PBX) with attached extension number (e.g. 967311). The following call routing configuration routes all calls with a destination number starting with 9673 to ISDN port bri2-1, and therefore to the ISDN PBX.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**.

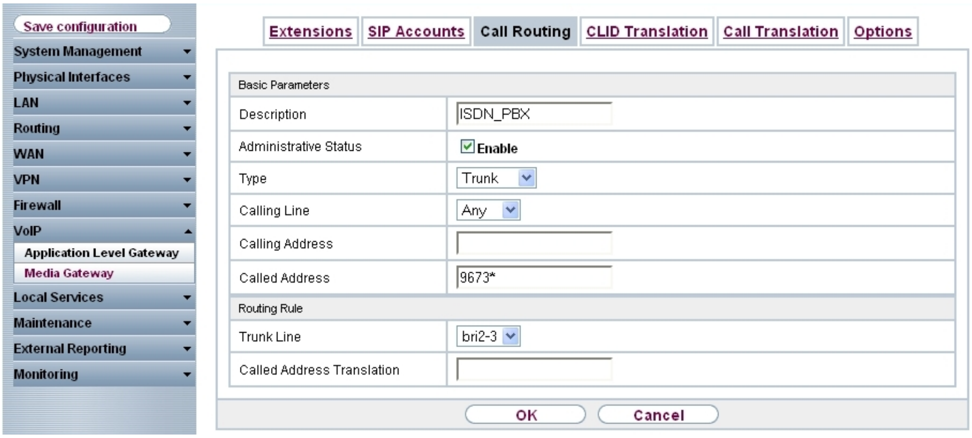

Fig. 60: **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**

Relevant fields in the Call Routing menu

Field	Meaning
Description	Here you give the entry a name.

Field	Meaning
Administrative Status	The entry is used with *Enable*.
Type	Select *Trunk* for calls that are routed to a PBX behind the media gateway.
Called Line	Here you enter the direction to which the entry is to apply. Select *Any* for incoming and outgoing calls (bidirectional).
Called Address	Here you can enter an address numerically (e.g. a subscriber number) or alphanumerically (e.g. for a trunk) that is to be compared with a dialled address. You can use wildcards. * means that at the end of a character string any number of characters may follow, If the configured address agrees with the signalled address, the routing entry is used.
Trunk Line	Defines the ISDN port for a call routed to the ISDN PBX.

An additional call routing entry is required for outgoing connections. If wildcards "*" are used in the **Called Address** option, all outgoing calls are routed via the SIP accounts/ISDN lines listed in the table. In the following configuration the media gateway mainly initiates outgoing connections over the Toplink SIP trunking account (Order 1). If the Toplink SIP trunking account fails (e.g. due to a failed SIP registration) the outgoing calls are routed over the ISDN backup line (Order 2).

The following section shows the call routing entries that are required for outgoing connections.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**.

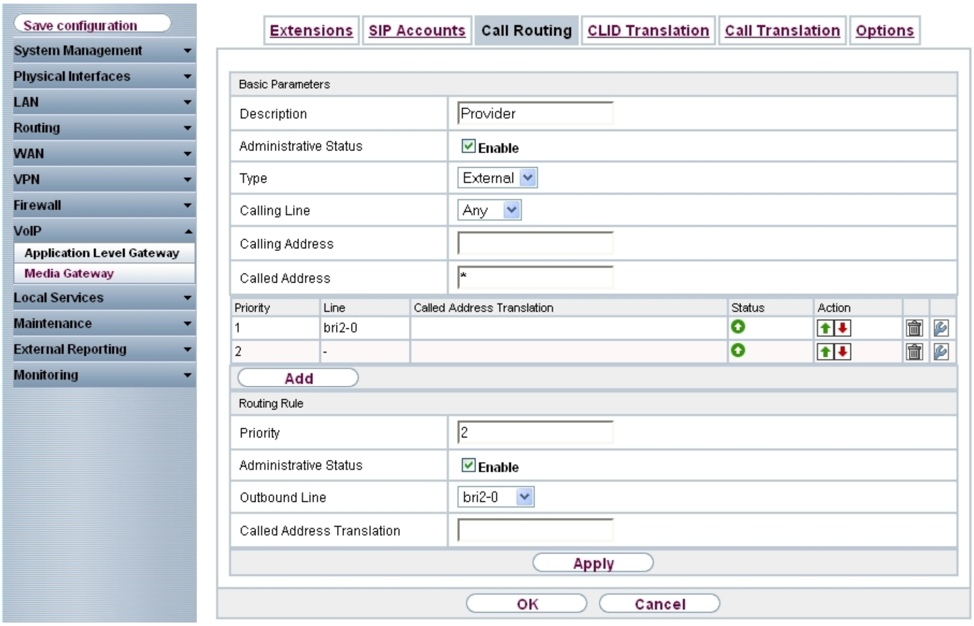

Fig. 61: **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**

Relevant fields in the Call Routing menu

Field	Meaning
Description	Here you give the entry a name.
Administrative Status	The entry is used with `Enable`.
Type	Select `External` for calls that are to be routed as outgoing external calls. This can be done using standard SIP accounts or SIP trunking accounts in DDI client mode.
Calling Line	Here you enter the direction to which the entry is to apply. Select `Any` for incoming and outgoing calls (bidirectional).
Called Address	Here you can enter an address that is compared with the dialled address. You can use wildcards. If wildcards * are used in the **Called Address** option, all calls that cannot be handled by another call routing are routed via the SIP accounts/ISDN lines listed in the table.

You can now create a list with connections over which outgoing calls can be sent. If the line (SIP provider or ISDN line) cannot be used with Order 1, the line with the next highest order will be used to establish the connection.

Use **Add** to create entries.

Relevant fields in the Routing Rule menu

Field	Meaning
Priority	Determines the order of the filter rules, starting with `1` in increasing numerical order.
Admin Status	The entry is used with `Enable`.
Outbound Line	Defines the PSTN line (PRI, BRI, FXO) or the SIP account used for an outgoing call.

4.2.5 Enabling the Application Level Gateway for dynamic monitoring of the NAT and firewall instance

To enable IP telephones to connect by SIP to a VoIP Provider your device has an **Application Level Gateway** (ALG), i.e. an appropriate proxy that implements the necessary NAPT and firewall releases.

In our example the media gateway is connected to the internet over an ADSL path. For security reasons **Network Address Translation** and the **Stateful Inspection Firewall** have been enabled. The **Application Level Gateway** must be enabled to prevent any negative interference by the firewall to VoIP calls (e.g. blocking the RTP stream). During a VoIP call the **Application Level Gateway** dynamically authorises access to the internet for the required SIP and RTP connections.

In the ex works state two proxy entries are predefined for the SIP **Application Level Gateway**.

In our example the `SIP UDP 5060` proxy entry is enabled.

(1) Go to **VoIP** -> **Application Level Gateway** -> **SIP-Proxies** -> **<SIP UDP 5060>**.

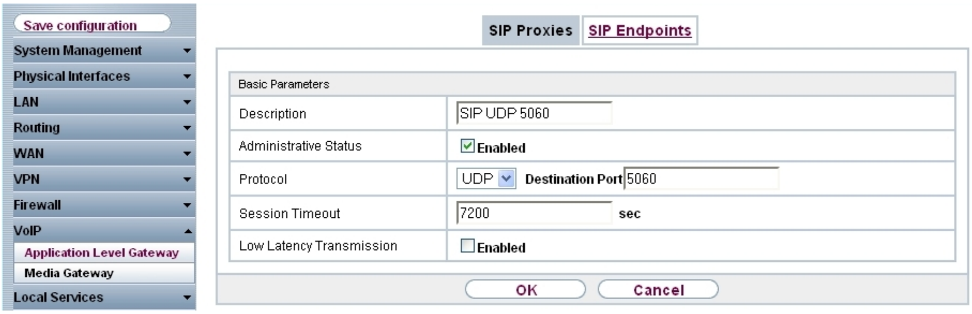

Fig. 62: **VoIP** -> **Application Level Gateway** -> **SIP-Proxies** -> **<SIP UDP 5060>**

Relevant fields in the SIP Proxies menu

Field	Meaning
Description	Name of the proxy entry.
Administrative Status	Set **Administrative Status** to `Active`.
Protocol	Defines the protocol to be used.
Destination Port	Here you enter the port to be supervised by the proxy.
Session Timeout	Shows the time in seconds for which a session stays up if no data packets are sent or received.
Low Latency Transmission	Mechanism to minimise the transit time of VoIP data packets between two subscribers. This guarantees good voice quality with high line load. Note that Low Latency Transmission does not have to be switched on if the media gateway supervises the VoIP connection. If `Enabled`, the voice quality is optimised, if `Disabled`, the voice quality is not optimised.

Real Time Jitter Control

If the internet connection of the router is used for other internet traffic or VPN connections in addition to VoIP data traffic, the QoS mechanism should be enabled. If the upload bandwidth of the Internet connection is under 1 Mbps, the **Controlled Interface** mechanism should also be enabled. The **Controlled Interface** function fragments the remaining (non-VoIP) traffic to prevent breaks in VoIP calls. If the internal ADSL modem of the router is not used, the maximum upload bandwidth must be configured manually. In this example an upload bandwidth of 512 kbps is used. Configuring the **Controlled Interface** function automatically enables the QoS mechanism.

Note

When using the media gateway, **Control Mode** should always be set on `Controlled RTP only`.

(1) Go to **WAN** -> **Real Time Jitter Control** -> **Controlled Interfaces** -> **New**.

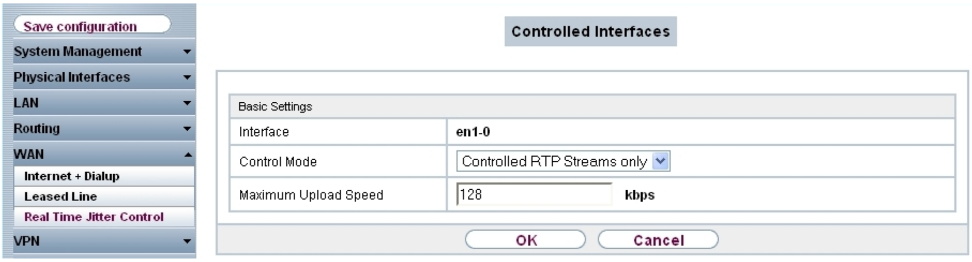

Fig. 63: **WAN** -> **Real Time Jitter Control** -> **Controlled Interfaces** -> **New**

Relevant fields in the Controlled Interfaces menu

Field	Meaning
Interface	Here you select the connection on which the voice transmission is to be optimised.
Control Mode	Select the mode for the optimisation. Select `Controlled RTP Streams only`: By means of the data routed through the media gateway, the system recognises VoIP data traffic and optimises the voice transmission. This setting should always be used together with the media gateway.
Maximum Upload Speed	Enter the maximum available upstream bandwidth in kbp/s for the selected interface.

4.3 Overview of configuration steps

Configuring the external ISDN interface

Field	Menu	Value
Port Name	**Physical Interfaces** -> **ISDN Ports** -> **<bri2-0 (TE)>**	*bri2-0 (TE)*
Autoconfiguration on Bootup	**Physical Interfaces** -> **ISDN Ports** -> **<bri2-0 (TE)>**	*Aktiviert*
Result of Autoconfiguration	**Physical Interfaces** -> **ISDN Ports** -> **<bri2-0 (TE)>**	*Port Usage: Dialup (Euro-ISDN), ISDN Configuration Type; Point-to-point*
Port Name	**Physical interfaces** -> **ISDN Ports** -> **<bri2-1 (NT)>**	*bri2-1 (NT)*
Port Usage	**Physical interfaces** -> **ISDN Ports** -> **<bri2-1 (NT)>**	*Dialup (Euro ISDN)*
ISDN Configuration Type	**Physical interfaces** -> **ISDN Ports** -> **<bri2-1 (NT)>**	*Point-to-point*

Configuration of SIP accounts

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *Toplink*
Administrative Status	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Aktiviert*
Trunk Mode	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	Client
Registrar	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *toplink-voice.de*
Protocol	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *UDP*
Port	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*5060*
User Name	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *D1093941000*
Password	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *secret*

Field	Menu	Value
Registration	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Aktiviert*
Expire Time	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*600* Sec
SIP Header Field(s) for Caller Address	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*P-Preferred*
Codec Proposal Sequence	**VoIP** -> **Media Gateway** -> **SIP Accounts** ->**New Advanced Settings**	*Low Bandwidth*
Echo cancellation	**VoIP** -> **Media Gateway** -> **SIP Accounts** ->**New Advanced Settings**	*Aktiviert*
Comfort Noise Generation	**VoIP** -> **Media Gateway** -> **SIP Accounts** ->**New Advanced Settings**	*Aktiviert*
Packet Size	**VoIP** -> **Media Gateway** -> **SIP Accounts** ->**New Advanced Settings**	*30* ms

Call Translation

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *1?<->Toplink*
Direction	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	*Both*
Associated Line	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *Toplink*
Local Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *96731?*
External Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *+49911647494?*
Description	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *1?<->ISDNBackup*
Direction	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	*Both*
Associated Line	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *bri2-0*
Local Address	**VoIP** -> **Media Gateway** ->	e.g. *96731?*

Field	Menu	Value
	Call Translation -> **New**	
External Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *7660069?*
Description	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *50<->Toplink*
Direction	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	*Both*
Associated Line	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *Toplink*
Local Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *967350*
External Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *+499116474946*
Description	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *50<->ISDNBackup*
Direction	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	*Both*
Associated Line	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *bri2-0*
Local Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *967350*
External Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *76600696*
Description	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *80<->Toplink*
Direction	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	*Both*
Associated Line	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *Toplink*
Local Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *967380*
External Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *+499116474947*
Description	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *80<->ISDNBackup*
Direction	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	*Both*

Field	Menu	Value
Associated Line	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *bri2-0*
Local Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *967380*
External Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *76600697*
Description	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *99<->Toplink*
Direction	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	*Both*
Associated Line	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *Toplink*
Local Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *967399*
External Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *+499116474948*
Description	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *99<->ISDNBackup*
Direction	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	*Both*
Associated Line	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *bri2-0*
Local Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *967399*
External Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *76600698*

Configuration of CLID translation

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *Toplink->PBX*
Call number	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	*Toplink*
Called Line	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	*Any*
Calling Address Translation	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *<:0>;*

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *ISDN->PBX*
Call number	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *bri2-0*
Called Line	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	*Any*
Calling Address Translation	**VoIP** -> **Media Gateway** -> **CLID Translation** -> **New**	e.g. *<:0>;*

Configuration of call routing

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *ISDN_PBX*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Enable*
Type	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Trunk*
Calling Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Any*
Called Address	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *9673**
Trunk Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *bri2-3*
Description	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *Provider*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Enable*
Type	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*External*
Calling Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Any*
Called Address	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. **
Priority	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*1*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*Enable*

Field	Menu	Value
Outbound Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	e.g. *Toplink*
Priority	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*2*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*Enable*
Outbound Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	e.g. *bri2-0*

Application Level Gateway

Field	Menu	Value
Description	**VoIP** -> **Application Level Gateway** -> **<SIP UDP 5060>**	e.g. *SIP UDP 5060*
Administrative Status	**VoIP** -> **Application Level Gateway** -> **<SIP UDP 5060>**	*Aktiviert*
Protocol	**VoIP** -> **Application Level Gateway** -> **<SIP UDP 5060>**	*UDP*
Destination Port	**VoIP** -> **Application Level Gateway** -> **<SIP UDP 5060>**	*5060*
Session Timeout	**VoIP** -> **Application Level Gateway** -> **<SIP UDP 5060>**	*7200*
Low Latency Transmission	**VoIP** -> **Application Level Gateway** -> **<SIP UDP 5060>**	*Disabled*

Real Time Jitter Control

Field	Menu	Value
Interface	**WAN** -> **Real Time Jitter Control** -> **Controlled Interfaces** -> **New**	e.g. *en1-0*
Control Mode	**WAN** -> **Real Time Jitter Control** -> **Controlled Interfaces** -> **New**	*Controlled RTP only*

Field	Menu	Value
Maximum Upload Speed	**WAN** -> **Real Time Jitter Control** -> **Controlled Interfaces** -> **New**	e.g. *128* kbit/s

Chapter 5 Media Gateway - Connecting an ISDN PBX to a sipgate VoIP account

This chapter describes how to configure a bintec media gateway to connect an existing ISDN PBX to a sipgate VoIP account. By using a different trunk prefix outgoing connections can be sent over the existing ISDN connection or via VoIP/sipgate. The extensions of the existing ISDN line are used as outgoing lines for both connections. Incoming connections are always accepted over ISDN.

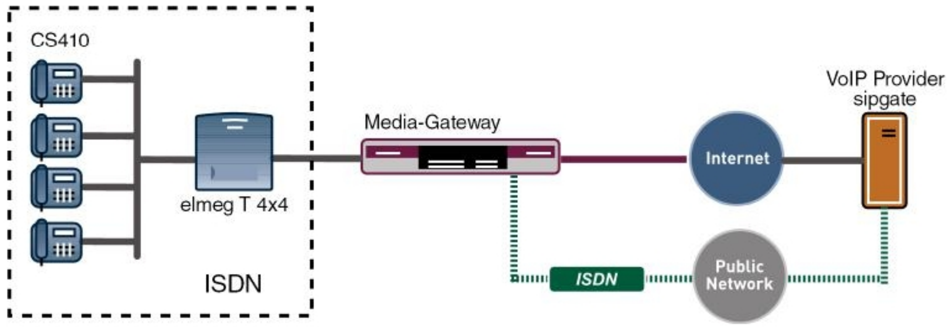

Fig. 64: Example scenario

Requirements

The following are required for the configuration:

- The ISDN PBX is configured for operation on an ISDN point-to-multipoint connection. In this example the following subscriber numbers are used: (0911)7660069-0 to (0911)7660069-9
- The bintec media gateway is connected to the internet.
- An account has been set up with VoIP provider sipgate
- A bintec media gateway (e.g. **R1200**, **R3000**, **R4100**) with at least two ISDN BRI interfaces is required.
- The optional DSP module and any ISDN BRI licences that are required must be installed

Configuration in this scenario is carried out using the **GUI** (Graphical User Interface).

5.1 Configuration

5.1.1 Configuring the sender number for the sipgate VoIP account

The settings of the sipgate account must be changed so that the correct subscriber number can be determined for outgoing calls. The telephony settings must be changed as follows:

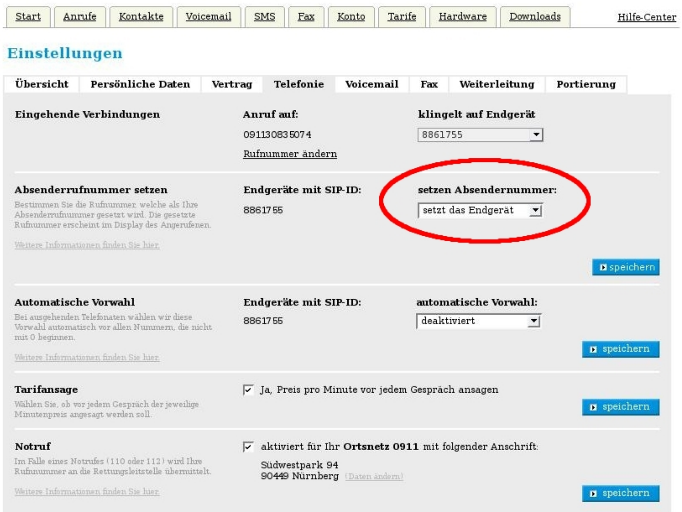

Fig. 65: Settings

5.1.2 Configuring the ISDN interfaces

The external ISDN 50 port on the PBX (point-to-multipoint connection) is connected to a BRI port on the media gateway. The ISDN mode for this BRI port must be changed to `NT Mode`. To do this, the housing on the media gateway must be opened. The link plugs for the ISDN-0 and ISDN-1 interfaces can be found on all devices on the main PCB behind the ISDN terminal block.

For further information on setting the ISDN interfaces see Release Notes 7.5.1 (Chapter: 2.2 Variable switching for ISDN S0 interfaces).

Next, the **ISDN configuration type**can be set on *`Dialup (Euro-ISDN) Point-to-Multipoint`*.

(1) Go to **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-0 (NT)**

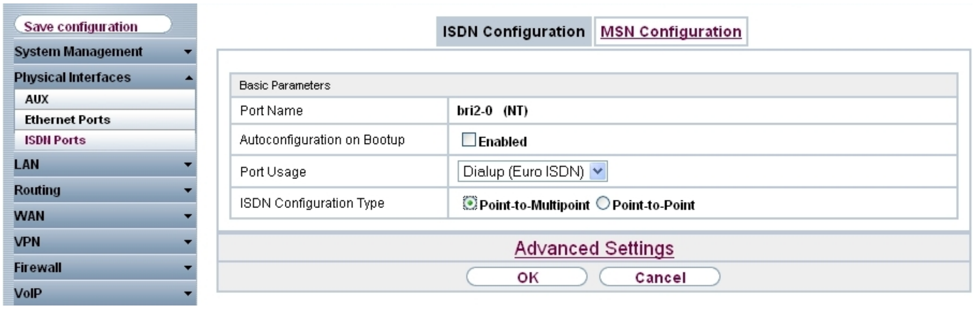

Fig. 66: **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-0 (NT)**

Relevant fields in the ISDN Configuration menu

Field	Meaning
Port Name	Shows the name of the ISDN port.
Port Usage	Select the protocol that you want to use for the ISDN port. Select `Dialup (Euro-ISDN)`.
ISDN Configuration Type	Here, select the ISDN access configuration `Point-to-Multipoint`.

An additional ISDN port on the media gateway is connected with the NTBA for the external ISDN line. The ISDN ports of the media gateway are already enabled in ISDN TE mode in the ex works state and the ISDN switch type is recognised automatically when starting the media gateway. As a result, no changes have to be made for this ISDN port.

(1) Go to **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-1 (TE)**

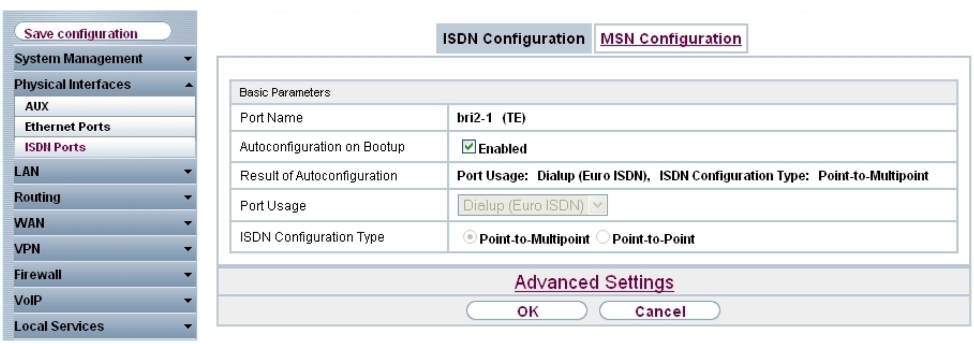

Fig. 67: **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-1 (TE)**

Relevant fields in the ISDN Configuration menu

Field	Meaning
Port Name	Shows the name of the ISDN port.
Autoconfiguration on Bootup	Here, select whether the ISDN switch type should be automatically recognised.
Result of Autoconfiguration	The status of the ISDN autoconfiguration is displayed here. Automatic D-channel recognition runs until a setting is found. This field cannot be edited.
Port Usage	If the ISDN protocol is not automatically recognised, you must select the port here manually. For this, you must first disable **Automatic Configuration at Start**. Select *Dialup (Euro-ISDN)*.
ISDN Configuration Type	Here, select the ISDN access configuration *Point-to-Multipoint*.

5.1.3 Configuring the sipgate VoIP account

The login data for registering the SIP accounts with provider sipgate are entered in the **SIP Accounts** menu.

Additional settings are required in the **Trunk Settings** submenu to configure a SIP trunking account. With outgoing calls sipgate allows a modified calling party number (caller number) to be transmitted. With outgoing calls sent via the sipgate account, the calling party number of the previously used ISDN point-to-multipoint connection is indicated. With the setting *Display Only* your own subscriber number is indicated in the SIP header field of the SIP INVITE message.

To create the account, add a new entry and configure the account as indicated below.

(1) Go to **VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**.

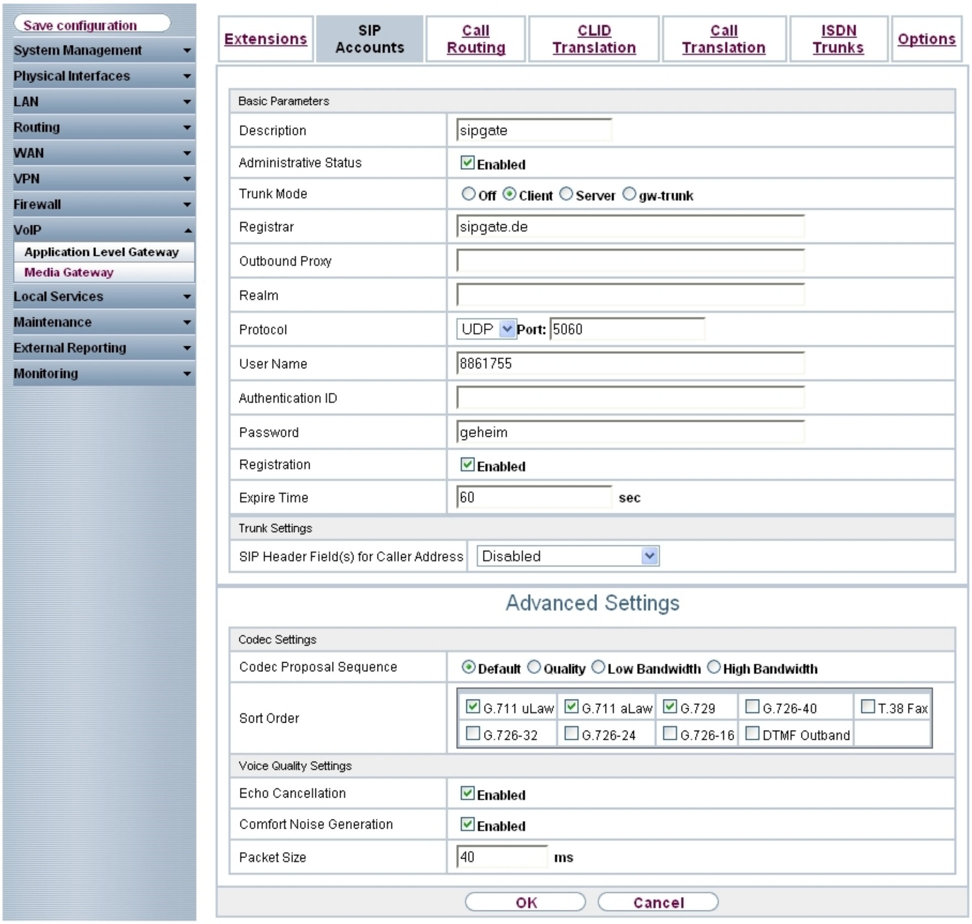

Fig. 68: **VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**

Relevant fields in the SIP Accounts menu

Field	Meaning
Description	Here, assign a name to the account. Maximum number of characters: 40.
Administrative Status	Enable the administrative status of the account.
Trunk Mode	Select the trunk mode to be used. If you select `Client`, the media gateway is run as a SIP client.
Registrar	Enter the IP address of the remote SIP terminal (client or server) here. Maximum number of characters: 40.
Protocol	Select the protocol to be used for the connection to the server or proxy.
Port	Number of the TCP or UDP port to be used for the connection

Field	Meaning
	to the server or proxy.
User Name	Here, enter the username for authentication if your VoIP provider has assigned one to you.
Authentication ID	Enter a name that is to be used for authentication. If you do not enter a name, the name in the **User Name** field is used.
Password	The VoIP provider gives you a PIN or password for authentication. You must enter this value here. Maximum number of characters: 40.
Registration	Enables or disables the SIP REGISTER registration mechanism.
Expire Time	Shows the time in seconds after which the current registration becomes invalid and a new registration request is therefore sent.
SIP Header Field(s) for Caller Address	This option defines where and how the DDI sender (caller) address is sent for outgoing calls. Select *Display Only*. the sender address is placed in the **Display** field of the SIP header.

In the **Advanced Settings** menu, perform the settings for the SIP protocol and other specific settings. In the **Codec Settings** submenu you can define which codecs are used for the chosen account. The settings can be applied without changes.

Some fields are optional and only have to be set if required for the corresponding account.

Relevant fields in the menu Advanced Settings

Field	Meaning
Codec Proposal Sequence	Determine the order in which the codecs are offered for use by the media gateway. If the first codec cannot be applied, an attempt is made to use the second codec, and so on. Set **Codec Proposal Sequence** to *default*. The codec in the first position will be used. You can sort the codecs according to quality or bandwidth.
Sort Order	Select the codecs to be proposed for the connection. The codecs chosen here are proposed in a certain order, depending on the setting in the Codec Proposal Sequence field.
Echo cancellation	Enable or disable echo cancellation. If *Enabled* is selected, echo feedback is suppressed.

Field	Meaning
Comfort Noise Generation	Specify whether Comfort Noise Generation should be used. The slight comfort noise generation prevents subscribers from thinking that the connection is lost during pauses.
Packet Size	The transmission time of an RTP data packet in milliseconds. Possible values: 10 ... 60.

If registration with the VoIP provider is successful, the status in the provider menu shows . The status of the VoIP connection is changed by pressing the button or button in the **Action** column.

