



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:

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

Call Assignment

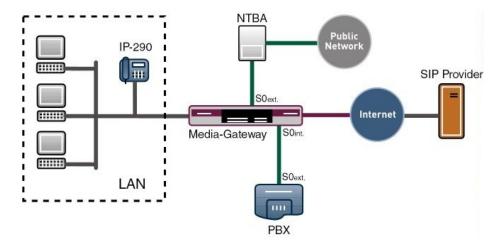


Fig. 1: Example scenario

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 -> <bri>2-0 (TE)>

Save configuration		ISDN Configuration MSN Configuration	
System Management			
Physical Interfaces	Basic Parameters		
AUX Ethernet Ports	Port Name		
ISDN Ports	Autoconfiguration on Bootup	✓ Enabled	
LAN	Result of Autoconfiguration	Port Usage: Dialup (Euro ISDN), ISDN Configuration Type: Point-to-Multipoin	
Routing	Port Usage	Dialup (Euro ISDN) 🐱	
WAN	ISDN Configuration Type	Point-to-Multipoint Point-to-Point	
Firewall •		Advanced Settings	
VoIP		OK Cancel	



Field	Meaning		
Port Name	Shows the name of the ISDN port.		
Autoconfiguration on Bootup	Here, select whether the ISDN switch type should be automatic- ally recognised.		
Result of Autoconfigura- tion	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 <i>Point-</i> <i>to-Multipoint</i> .		

Relevant fields in the ISDN Configuration menu

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.

Save configuration		ISDN Configuration MSN Configuration
Assistants -		
System Management 🔹		
Physical Interfaces	Basic Parameters	
AUX	ISDN Port	bri2-3 🗸
Ethernet Ports		
ISDII Ports	Service	PPP (Routing) 💌
LAN	MSN	999999
Wireless LAN Controller 🛛 🔻		
Networking -	MSN Recognition	
Routing Protocols 🗸	Bearer Service	O Data + Voice ○ Data ○ Voice
Multicast -		OK Cancel
WAN		

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

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 num- ber below.
MSN	Enter the call number.
MSN Recognition	Select the mode your device is to use for the number comparis- on for MSN with the <i>called party number</i> of the incoming call. By default, the value is set on <i>right to left</i> . Select the value <i>left to right (DDI)</i> if your device is connected to a point-to-point connection.
Service attribute	Select the type of incoming call (service detection). <i>Data</i> + <i>Voice</i> specifies that both data and voice calls (default value) are executed.

Relevant fields in the MSN Configuration menu

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)>

Save configuration	•		ISDN Configuration MSN Configuration
System Management	-		
Physical Interfaces	•	Basic Parameters	
AUX		Port Name	bri2-3 (NT)
Ethernet Ports ISDN Ports		Autoconfiguration on Bootup	Enabled
LAN	-	Port Usage	Dialup (Euro ISDN) 💌
Wireless LAN Controller	-	ISDN Configuration Type	
Networking	•		
Routing Protocols	•	Advanced Settings	
Multicast	•	X.31 (X.25 in D Channel)	
WAN	-		
VPN	-		OK Cancel

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

Relevant fields in the ISDN Configuration menu

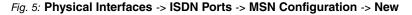
Field	Meaning	
Port Name	Shows the name of the ISDN port.	
Port Usage	Here, select Dialup (Euro ISDN).	
ISDN Configuration	Select the ISDN access configuration	
Туре	Point-to-Multipoint.	

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.

Save configuration		ISDN Configuration MSN Configuration
Assistants 👻		
System Management 🛛 👻		
Physical Interfaces	Basic Parameters	
AUX	ISDN Port	bri2-3 🗸
Ethernet Ports		
ISDN Ports	Service	PPP (Routing) 🔽
LAN 🔻	MSN	999999
Wireless LAN Controller 🛛 👻		
Networking -	MSN Recognition	
Routing Protocols 🔹	Bearer Service	🖲 Data + Voice 🔿 Data 🔿 Voice
Multicast 👻		
WAN		



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 num- ber below.