(1) Go to **VoIP** -> **Media Gateway** -> **SIP Accounts**.

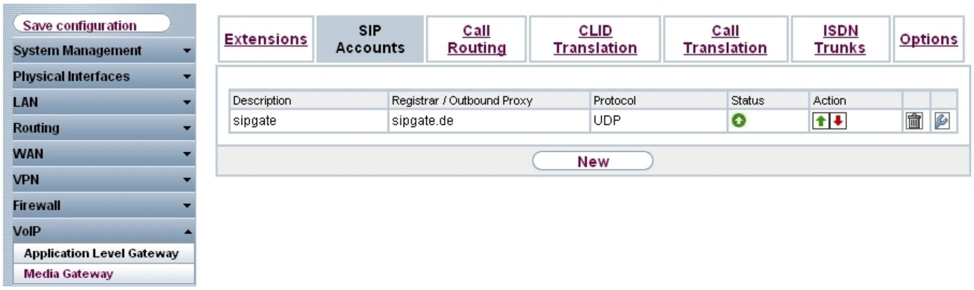

Fig. 69: **VoIP** -> **Media Gateway** -> **SIP Accounts**

5.1.4 Configuring the internal extensions

In this example the internal ISDN port to connect to the ISDN PBX is operated as an ISDN point-to-multipoint connection. It is therefore necessary to store the external MSN numbers for the PBX. If more than one ISDN port is used in ISDN NT Mode (point-to-multipoint) on the media gateway, the respective ISDN port can be selected.

(1) Go to **VoIP** -> **Media Gateway** -> **Extensions** -> **New**.

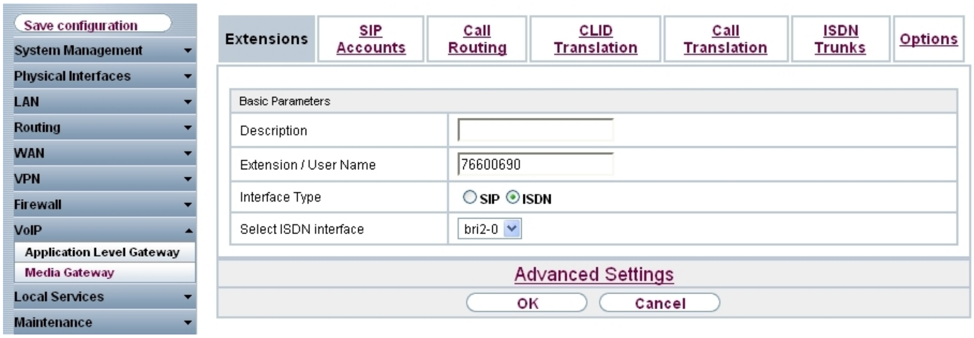

Fig. 70: **VoIP** -> **Media Gateway** -> **Subscriber** -> **New**

Relevant fields in the Extensions menu

Field	Meaning
Extension / User Name	Enter the subscriber number here. Maximum number of characters: 40.
Interface Type	Terminal type, an internal PBX is used for the call. The `ISDN` setting can only be selected if ISDN interfaces with the **ISDN Configuration Type** = `Dialup (Euro ISDN) point to multipoint (TE Mode)` is set.
Select ISDN interface	Select an interface. The ISDN interface you can select depends on the device used.

In our example, the 10 external MSN subscriber numbers for the PBX are connected to the ISDN port `bri2-0` on the media gateway. The complete configuration looks like this:

(1) Go to **VoIP** -> **Media Gateway** -> **Extension**.

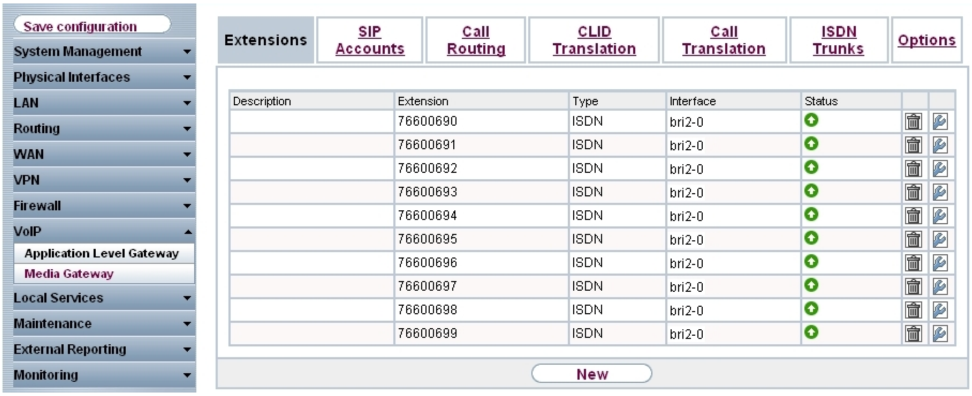

Fig. 71: **VoIP** -> **Media Gateway** -> **Extension**

5.1.5 Extension assignment - Call routing - Call translation

An MSN subscriber number of the PBX should be stored as the **Default Extension** in the **PBX Configuration** menu. Here you can nominate an extension to receive calls that cannot be routed because there is no valid routing entry for them.

(1) Go to **VoIP** -> **Media Gateway** -> **Options**.

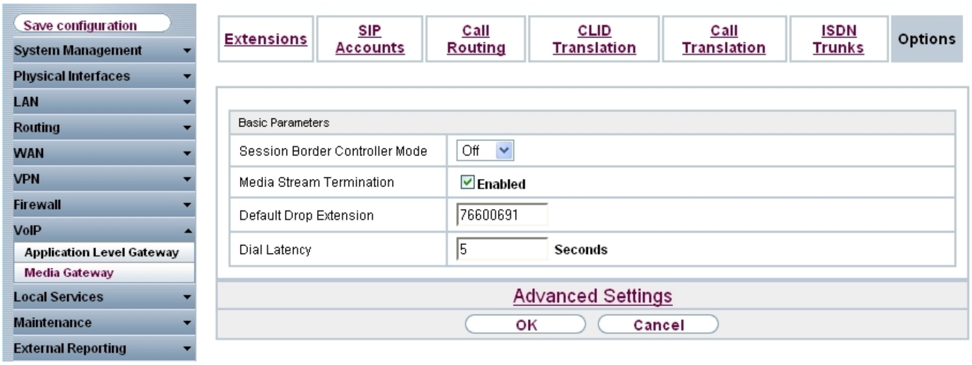

Fig. 72: **VoIP** -> **Media Gateway** -> **Options**

Relevant fields in the Options menu

Field	Meaning
Session Border Controller Mode	Determines the behaviour of the media gateway in combination with a session border controller. Select `off`: call routing is handles exclusively by the media gateway in accordance with the configured call routing and the local extensions. For calls that are to be routed via a particular provider (account), you must configure a corresponding call routing entry. Internal calls (from internal extension to internal extension) that are only to be routed internally do not require an additional call routing entry.
Media Stream Termination	Determines how RTP sessions are controlled by the system. Select `on`: RTP sessions are terminated on the media gateway, i.e. all RTP streams are controlled by the media gateway and routed via the media gateway. The participating terminal devices (e.g. SIP telephones) are not connected directly with one another. Note that, for VoIP to VoIP connections, there is no code trans-

Field	Meaning
	lation for different VoIP terminal codecs. This is why the codecs from media gateway and VoIP terminals must match; the RTP sessions are not terminated on the media gateway, i.e. all RTP streams are routed from the media gateway without termination. The RTP data packets can be routed in complex networks and thus also via other gateways.
Default Drop Extension	Here you can nominate an extension to receive calls that cannot be routed because there is no valid routing entry for them.
Dialling break	Maximum delay time before the system assumes the telephone number entered is complete and starts the SIP dialling process (sends the SIP INVITE message). This timeout is reset each time that a button is pressed. If you terminate the number entered with #, dialling is immediate.

Call Routing

The **Call Routing** menu determines whether outgoing connections are routed over the ISDN line or over the sipgate VoIP account. Here you can decide over which line the outgoing call is initiated for each called or calling party number (with a special number as a trunk prefix).

Our example shows the call routing entry through which all outgoing calls with an international destination number (e.g. 0043, 0033) are initiated over the sipgate VoIP account. The **Calling Line** option indicates the ISDN port of the media gateway that is connected to the ISDN PBX.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**.

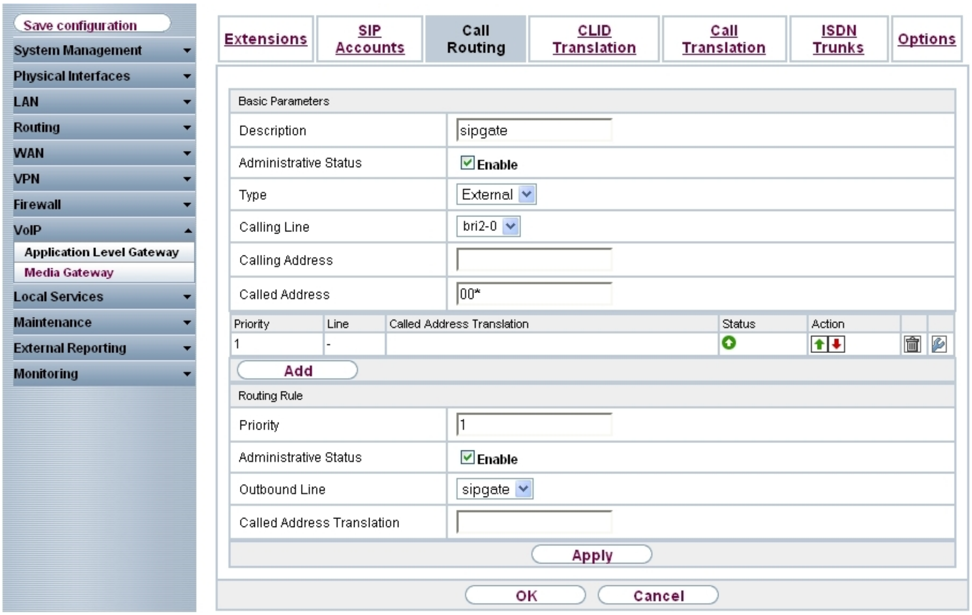

Fig. 73: **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**

Relevant fields in the Call Routing menu

Field	Meaning
Description	Here, enter the name of the call routing entry.
Administrative Status	The entry is used with *enabled*.
Type	Select *External* for calls that are to be routed as outgoing, external calls.
Calling Line	Here you can restrict the routing entry to the line on which the call comes in.
Calling Address	Here you can restrict the routing entry to a particular caller. To do this, you must specify the subscriber number exactly (no wildcards).
Called Address	Here, you can enter an address (call number) that is compared with the dialled address. You can use wildcards here. For example *00** means that at the end of a character string an arbitrary number of any characters can follow.

You can now select the ISDN line or SIP provider account to be used for this call entry (for outgoing connections).

Use **Add** to create entries.

Relevant fields in the Routing Rule menu

Field	Meaning
Priority	Determines the order of the filter rules, starting with *1* in increasing numerical order.
Administrative Status	The entry is used with *Enable*.
Outbound Line	Defines the PSTN line (PRI, BRI, FXO) or the SIP account used for an outgoing call.

An additional call routing entry is required for outgoing connections (without international prefix).

The **Calling Line** option indicates the ISDN port of the media gateway that is connected to the ISDN PBX.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**.

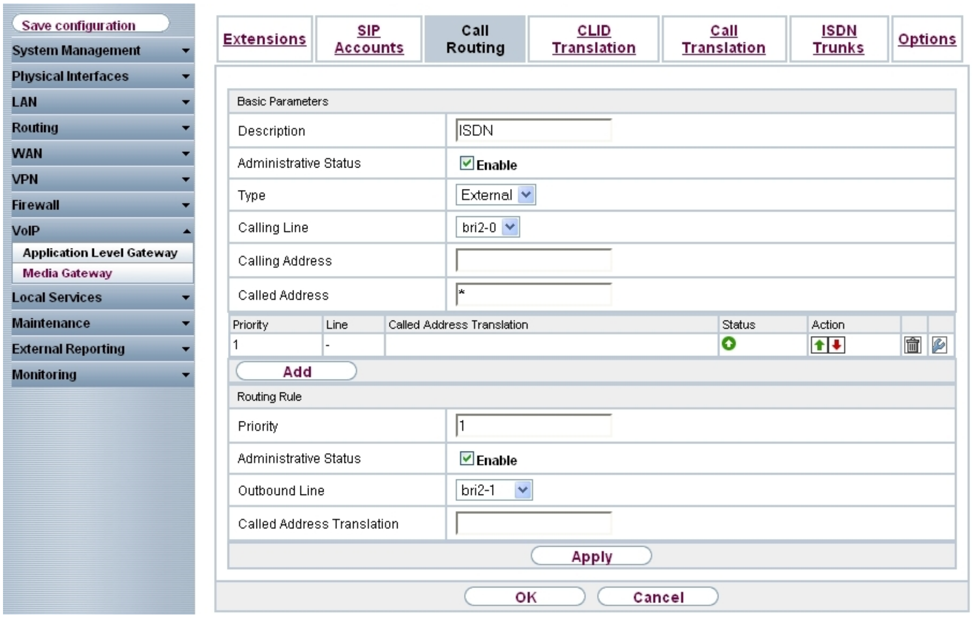

Fig. 74: **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**

If you select **Add** the external connection (ISDN line or SIP provider account) used for this entry is selected. In our example, the ISDN port *bri2-1* has been connected with the external ISDN point-to-multipoint exchange connection.

The complete configuration looks like this:

(1) Go to **VoIP** -> **Media Gateway** -> **Call Routing** .

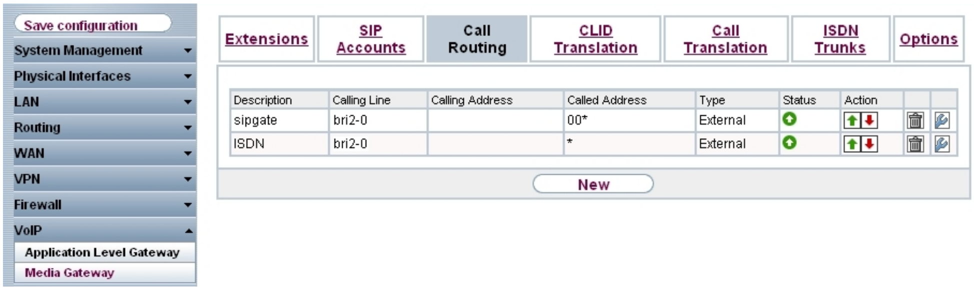

Fig. 75: **VoIP** -> **Media Gateway** -> **Call Routing**

Call Translation

Call translation is necessary to ensure that the calling party number (outgoing line) is transmitted correctly for outgoing calls initiated over the sipgate VoIP account. This call translation is configured in the **Call Translation** menu. In the following configuration the MSN subscriber numbers for the ISDN PBX are preceded with the prefix *49911* for outgoing calls. If, for example, a call is initiated over sipgate with the calling party number 76600695, the call is signalled with the subscriber number 4991176600695.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Translation** -> **New**.

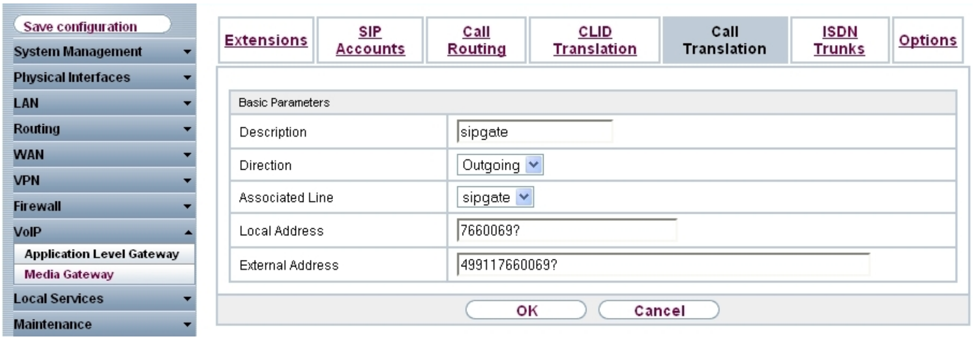

Fig. 76: **VoIP** -> **Media Gateway** -> **Call Translation** -> **New**

Relevant fields in the Call Translation menu

Field	Meaning
Description	Give the number translation a name.
Direction	Here you enter the direction to which the entry is to apply. Select *Outgoing* for outgoing calls.
Associated Line	Determines the line or SIP account via which the calls are to be routed.

Field	Meaning
Local Address	Here you enter the internal number (e.g. extension or PBX number). For outgoing calls, the signalled Calling Party Number (corresponds in the menu to the **Local Address** field) is translated to the **External Address**. Numerical and alphanumerical characters are permissible. ? is a placeholder for an arbitrary digit. Note **Local Address** and **External Address** must contain the same number of wildcards.
External Address	Enter the external number here. For outgoing calls, the signalled called party number (corresponding in the menu to the **Local Address** field) is translated to the **External Address**.

5.2 Overview of configuration steps

ISDN interface configuration

Field	Menu	Value
Port Usage	**Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-0 (NT)**	*Dialup (Euro ISDN)*
ISDN Configuration Type	**Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-0 (NT)**	*Point-to-multipoint*

Configuring the second ISDN interface

Field	Menu	Value
Autoconfiguration on Bootup	**Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-1 (TE)**	*Aktiviert*
Result of Autoconfiguration	**Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-1 (TE)**	*Port Usage: Dialup (Euro ISDN), ISDN Configuration Type: Point-to-multipoint*

SIP Account Configuration

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *sipgate*
Administrative Status	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Aktiviert*
Trunk Mode	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Client*
Registrar	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *sipgate.de*
Protocol	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *UDP*
Port	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*5060*
User Name	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *8861755*
Password	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *secret*
Registration	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Aktiviert*
Expire Time	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*60* Sec
SIP Header Field(s) for Caller Address	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Display only*
Codec Proposal Sequence	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**-> **Advanced Settings**	*Default*
Echo cancellation	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**-> **Advanced Settings**	*Aktiviert*
Comfort Noise Generation (CNG)	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**-> **Advanced Settings**	Aktiviert
Packet Size	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**-> **Advanced Settings**	*40* ms

Configuring the internal extension

Field	Menu	Value
Extension / User Name	**VoIP** -> **Media Gateway** ->	e.g. *76600690*

Field	Menu	Value
	Subscriber -> **New**	
Interface Type	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	*ISDN*
Select ISDN interface	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	e.g. *bri2-0*

Call Assignment

Field	Menu	Value
Session Border Controller Mode	**VoIP** -> **Media Gateway** -> **Options**	*Off*
Media Stream Termination	**VoIP** -> **Media Gateway** -> **Options**	*Aktiviert*
Default Drop Extension	**VoIP** -> **Media Gateway** -> **Options**	e.g. *76600691*
Dialling break	**VoIP** -> **Media Gateway** -> **Options**	e.g. *5* seconds

Call Routing

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *sipgate*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Enable*
Type	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*External*
Calling Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*bri2-0*
Called Address	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *00**
Priority	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*1*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*Enable*
Outbound Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	e.g. *sipgate*
Description	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *ISDN*

Field	Menu	Value
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Enable*
Type	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*External*
Calling Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*bri2-0*
Called Address	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. ***
Priority	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*1*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*Enable*
Outbound Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	e.g. *bri2-1*

Call Translation

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *sipgate*
Direction	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	*Outgoing*
Associated Line	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *sipgate*
Local Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *7660069?*
External Address	**VoIP** -> **Media Gateway** -> **Call Translation** -> **New**	e.g. *499117660069?*

Chapter 6 Media Gateway - Configuration for connection of a SwyxWare IP-PBX to an ISDN point-to-multipoint

6.1 Introduction

This chapter describes configuration of the **bintec R4100** as a media gateway for connection of a **SwyxWare** IP PBX to an ISDN point-to-multipoint.

The ISDN point-to-multipoint connection was wired with the block of numbers (MSN) 6898924 to 6898927. The **SwyxWare** IP PBX is connected to the media gateway via a SIP gateway trunk. All incoming calls are delivered to the **SwyxWare** IP PBX. Outgoing calls are routed into the ISDN network.

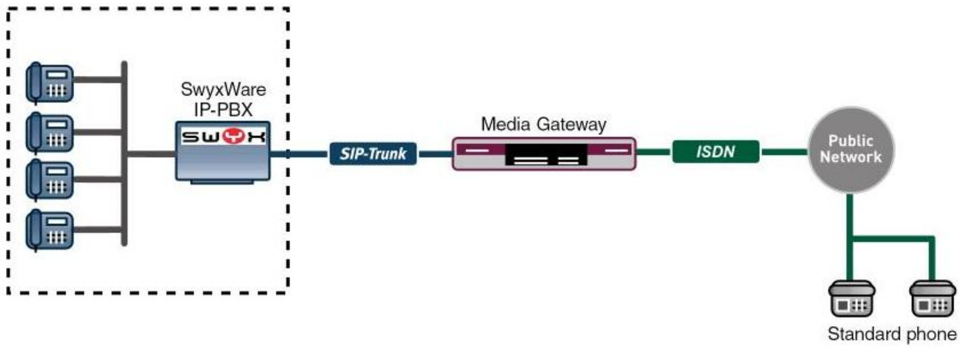

Fig. 77: Example scenario

Requirements

The following are required for the configuration:

- A **bintec R4100** with 7.8.4 system software
- A DSP module (4-way) must be installed.
- The ISDN port `ISDN-0` must be connected to the ISDN point-to-multipoint.
- The **SwyxWare** IP-PBX must already be preconfigured (user and terminal configuration , etc.).

Configuration of the **bintec R4100** is performed using the **GUI** (Graphical User Interface).

6.2 Configuration

6.2.1 Configuration of a trunk group in the SwyxWare administrator

To create a new SIP gateway trunk in the **SwyxWare** administrator, a new trunk group must first be configured. Trunk groups consist of one or more trunks possessing similar characteristics.

The assistant to create a new trunk group is launched in the **SwyxWare** administrator. For this, go to the following menu:

(1) Go to **Trunk Groups** -> **Add Trunk Group...** .

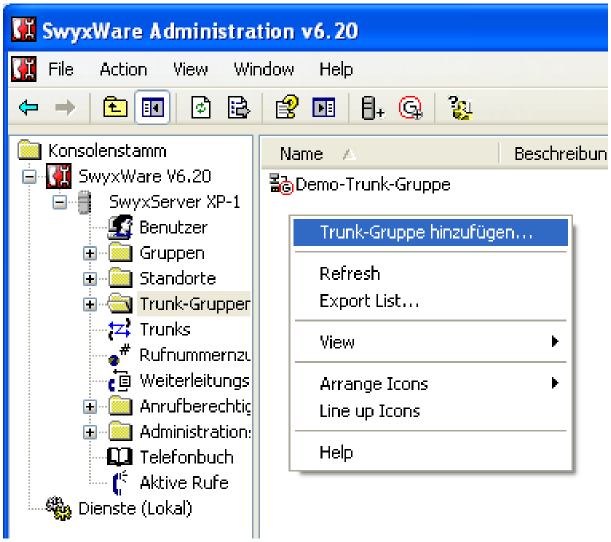

Fig. 78: Create new trunk group

The **assistant to add a trunk group** opens. Follow the assistant's instructions.

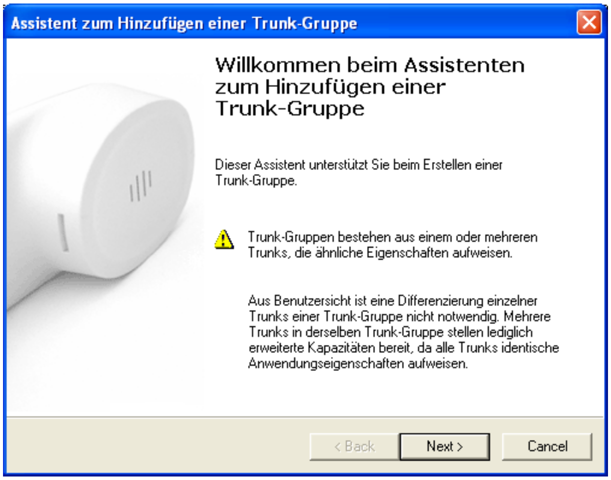

Fig. 79: Assistant for adding a trunk group

Click **Next**.

First, a name must be assigned to the trunk group.

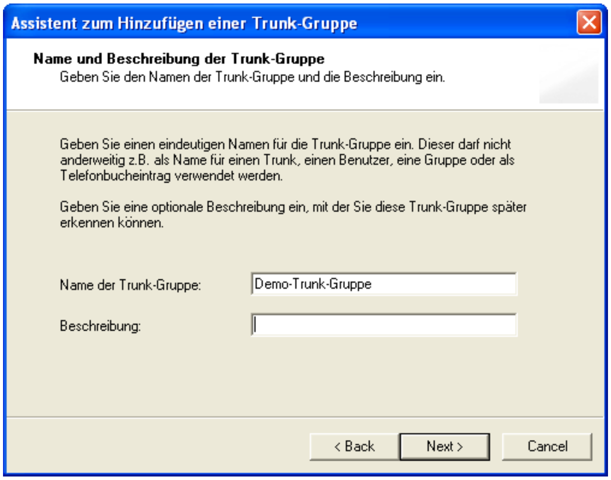

Fig. 80: Name of trunk group

(1) Under **Name of Trunk Group** enter `Demo Trunk Group`, for example.

In the next step, the type of trunk group is defined.

Fig. 81: Type of trunk group

(1) To connect a bintec media gateway, select `SIP gateway` in **Type of Trunk Group** .

(2) In **Profil**, select `SwyxConnect`.

In our example, all outgoing calls are routed over the bintec media gateway.

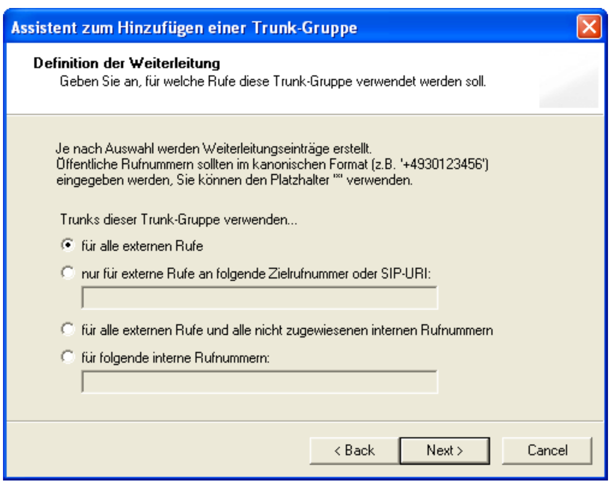

Fig. 82: Definition of forwarding

(1) In **Use trunks from this group...**, select `for all external calls`.

In this example, incoming calls are not subject to call restrictions.

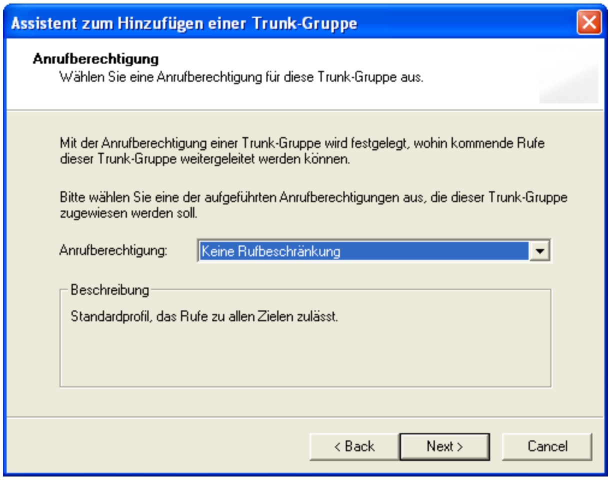

Fig. 83: Call authorisation

(1) In **Call Authorisation**, select `No Call Restriction`.

As the last step of this assistant, the trunk group is assigned a locality.

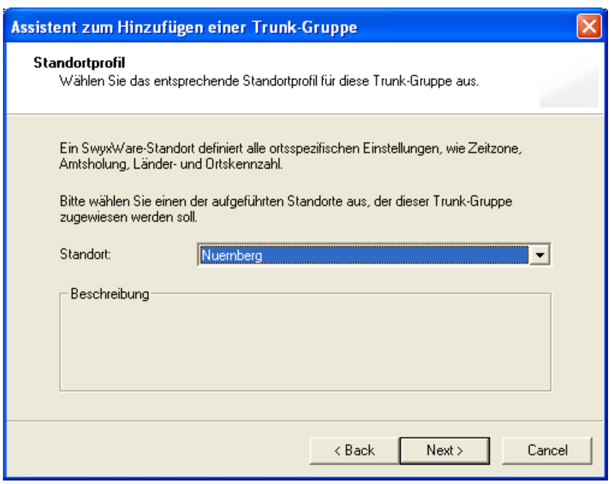

Fig. 84: Locality profile

(1) In **Locality**, select one of the localities configured in the **SwyxWare** administrator.

Fig. 85: Close assistant

Configuration of the trunk group is thus complete. Click **Finish**.

6.2.2 Configuration of a SIP trunk in the SwyxWare administrator

After creation of a trunk group, configuration of a SIP trunk for connection of the bintec media gateway can begin.

The assistant to create a trunk is launched in the **SwyxWare** administrator. For this, click on the associated trunk group:

(1) Go to **Trunk Groups** -> **Demo Trunk Group...** -> **Add Trunk...** .

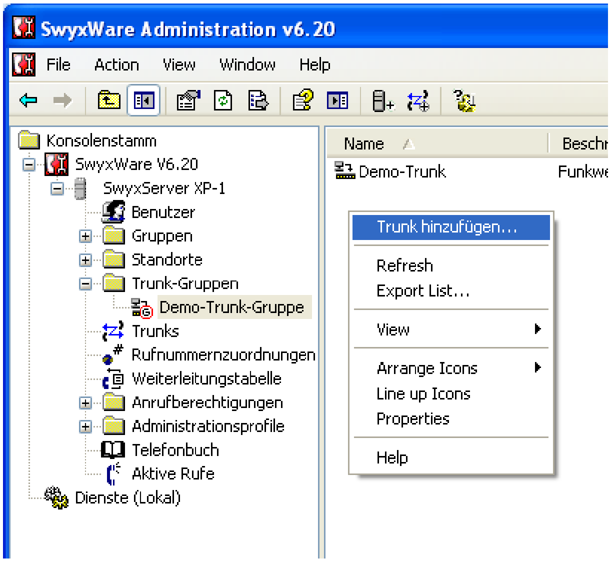

Fig. 86: Add trunk

The **assistant to add a trunk**opens. Follow the assistant's instructions.

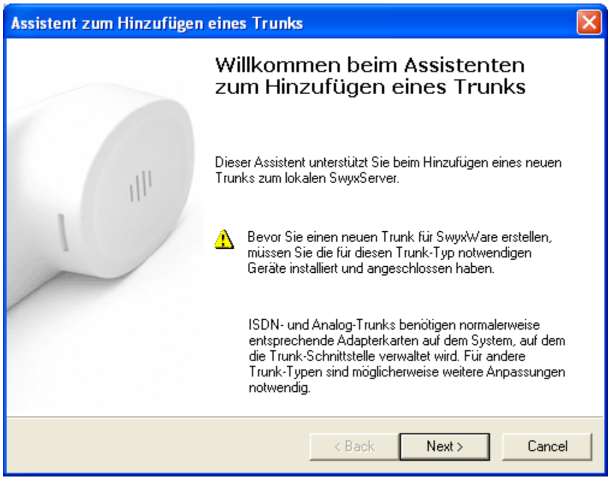

Fig. 87: Assistant for adding a trunk

Click **Next**.

At the beginning of the assistant, a name and description must be assigned to the trunk for connection to the bintec media gateway.

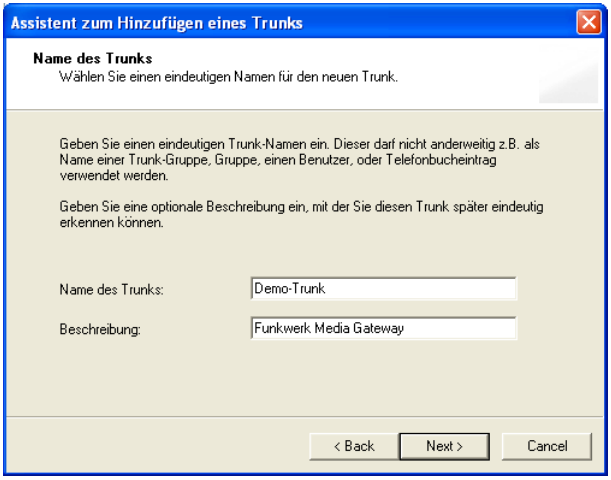

Fig. 88: Trunk name

(1) Under **Trunk name** enter `Demo Trunk`, for example.

(2) For example, under **Description** enter `bintec Media Gateway`.

At the next step of the assistant, the SIP user and SIP authentication are entered. These data are required for registration of the bintec media gateway.

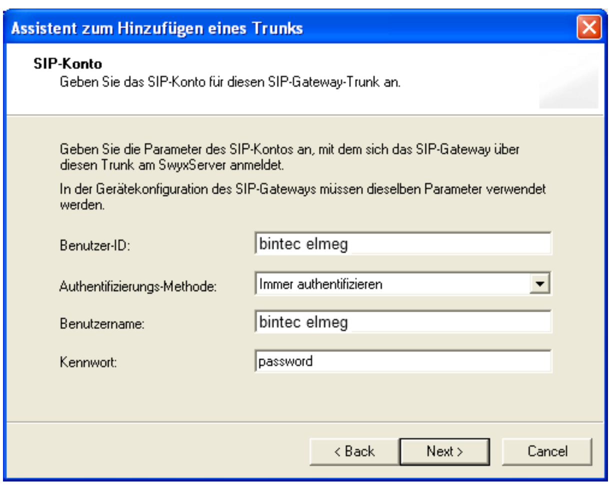

Fig. 89: SIP account

(1) In **User ID** enter an elective value, e. g. `bintec elmeg`.

(2) Set **Authentication Method** to `Always Authenticate`.

(3) In **User Name**, enter an elective value, e. g. `bintec elmeg`.

(4) Under**Password** enter your password.

In our example, the media gateway is operated on an ISDN point-to-multipoint connection with a continuous block of numbers (MSN: 6898924-6898927). This block of numbers is assigned to the trunk group.

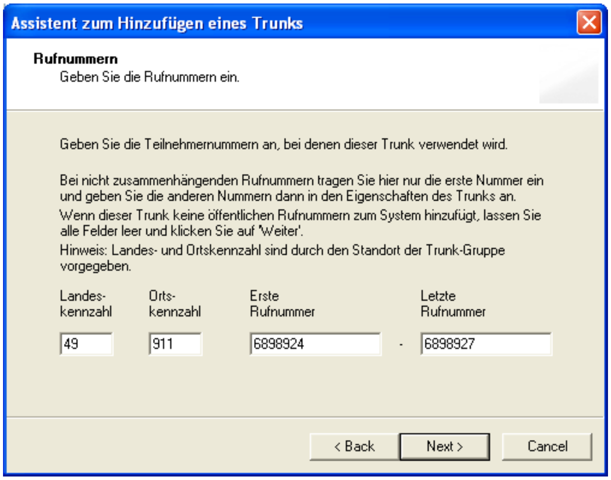

Fig. 90: Subscriber numbers

(1) With continuous numbers, under **First Number** enter the first number of the block of numbers (here, e.g., `6898924`).

(2) Under **Last Number**, enter the last number of the block of numbers (here, e.g., `6898927`).

The codec selection is taken over unchanged.

Fig. 91: Codecs

In our example, the bintec media gateway is operated on an ISDN point-to-multipoint connection. For this reason, the number of simultaneous calls is limited to `two`.

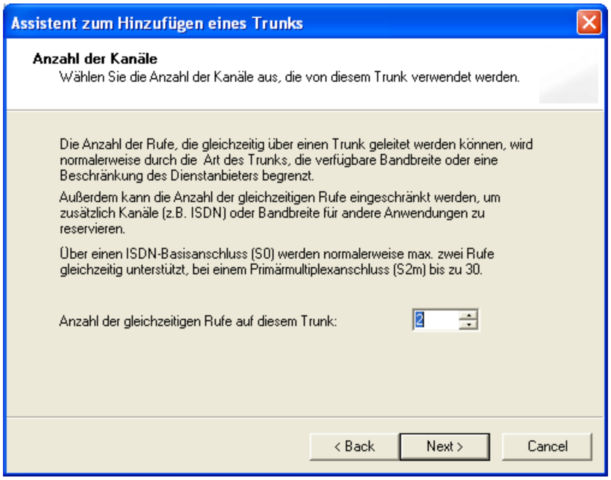

Fig. 92: Number of channels

As the last step of the assistant, the trunk to the bintec media gateway is assigned to the **SwyxWare** server.

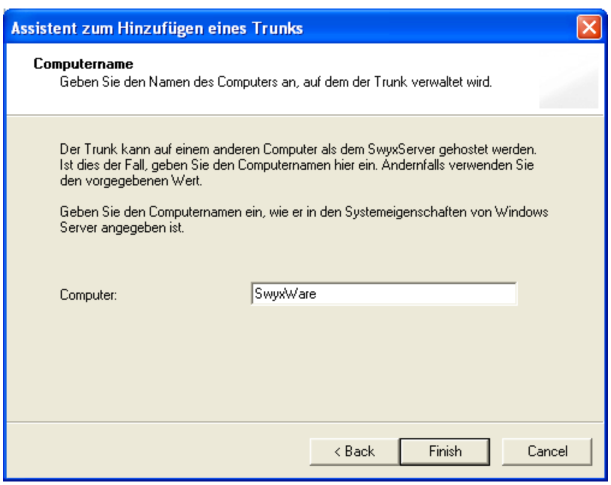

Fig. 93: Computer name

(1) Under **Computer**, enter the computer name of the **SwyxWare** server.

(2) This concludes trunk configuration. Click **Finish**.

6.2.3 Configuration of the bintec media gateway

ISDN interface configuration

The ISDN port `ISDN-0` of the media gateway is connected to the NTBA of the point-to-multipoint. Configuration of the ISDN interface is already wired in ISDN TE mode ex-works, and the ISDN switch type is automatically recognised at media gateway startup.

Do not perform any modifications for this port in menu **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-0 (TE)** .

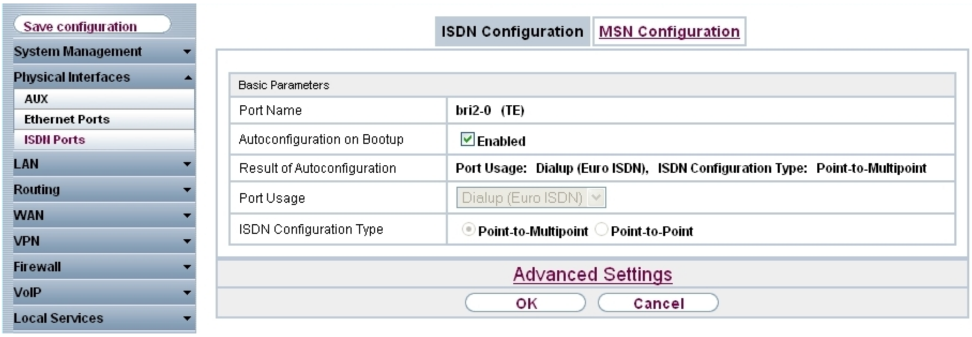

Fig. 94: **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-0 (TE)**

SIP Account Configuration

A SIP account must be created at gateway configuration for connection of the bintec media gateway.

To save the login data for registering the media gateway with the **SwyxWare** IP PBX, go to the following menu:

(1) Go to **VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**.

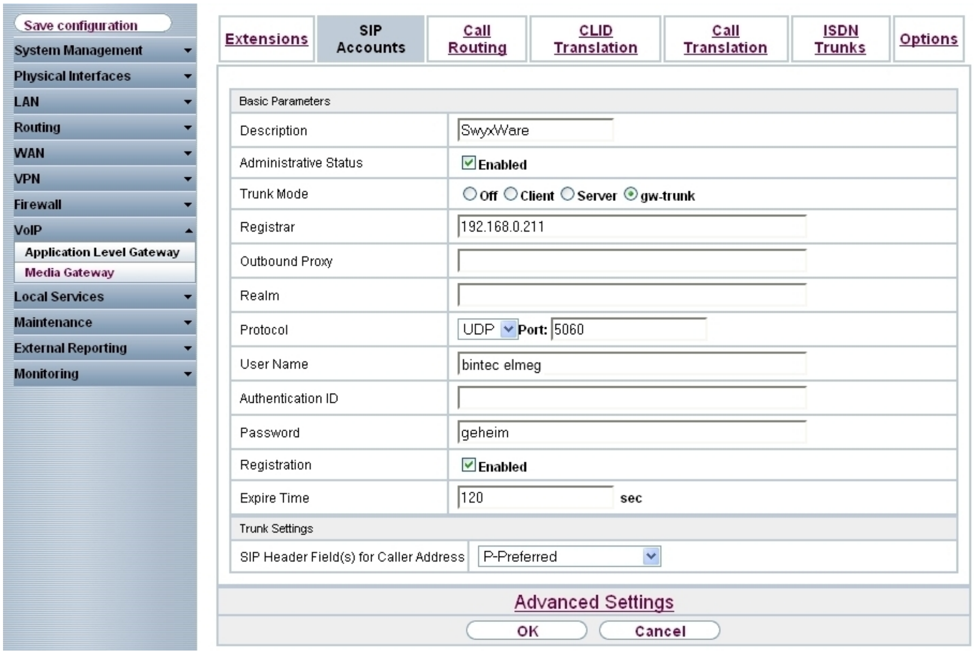

Fig. 95: **VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**

Relevant fields in the SIP Accounts menu

Field	Description
Description	Here, assign a name to the account. Maximum number of characters: 40.
Administrative Status	Enable the administrative status of the account.
Trunk Mode	Select the trunk mode to be used. If you select `gw-trunk`, the gateway trunk is used.
Registrar	Here, enter the IP address of the **SwyxWare** server. Maximum number of characters: 40.
Protocol	Select the protocol to be used for the connection to the server or proxy.
Port	Number of the TCP or UDP port to be used for the connection to the server or proxy.
User Name	Here, enter the username for authentication if your VoIP provider has assigned one to you.
Authentication ID	Enter a name that is to be used for authentication. If you do not enter a name, the name in the **User Name** field is used.
Password	The VoIP provider gives you a PIN or password for authentication. You must enter this value here. Maximum number of char-

Field	Description
	acters: 40.
Registration	Enables or disables the SIP REGISTER registration mechanism.
Expire Time	Shows the time in seconds after which the current registration becomes invalid and a new registration request is therefore sent. Here, the SIP expire time is matched to the **SwyxWare** IP PBX. `120` seconds are used as the default value.
SIP Header Field(s) for Caller Address	With a SIP trunk to the **SwyxWare** IP PBX, the outgoing number is indicated via the SIP header field *P-Preferred* (according to RFC 3325) in the SIP INVITE message.

Call Routing

In this example, all incoming calls over the ISDN line are routed to the **SwyxWare** IP PBX. All outgoing calls (**SwyxWare IP PBX** -> **ISDN**) are routed by the media gateway to the ISDN line. Two routing entries are necessary for this. Shown below is the configuration of the call routing entry for outgoing connections.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**.

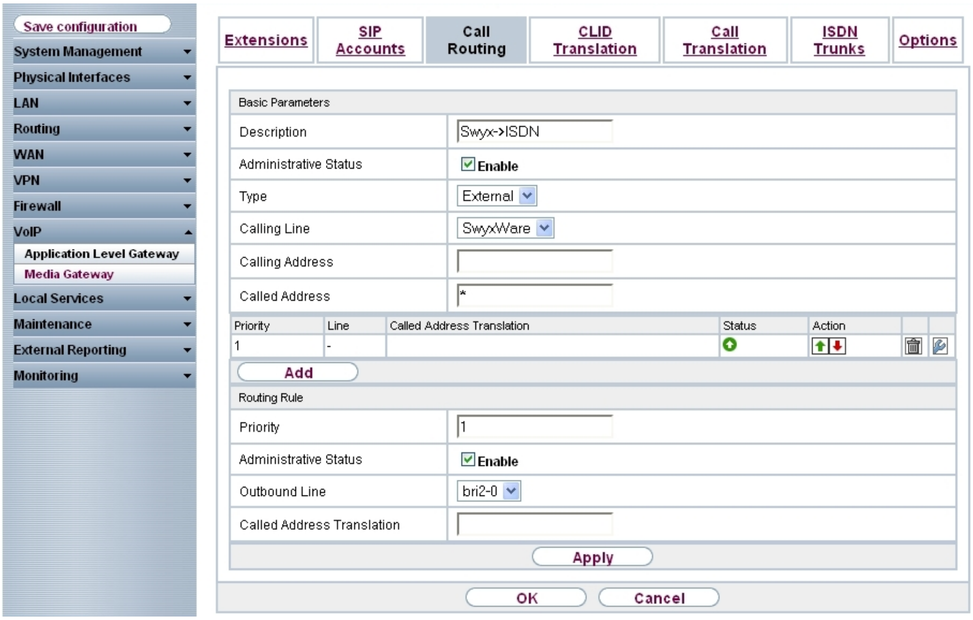

Fig. 96: **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**

Relevant fields in the Call Routing menu

Field	Description
Description	Here, enter the name of the call routing entry.
Administrative Status	The entry is used with *enabled*.
Type	Select *External* for calls that are to be routed as outgoing, external calls.
Calling Line	Here you can restrict the routing entry to the line on which the call comes in. The selection depends on the interfaces available and on the SIP accounts that have been created. If you select *SwyxWare*, the routing entry is restricted to the selected SIP account.
Calling Address	Here you can restrict the routing entry to a particular caller. To do this, you must specify the subscriber number exactly (no wildcards).
Called Address	Here, you can enter an address (call number) that is compared with the dialled address. You can use wildcards here. For example, *** means that at the end of a character string any number of additional characters can follow. If the configured address agrees with the signalled address, the routing entry is used.

You can now create a list with rules that are assigned to the currently selected routing entry, and that serve to manipulate the signalled destination number. You can also delete routing entries.

Use **Add** to create entries.

Relevant fields in the Routing Rule menu

Field	Description
Priority	Determines the order of the filter rules, starting with *1* in increasing numerical order.
Administrative Status	The entry is used with *Enable*.
Outbound Line	Defines the PSTN line (PRI, BRI, FXO) or the SIP account used for an outgoing call.

A call routing entry for incoming calls must then be configured.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**.

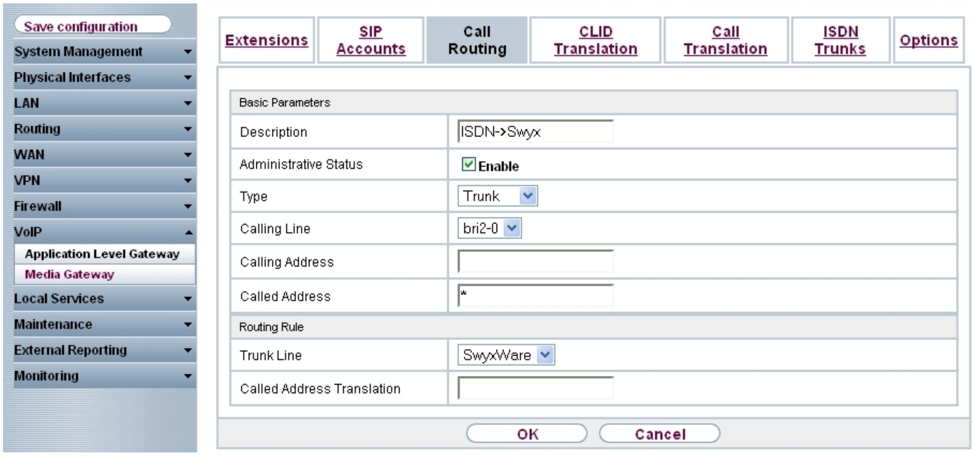

Fig. 97: **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**

Relevant fields in the Call Routing menu

Field	Description
Description	Here, enter the name of the call routing entry.
Administrative Status	The entry is used with `enabled`.
Type	Select `Trunk` for calls that are routed to a PBX behind the media gateway.
Calling Line	Here you can restrict the routing entry to the line on which the call comes in. The selection depends on the interfaces available and on the SIP accounts that have been created. If you select `SwyxWare`, the routing entry is restricted to the selected SIP account.
Calling Address	Here you can restrict the routing entry to a particular caller. To do this, you must specify the subscriber number exactly (no wildcards).
Called Address	Here, you can enter an address (call number) that is compared with the dialled address. You can use wildcards here. For example, `*` means that at the end of a character string any number of additional characters can follow. If the configured address agrees with the signalled address, the routing entry is used.

After creation of both entries, these are displayed in the **Routing** menu.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Routing** .

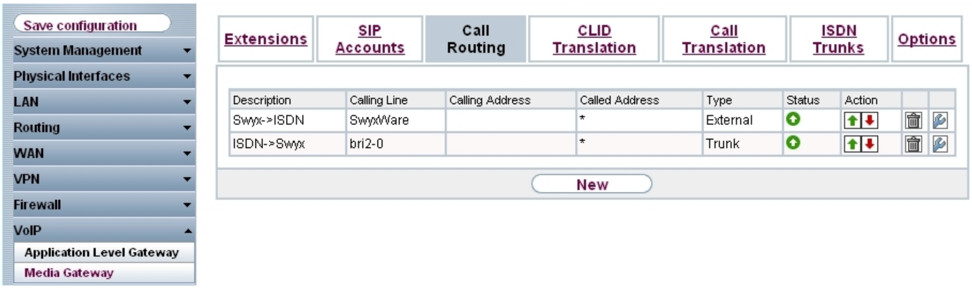

Fig. 98: **VoIP** -> **Media Gateway** -> **Call Routing**

6.3 Overview of configuration steps

Add trunk group

Field	Menu	Value
SwyxWare Adminis-tration	**SwyxWare** -> **Swyx Server** -> **Trunk Groups** -> **Add Trunk Group...**	e. g. `Demo Trunk Group`

Assistant

Field	Menu	Value
Assistant	**Assistant for adding a trunk group**	**Next**
Name of trunk group	**Assistant for adding a trunk group**	e. g. `Demo Trunk Group`
Type of trunk group	**Assistant for adding a trunk group**	e.g. `SIP Gateway`
Profile	**Assistant for adding a trunk group**	e. g. `SwyxConnect`
Definition of forward-ing	**Assistant for adding a trunk group**	`for all external calls`
Call authorisation	**Assistant for adding a trunk group**	`No call restric-tion`
Locality profile	**Assistant for adding a trunk group**	e. g. `NÃ¼rnberg`

Add trunk

Field	Menu	Value
SwyxWare Adminis-tration	**SwyxWare** -> **Swyx Server** -> **Trunk Groups** -> **Demo Trunk Group** -> **Add Trunk...**	e. g. `Demo Trunk`

Assistant

Field	Menu	Value
Assistant	**Assistant for adding a trunk**	**Next**
Trunk name	**Assistant for adding a trunk** -> **Trunk Name**	e. g. *Demo Trunk*
Description	**Assistant for adding a trunk** -> **Trunk Name**	e.g. *bintec Media Gateway*
User ID	**Assistant for adding a trunk** ->**SIP Account**	e. g. *bintec elmeg*
Authentication method	**Assistant for adding a trunk** ->**SIP Account**	e.g. *Always authenticate*
User Name	**Assistant for adding a trunk** ->**SIP Account**	e. g. *bintec elmeg*
Password	**Assistant for adding a trunk** ->**SIP Account**	Password
First call number	**Assistant for adding a trunk** ->**Call number**	e.g. *6898924*
Last call number	**Assistant for adding a trunk** ->**Call number**	e.g. *6898927*
Number of channels	**Assistant for adding a trunk**	*2*
Computer name	**Assistant for adding a trunk**	e. g. *SwyxWare*

ISDN interface configuration

Field	Menu	Value
Autoconfiguration on Bootup	**Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-0 (TE)**	*Aktiviert*
Result of Autoconfiguration	**Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<bri2-0 (TE)**	*Port Usage: Dialup (Euro ISDN), ISDN Configuration Type: Point-to-multipoint*

Configuration of SIP accounts

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e. g. *SwyxWare*
Administrative Status	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Aktiviert*

Field	Menu	Value
Trunk Mode	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	gw-trunk
Registrar	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e. g. *192.168.0.211*
Protocol	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *UDP*
Port	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*5060*
User Name	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e. g. *bintec elmeg*
Password	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *secret*
Registration	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*Aktiviert*
Expire Time	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	*120* Sec
SIP Header Field(s) for Caller Address	**VoIP** -> **Media Gateway** -> **SIP Accounts** -> **New**	e.g. *P-Preferred*

Call routing for outgoing calls

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e. g. *Swyx->ISDN*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Enable*
Type	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*External*
Calling Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e. g. *SwyxWare*
Called Address	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. ***
Priority	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*1*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	*Enable*
Outbound Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	e.g. *bri2-0*

Call routing for incoming calls

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e. g. *ISDN->Swyx*
Administrative Status	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Enable*
Type	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Trunk*
Calling Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *bri2-0*
Called Address	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e.g. *
Trunk Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	e. g. *SwyxWare*

Chapter 7 Media Gateway - Connection of a virtualised serVonic IXI-UMS server to a bintec R1200

7.1 Introduction

This chapter describes connection of a serVonic **IXI-UMS**solution operated in a VMware environment to the ISDN network.

For this, the serVonic **IXI-UMS** kernel uses the remote CAPI interface of the **bintec R1200** router. For provision of all required CAPI protocols (e.g., T.30 modem for FAX G3), the **bintec R1200**/**R3000**/**R4100** must be equipped with the optional VoIP DSP module.

In our example, a ISDN point-to-multipoint is used for the ISDN connection.

The **GUI** (Graphical User Interface) is used here for configuration of the **bintec R1200**.

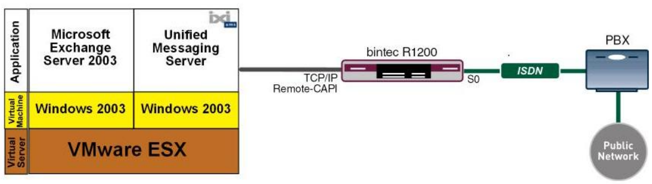

Fig. 99: Example scenario

Requirements

The following are required for the configuration:

- A **bintec R1200** with 7.8.4 system software.
- A DSP module (4-way) must be installed.
- Basic installation of the **IXI-UMS** kernel is assumed, along with an existing VMware environment.
- During installation, the router must already be connected to the ISDN point-to-multipoint.

7.2 Configuration

7.2.1 Configuration of the bintec R1200 as a remote CAPI server

Installation of the DSP module

Detailed information on installing the DSP module can be found in the release notes for the system software version 7.5.1., chapter: 2.1 DSP Module - Installation and function testing (*http://www.bintec-elmeg.com* / portal/downloadcenter/dateien/r1200/r7501p01/relnote_751_de.pdf).

Following successful installation of the DSP module, the module type is displayed on the **GUI** status page.

In our example, *`4 Channel VINETIC`*.

(1) Go to **System Administration** -> **Status**.

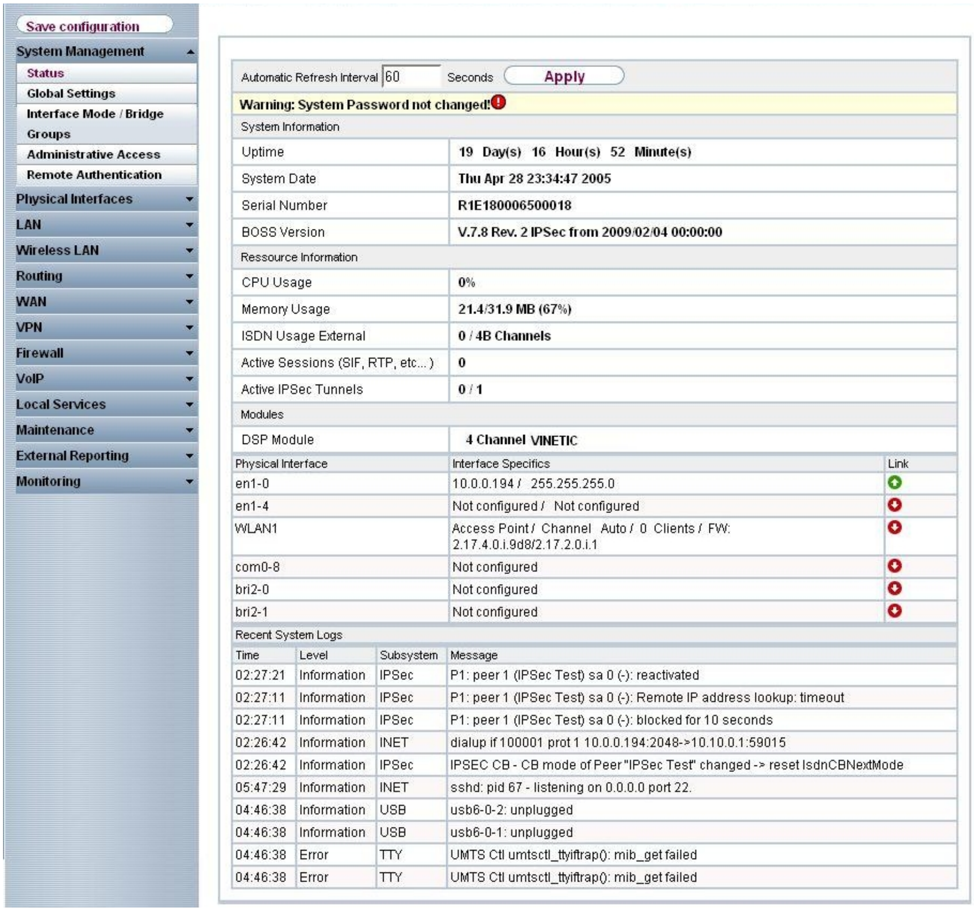

Fig. 100: **System Administration** ->**Status**

ISDN interface configuration

The router's ISDN port must already be connected with the ISDN point-to-multipoint during the configuration. At startup of the **bintec R1200** the router performs an ISDN auto-recognition and displays the result.

In the **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** menu, a list of all ISDN ports and their configurations is shown.

Here, the ISDN point-to-multipoint connection was successfully recognised.