Field	Meaning
MSN	Enter the call number.
MSN Recognition	Select the mode your device is to use for the number comparis- on for MSN with the <i>called party number</i> of the incoming call. By default, the value is set on <i>right to left</i> .
Service attribute	Select the type of incoming call (service detection). <i>Data</i> + <i>Voice</i> 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.



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

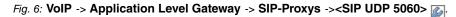
In the **VoIP**-> **Application Level Gateway** menu, a list is shown of all configured Application Level Gateway entries. These entries enable the ALG. Each entry defines a particular TCP or UDP destination port that is to be supervised by the ALG. In the ex works state, there are two entries configured 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> [].

Save configuration		SIP Proxies SIP Endpoints
System Management 🔹 👻		
Physical Interfaces 🔹	Basic Parameters	
LAN 🔫	Description	SIP UDP 5060
Routing 👻		
WAN -	Administrative Status	Enabled
VPN -	Protocol	UDP 💌 Destination Port 5060
Firewall 👻	Session Timeout	7200 sec
VolP 🔺		
Application Level Gateway	Low Latency Transmission	✓ Enabled
Media Gateway		
Local Services 👻		OK Cancel



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 Transmis- sion	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 connec- tion.
	The voice quality is optimised by choosing <i>Enabled</i> .

Relevant fields in the SIP Proxies menu

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.

PAccounts	can rouding	CLID ITal	station C	all Translation	Options
sir	sipgate				
1	Enabled				
۲	Off O Client O	Server			
sip	ogate.de				
Г					
U	DP 🔽 Port: 508	50			
ne [1839681					
ge	geheim				
V	✓ Enabled				
60	600 sec				
	Advanc	ed Settin	gs		
nce 📀) Default 🔿 Qual	ity 🔿 Low Bai	ndwidth 🔿 Hi	igh Bandwidth	
5	G.711 uLaw	G.711 aLaw	🗹 G.729	G.726-40	1
		G.726-24		DTMF Outban	4
Voice Quality Settings					
	✓ Enabled				
on 🔽	✓ Enabled				
40	40 ms				
			40 ms	40 ms	[40] ms

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 <i>Off</i> , the trunk mode is not used.
Registrar	Enter the IP address of the remote SIP terminal (client or serv- er) 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 pro- vider 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 authentica- tion. You must enter this value here. Maximum number of char- acters: 40.
Registration	Enables or disables the SIP REGISTER registration mechan- ism.
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 Se- quence	Determine the order in which the codecs are offered for use by the media gateway. If the first codec cannot be applied, an at- tempt is made to use the second codec, and so on. Set Codec Proposal Sequence to <i>default</i> . The codec in the first posi- tion 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 co- decs 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 <i>Enabled</i> is selected, echo feedback is suppressed.

Field	Meaning
Comfort Noise Genera- tion (CNG)	Specify whether Comfort Noise Generation should be used. The slight comfort noise generation prevents subscribers from think- ing 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

 \bigcirc . The status of the VoIP connection is changed by pressing the \frown button or \bigcirc button in the **Action** column.

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

Save configuration	_	Extensions	SIP Accounts	Call Routing	CLID Translation	Call Tran	islation	Options	
System Management	-								
Physical Interfaces	-	Description	Registrar / Outl	ound Proxy	Protocol	Status	Action		T
LAN	-	sipgate	sipgate.de		UDP	0	+	1	
Routing	-	-	5.0 00000						and miner (such
WAN	-				New				
VPN	-								
Firewall	-								
VolP									
Application Level Gate	Nay								
Media Gateway									

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.