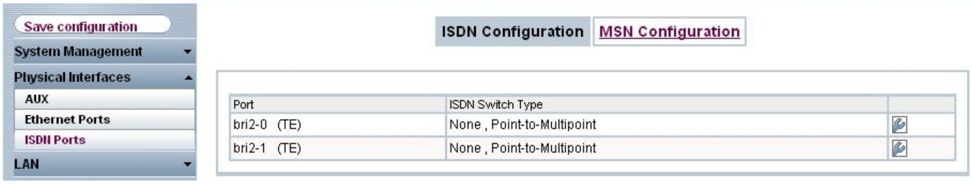

Fig. 101: **Physical Interfaces** -> **ISDN Ports**-> **ISDN Configuration**

Ex works (or without MSN configuration), the router accepts all incoming ISDN connections, thus permitting remote configuration via ISDN login. As the router in our example routes all connections to the serVonic **IXI-UMS**solution, a "dummy call number" must be configured. As soon as an entry exists, the incoming calls not assigned to any entry are forwarded to the CAPI service.

(1) Go to **Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**.

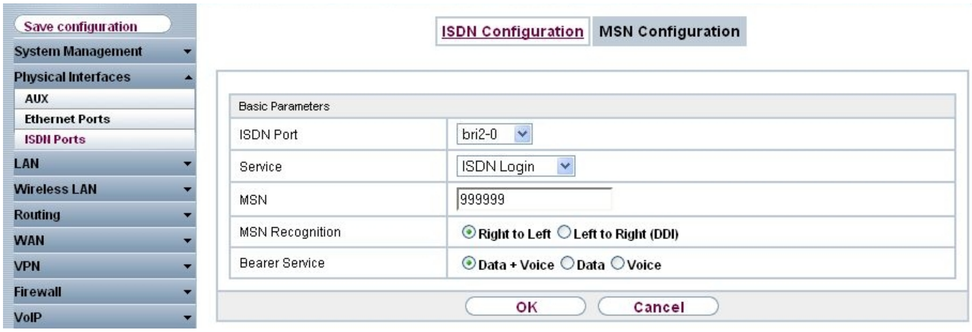

Fig. 102: **Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**

Relevant fields in the MSN Configuration menu

Field	Description
ISDN Port	Select the ISDN port for which the MSN is to be configured.
Service	Select the service to which a call is to be assigned on the MSN.
MSN	Enter any number here (dummy call number).
MSN Recognition	Select the mode your device is to use for the number comparison of **MSN** with the called party number of the incoming call. Possible values: • `Right to Left` (default value) • `Left to Right (DDI)`: If your device is connected to a point-to-point connection.
Service attribute	Select the type of incoming call.

Remote CAPI server configuration

The remote CAPI server of the **bintec R1200** is already enabled ex-works.

(1) Go to **Local Services** -> **CAPI Server**->**Options**.

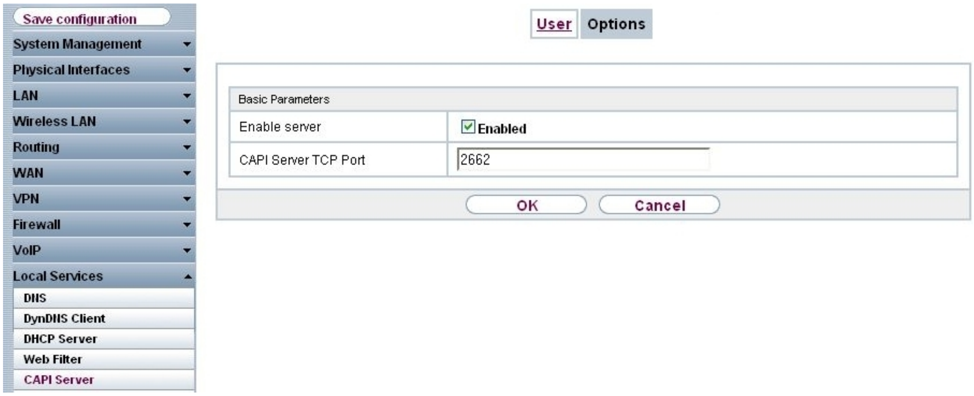

Fig. 103: **Local Services** ->**CAPI Server** ->**Options**

Relevant fields in the Options menu

Field	Description
Enable server	The function is activated by selecting `Enabled`. The function is enabled by default.
CAPI Server TCP Port	The field can only be edited if **Enable Server** is enabled. Enter the TCP port number for remote CAPI connections. The default value is `2662`.

For security reasons, access to the remote CAPI interface should be protected with a user name and password.

Note

Ex works, a user with the user name `default` and no password is always entered for the CAPI subsystem. All calls to the CAPI are offered to all CAPI applications in the LAN. Use the Settings menu to distribute incoming calls for the CAPI subsystem to defined users with password. You should then delete the user `default` without password.

(1) Go to **Local Services** -> **CAPI Server**->**User**->**New**.

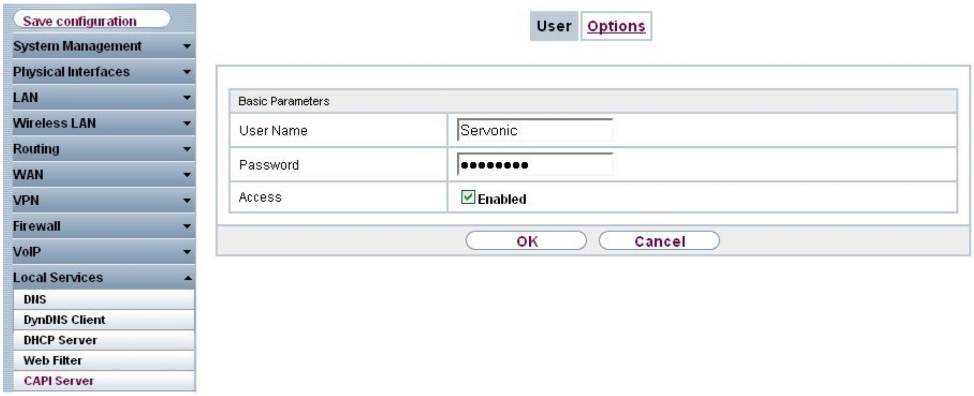

Fig. 104: **Local Services** ->**CAPI Server** ->**User**-> **New**.

Relevant fields in the User menu

Field	Description
User Name	Enter the user name for which access to the CAPI service is to be allowed.
Password	Enter the password which the user shall use for identification to gain access to the CAPI service.
Access	Select whether access to the CAPI service is to be permitted or denied for the user. The function is activated by selecting `Enabled`.

7.2.2 Configuration of remote CAPI client software

The remote CAPI client software is a component of the **BRICKware** software package. The latter is located on the provided companion CD, or can be accessed in the download area at *www.bintec-elmeg.com* . The remote CAPI client software is installed in the **BRICKware** program group.

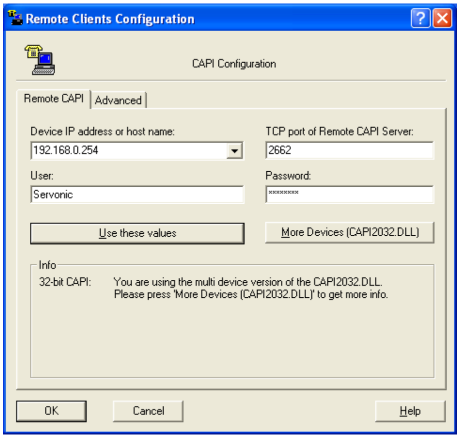

Fig. 105: Remote Clients Configuration

(1) To log in the remote CAPI client, the **Device IP address or host name** of the **bintec R1200**must be saved.

(2) Under **User** enter `Servonic`, for example.

(3) Enter the **password**.

(4) Apply the configuration with **Use these values**.

(5) As confirmation, a corresponding message appears in the info area of the remote CAPI client software.

Detailed in information about the configured CAPI servers and their CAPI controllers is provided under **Remote Multi CAPI Client Configuration**.

Following login of the remote CAPI client software bintec router, which functions as CAPI server, one CAPI controller is displayed per ISDN interface.

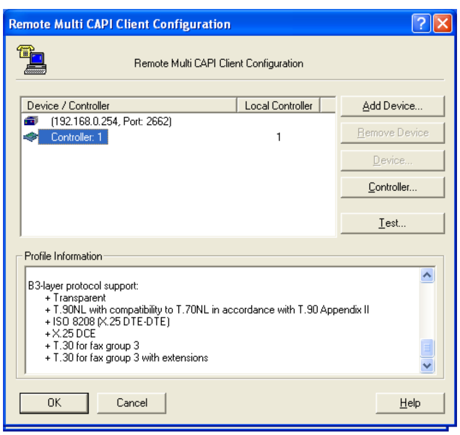

Fig. 106: Remote Multi CAPI Client Configuration

7.2.3 IXI-UMS kernel configuration for the remote CAPI interface

With basic installation of the serVonic **IXI-UMS** kernel, the **SerVonic**->**IXI-UMS**program group was created on your server. There, you will find the **IXI-UMS Kernel Configuration** Microsoft Management Console. In this Management Console, the dialog for configuration of the ISDN hardware is launched.

(1) Go to **IXI-UMS Kernel Configuration** -> **Hardware** -> **Properties** .

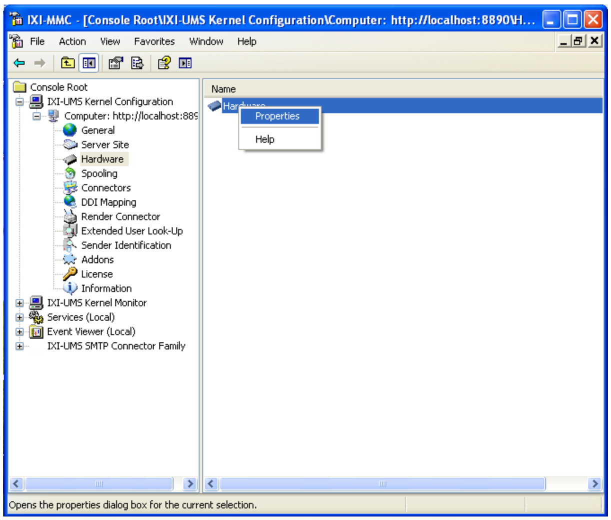

Fig. 107: **IXI-UMS** *Kernel Configuration*

With the **Hardware Detection** of the IXI-UMS kernel, the remote CAPI interface is recognised as ISDN hardware.

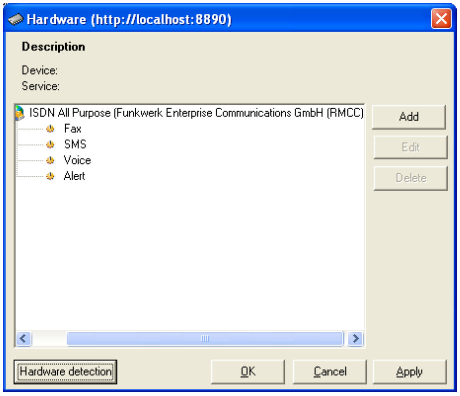

Fig. 108: Hardware

Under **Add ISDN device**, an ISDN controller with two B-channels is displayed. This dialog also offers the option of limiting the number of usable CAPI controllers.

With **Add**, you can modify the properties of the respective CAPI controllers (e.g., number of available B-channels).

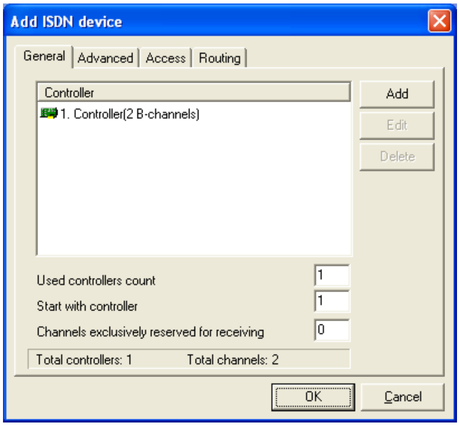

Fig. 109: Add ISDN device

After confirmation, the **IXI-UMS** kernel services are restarted. Following this, two available ISDN B-channels are displayed in**IXI-UMS Kernel Monitor** under **Channels** with Ready status.

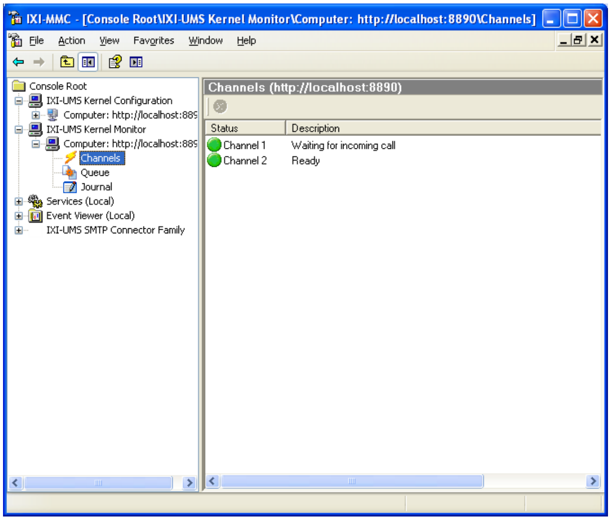

Fig. 110: **IXI-UMS** *Kernel Monitor*

In **IXI-Kernel Monitor** a test fax can now be sent.

(1) Go to **IXI-UMS Kernel Monitor** -> **Queue**.

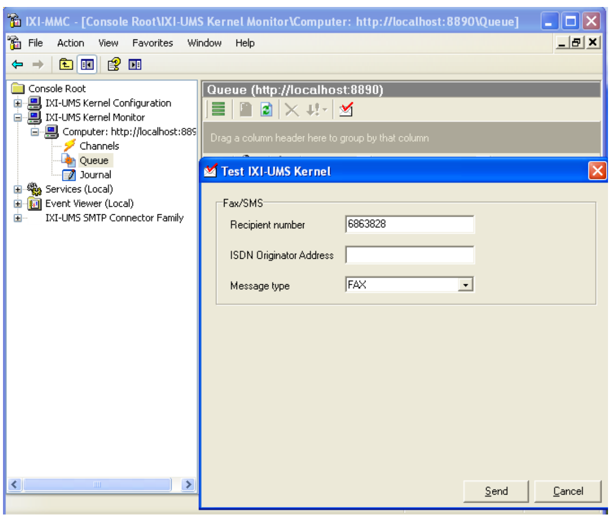

Fig. 111: Test **IXI-UMS** *Kernel*

Further configuration of the serVonic **IXI-UMS**solution will not be discussed here. For connections to various server systems, such as Microsoft Exchange Server, Lotus Domino, etc., we refer you to our Technology Partner, serVonic.

7.3 Overview of configuration steps

MSN Configuration

Field	Menu	Value
ISDN Port	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*bri2-0*
Service	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*ISDN Login*
MSN	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	e.g. *999999*
MSN Recognition	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*Right to Left*
Service attribute	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*Data + Voice*

Remote CAPI server configuration

Field	Menu	Value
Enable server	**Local Services** ->**CAPI Server** ->**Options**	*Aktiviert*
CAPI Server TCP Port	**Local Services** ->**CAPI Server** ->**Options**	e.g. *2662*
User Name	**Local Services** ->**CAPI Server** ->**User**-> **New**.	e.g. *Servonic*.
Password	**Local Services** ->**CAPI Server** ->**User**-> **New**.	*Password*
Access	**Local Services** ->**CAPI Server** ->**User**-> **New**.	*Aktiviert*

Configuration of remote CAPI client software

Field	Menu	Value
Device IP address or host name	**Remote Clients Configuration**	e.g. *192.168.0.254*
User	**Remote Clients Configuration**	*Servonic*
Password	**Remote Clients Configuration**	*Password*

Chapter 8 Media Gateway - Connection of a virtualised serVonic Tobit David server to a bintec R1200

8.1 Introduction

This chapter describes connection of a serVonic **Tobit David** server operated in a VMware environment to the ISDN network. For this, the David server uses the remote CAPI interface of the **bintec R1200** router. For provision of all required CAPI protocols (e.g., T.30 modem for FAX G3), the **bintec R1200**/**R3000**/**R4100** must be equipped with the optional VoIP DSP module.

In our example, a ISDN point-to-multipoint is used for the ISDN connection.

The **GUI** (Graphical User Interface) is used here for configuration of the **bintec R1200**.

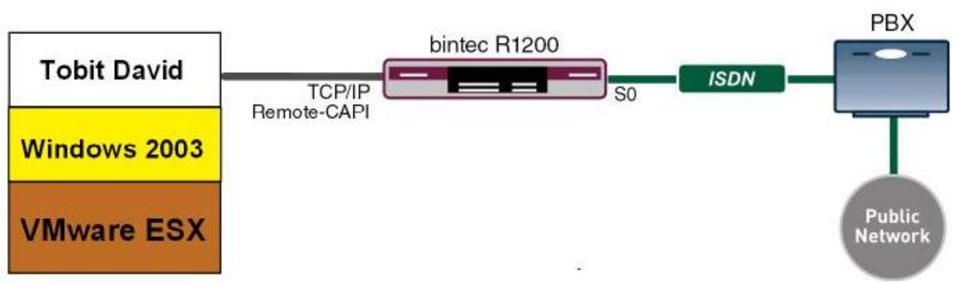

Fig. 112: Example scenario

Requirements

The following are required for the configuration:

- A **bintec R1200** with 7.8.4 system software.
- A DSP module (4-way) must be installed.
- Basic installation of the **Tobit David** server is assumed, along with an existing VMware environment.
- During installation, the router must already be connected to the ISDN point-to-multipoint.

8.2 Configuration

8.2.1 Configuration of the bintec R1200 as a remote CAPI server

Installation of the DSP module

Detailed information on installing the DSP module can be found in the release notes for the system software version 7.5.1., chapter: 2.1 DSP Module - Installation and function testing (*http://www.bintec-elmeg.com* / portal/downloadcenter/dateien/r1200/r7501p01/relnote_751_de.pdf).

Following successful installation of the DSP module, the module type is displayed on the **GUI** status page.

In our example, `4 Channel VINETIC`.

(1) Go to **System Administration** -> **Status**.

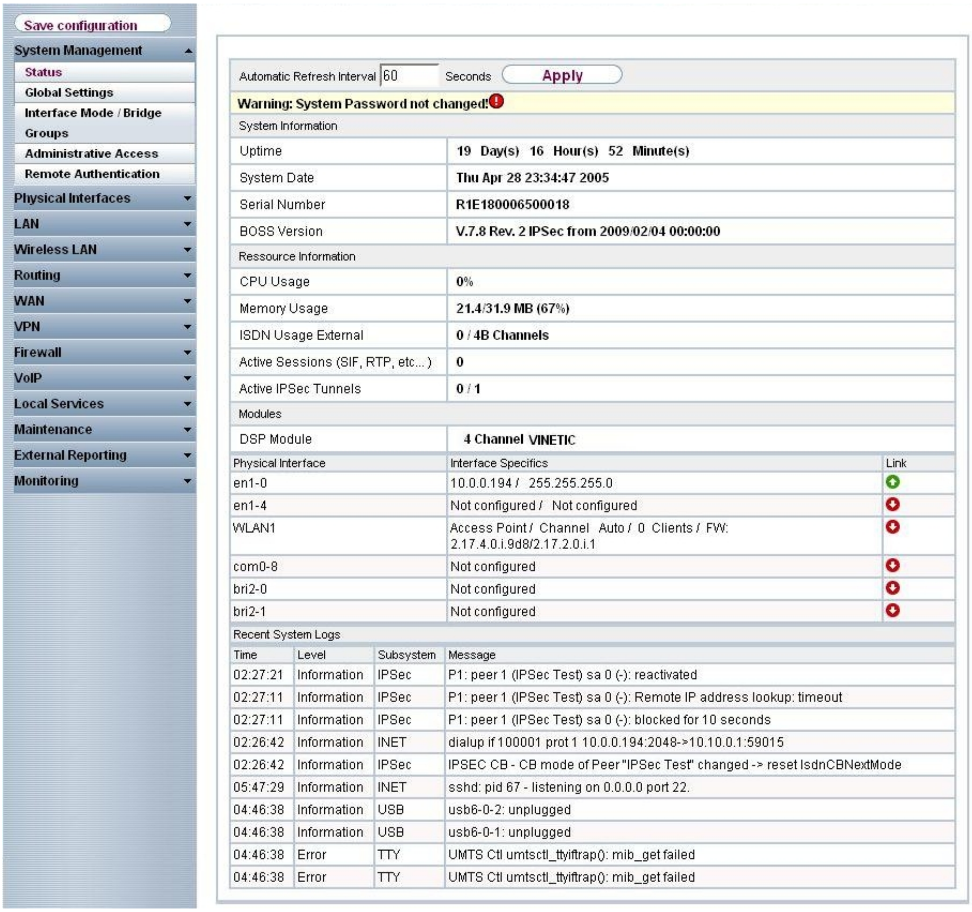

Fig. 113: **System Administration** ->**Status**

ISDN interface configuration

The router's ISDN port must already be connected with the ISDN point-to-multipoint during the configuration. At startup of the **bintec R1200**, the router performs an ISDN auto-recognition and displays the result.

In the **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** menu, a list of all ISDN ports and their configurations is shown.

Here, the ISDN point-to-multipoint connection was successfully recognised.

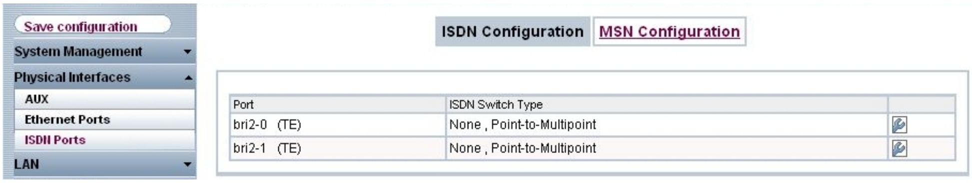

Fig. 114: **Physical Interfaces** -> **ISDN Ports**-> **ISDN Configuration**

Ex works (or without MSN configuration), the router accepts all incoming ISDN connections, thus permitting remote configuration via ISDN login. As the router in our example routes all connections to the serVonic **Tobit David** Server, a "dummy call-number" must be configured. As soon as an entry exists, the incoming calls not assigned to any entry are forwarded to the CAPI service.

(1) Go to **Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**.

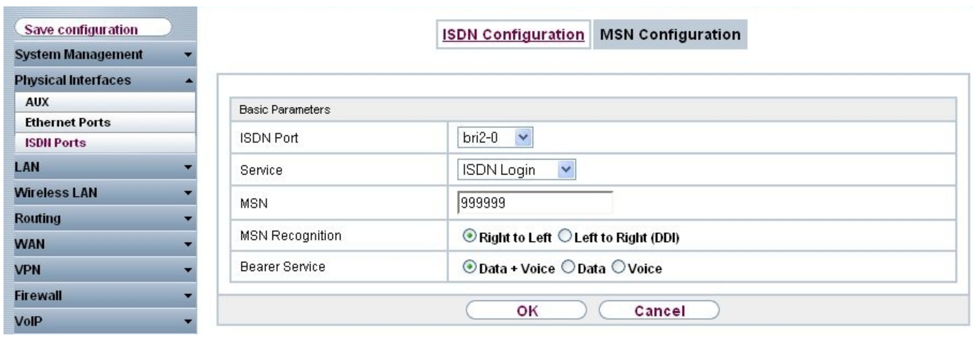

Fig. 115: **Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**

Relevant fields in the MSN Configuration menu

Field	Description
ISDN Port	Select the ISDN port for which the MSN is to be configured.
Service	Select the service to which a call is to be assigned on the MSN.
MSN	Enter any number here (dummy call number).
MSN Recognition	Select the mode your device is to use for the number comparison of **MSN** with the called party number of the incoming call. Possible values: • `Right to Left` (default value) • `Left to Right (DDI)`: If your device is connected to a point-to-point connection.
Service attribute	Select the type of incoming call.

Remote CAPI server configuration

The remote CAPI server of the **bintec R1200** is already enabled ex-works.

(1) Go to **Local Services** -> **CAPI Server**->**Options**.

Fig. 116: **Local Services** ->**CAPI Server** ->**Options**

Relevant fields in the Options menu

Field	Description
Enable server	The function is activated by selecting `Enabled`. The function is enabled by default.
CAPI Server TCP Port	The field can only be edited if **Enable Server** is enabled. Enter the TCP port number for remote CAPI connections. The default value is `2662`.

For security reasons, access to the remote CAPI interface should be protected with a user name and password.

Note

Ex works, a user with the user name `default` and no password is always entered for the CAPI subsystem. All calls to the CAPI are offered to all CAPI applications in the LAN. Use the Settings menu to distribute incoming calls for the CAPI subsystem to defined users with password. You should then delete the user `default` without password.

(1) Go to **Local Services** -> **CAPI Server**->**User**->**New**.

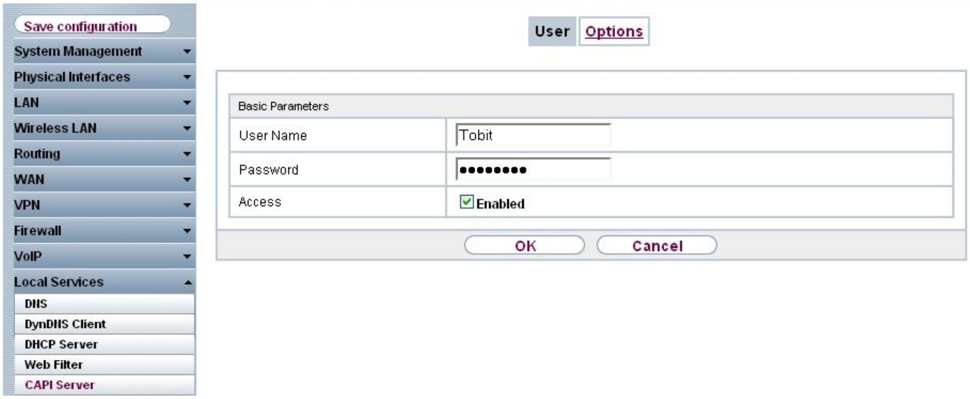

Fig. 117: **Local Services** ->**CAPI Server** ->**User**-> **New**.

Relevant fields in the User menu

Field	Description
User Name	Enter the user name for which access to the CAPI service is to be allowed.
Password	Enter the password which the user shall use for identification to gain access to the CAPI service.
Access	Select whether access to the CAPI service is to be permitted or denied for the user. The function is activated by selecting `Enabled`.

8.2.2 Configuration of remote CAPI client software

The remote CAPI client software is a component of the **BRICKware** software package. The latter is located on the provided companion CD, or can be accessed in the download area at *www.bintec-elmeg.com* . The remote CAPI client software is installed in the **BRICKware** program group.

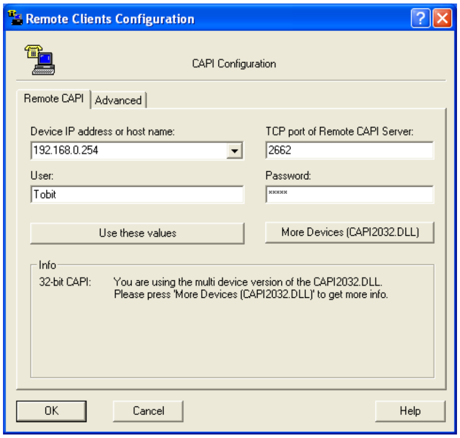

Fig. 118: Remote Clients Configuration

(1) To log in the remote CAPI client, the **Device IP address or host name** of the **bintec R1200**must be saved.

(2) Under **User** enter `Tobit`, for example.

(3) Enter the **password**.

(4) Apply the configuration with **Use these values**.

(5) As confirmation, a corresponding message appears in the info area of the remote CAPI client software.

Detailed in information about the configured CAPI servers and their CAPI controllers is provided under **Remote Multi CAPI Client Configuration**.

Following login of the remote CAPI client software bintec router, which functions as CAPI server, one CAPI controller is displayed per ISDN interface.

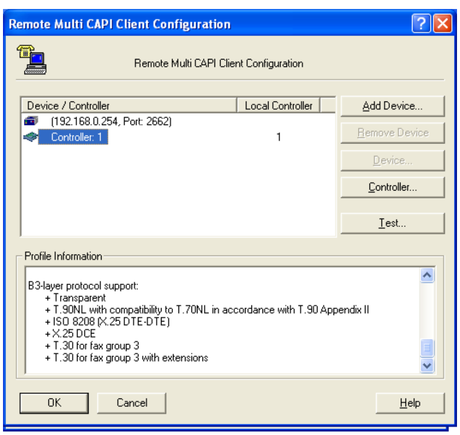

Fig. 119: Remote Multi CAPI Client Configuration

8.2.3 CAPI port configuration for the Remote CAPI interface

With basic installation of the **Tobit David**, the **Tobit Software** -> **David** program group was installed on your server. There you will find the **David Administrator** for configuration of the David. In this section, the dialog for configuration of the ISDN hardware is launched via the option **Ports** -> **add Port**. Follow the administrator's instructions.

(1) Go to **David** -> **System** -> **Ports**.

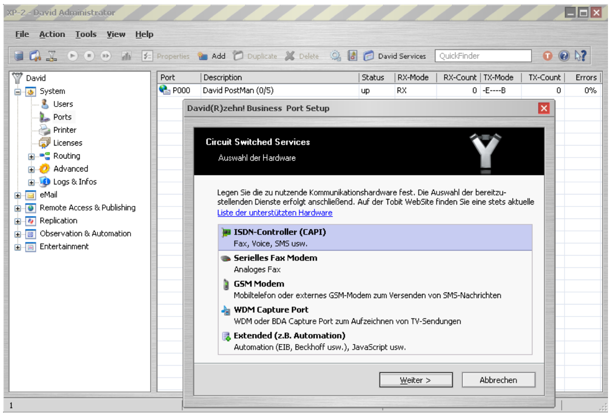

Fig. 120: XP-2 - David Administrator

With **Hardware Detection**, **Port Setup** locates the remote CAPI controller.