stem Management 🛛 👻	Extensions SIP Acc							
nysical Interfaces 🔹								
	Basic Parameters							
N +	Description	IP-Telefon						
outing 🔹	Extension / User Name	10	_					
AN -	Extension / Oser Name	1						
PN 🔻	Interface Type	SIP ○ ISDN						
rewall 🔻	Registration	Enabled						
DIP 🔺	Expire Time	60						
Application Level Gateway	Expire time	1	sec					
Media Gateway	Authentication ID	10						
ocal Services 🔹 🔻	Password	geheim	_					
aintenance 🔹		1-	_					
dernal Reporting 🔹 👻	Protocol	UDP 💌						
onitoring 🔹 🔻	Port	5060						
	Advanced Settings							
	Codec Settings							
	Codec Proposal Sequence I Couality Couvers Courses Couvers							
	Sort Order		711 aLaw 🗹 G.729	G.726-40				
	Son Order	G.726-32 G.	726-24 G.726-	16 DTMF Outband				
	Voice Quality Settings							
	Echo Cancellation	✓ Enabled						
	Comfort Noise Generation	Enabled	✓ Enabled					
	Packet Size	40	ms					

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

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 mechan- ism.
Expire Time	Enter the time in seconds after which the current registration be- comes 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 tele- phone. 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 Se- quence	Determine the order in which the codecs are offered for use by the media gateway. If the first codec cannot be applied, an at- tempt is made to use the second codec, and so on. Set Codec Proposal Sequence to <i>default</i> . The codec in the first posi- tion 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 co- decs 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 <i>Enabled</i> is selected, echo feedback is suppressed.
Comfort Noise Genera- tion (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.

Save configuration		Extensions	SIP Accounts	Call Routing	CLID Translation	Call Translation	Options	
System Management	•							
Physical Interfaces	-	Basic Parameters						
LAN	-	Description	lis	DN port				
Routing	-	Description	1					
WAN	-	Extension / User Name		20				
VPN	-	Interface Type	C	SIP 💿 ISDN				
Firewall	-	Select ISDN interface	b b	ri2-0 💌				
VolP	-							
Application Level Gatewa	iy 🛛			Advanc	ed Settings			
Media Gateway				-				
Local Services	-		(OK	Cancel)		

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

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.

Save configuration)	Extensions	SIP Accounts	Call Routing	CLID Translation	Call Translation	Options	
System Management	-						-	
Physical Interfaces	-							
LAN	-	Basic Parameters						
Routing	-	Session Border Con	troller Mode 🛛 🗛	vuto 🔽				
WAN	-	Media Stream Termination						
VPN	•							
Firewall	-	Default Drop Extensi	on					
VolP	•	Dial Latency	5	Sec	onds			
Application Level Gatew	ay							
Media Gateway				Ad∨anc	ed Settings			
Local Services	-		(ок	Cancel)		
Maintenance	-			51	Cancer			



Relevant fields in the Options menu					
Field	Meaning				
Session Border Control- ler Mode	Determines the behaviour of the media gateway in combination with a session border controller.				
	• <i>Auto</i> : 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).				
	• <i>Off</i> : 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</sip> 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 Termina- tion	Determines how RTP sessions are controlled by the system.				

Relevant fields in the Options menu

Field	Meaning
	• <i>Enabled</i> : 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.
	• <i>Disabled</i> (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.

Save configuration	Extensions	SIP Accounts	Call Routing	CLID Translation	Call Translation	Options	
System Management 🛛 👻							
Physical Interfaces 🔹 👻	Basic Parameters						
LAN 👻	Description	6	ipgate				
Routing 🔹							
WAN -	Administrative Statu	в <u>Ц</u>	Z Enable				
VPN -	Туре	E	External 💌				
Firewall 👻	Calling Line	7	Any 🔽				
VolP 🔺	Calling Address		· · · · · · · · · · · · · · · · · · ·				
Application Level Gateway	Calling Address	I.					
Media Gateway	Called Address		9*				
Local Services 👻	Priority Line Called Add		iress Translation St		Status Actio	n	
Maintenance 🔹 👻	1 -				0 14	. 💼 🖉	
External Reporting 🗾 👻	Add						
Monitoring 👻	Routing Rule						
	Priority	1					
	Administrative Status						
	Outbound Line	ŀ	bri2-0 💌				
	Called Address Translation <9:0049>;						
	Apply						
			ок	Cancel)		

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

Relevant fields in the Ca	II Routing menu

Field	Meaning
Description	Here, enter the name of the call routing entry.
Administrative Status	The entry is used with enabled.
Туре	Select <i>External</i> 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.

FieldMeaningPriorityDetermines the order of the filter rules, starting with 1 in increasing numerical order.Administrative StatusThe entry is used with Enable.Outbound LineDefines the PSTN line (PRI, BRI, FXO) or the SIP account used for an outgoing call.Called Address TranslationThe 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.

Relevant fields in the Routing Rule menu

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.

Save configuration	Extensions SIP Accou	nts Call Routing CLID Translation Call Translation Options					
System Management 🔹 👻							
Physical Interfaces 🔹 👻	Basic Parameters						
_AN 👻	Description	2557435->10					
Routing 👻	Description	2337435-910					
WAN -	Calling Line	bri2-0 🔽					
VPN -	Called Line	Any 💌					
irewall 👻	Called Address	2557435					
/oIP 🔺							
Application Level Gateway	Calling Address Translation	<2557435:10>;					
Media Gateway							
Local Services 👻		OK Cancel					

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

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.

Relevant fields in the CLID Translation menu

Field	Meaning
	Select <i>Any</i> 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.

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 .

Save configuration		Extensions	SIP Accounts	Call Routing	CLID Translation	Call Translation	Options		
System Management	-							-	
Physical Interfaces	-	Description	Calling L	ine	Called Address	Called Line		T	
LAN	• 2	2557435->10	bri2-0		2557435	Any	1	1	
Routing	- 1	839681->20	sipgate 1		1839681	Any	1	7	
WAN	- 2	2558296->20	bri2-0		2558296	Any	1	1	
/PN	-								
Firewall	-	()							
VoIP	-								
Application Level Gateway	/								
Media Gateway									

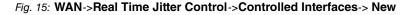
Fig. 14: VoIP -> Media Gateway -> CLID Translation

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.

Save configuration		Controlled Interfaces				
System Management 🔹 👻						
Physical Interfaces 🔹 👻						
LAN 🔫	Basic Settings					
Routing 👻	Interface	en1-0 🗸				
WAN 🔺	Control Mode	Controlled RTP Streams only				
Internet + Dialup	Control Mode					
Leased Line	Maximum Upload Speed	128 kbps				
Real Time Jitter Control		,				
VPN 👻		OK Cancel				
Firewall 👻						



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 <i>Controlled RTP Streams only</i> : 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.

Relevant fields in the Controlled Interfaces menu

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.

Save configuration		Filter Rules QoS Options
System Management 🔹 👻		
Physical Interfaces 🔹		
LAN 👻	Basic Parameters	
Routing 🗸 🗸	Source	ANY
WAN -	Destination	ANY
VPN -		
Firewall	Service	any
Policies	Action	Access 💌
Interfaces	Apply QoS	✓ Enabled
Addresses Services		
VoIP -	Traffic Priority	High
Local Services 🔹 🔻		OK Cancel
Maintenance 🔹		

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 <i>ANY</i> 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 <i>Access</i> 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.

Save configuration							- 1	Statistics					
System Management	-												
Physical Interfaces	-												
LAN	-	5	Show Tran	sfer Tota	ls 🗸	Automatic I	Refresh Inte	rval 300	Secor	nds	Appl	v	
Routing	-					_		,	_				
WAN	-	Vi	iew 20	per page				equal	~		Go		
VPN	-	#	Description	Туре	Tx Packets	Tx Bytes	Tx Errors	Rx Packets	Rx Bytes	R× Errors		Unchanged for	Action
		1	en1-0	Ethernet	592	588.23K	0	405	81.80K	0	0	0d 0h 29m 50s	1 +
Firewall	-	2	en1-4	Ethernet	0	0	0	0	0	0	0	0d 0h 29m 52s	1
VolP	-	Pa	age: 1, Items:	1 - 2									
Local Services	-												
Maintenance	-												
External Reporting	-												
Monitoring	-												
Internal Log													
IPSec													
ISDN/Modem	-												
Interfaces													