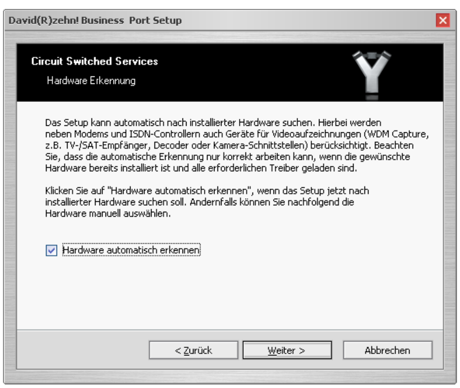

Fig. 121: Hardware detection

(1) Enable `Autodetect Hardware`.

The next **Port Setup** step allows selection of services assigned to this CAPI port.

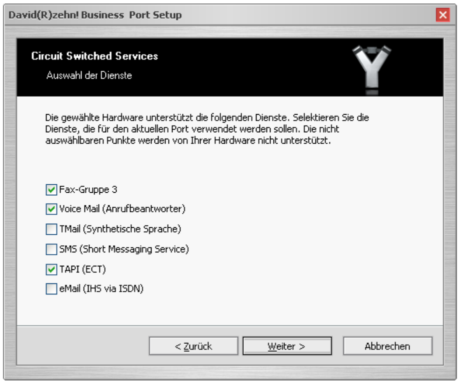

Fig. 122: Selection of services

(1) For the remote CAPI port of the **bintec R1200**, select the services *`Fax group 3`*, *`Voice Mail (answering machine)`* and *`TAPI (ECT)`*.

In the next step, a unique name is assigned to the port.

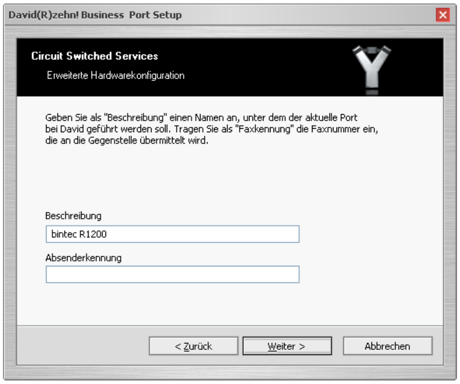

Fig. 123: Advanced hardware configuration

(1) Under **Description** enter *`bintec R1200`*, for example.

The **Operation Mode** *`send and receive (TX/RX)`* as well as the option *`ISDN point-to-multipoint connection`* may be transferred without modification.

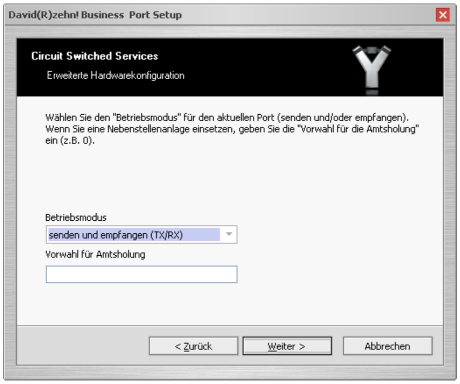

Fig. 124: Advanced hardware configuration

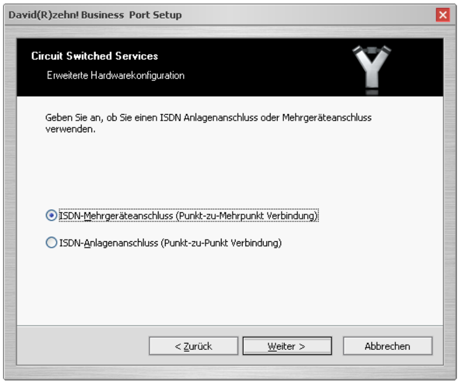

Fig. 125: Advanced hardware configuration

In the final step of **Port Setup**, an ISDN Multi Subscriber Number (MSN) is assigned to the port. This call number is used for incoming connections after completed configuration.

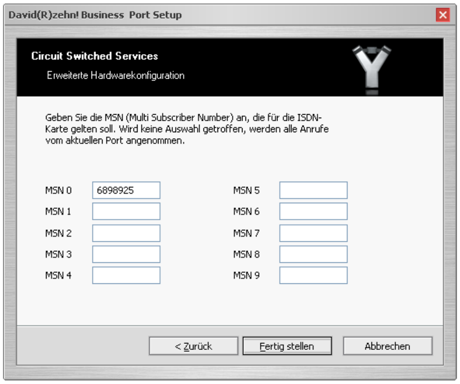

Fig. 126: Advanced hardware configuration

(1) Under **MSN 0** enter `6898925`, for example.

(2) This concludes the configuration. Click on **Complete**.

To be able to use both B-channels of the ISDN point-to-multipoint connection, an additional port must be created. In this example, the existing port is duplicated.

(1) Go to **David** -> **Ports**-> **Duplicate...**.

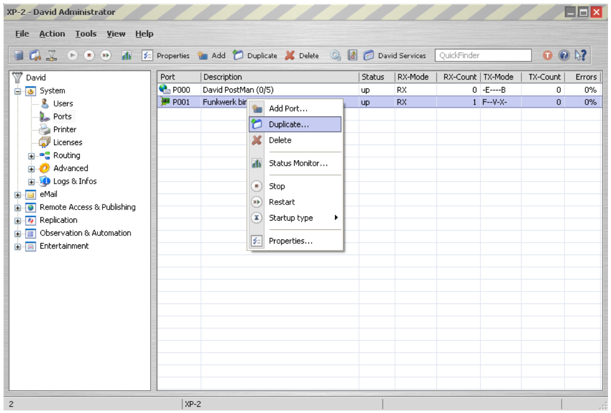

Fig. 127: XP-2 - David Administrator

In the properties of the configured port you can, among other things, set which user receives an incoming fax message.

(1) Go to **PORT 011 - bintec R1200** -> **Advanced**

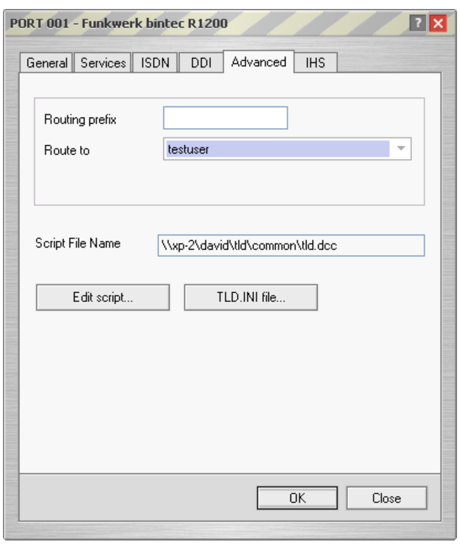

Fig. 128: PORT 001 - bintec R1200

(1) Under **Route to** select `testuser`, for example.

You can now send a test fax in the **David InfoCenter**.

Enter the recipient's fax number in the address field, and create a message.

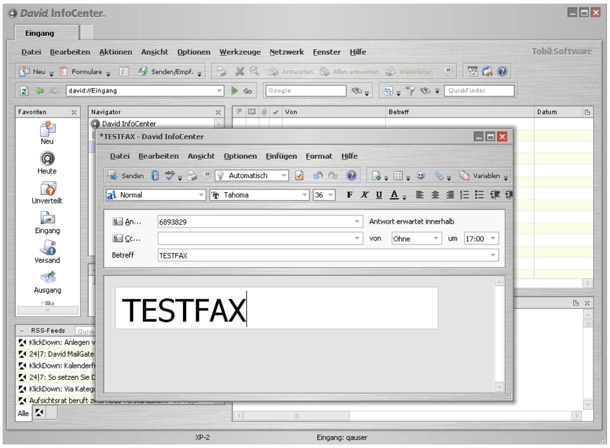

Fig. 129: Test fax

Further configuration of the David by Tobit will not be addressed here. For this, we refer you to our Technology Partner, Tobit Software.

8.3 Overview of configuration steps

MSN Configuration

Field	Menu	Value
ISDN Port	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*bri2-0*
Service	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*ISDN Login*
MSN	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	e.g. *999999*
MSN Recognition	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*Right to Left*
Service attribute	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*Data + Voice*

Remote CAPI server configuration

Field	Menu	Value
Enable server	**Local Services** ->**CAPI Server** ->**Options**	*Aktiviert*
CAPI Server TCP Port	**Local Services** ->**CAPI Server** ->**Options**	e.g. *2662*
User Name	**Local Services** ->**CAPI Server** ->**User**-> **New**.	e.g. *Tobit*
Password	**Local Services** ->**CAPI Server** ->**User**-> **New**.	*Password*
Access	**Local Services** ->**CAPI Server** ->**User**-> **New**.	*Aktiviert*

Configuration of remote CAPI client software

Field	Menu	Value
Device IP address or host name	**Remote Clients Configuration**	e.g. *192.168.0.254*
User	**Remote Clients Configuration**	*Tobit*
Password	**Remote Clients Configuration**	*Password*

Hardware detection

Field	Menu	Value
Detect hardware automatically	**Port Setup**	enable
Fax group 3	**Port Setup**	enable
Voice Mail (answering machine)	**Port Setup**	enable
TAPI (ECT)	**Port Setup**	enable
Description	**Port Setup**	e.g. *bintec R1200*
Operation Mode	**Port Setup**	*sending and receiving (TX/RX)*
ISDN point-to-multipoint connection	**Port Setup**	enable
MSN0	**Port Setup**	e.g. *6898925*

Duplicate port

Field	Menu	Value
Duplicate...	**David** -> **System** -> **Ports**	enable

Field	Menu	Value
Route to	**bintec R1200** -> **Advanced**	e.g. *test user*

Chapter 9 Media Gateway - Connecting a virtualised Tobit David.fx server to a Primary Rate Interface with a bintec RT4402

9.1 Introduction

This chapter describes how to connect, running in a virtual environment, Tobit **David.fx** 2011 servers to a 30-channel Primary Rate interface (point-to-point connection) with a 3-digit direct dialling range. The **David.fx** server uses the remote CAPI interface in the **bintec RT4402** gateway to communicate.

The **GUI** (Graphical User Interface) is used here to configure the **bintec RT4402**.

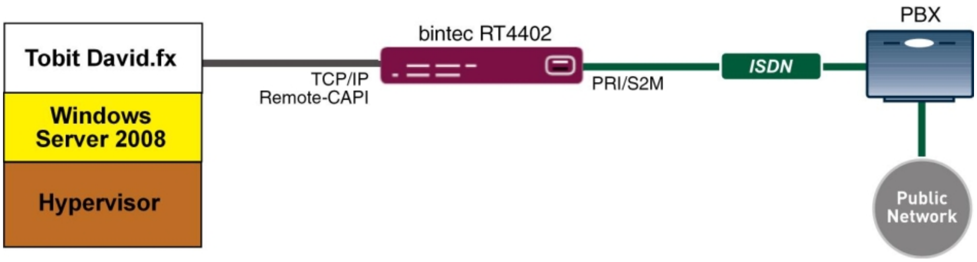

Fig. 130: Example scenario

Requirements

- A **bintec RT4402** gateway with system software 7.10.1
- A fax licence for the **bintec RT4402** gateway
- Basic installation of the Tobit **David.fx** server is assumed, along with an existing VMware environment.
- A Primary Rate interface

9.2 Configuration

9.2.1 Configuring the bintec RT4402

Activate the fax licence

With the **bintec RT4402**, the T.30 protocol for G3 fax needs to be activated by a fax licence. Once this licence has been authorised in the Service area on our website *www.bintec-elmeg.com* , the licence can be entered in the web interface on the **bintec RT4402**.

Note

The fax licence must be activated by rebooting the **bintec RT4402** gateway.

(1) Go to **System Administration** -> **Global Settings** -> **System Licences**.

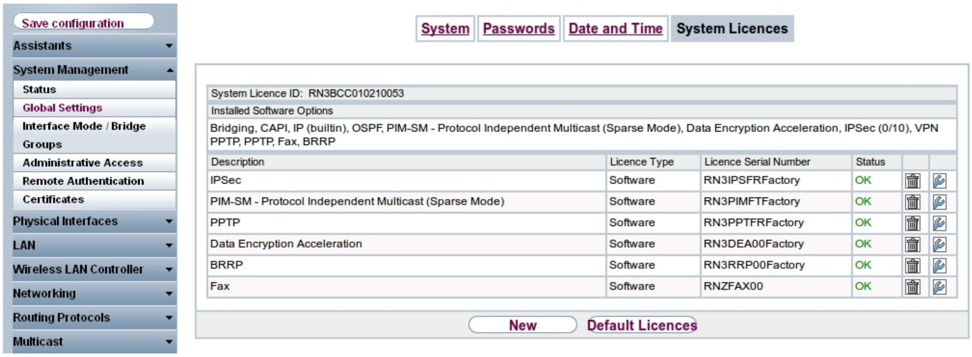

Fig. 131: **System Management** -> **Global Settings** -> **System Licences**

Configure the ISDN Primary Rate interface

The **bintec RT4402** gateway can be connected to a provider's Primary Rate interface or to an internal S2M bus in a telephone system which is wired with the DSS1 protocol.

(1) Go to **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<pri2-4 (TE)>**

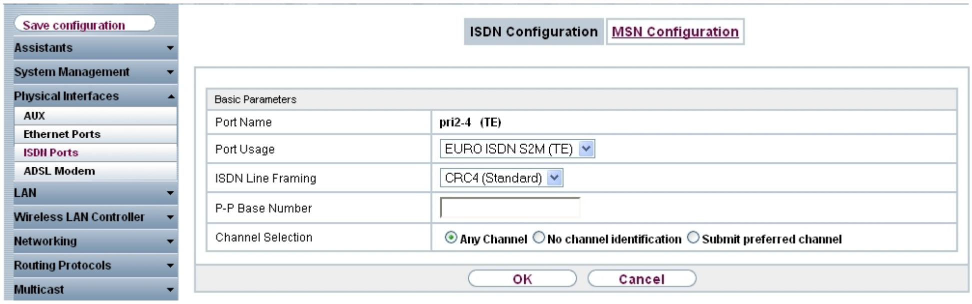

Fig. 132: **Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<pri2-4 (TE)>**

Proceed as follows to edit the ISDN port configuration:

(1) For **Port Usage**, select *EURO ISDN S2M (TE)*.

(2) Leave the **ISDN Line Framing** set to *CRC4 (Standard)*.

(3) Under **Channel Selection**, select *Any Channel*. The device tells the PABX that all channels are available. The exchange of the PABX selects the channel to be used.

(4) Confirm with **OK**.

Ex works, the router accepts all incoming ISDN connections, thus permitting remote configuration via ISDN login. This needs to be prevented, for security reasons. To do this, go to the following menu:

(1) Go to **Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**.

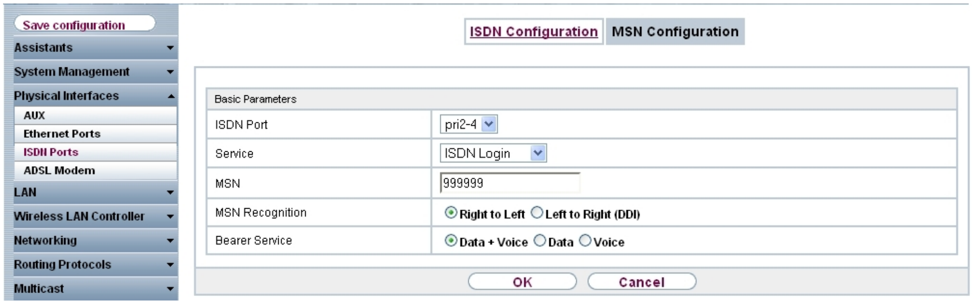

Fig. 133: **Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**

Proceed as follows:

(1) Select the **ISDN port** for which the MSN is to be configured, e. g. *pri-4*.

(2) For **Service**, select *ISDN Login*. This enables logins with ISDN login.

(3) For **MSN**, enter the number which is used to check the called party number, e. g. *999999*.

(4) For **MSN Recognition**, select the mode your device is to use to do the numbers com-

parison for MSN with the called party number of the incoming call, here `Right to Left`.

(5) For **Service attribute**, select the type of the incoming call (service recognition), here e. g. `Data + voice`.

(6) Confirm with **OK**.

Remote CAPI server configuration

The CAPI service allows connection of incoming and outgoing data and voice calls to communications applications on hosts in the LAN that access the Remote CAPI interface of your device. This enables hosts connected to your device to receive and send faxes.

The remote CAPI server of the **bintec RT4402** is already enabled ex-works.

(1) Go to **Local Services** -> **CAPI Server**->**Options**.

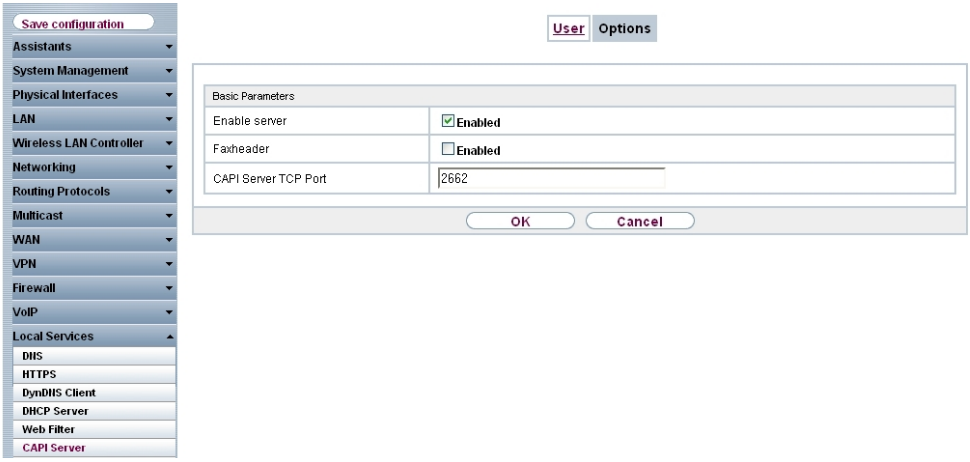

Fig. 134: **Local Services** ->**CAPI Server** ->**Options**

For security reasons, access to the remote CAPI interface should still be protected with a user name and password.

You use the symbol to edit the existing user "default".

(1) Go to **Local Services** -> **CAPI Server** -> **User** -> **<default>** .

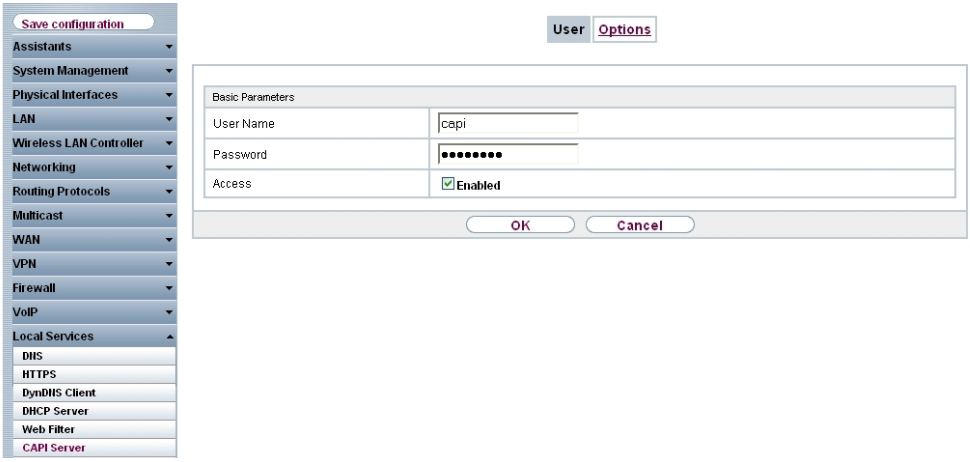

Fig. 135: **Local Services** -> **CAPI Server** -> **User** -> **<default>**

Proceed as follows to protect the CAPI interface:

(1) For **Username**, enter the name for which access to the CAPI service is to be allowed or denied, e. g. `capi`.

(2) Enter the **Password** that the user will use to identify themselves to gain access to the CAPI service, e. g. `supersecret`.

(3) Enable the **Access** option. Now the user is permitted to access the CAPI service.

(4) Confirm with **OK**.

9.2.2 Configuration of remote CAPI client software

To install the remote CAPI interface, use the latest installation pack **Remote CAPI for MS-WINDOWS**. You can get this, in both 32-bit and 64-bit versions, from the download area on our website *www.bintec-elmeg.com*.

The LAN CAPI configuration software is installed in the program group **Bintec Brickware**.

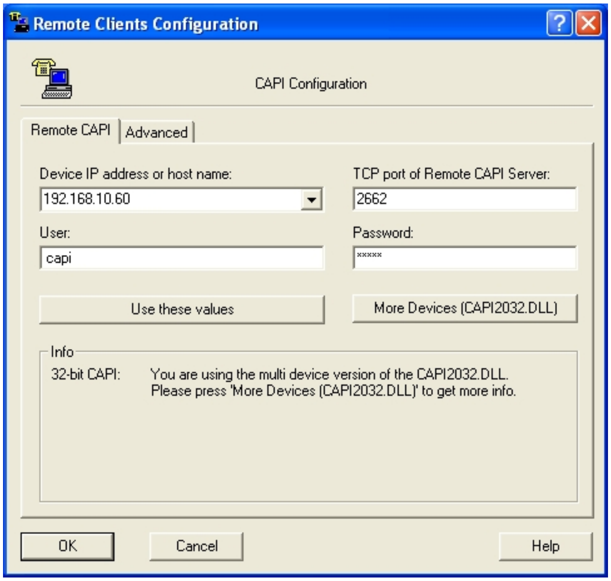

Fig. 136: Remote Clients Configuration

(1) To log in the remote CAPI client, the **IP address or host name** of the **bintec RT4402** gateway must be saved.

(2) Under **User Name** enter e. g. `capi`.

(3) Enter the **Password**, e. g. `supersecret`.

(4) Apply the configuration with **Apply**.

(5) As confirmation, a corresponding message appears in the info area of the remote CAPI client software.

Detailed information about the configured CAPI servers and their CAPI controllers is provided under the **Multiple Devices (CAPI 2032.dll)** option.

In our example, there must be a CAPI controller available with 30 channels.

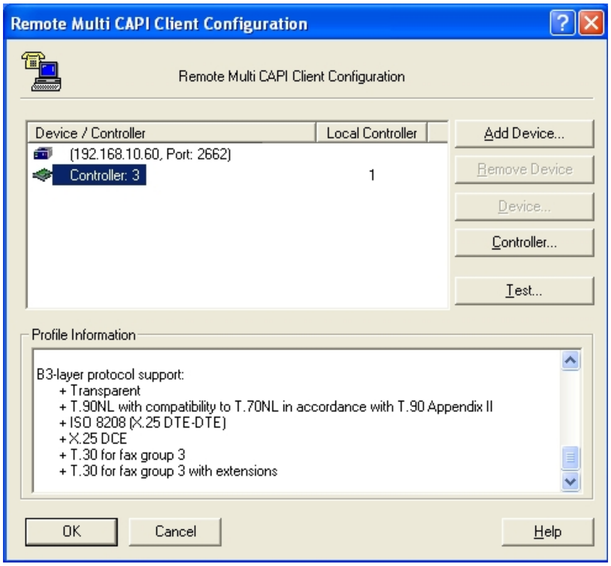

Fig. 137: Remote Multi CAPI Client Configuration

9.2.3 CAPI port configuration for the Remote CAPI interface

When the basic installation of the Tobit David was done, the **Tobit Software** -> **David.fx** program group was created on your server. There you will find the **David Administrator** for configuring the Tobit **David.fx** ports to communicate with the bintec Remote CAPI. In this menu, the dialog for configuring the ISDN hardware is opened via the **Ports** -> **Add Ports** option.

(1) Go to **David** -> **System** -> **Ports**.

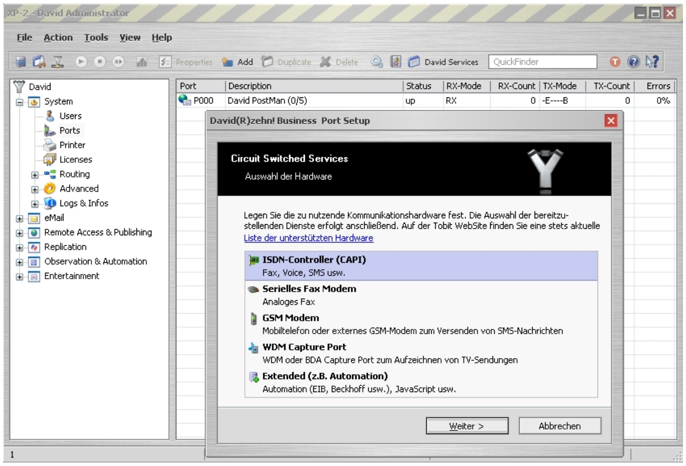

Fig. 138: David.fx Administrator

With **Hardware Detection**, **Port Setup** locates the remote CAPI controller.

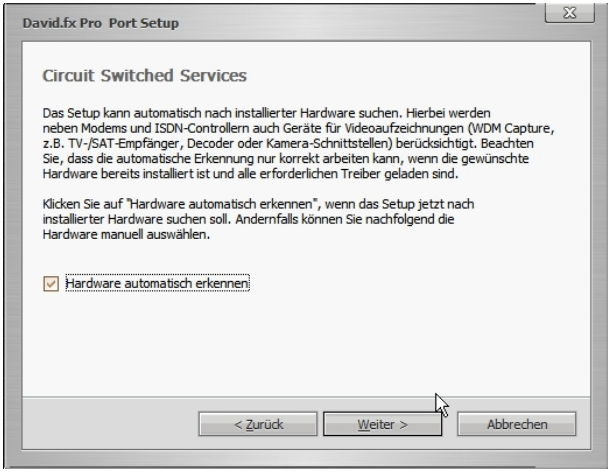

Fig. 139: Hardware detection

(1) Enable `Autodetect Hardware`.

The next **Port Setup** step allows selection of services assigned to this CAPI port.

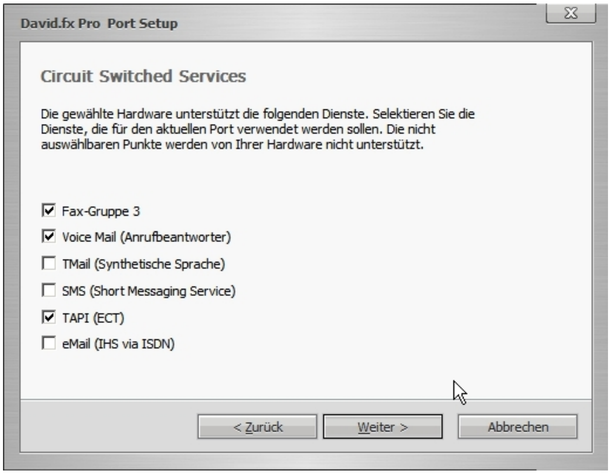

Fig. 140: Selection of services

(1) For the remote CAPI port of the **bintec RT4402**, select the services `Fax group 3`, `Voice Mail (answering machine)` and `TAPI (ECT)`.

The **Operating Mode** is set to `send and receive (TX/RX)` and the **Access Configuration** is set to `ISDN access configuration (point-to-point connection)` due to the Primary Rate interface.

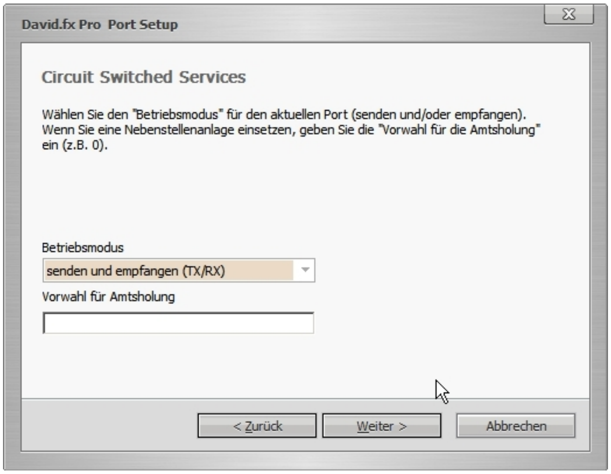

Fig. 141: Advanced hardware configuration

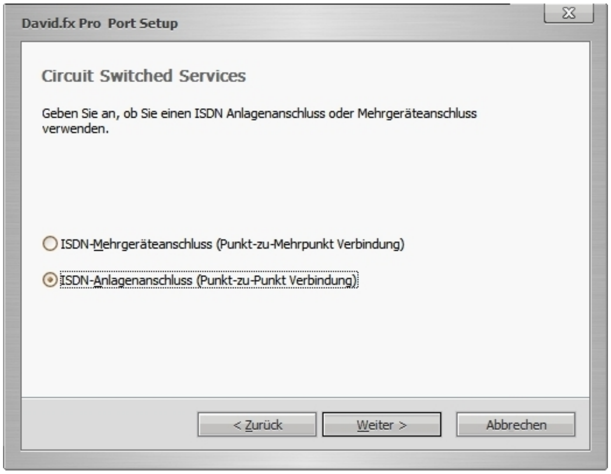

Fig. 142: Advanced hardware configuration

In the final step in the **port setup**, the call number length of your own number including the DDI block (e. g. 8 digits) is specified. This results in incoming connections being accepted immediately, as soon as the target number has reached a length of 8 digits.

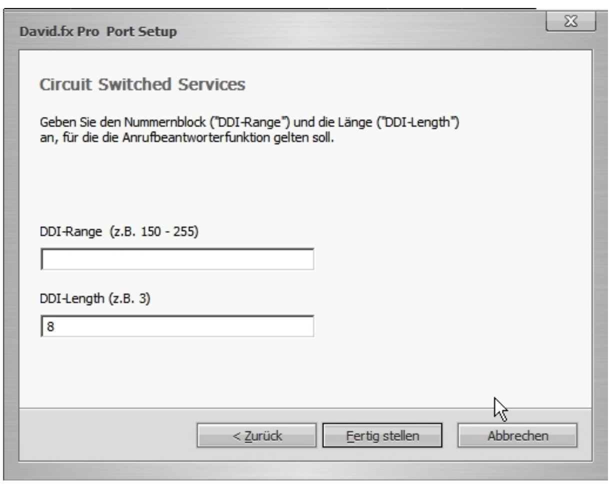

Fig. 143: Advanced hardware configuration

Once the first CAPI port has been created using the wizard, its properties need to be adjusted. Right-click on the CAPI port you have created to be able to edit the port's properties. The `Take call when DDI length reached` option must be enabled in the **DDI** menu.

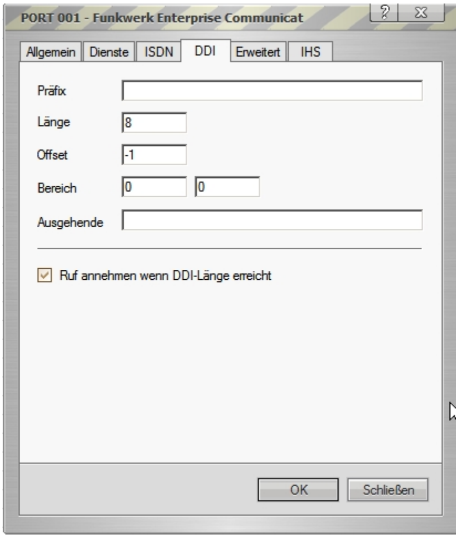

Fig. 144: Advanced hardware configuration

When the wizard has finished creating a new Tobit **David.fx** CAPI port, the installation can be used for incoming and outgoing connections via an ISDN channel. To be able to use all the channels, one more CAPI port needs to be created for each available ISDN channel. In this example, the existing port is duplicated.

(1) Go to **David** -> **Ports**-> **Duplicate...**.

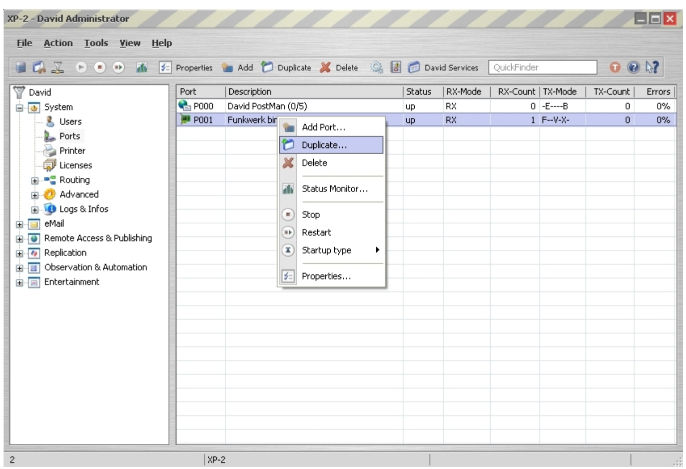

Fig. 145: David Administrator

You can now send a test fax in the **David client**.

Enter the recipient's fax number in the address field, and create a message.

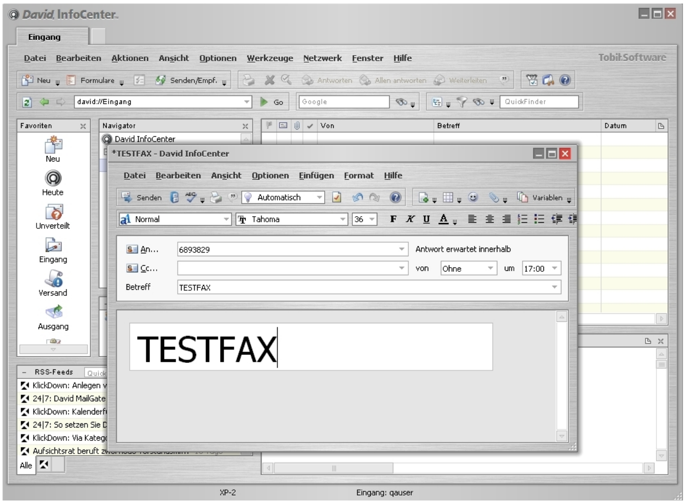

Fig. 146: Test fax

Further configuration of the David by Tobit will not be addressed here. For this, we refer you to our Technology Partner, Tobit Software.

The console or the Telnet or SSH access should be used to debug the **bintec RT4402** fax connections, because the messages cannot be seen in real time here. After logging in, use the `debug isdn modem capi fax` command to enable the outputting of log messages.

```
Bespiel Debug Ausgaben während eines eingehenden Faxes:
rt4402:> debug isdn modem capi fax
11:20:01 DEBUG/CAPI: DBG(000555492) APPL34:33 PLCI 0x0403 dialin from <0911908070> to local number <091196731550>
11:20:02 INFO/CAPI: INF(000556429) APPL34:33 PLCI 0x0403 incoming call accepted
11:20:02 INFO/MODEM: ID:4 Allocate FAX Modem on B-Chan:1 using Timeslot:4
11:20:02 DEBUG/MODEM: ID:4 Open DSP FAX Resource
11:20:02 DEBUG/MODEM: slot 2, unit 4, chan 1: modem connect 64000
11:20:29 DEBUG/ISDN: faxdbg(0519):(41/9/0) dl_disconnect_ind reason: no error
11:20:29 DEBUG/CAPI: DBG(000583445) APPL34:33 PLCI 0x0403 Fax disconnected: 0 no error
11:20:29 DEBUG/MODEM: slot 2, unit 4, chan 1: modem local hangup
11:20:29 DEBUG/ISDN: stack 2: disconnect cause: normal, unspecified (0x9f)

Bespiel Debug Ausgaben während eines ausgehenden Faxes:
rt4402:> debug isdn modem capi fax
11:20:01 DEBUG/CAPI: DBG(000555492) APPL34:33 PLCI 0x0403 dialin from <0911908070> to local number <091196731550>
11:20:02 INFO/CAPI: INF(000556429) APPL34:33 PLCI 0x0403 incoming call accepted
11:20:02 INFO/MODEM: ID:4 Allocate FAX Modem on B-Chan:1 using Timeslot:4
11:20:02 DEBUG/MODEM: ID:4 Open DSP FAX Resource
11:20:02 DEBUG/MODEM: slot 2, unit 4, chan 1: modem connect 64000
11:20:29 DEBUG/ISDN: faxdbg(0519):(41/9/0) dl_disconnect_ind reason: no error
11:20:29 DEBUG/CAPI: DBG(000583445) APPL34:33 PLCI 0x0403 Fax disconnected: 0 no error
11:20:29 DEBUG/MODEM: slot 2, unit 4, chan 1: modem local hangup
11:20:29 DEBUG/ISDN: stack 2: disconnect cause: normal, unspecified (0x9f)
```

9.3 Overview of Configuration Steps

ISDN Configuration

Field	Menu	Value
Port Usage	**Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<pri2-4 (TE)**	*EURO ISDN S3M (TE)*
ISDN Line Framing	**Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<pri2-4 (TE)**	*CRC4 (Standard)*
Channel Selection	**Physical Interfaces** -> **ISDN Ports** -> **ISDN Configuration** -> **<pri2-4 (TE)**	*Any channel*

MSN Configuration

Field	Menu	Value
ISDN Port	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*pri2-4*
Service	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*ISDN Login*
MSN	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	e. g. *999999*
MSN Recognition	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*Right to Left*
Service attribute	**Physical Interfaces** -> **ISDN Ports** -> **MSN Configuration** -> **New**	*Data + Voice*

Remote CAPI server configuration

Field	Menu	Value
Enable server	**Local Services** ->**CAPI Server** ->**Options**	*Enabled*
User Name	**Local Services** ->**CAPI Server** ->**User**-> **New**.	e. g. *capi*.
Password	**Local Services** ->**CAPI Server** ->**User**-> **New**.	e. g. *supersecret*
Access	**Local Services** ->**CAPI Server** ->**User**-> **New**.	*Enabled*

Configuration of remote CAPI client software

Field	Menu	Value
IP address or host name of the device	**Remote Clients Configuration**	e. g. *192.168.10.60*
User Name	**Remote Clients Configuration**	e. g. *capi*.
Password	**Remote Clients Configuration**	e. g. *supersecret*

Hardware detection

Field	Menu	Value
Detect hardware automatically	**Port Setup**	enable
Fax group 3	**Port Setup**	enable
Voice Mail (answering machine)	**Port Setup**	enable
TAPI (ECT)	**Port Setup**	enable
Operation Mode	**Port Setup**	*sending and receiving (TX/RX)*
ISDN access (point-to-multipoint connection)	**Port Setup**	enable
DDI Length	**Port Setup**	e. g. *8*

Duplicate port

Field	Menu	Value
Duplicate...	**David** -> **System** -> **Ports**	enable

Chapter 10 Media Gateway - bintec R1200 VoIP/R4100 VoIP as Unified Messaging Gateway for Microsoft Exchange Server 2007

10.1 Introduction

The present chapter describes connection of the unified messaging roll for Microsoft Exchange Server 2007 to the public telephone network or a PBX with a **bintec R1200** VoIP or **bintec R4100** VoIP media gateway.

The unified messaging roll for Microsoft exchange server 2007 offers the following functions:

- Access to e-mails and voice messages, appointments and contacts by voice control/tone dialling
- Server for fax reception
- Answering machine function with message delivery by e-mail
- Auto Attendant / call relay

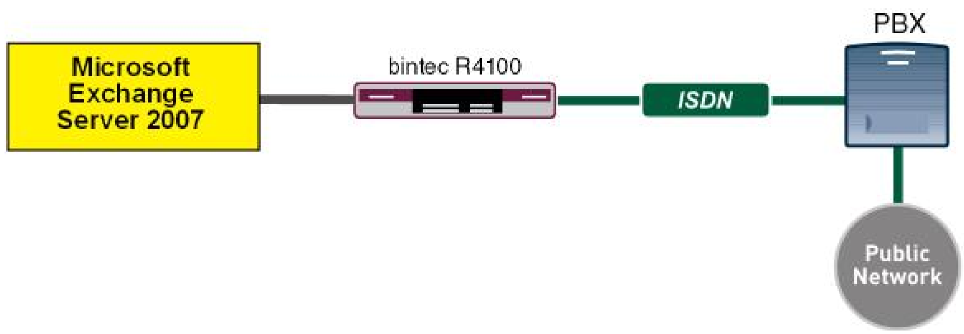

Fig. 147: Example scenario

Requirements

- A **bintec R1200** VoIP or **bintec R4100** VoIP
- Microsoft Exchange Server 2007 with Unified Messaging Roll
- Access to the public telephone network or a PBX

10.2 Configuration

10.2.1 Configuration steps on Microsoft Exchange server

Configuration of the Microsoft Exchange server is performed with the **exchange administration console** :

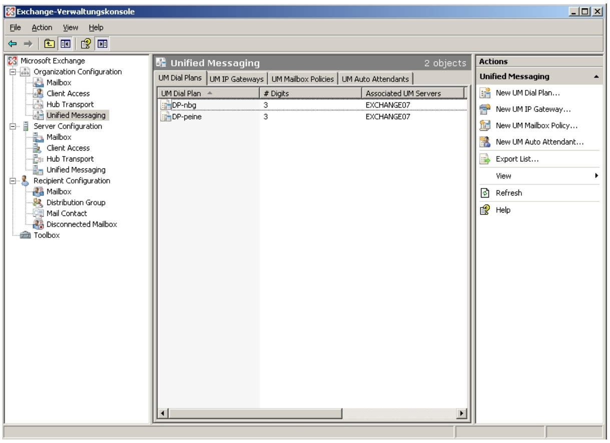

Fig. 148: **Exchange administration console**

Creation of a dial plan

In the **Unified Messaging** menu, you can launch the wizard to create a new UM dial plan.

(1) Go to **Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan...**

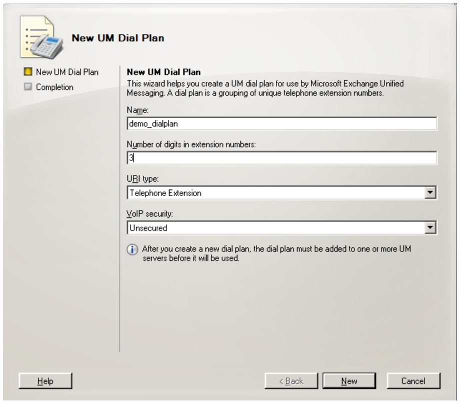

Fig. 149: New UM dial plan

To create a new UM dial plan, proceed as follows:

(1) Enter the dial plan name, e. g. *`demo_dialplan`*.

(2) In **Number of digits in extension numbers** set the number of direct dial-in numbers, e.g., *3*.

(3) In **URI type** select a designation for the resources, e.g. *`Telephone Extension`*.

(4) In **VoIP security** select *`Unsecured`*.

(5) With the option **New**, you create the new dial plan.

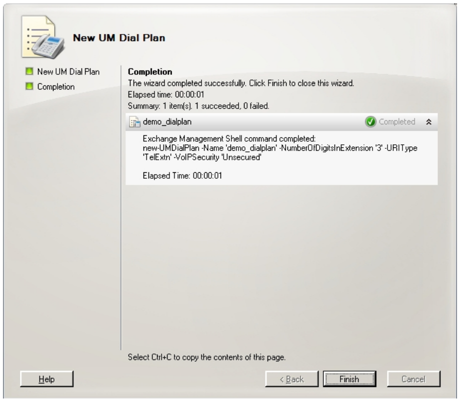

Fig. 150: New UM dial plan

Click on **Finish** to close the wizard.

After the wizard is closed, dial plan properties must be edited.

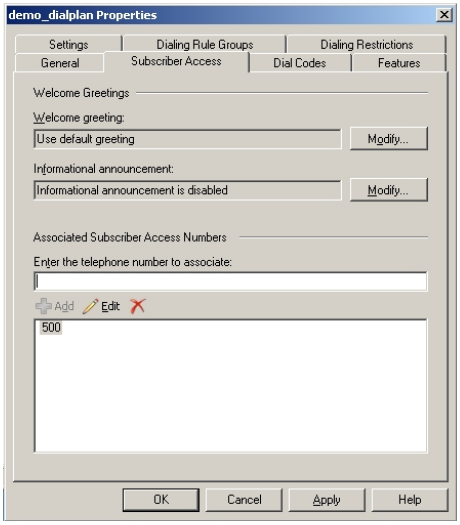

Fig. 151: Subscriber Access

Under **demo_dialplan Properties** -> **Subscriber Access** the call number under which the system may later be reached is saved, e.g., `500`.

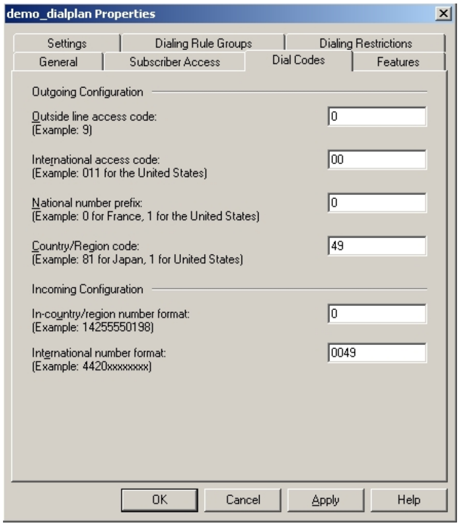

Fig. 152: Dial Codes

Under **demo_dialplan Properties** -> **Dial Codes** national and other prefixes are saved.

To save the prefixes, proceed as follows:

First, enter the numbers for outgoing calls.

(1) In the **Outside line access code**field, you can save a number for an outside line.

(2) In **International access code** enter the international access number *00*.

(3) In **National number prefix** enter the national prefix, here *0*.

(4) In **Country/Region code** enter the country code, e.g., *49* for Germany.

Now enter the numbers for incoming calls.

(1) In **In-country/region number format** enter *0*.

(2) In **International number format** enter the prefix, e.g., *0049* for Germany.

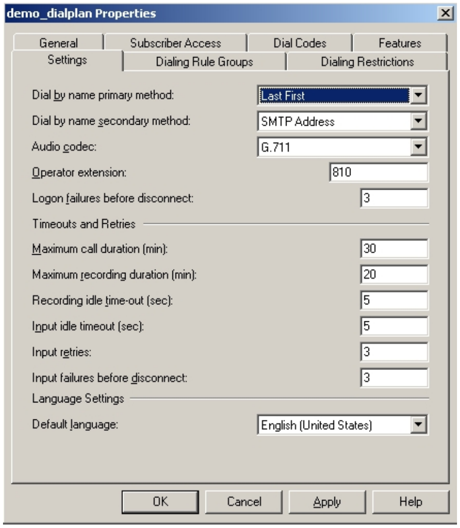

Fig. 153: Settings

In the **Settings** submenu, notably the language codecs and the language with which the system shall respond are saved.

To save additional settings, proceed as follows:

(1) In **Dial by name primary method** select, for example, *Last First*.

(2) In **Dial by name secondary method** select *SMTP Address*.

(3) In **Audio codec** enter language codec *G.711*.

(4) For **Operator extension** enter the switchboard number, for example, *810*.

(5) In **Default language** select the language in which the system shall subsequently answer, e.g., *English (United States)*.

In the submenu **Dialing Rule Groups** a UM dial plan is defined. This determines which type of calls the UM-enabled user can make. In our example, national and international connections are permitted. **Dialing Rule Groups** also allow transformation of destination numbers (e.g. setting of a specific prefix).

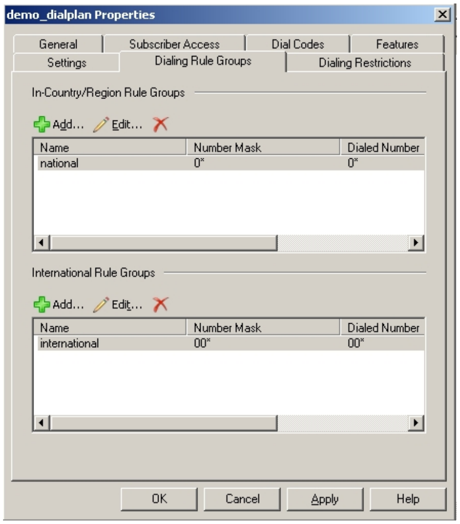

Fig. 154: Dialing Rule Groups

In the submenu **Dialing Restrictions**, it is determined which kinds of calls are permitted or, as the case arises, prohibited.

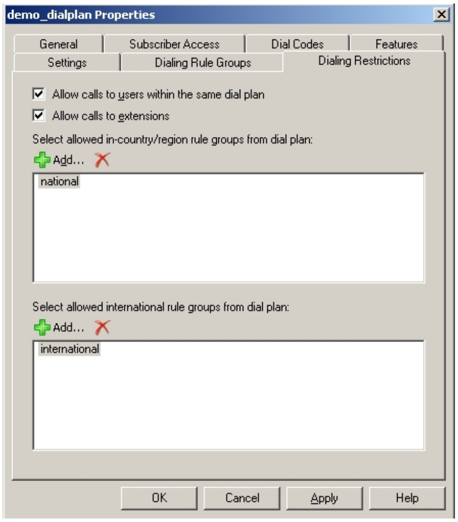

Fig. 155: Dialing Restrictions

The newly-created dial plan is subsequently allocated to a UM server. The dial plan can be added in Server Properties **UM Settings**. Here are administered the installed language packs and the restriction on the maximum possible number of voice and fax connections.

(1) Go to **Server Configuration** -> **Unified Messaging** -> **UM Settings**.

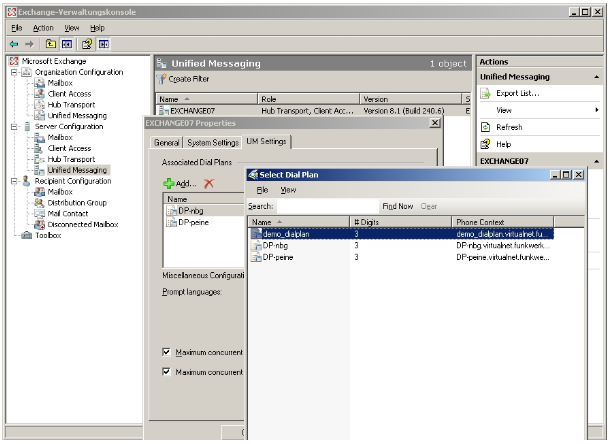

Fig. 156: UM Settings

Creation of a UM IP Gateway

A new UM IP gateway is created with the assistant in the **Unified Messaging** submenu.

(1) Go to **Organization Configuration** -> **Unified Messaging** -> **New UM IP Gateway**.

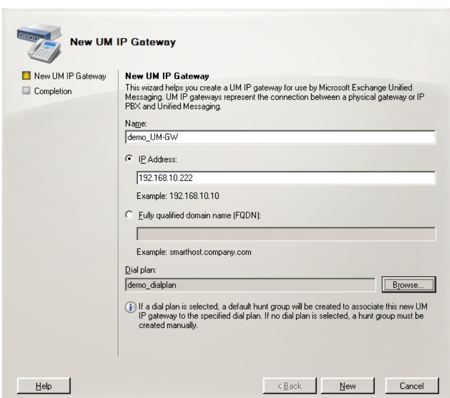

Fig. 157: New UM IP gateway

To create a new UM IP gateway, proceed as follows:

(1) In **Name** enter, for example, `demo_UM-GW`.

(2) Enter the IP address at which the UM gateway is accessible, e.g. `192.168.10.222`.

(3) In **Fully qualified domain name (FQDN)** you can enter the name under which the UM gateway is accessible.

(4) Next, the previously-created **Dial Plan** is assigned.

Creation of a UM hunt group

The **Hunt Groups** are required for drive of the exchange server by the UM gateway . The assistant for creation of a new UM hunt group is launched on the **exchange administration console**.

(1) Go to **Organization Configuration** -> **Unified Messaging** -> **New UM Hunt Group**.

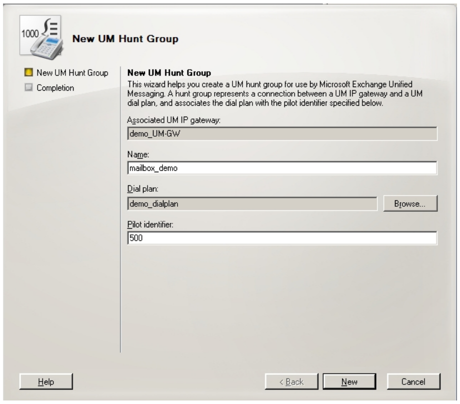

Fig. 158: New UM Hunt Group

To create a new UM hunt group, proceed as follows:

(1) In **Name** enter the name of the hunt group, e.g., *`mailbox_demo`*.

(2) In **Dial plan** select *`demo_dialplan`*.

(3) The number of the **Pilot identifier**, here *`500`*, for example, is later saved at the UM gateway as a VoIP extension in order to create a connection to the Exchange Server 2007.

You can view the completed configuration in the menu **Organization Configuration** -> **Unified Messaging** -> **UM IP Gateways**.

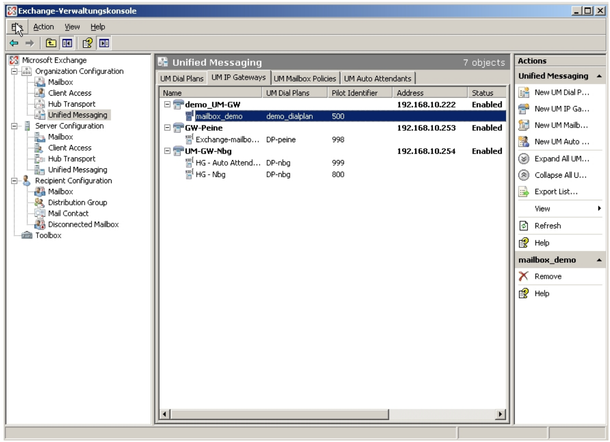

Fig. 159: UM IP Gateways

Configuration of a UM Mailbox Policy

Already when creating a **Dial Plan** a standard **UM Mailbox Policy** is created.

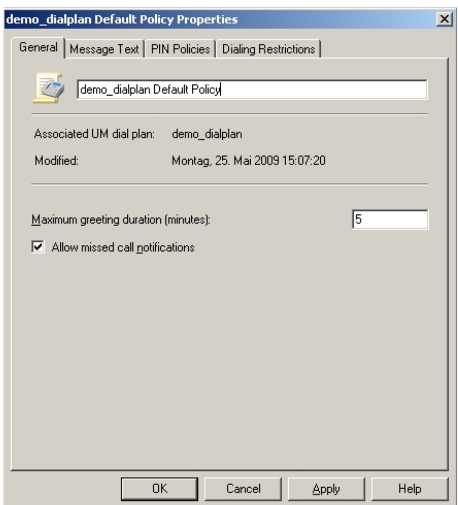

Fig. 160: Default Policy Properties

In properties of **UM Mailbox Policy**, in the **Message Text** submenu, various text templates can be saved; these can be sent to the UM user per e-mail (e.g., when activating the unified messaging mailbox or when resetting the unified messaging PIN).

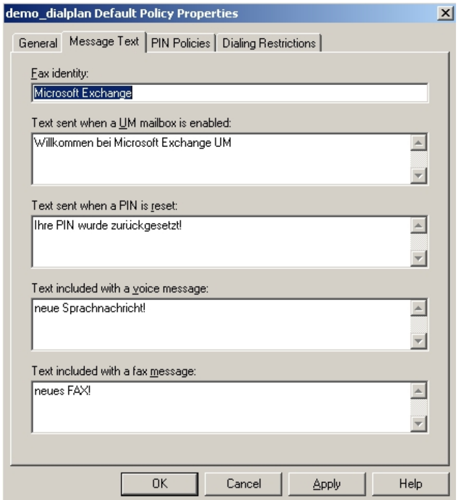

Fig. 161: Message Text

In the submenu **PIN Policies**, different properties of the UM PIN (e.g., PIN length) requested when accessing the UM system can be modified.

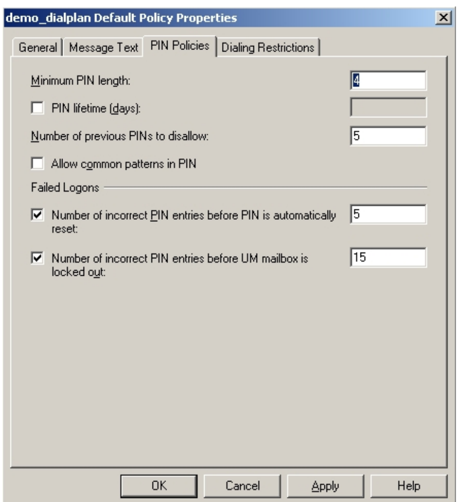

Fig. 162: PIN Policies

In the submenu **Dialing Restrictions**, it is determined which kinds of calls are permitted or, as the case arises, prohibited.

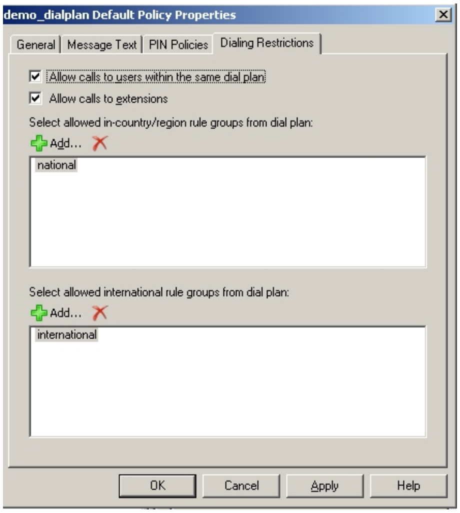

Fig. 163: Dialing Restrictions

Auto Attendants (optional)

Configuration of an **Auto Attendant**, a type of electronic telephone switchboard, is optional. For the **Auto Attendant** an additional **Hunt Group** should be created, under whose **Pilot Identifier** (extension number) the electronic switchboard position can be reached.

Activation of unified messaging for an exchange mailbox

In the **Mailbox** submenu, the unified messaging functions for an exchange mailbox/exchange user can be activated via an assistant. For this, the previously configured **Unified Messaging Mailbox Policy** must be saved, along with a **PIN** (for authentication).

(1) Go to **Organization Configuration** -> **Recipient Configuration** -> **Mailbox**.

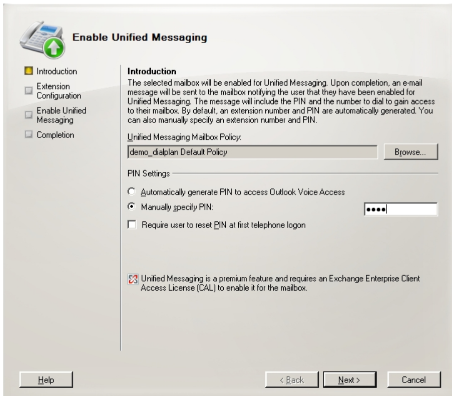

Fig. 164: Mailbox

In the assistant's second step, a **Mailbox Extension** (mailbox number) for the user must be saved. The **Mailbox Extension** should match the user's direct dial-in number.