Fig. 17: Monitoring -> Interfaces -> Statistics

You change the state of the interface by pressing the $\textcircled{\bullet}$ button or $\textcircled{\bullet}$ button in the Action column. Press the $\textcircled{\bullet}$ 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 Configura- tion -> <bri2-0 (te)<="" td=""><td>Aktiviert</td></bri2-0>	Aktiviert
Result of Autoconfiguration	Physical Interfaces -> ISDN Ports -> ISDN Configura- tion -> <bri2-0 (te)<="" th=""><th>Port Usage: Dialup (Euro ISDN), ISDN Configuration Type: Point-to-multipoint</th></bri2-0>	Port Usage: Dialup (Euro ISDN), ISDN Configuration Type: Point-to-multipoint

MSN Configuration

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

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

Configuring the internal ISDN interface

Field	Menu	Value
Port Usage	Physical Interfaces -> ISDN Ports -> ISDN Configura- tion -> <bri2-3 (nt)<="" td=""><td>Dialup (Euro ISDN)</td></bri2-3>	Dialup (Euro ISDN)
ISDN Configuration Type	Physical Interfaces -> ISDN Ports -> ISDN Configura- tion -> <bri2-3 (nt)<="" td=""><td>Point-to-multipoint</td></bri2-3>	Point-to-multipoint

MSN Configuration

Field	Menu	Value
ISDN Port	Physical Interfaces -> ISDN Ports -> MSN Configura- tion -> New	bri2-3
Service	Physical Interfaces -> ISDN Ports -> MSN Configura- tion -> New	e.g. <i>PPP</i> (routing)
MSN	Physical Interfaces -> ISDN Ports -> MSN Configura- tion -> New	e.g. 999999
MSN Recognition	Physical Interfaces -> ISDN Ports -> MSN Configura- tion -> New	Right to Left
Service attribute	Physical Interfaces -> ISDN Ports -> MSN Configura- tion -> 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. <i>sipgate</i>
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. <i>sipgate.de</i>
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-> Ad- vanced Settings	
Echo cancellation	VoIP -> Media Gateway -> SIP Accounts -> New-> Ad- vanced Settings	Aktiviert
Comfort Noise Generation (CNG)	VoIP -> Media Gateway -> SIP Accounts -> New-> Ad- vanced Settings	Aktiviert
Packet Size	VoIP -> Media Gateway -> SIP Accounts -> New-> Ad- vanced 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-> Ad- vanced Settings	Default
Sort Order	VoIP -> Media Gateway ->	G.711 uLaw, G.711 aLaw,

Field	Menu	Value
	Extensions -> New-> Ad- vanced Settings	G.729, DTMF Outband
Echo cancellation	VoIP -> Media Gateway -> Extensions -> New-> Ad- vanced Settings	Aktiviert
Comfort Noise Generation (CNG)	VoIP -> Media Gateway -> Extensions -> New-> Ad- vanced Settings	Aktiviert
Packet Size	VoIP -> Media Gateway -> Extensions -> New-> Ad- vanced 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. <i>sipgate</i>
Administrative Status	VoIP -> Media Gateway -> Call Routing -> New	Enable

Field	Menu	Value
Туре	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 Inter- faces->New	e.g . en1-0
Control Mode	WAN->Real Time Jitter Control->Controlled Inter- faces->New	Controlled RTP only
Maximum Upload Speed	WAN->Real Time Jitter Control->Controlled Inter- faces->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).