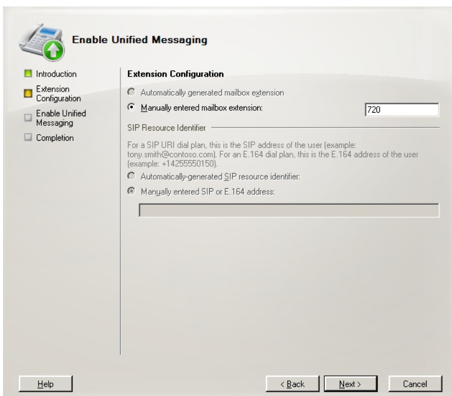

Fig. 165: Mailbox Extension

10.2.2 Configuration of the bintec media gateway

In our example, the bintec media gateway is connected to a PBX internal ISDN port with extension number 500 via the external ISDN S0 interface (e.g. ISDN-0). The ISDN auto-recognition provides detection of a point-to-point or point-to-multipoint connector.

(1) Go to **Physical Interfaces** ->**ISDN Ports**->**<bri2-0 (TE)>** .

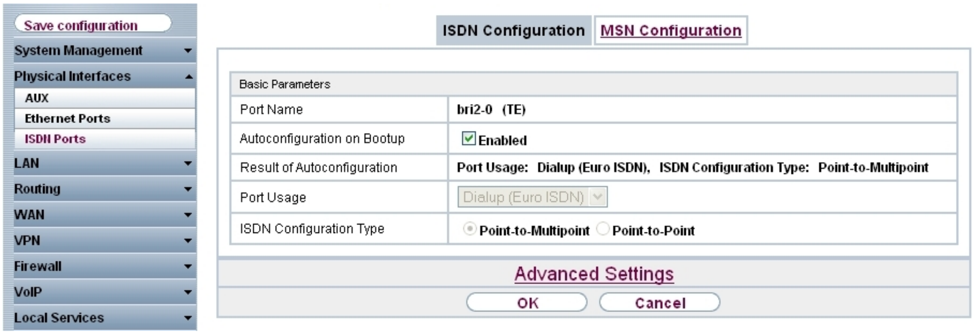

Fig. 166: **Physical Interfaces** -> **ISDN Ports** -> **<bri2-0 (TE)>** .

Relevant fields in the ISDN Configuration menu

Field	Meaning
Port Name	Shows the name of the ISDN port.
Autoconfiguration on Bootup	Autoconfiguration at bootup is enabled.
Result of Autoconfiguration	Shows the status of the ISDN Auto Config.

Connection of the exchange server as VoIP/SIP subscriber

Microsoft Exchange Server 2007 is configured at the media gateway as SIP extension.

(1) Go to **VoIP** -> **Media Gateway** -> **Extensions** -> **New**.

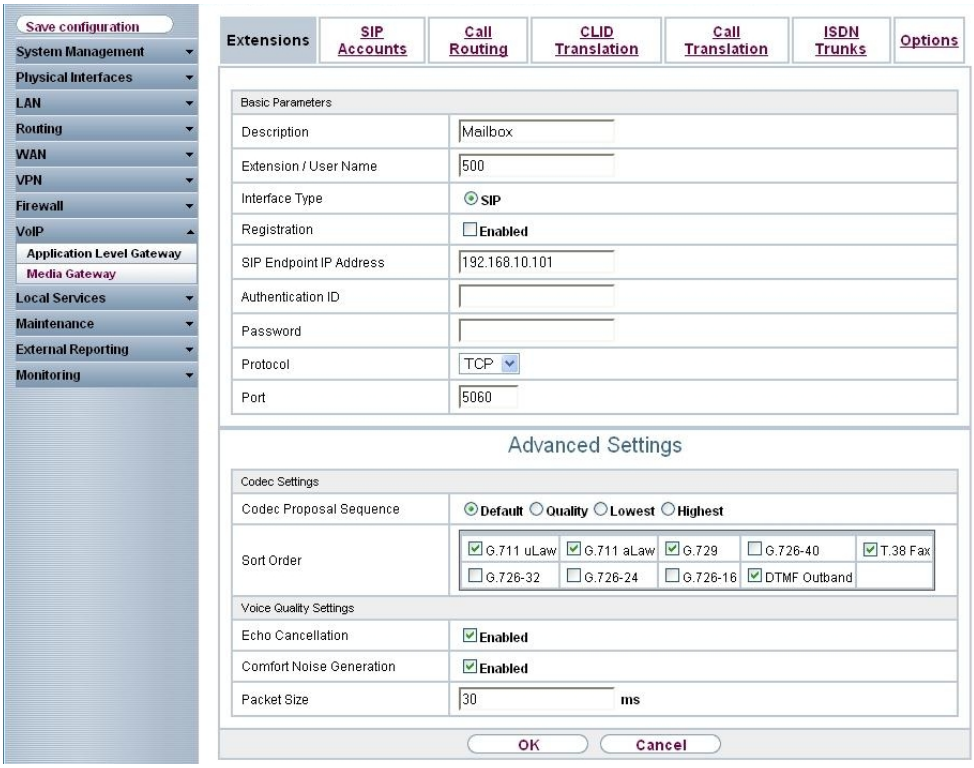

Fig. 167: **VoIP** -> **Media Gateway** -> **Subscriber** -> **New**

Relevant fields in the Extensions menu

Field	Meaning
Description	Enter the name of the terminal, e.g. *`Mailbox`*.
Extension / User Name	Here the number under which the system can be accessed is saved, in this case *`500`*.
Registration	Disable the registration mechanism.
SIP Endpoint IP Address	Here, the IP address of the Microsoft exchange server must be saved, e.g. *`192.168.10.101`*.
Protocol	Select protocol *`TCP`* to be used for data transmission.
Port	For connection to the Microsoft exchange server identify port *`5065`*.

The **Advanced Settings** menu consists of the following fields:

Relevant fields in the menu Advanced Settings

Field	Meaning
Sort Order	Enable options `DTMF Outband` and `T.38 Fax`.

Call routing configuration at the media gateway

To allow outgoing connections toward the PBX/PSTN, a route in menu **Call Routing** must be created. With this routing entry, all calls are routed to the ISDN PBX via the ISDN interface.

(1) Go to **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**.

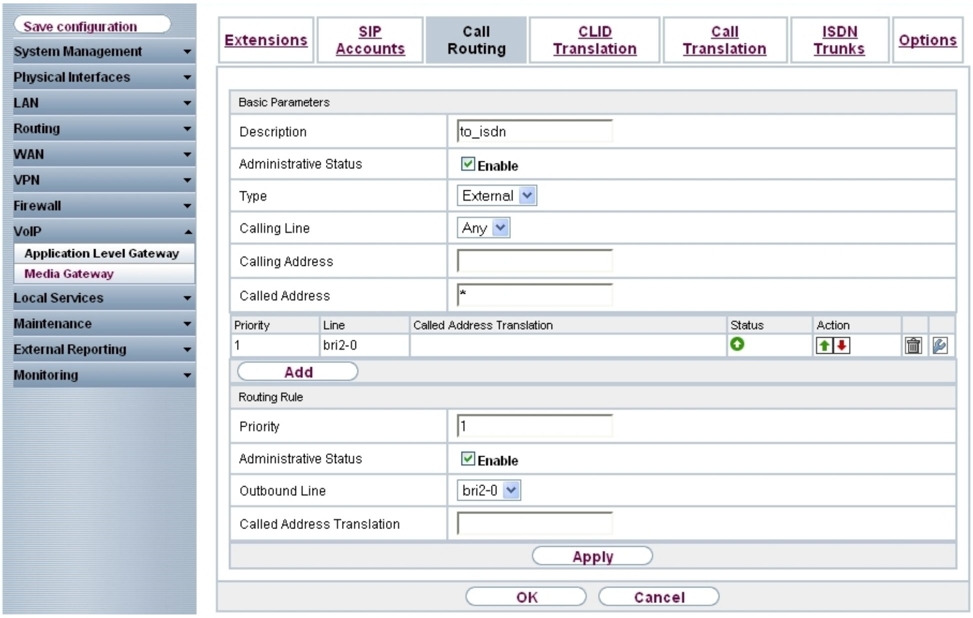

Fig. 168: **VoIP** -> **Media Gateway** -> **Call Routing** -> **New**

Relevant fields in the Call Routing menu

Field	Meaning
Description	Enter the name of the call routing entry, e.g. `to_isdn`.
Type	Specify how calls are to be routed. With `External` for calls that are to be routed as outgoing external calls.
Calling Line	You can restrict the routing entry to the line on which the call comes in. With `Any` there is no restriction of routing entries.
Called Address	You can enter an address numerically (e.g. a subscriber number) or alphanumerically (e.g. for a trunk) that is to be compared with a dialled address. You can use wildcards here.

Field	Meaning
	* means that at the end of a character string any number of characters may follow,

In the **Routing Rules** area, you can define the line/provider over which outgoing connections are routed

10.2.3 Function test

At the first function test, it is possible to call from the telephone extension of the unified messaging user (e.g., demo user *John Everyman* with extension number *720*) to the extension of the exchange server (e.g., extension *500*). Microsoft Exchange server 2007 should respond with a PIN request and permit access to e-mails, contacts, etc.

At the second function test, a unified messaging user (e.g., demo user *John Everyman* with extension number *720*) should configure a call diversion to the Microsoft Exchange extension (call number *500*). With an incoming call to the user call number, the call/fax is put through to the user mailbox on the Microsoft Exchange server.

10.3 Overview of configuration steps

Creation of a dial plan

Field	Menu	Value
Name	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**	e.g. *demo_dailplan*
Number of digits in extension numbers	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**	e.g. *3*
URI type	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**	*Telephone Exten-sion*
VoIP security	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**	*Unsecured*
Subscriber Access	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Subscriber Access**	e.g. *500*
Outside line access code	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Dial Codes**	*0*
International access code	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Dial Codes**	*00*
National number prefix	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Dial Codes**	*0*
Country/Region code	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Dial Codes**	*49*
In-country/region number format	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Dial Codes**	*0*
International number format	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Dial Codes**	*0049*
Dial by name primary method	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Settings**	e.g. *Last First*
Dial by name sec-ondary method	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**->	*SMTP Address*

Field	Menu	Value
	Settings	
Audio codec	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Settings**	*G.711*
Operator extension	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Settings**	e.g. *810*
Logon failures before disconnect	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Settings**	e.g. *3*
Default language	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Settings**	e.g. *English (United States)*
In-Country/Region Rule Groups	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Dialing Rule Groups**	*national*, *0**, *0**
International Rule Groups	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Dialing Rule Groups**	*international*, *00**, *00**
Allow calls to uses within the same dial plan	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Dialing Restrictions**	*Aktiviert*
Allow calls to extensions	**Organization Configuration** -> **Unified Messaging** -> **New UM Dial Plan..**-> **Dialing Restrictions**	Aktiviert

Creation of a UM IP Gateway

Field	Menu	Value
Name	**Organization Configuration** -> **Unified Messaging** -> **New UM IP Gateway**	e.g. *demo_UM-GW*
IP Address	**Organization Configuration** -> **Unified Messaging** -> **New UM IP Gateway**	e.g. *192.168.10.222*
Dial plan	**Organization Configuration** -> **Unified Messaging** -> **New UM IP Gateway**	*demo_dialplan*

Creation of a UM hunt group

Field	Menu	Value
Associated UM IP gateway	**Organization Configuration** -> **Unified Messaging** -> **New UM Hunt Group**	e.g. *demo_UM-GW*

Field	Menu	Value
Name	**Organization Configuration** -> **Unified Messaging** -> **New UM Hunt Group**	e.g. *mailbox_demo*
Dial plan	**Organization Configuration** -> **Unified Messaging** -> **New UM Hunt Group**	e.g. *demo_dialplan*
Pilot identifier	**Organization Configuration** -> **Unified Messaging** -> **New UM Hunt Group**	e.g. *500*

Configuration of a UM Mailbox Policy

Field	Menu	Value
Fax identity	**Organization Configuration** -> **Unified Messaging** -> **New UM Mailbox Policy** -> **Message Text**	*Microsoft Exchange*
Text send when a UM mailbox is enabled	**Organization Configuration** -> **Unified Messaging** -> **New UM Mailbox Policy** -> **Message Text**	e.g. *Welcome to Microsoft Exchange UM*
Text send when a PIN is reset	**Organization Configuration** -> **Unified Messaging** -> **New UM Mailbox Policy** -> **Message Text**	e.g. *Your PIN has been reset!*
Text included with a voice message	**Organization Configuration** -> **Unified Messaging** -> **New UM Mailbox Policy** -> **Message Text**	z. B. *new voice message!*
Text included with a fax message	**Organization Configuration** -> **Unified Messaging** -> **New UM Mailbox Policy** -> **Message Text**	e.g. *new fax!*
Minimum PIN length	**Organization Configuration** -> **Unified Messaging** -> **New UM Mailbox Policy** -> **PIN Policies**	e.g. *4*
Number of previous PINs to disallow	**Organization Configuration** -> **Unified Messaging** -> **New UM Mailbox Policy** -> **Message Text**	e.g. *5*
Number of incorrect PIN entries before PIN is automatically reset	**Organization Configuration** -> **Unified Messaging** -> **New UM Mailbox Policy** -> **Message Text**	e.g. *5*
Number of incorrect PIN entries before UM mailbox is locked out	**Organization Configuration** -> **Unified Messaging** -> **New UM Mailbox Policy** -> **Message Text**	e.g. *15*
Allow calls to uses	**Organization Configuration** -> **Unified**	Aktiviert

Field	Menu	Value
within the same dial plan	**Messaging** -> **New UM Mailbox Policy** -> **Dialing Restrictions**	
Allow calls to extensions	**Organization Configuration** -> **Unified Messaging** -> **New UM Mailbox Policy** -> **Dialing Restrictions**	Aktiviert

Activation of unified messaging for an exchange mailbox

Field	Menu	Value
Unified Messaging Mailbox Policy	**Organization Configuration** -> **Recipient Configuration** -> **Mailbox**	e.g. *demo_dialplan Default Policy*
Manually specify PIN	**Organization Configuration** -> **Recipient Configuration** -> **Mailbox**	Your PIN
Manually entered mailbox extension	**Organization Configuration** -> **Recipient Configuration** -> **Mailbox**	e.g. *720*

ISDN Configuration

Field	Menu	Value
Autoconfiguration on Bootup	**Physical Interfaces** -> **ISDN Ports** -> **<bri2-0 (TE)>** .	*Aktiviert*

SIP extension configuration

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	e.g. *Mailbox*
Extension / User Name	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	*500*
SIP Endpoint IP Address	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	e.g. *192.168.10.101*
Protocol	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	*TCP*
Port	**VoIP** -> **Media Gateway** -> **Subscriber** -> **New**	*5065*
Sort order	**VoIP** -> **Media Gateway** -> **Extensions** -> **New**-> **Advanced Settings**	*T.38 Fax*, *DTMF Outband*

Configuration of call routing

Field	Menu	Value
Description	**VoIP** -> **Media Gateway** -> **Call Routing**	e.g. *to_isdn*

Field	Menu	Value
	-> **New**	
Type	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*External*
Calling Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*Any*
Called Address	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**	*
Priority	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	e.g. *1*
Outbound Line	**VoIP** -> **Media Gateway** -> **Call Routing** -> **New**-> **Add**	e.g. *bri2-0*

Chapter 11 Media Gateway - Connecting the IP PBX hybird 300 to an SIP provider via bintec RS232b gateway

11.1 Introduction

Below is a description of how to connect the IP PBX **elmeg hybird 300** to a VoIP provider. Access to the Internet is established with the aid of a **bintec RS232b** gateway. The VoIP provider (e.g. sipgate) can be accessed via the Internet.

The **GUI** (Graphical User Interface) is used for configuration.

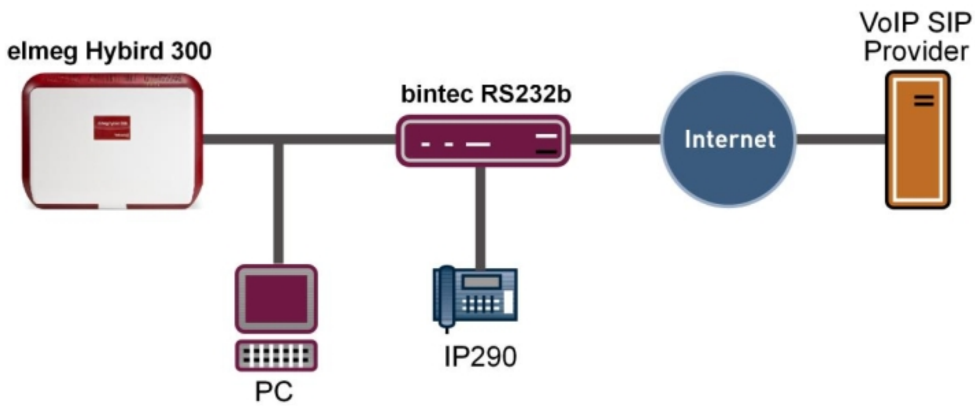

Fig. 169:

Requirements

- An **elmeg hybird 300**
- A **bintec RS232b** gateway with system software of 7.9.5 upwards
- Internet access
- Configuring the local IP address with the aid of the **Dime Manager**
- Setting up the Internet access at the **bintec RS232b** gateway
- Adapting firewall and quality of service to the internal **bintec RS232b** gateway
- Configuring the VoIP provider settings of the **elmeg hybird 300**
- Checking the QoS function at the **bintec RS232b** gateway

11.2 Configuration

11.2.1 Configuring the local IP address with the aid of the Dime Manager

After the the **elmeg hybird 300** and the **bintec RS232b** have been integrated in the local network, they can be located via the **Dime Manager**. The **Dime Manager** now offers the option of setting the local IP address via the context menu.

(1) Go to **Dime Manager** -> **IP Settings**.

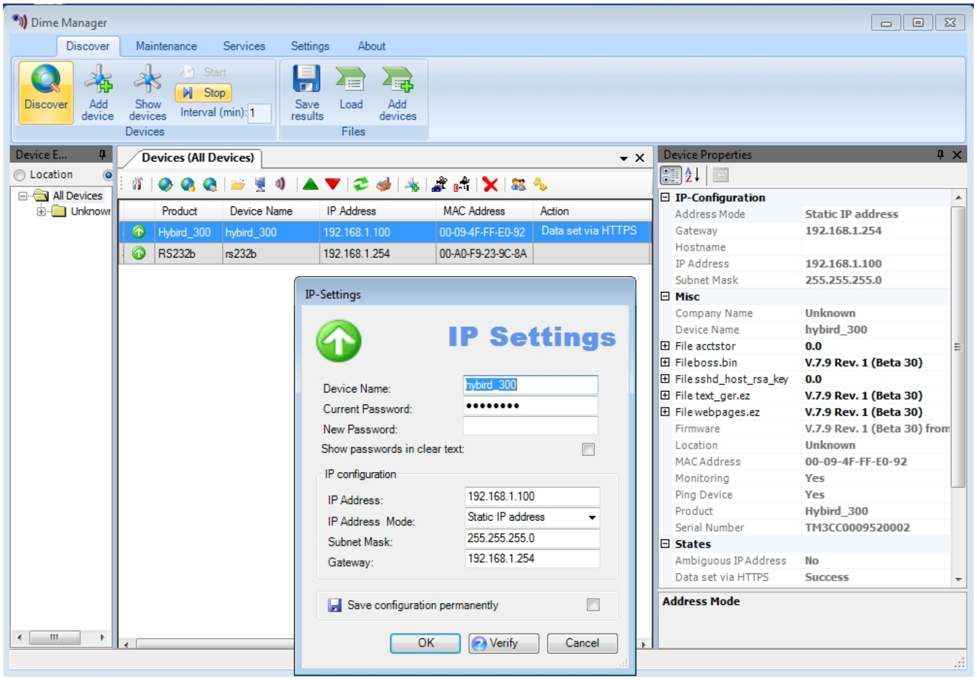

Fig. 170: **Dime Manager** -> **IP Settings**

In this workshop, the address 192.168.1.254/24 is assigned to the **bintec RS232b** gateway and the address 192.168.1.100/24 with standard gateway 192.168.1.254 for the IP PBX **elmeg hybird 300**.

11.2.2 Setting up the Internet access at the bintec RS232b gateway

To configure the Internet access at the **bintec RS232b** gateway the **GUI** has a wizard. Go to the following menu:

(1) Go to **Assistants** -> **Internet access** -> **Internet connections** -> **New**.

(2) At **Connection type**, select `Internal ADSL modem` .

(3) Click on **Continue** in order to configure a new Internet connection.

Enter the required data for the Internet connection.

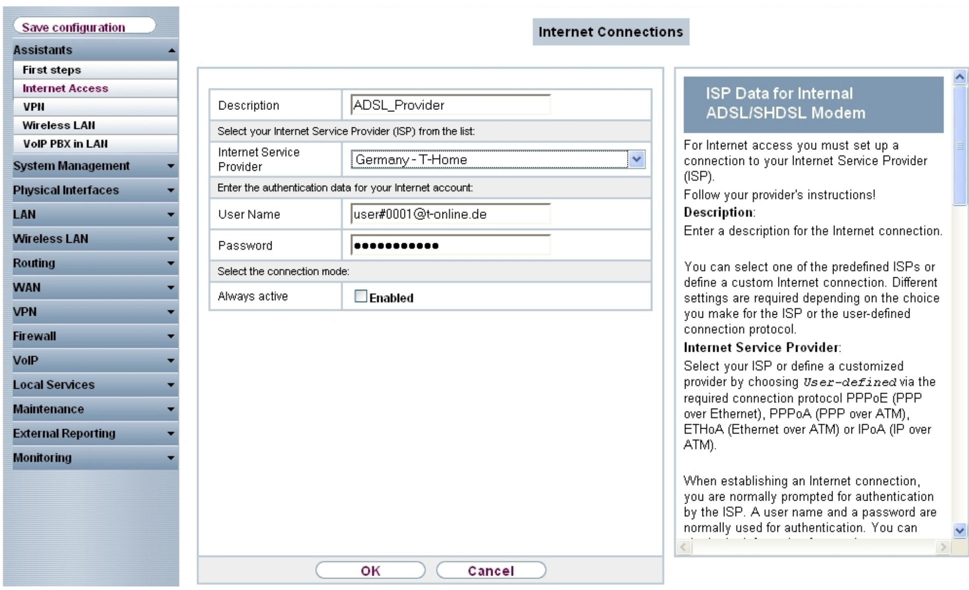

Fig. 171: **Assistants** -> **Internet access** -> **Internet connections** -> **Continue**.

Proceed as follows, to configure a new Internet connection:

(1) At **Description**, enter any designation for the Internet connection, e.g. `AD-SL_Provider`.

(2) Under **Internet Service Provider**, select the profile `Germany-T-Home`.

(3) For **Username**, enter the access data given to you by your provider, e.g. `user#0001@t-online.de`.

(4) Enter the **Password** given to you by your provider, e.g `supersecretgeheimkey`.

(5) Confirm the details with **OK**.

11.2.3 Adapting Firewall and Quality of Service to the internal bintec RS232b gateway

The **bintec RS232b** gateway uses, among others, Network Address Translation (NAT) as Firewall mechanism (Symetric NAT) to block undesirable data from the Internet. To ensure an uninterrupted VoIP connection, it must be guaranteed that the VoIP PBX in the LAN (**elmeg hybird 300**) uses a different NAT type (Full_cone NAT). In addition, the VoIP data (call signalling and the pure voice data) are to be prioritised with Quality of Service (QoS).

(1) Go to **Assistants** -> **VoIP PBX in LAN** -> **VoIP PBX in LAN** -> **New**.

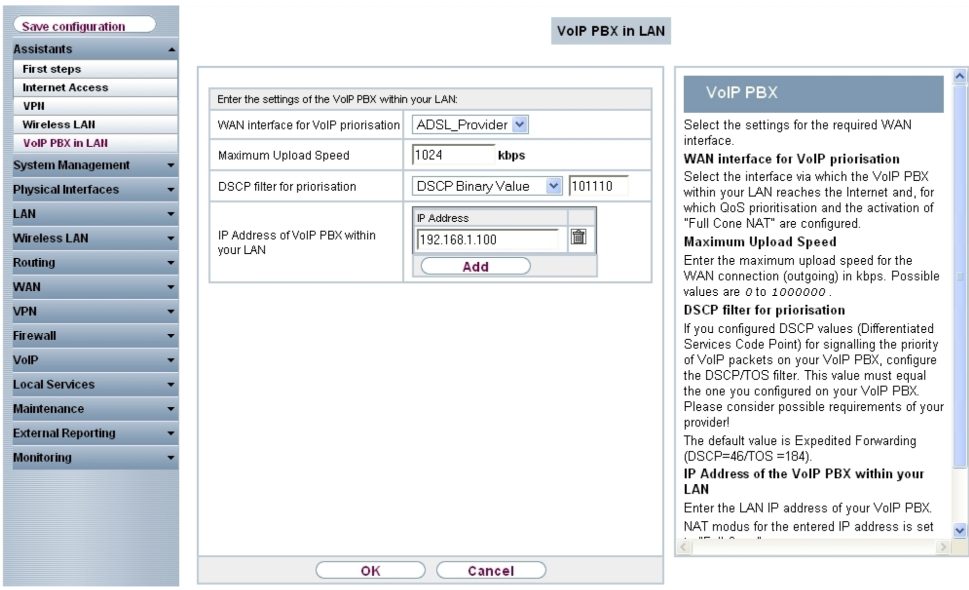

Fig. 172: **Assistants** -> **VoIP PBX in LAN** -> **VoIP PBX in LAN** -> **New**.

Proceed as follows in order to make the necessary settings:

(1) For **WAN interface VoIP prioritisation** , for example, select `ADSL_Provider`.

(2) For **Maximum Upload Speed**, for example, enter `1024` kbit/s.

(3) With the setting **DSCP filter for prioritisation** on `DSCP binary value 101110`, a filter for recognising and subsequently prioritising the VoIP data traffic of the IP PBX VoIP is configured. The **elmeg hybird 300** signals the VoIP data traffic with DSCP value 46 (decimal format) or 101110 (binary format).

(4) With option **IP Address of VoIP PBX** , the IP address of the local IP PBX (**elmeg hybird 300**) is queried. With **Add**, you add an entry, e.g., `192.168.1.100`. This entry automatically changes the Network Address Translation Mode.

(5) Confirm the entry with **OK**.

11.2.4 Configuring the VoIP provider settings of the elmeg hybird 300

Before the VoIP settings of the SIP provider are stored in the web interface of the **elmeg hybird 300**, their network settings must have been completed.

(1) Go to **Assistants** -> **First Steps**.

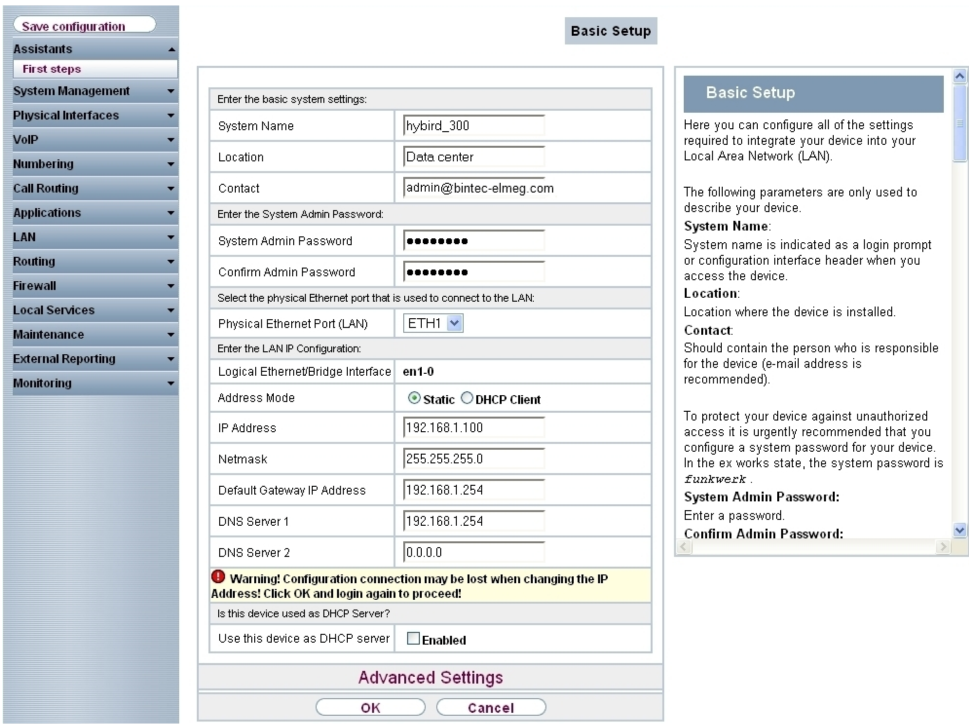

Fig. 173: **Assistants** -> **First Steps**

Proceed as follows in order to complete the network settings:

(1) The **System Name** can be changed in the assistant, here `hybird_300`, for example.

(2) For **Location**, for example, enter `Computing Centre`.

(3) For **Contact**, for example, enter `admin@bintec-elmeg.com`.

(4) Also the **password** can be changed in the assistant, here `supersecretgeheimkey`, for example.

(5) In the sector **LAN IP Configuration**, enter the **Standard gateway IP Address**, here `192.168.1.254`.

(6) For **DNS Server 1**, enter the address of the **bintec RS232b**, in this case

`192.168.1.254`.

(7) Confirm the entry with **OK**.

For every connection set-up of a VoIP call, the IP PBX **elmeg hybird 300** calculates which speech coding (e.g. G.711) is used and whether the connection can be developed with the available bandwidth. A location profile for the VoIP provider is created for this.

(1) Go to **VoIP** -> **Settings** -> **Locations** -> **New**.

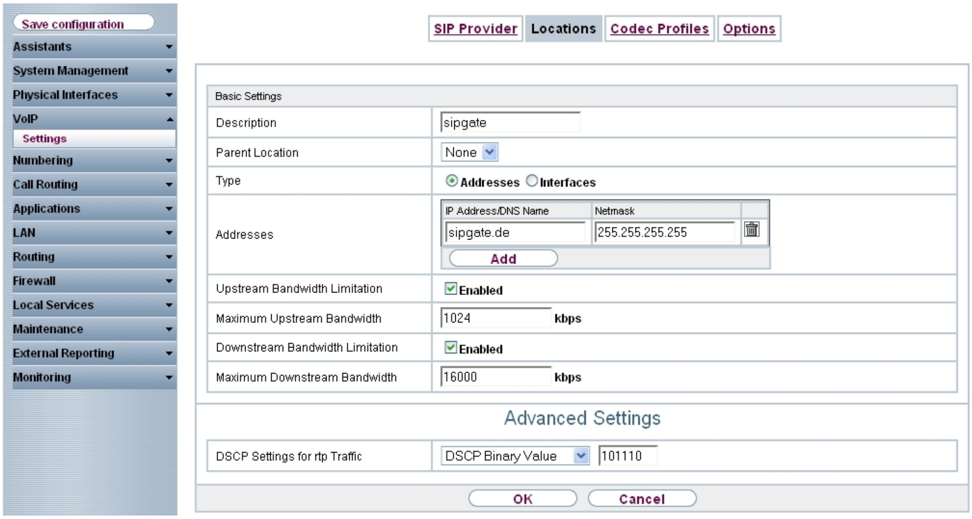

Fig. 174: **VoIP** -> **Settings** -> **Locations** -> **New**.

Proceed as follows in order to set up the location profile for the VoIP provider:

(1) For **Description**, for example, enter `sipgate`.

(2) Click on **Add**, in order to configure new addresses. Enter the **IP Address** of the SIP server, e.g. `sipgate.de`.

(3) Enable the option **Bandwidth Limitation Upstream**.

(4) At **Maximum Upstream Bandwidth**, enter the maximum data rate in kbit/s per second in the direction of transmission, e.g. `1024`.

(5) Enable the option **Bandwidth Limitation Downstream**.

(6) For **Maximum Downstream Bandwidth**, enter the maximum data rate in kbit/s per second in the direction of transmission, e.g. `16000`.

(7) Click on **Advanced Settings**.

(8) In the case of the **DSCP Settings for RTP Data** option, the DSCP value for marking language packs is entered, in this case `DSCP Binary Value`. With this option, the same value is assigned that has been configured at **bintec RS232b** gateway in the **Assistants** -> **VoIP PBX menu in the LAN**. This DSCP value can be entered in dif-

ferent formats, e.g. in decimal (46) or in binary format (101110).

(9) Confirm the entry with **OK**.

Following this, the login data of the **SIP Provider** (here shown with the example of VoIP provider sipgate) can be stored.

(1) Go to **VoIP** -> **Settings** -> **SIP Provider** -> **New**.

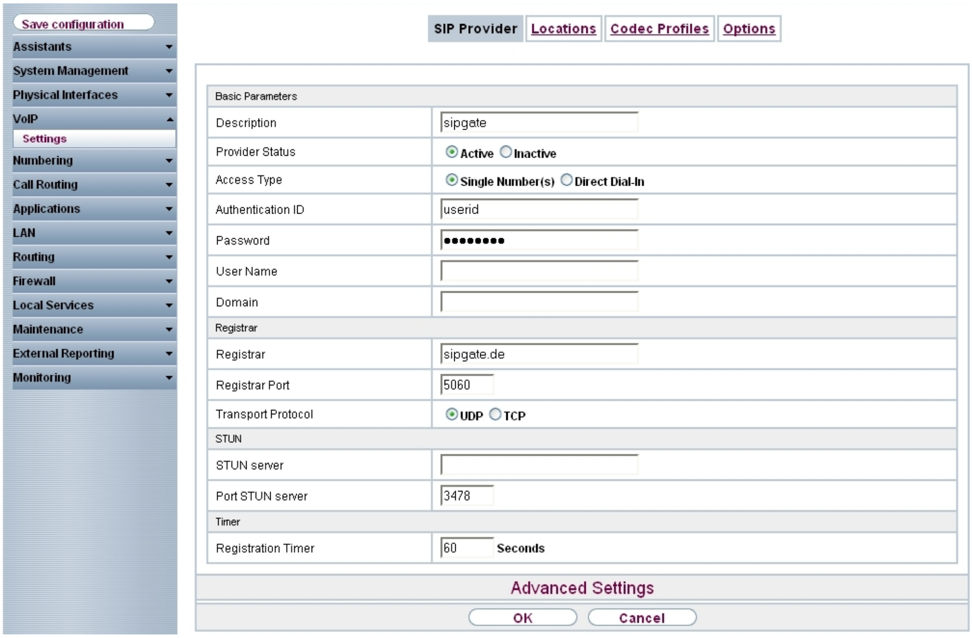

Fig. 175: **VoIP** -> **Settings** -> **SIP Provider** -> **New**.

Proceed as follows, in order to store the SIP provider's login data:

(1) For **Description**, for example, enter a description for the SIP provider, e.g. `sipgate`.

(2) Enter your provider's **Authentication Address**, e.g. `userid`.

(3) Enter the **User Name** given to you by your VoIP provider, e.g `userid`.

(4) Specify the IP address or the domain name of the SIP **Registrar**, e.g. `sipgate.de`.

(5) Enter the IP address or the domain name of the **STUN server**, e.g. `stun.sipgate.de`. Since the connection to the VoIP provider is established via an upstream NAT gateway (**bintec RS232b**), a STUN server must also be configured in the provider settings. The IP PBX **elmeg hybird 300** determines the WAN IP address of the **bintec RS232b** gateway with the aid of the STUN server.

(6) Confirm your entry with **OK**.

11.2.5 Checking the QoS function at the bintec RS232b gateway

Go to the following menu to check the quality of service settings on the **bintec RS232b** gateway:

(1) Go to **Monitoring** -> **QoS** -> **QoS**.

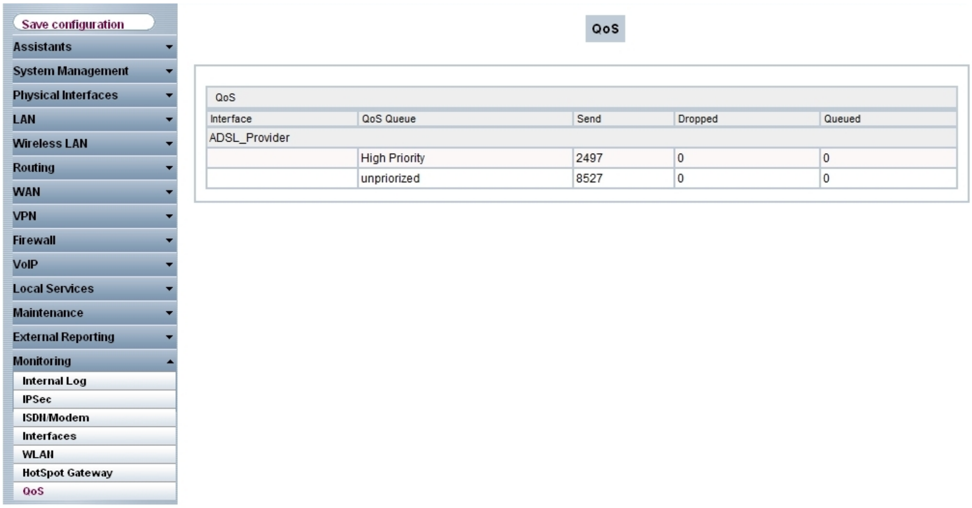

Fig. 176: **Monitoring** -> **QoS** -> **QoS**

A list of all the interfaces for which the QoS has been configured is displayed in this menu. Here the IP packages that are sent to the Internet are counted. In the case of VoIP calls, the entry `High Priority` (High Priority QoS Queue) must be incremented. This ensures that the call data have a higher priority than the remaining data traffic.

11.3 Overview of configuration steps

Configuring the local IP address

Field	Menu	Value
IP address	**Dime Manager** -> **IP Settings**	`192.168.1.100`
gateway	**Dime Manager** -> **IP Settings**	`192.168.1.254`

Select connection type

Field	Menu	Value
Connection type	**Assistants**-> **Internet Access** -> **Internet Connections** -> **New**.	e. g. *internal ADSL modem*

Setting up the Internet connection

Field	Menu	Value
Description	**Assistants** -> **Internet Access** -> **Internet Connections** -> **Continue**.	e.g. *ADSL_Provider*
Internet Service Provider	**Assistants** -> **Internet Access** -> **Internet Connections** -> **Continue**.	e.g. *Germany-T-Home*
Username	**Assistants** -> **Internet Access** -> **Internet Connections** -> **Continue**.	e.g. *user#0001@t-online.de*
Password	**Assistants** -> **Internet Access** -> **Internet Connections** -> **Continue**.	e.g. *supersecretgeheimkey*

Adaptation to the Internet gateway

Field	Menu	Value
WAN interface for VoIP prioritisation	**Assistants** -> **VoIP PBX in LAN** -> **VoIP PBX in LAN** -> **New**.	e.g. *ADSL_Modem*
Maximum Upload Speed	**Assistants** -> **VoIP PBX in LAN** -> **VoIP PBX in LAN** -> **New**.	e.g. *1024*
DSCP filter for prioritisation	**Assistants** -> **VoIP PBX in LAN** -> **VoIP PBX in LAN** -> **New**.	e.g. *DSCP Binary Value* and *101110*
IP Address of the VoIP PBX in the LAN	**Assistants** -> **VoIP PBX in LAN** -> **VoIP PBX in LAN** -> **New**.	e.g. *192.168.1.100*

Configuring VoIP Providers

Field	Menu	Value
System Name	**Assistants** -> **First Steps**	e.g. *hybird_300*
Location	**Assistants** -> **First Steps**	e. g. *Computing Centre*
Contact	**Assistants** -> **First Steps**	e.g. *ad-*

Field	Menu	Value
		min@bintec-elmeg.com
System Admin Password	**Assistants** -> **First Steps**	e.g. *supersecretge-heimkey*
Default gateway IP Address	**Assistants** -> **First Steps**	e.g. *192.168.1.254*
DNS Server 1	**Assistants** -> **First Steps**	e.g. *192.168.1.254*

Setting up a location profile for the VoIP provider

Field	Menu	Value
Description	**VoIP** -> **Settings** -> **Locations** -> **New**.	e.g. *sipgate*
Addresses	**VoIP** -> **Settings** -> **Locations** -> **New**.	e.g. *sipgate.de*
Maximum Upstream Bandwidth	**VoIP** -> **Settings** -> **Locations** -> **New**.	e.g. *1024*
Maximum Downstream Bandwidth	**VoIP** -> **Settings** -> **Locations** -> **New**.	e.g. *16000*
DSCP Settings for RTP Data	**VoIP** -> **Settings** -> **Locations** -> **New**-> **Advanced Settings**	e.g. *DSCP Binary Value* and *101110*

Storing SIP provider's login data

Field	Menu	Value
Description	**VoIP** -> **Settings** -> **SIP Provider** -> **New**.	e.g. *sipgate*
Authentication ID	**VoIP** -> **Settings** -> **SIP Provider** -> **New**.	e.g. *userid*
User name	**VoIP** -> **Settings** -> **SIP Provider** -> **New**.	e.g. *userid*
Registrar	**VoIP** -> **Settings** -> **SIP Provider** -> **New**.	e.g. *sipgate.de*
STUN Server	**VoIP** -> **Settings** -> **SIP Provider** -> **New**.	e.g. *stun.sipgate.net*

Chapter 12 Media Gateway - Settings on the elmeg hybird 300 to phone via an SIP provider (sipgate)

12.1 Introduction

The following describes configuration of an SIP provider using an **elmeg hybird 300**.

The pictured information is only provided as an example. Please use the data obtained from your SIP provider. When using an inland SIP provider, certain presets are important, e.g. in order not to have to enter the entire prefix and number in case of a local call, but rather only the number itself. When connecting the **elmeg hybird 300** to an existing network, configuration of the border router must be taken into account.

The **GUI** (Graphical User Interface) is used for configuration.

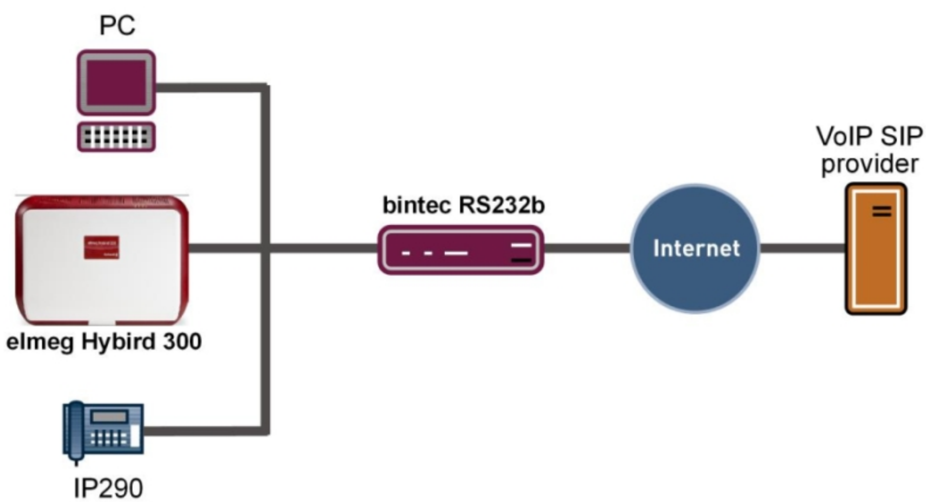

Fig. 177: Example scenario

Prerequisites

- An **elmeg hybird 300** with system software 7.9.1 patch 4
- A **bintec RS232b** with system software 7.9.5 patch 4

- An Internet access

12.2 Configuration

12.2.1 Configuring the bintec RS232bw

Session Initiation Protocol (SIP) serves as a translation instance between different telecommunications networks, e.g between the plain old phone network and the next generation networks (IP networks). To create SIP connections, go to the following menu:

(1) Go to **VoIP** -> **SIP** -> **Options**.

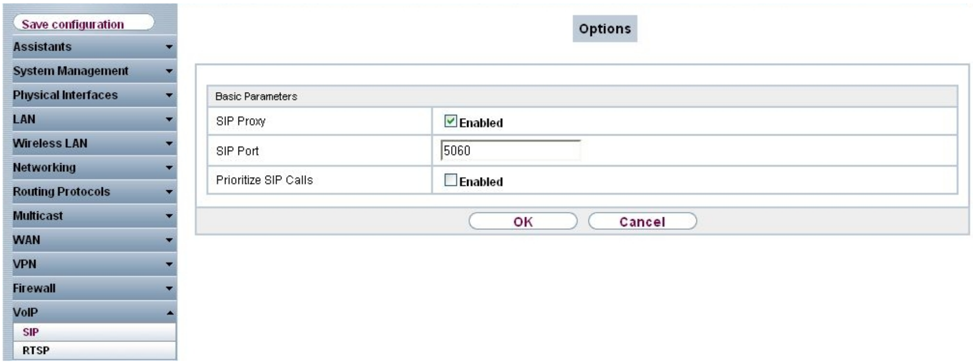

Fig. 178: **VoIP** ->**SIP**-> **Options**

Proceed as follows to perform the SIP settings:

(1) Activate the **SIP Proxy**. The SIP connections are forwarded.

(2) Activate **Prioritise SIP Calls**.

(3) Press **OK** to confirm your entry.

Note

These settings are important, as problems may otherwise arise in calls via a SIP provider.

12.2.2 Configuration of the elmeg hybird 300

Over the assistant you can configure the settings required for login to an external SIP provider.

(1) Go to **Assistants** -> **First Steps** -> **Basic Settings**.

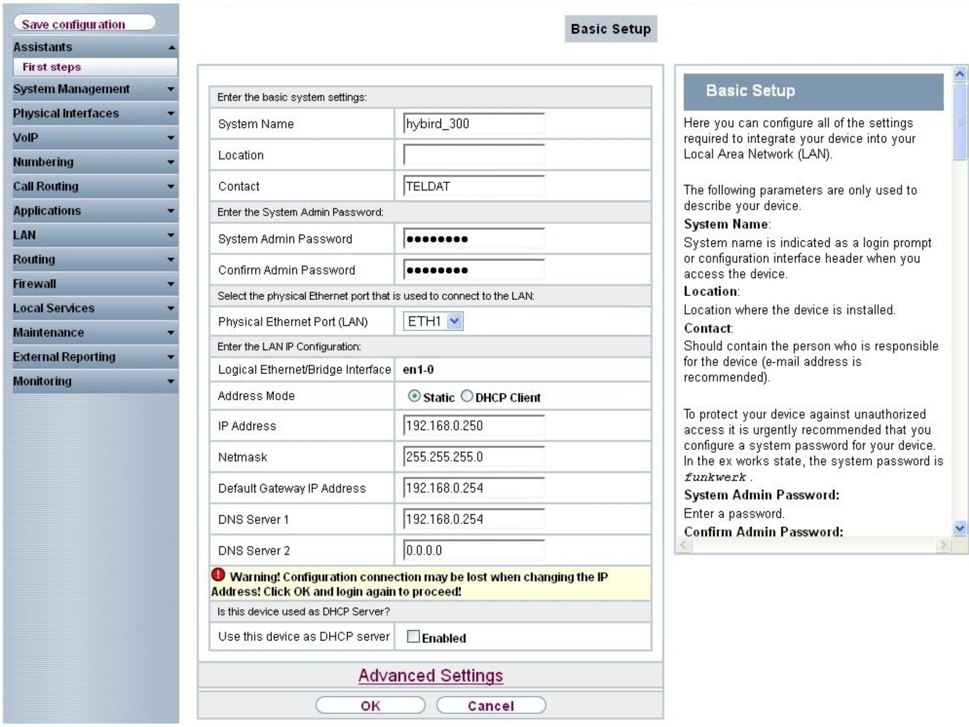

Fig. 179: Go to **Assistants** -> **First Steps** -> **Basic Settings**.

Proceed as follows to make the SIP settings:

(1) In **Assistants**, the **System name** can be modified, here, for example *hybird_300*.

(2) For **Standard Gateway IP Address** enter the IP address of your gateway providing Internet access, e.g. *192.168.0.254*.

(3) For **DNS Server 1** enter the IP address of the name server for the name resolution of Internet addresses *192.168.0.254*.

(4) Press **OK** to confirm your entry.

For automatic generation of international and national numbers, you must perform the following settings:

(1) Go to **System Administration** -> **Global Settings** -> **System**.

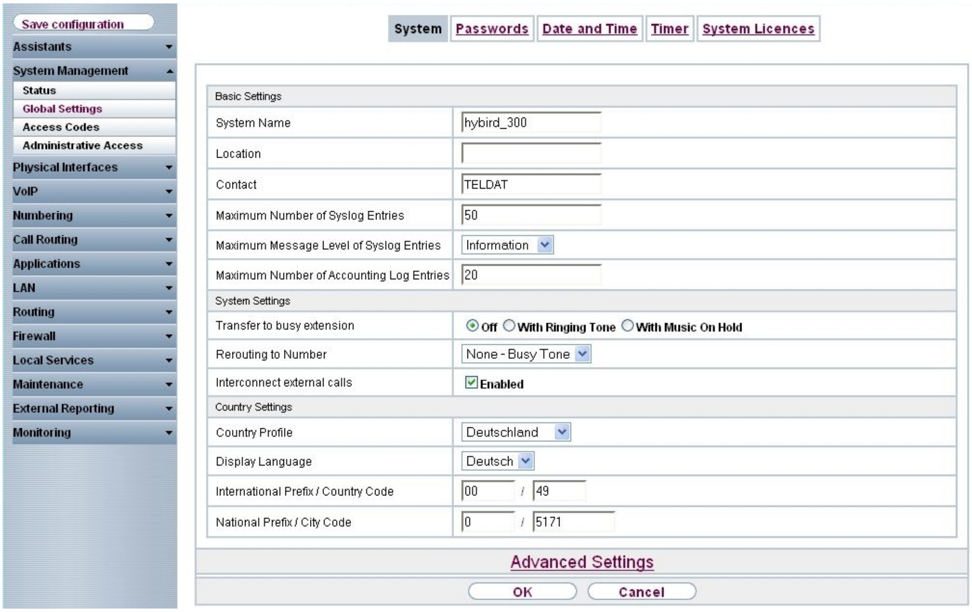

Fig. 180: **System Management** -> **Global Settings** -> **System**

Proceed as follows to configure the numbers:

(1) Under **International prefix / country code** enter the country code, e.g., `49` for Germany. Without the entry, for SIP providers the entire call number with country code must be dialled.

(2) Under **National Prefix / Area code** enter the area code for the location where your system is installed, e.g. `5171` . This area code is crucial for a point-to-point connection, as otherwise, automatic callback to an external number, for example, is not possible.

(3) Confirm with **OK**.

12.2.3 Registering bintec TR200 with provider sipgate

The login data for registering the VoIP provider accounts with provider sipgate are entered in the **SIP Providers** menu.

(1) Go to **VoIP** -> **Settings** -> **SIP Providers** -> **New**.

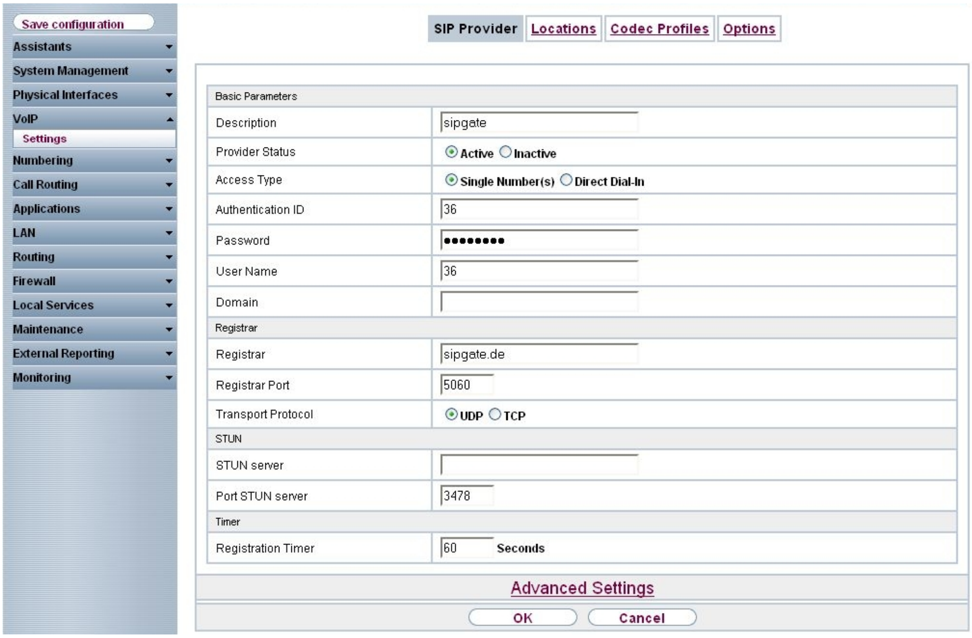

Fig. 181: **VoIP** -> **Settings** -> **SIP Providers** -> **New**.

To save SIP provider login data, proceed as follows:

(1) Under **Description** enter a description for the SIP provider, e.g. `sipgate`.

(2) Enter your provider's **Authentication ID** e.g. `36`.

(3) Enter the **User Name** you received from your VoIP provider, e.g. `36`.

(4) Indicate the IP address or domain name of the SIP **Registrar** e.g. `sipgate.de`.

(5) Press **OK** to confirm your entry.

In the menu **VoIP** ->**Settings** -> **SIP Providers** the current VoIP configuration is displayed. It takes less than a minute to register a new SIP provider account with the provider. As soon as the enrolment process has been completed successfully, the status is set automatically to (active).

Fig. 182: **VoIP** -> **Settings** -> **SIP Providers** .

You change the status of VoIP configuration by pressing the button or button in the **Action** column.

Next, login data of the SIP provider can be saved. For the prefix to be automatically added before the dialled number, the **Generate National Number** option must be enabled. For this, go to the following menu:

(1) Go to **VoIP** -> **Settings** -> **SIP Providers** -> **Advanced Settings**.

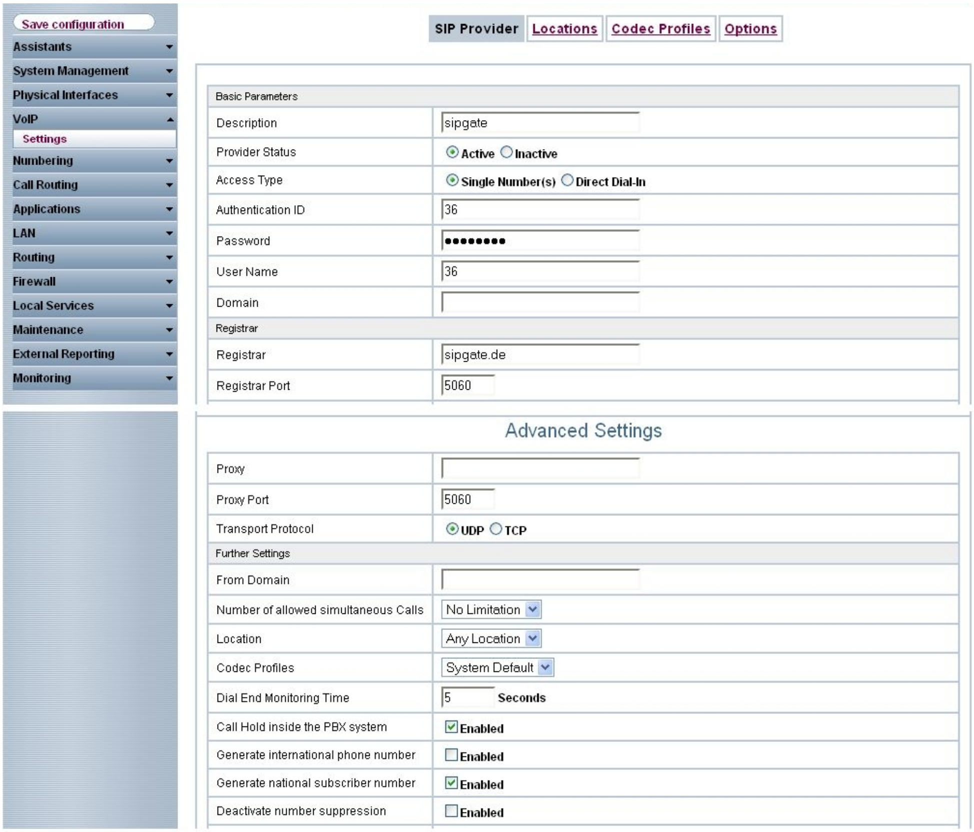

Fig. 183: **VoIP** -> **Settings** -> **SIP Providers** -> **Advanced Settings**.

To automatically generate the prefix, proceed as follows:

(1) Enable the option **Generate National Number**. The prefix you've entered under **National prefix / area code** is automatically generated (e. g. for 5171 the prefix 05171 is automatically added before the dialled number).

(2) Confirm with **OK**.

12.3 Overview of configuration steps

Configuring the bintec RS232bw

Field	Menu	Value
SIP Proxy	**VoIP** ->**SIP**-> **Options**	Activated
Prioritize SIP Calls	**VoIP** ->**SIP**-> **Options**	Activated

Configuration of the elmeg hybird 300

Field	Menu	Value
Standard gateway IP address	Go to **Assistants** -> **First Steps** -> **Basic Settings**.	e.g. *192.168.0.254*
DNS Server 1	Go to **Assistants** -> **First Steps** -> **Basic Settings**.	e.g. *192.168.0.254*

Generate number

Field	Menu	Value
International prefix/Country code	**System Management** -> **Global Settings** -> **System**	e.g. *49*
National prefix/Area code	**System Management** -> **Global Settings** -> **System**	e.g. *5171*

Login with the provider

Field	Menu	Value
Description	**VoIP** -> **Settings** -> **SIP Providers** -> **New**.	e.g. *sipgate*
Authentication ID	**VoIP** -> **Settings** -> **SIP Providers** -> **New**.	e.g. *36*
User Name	**VoIP** -> **Settings** -> **SIP Providers** -> **New**.	e.g. *36*
Registrar	**VoIP** -> **Settings** -> **SIP Providers** -> **New**.	e.g. *sipgate.de*

Automatically generate prefix

Field	Menu	Value
Create inland call number	**VoIP** -> **Settings** -> **SIP Providers** ->**New** -> **Advanced Settings**	Activated