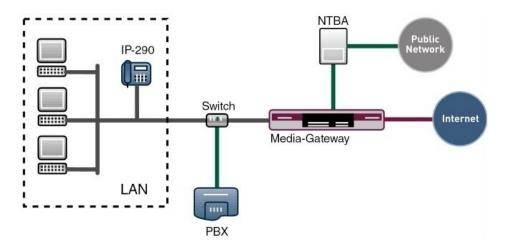


Fig. 18: Example scenario

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 -> <bri>2-0 (TE) .

Save configuration		ISDN Configuration MSN Configuration		
Physical Interfaces	Basic Parameters			
AUX Ethernet Ports	Port Name	bri2-0 (TE)		
ISDN Ports	Autoconfiguration on Bootup	Enabled		
LAN 🔻	Port Usage	Dialup (Euro ISDN) 🔽		
Routing 👻	ISDN Configuration Type	○ Point-to-Multipoint ⑧ Point-to-Point		
VAN -	Subscriber Number			
Firewall • VolP •		Advanced Settings		
Local Services 🗸		OK Cancel		

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

······································				
Field	Meaning			
Port Name	Shows the name of the ISDN port.			
Autoconfiguration on Bootup	Here, select whether the ISDN switch type should be automatic- ally recognised.			
Result of Autoconfigura- tion	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 .			

Relevant fields in the ISDN Configuration menu

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.

Basic Parameters Description	[isdn_TE					
	lisdn_TE					
Description	isdn_TE					
SDN Mode	external V					
obiv mode						
	♥ bri2-0					
	⊯ bri2-1					
	OK Cancel					

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. <i>External</i> : 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.

g D Ac	asic Parameters escription					
g · A	escription					
		asterisk				
T	dministrative Status	✓ Enabled				
- II	runk Mode	O off O Client ● Server				
I TR	ealm					
^	rotocol	UDP Port: 5060				
cation Level Gateway						
	ser Name	R4100				
Services · Au	uthentication ID					
	assword	geheim				
ring 👻 R	egistration	✓ Enabled				
E	xpire Time	600 sec				
	Trunk Settings					
	SIP Header Field(s) for Caller Address P-Preferred					
S	ubscriber Number					
		Advanced Settings				
Ca	odec Settings					
C	odec Proposal Sequence	● Default ○ Quality ○ Low Bandwidth ○ High Bandwidth				
	NY 1	☑ G.711 uLaw ☑ G.711 aLaw ☑ G.729				
5	ort Order	G.726-32 G.726-24 G.726-16 DTMF Outband				
V	Voice Quality Settings					
E	cho Cancellation	Enabled				
C	omfort Noise Generation	Enabled				
P	acket Size	30 ms				

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 <i>Server</i> , the me- dia 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 pro- vider. 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 pro- vider 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 authentica- tion. You must enter this value here. Maximum number of char- acters: 40.
Registration	Enables or disables the SIP REGISTER registration mechan- ism.
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) ad- dress is sent for outgoing calls. Select <i>P-Preferred</i> . 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 outgo- ing 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.

Field	Meaning
Codec Proposal Se- quence	Determine the order in which the codecs are offered for use by the media gateway. If the first codec cannot be applied, an at- tempt is made to use the second codec, and so on. Set Codec Proposal Sequence to <i>default</i> . The codec in the first posi- tion 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 co- decs 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 <i>Enabled</i> is selected, echo feedback is suppressed.
Comfort Noise Genera- tion (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.

Relevant fields in the menu Advanced Settings

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.

Save configuration		Extensions SIP A	counts	Call Routing	CLID Translation	Call Translation	Options
System Management	-						
Physical Interfaces	-						
LAN	-	Basic Parameters					
Routing	-	Session Border Controller Mo	ode O	ff 💌			
WAN	•	Media Stream Termination		Enabled			
VPN	-						
Firewall	-	Default Drop Extension					
VolP		Dial Latency	5	Sec	onds		
Application Level Gate	way		,				
Media Gateway				Ad∨anc	ed Settings		
Local Services	-		(OK	Cancel)	
Maintenance	-			UN	Cancer	/	



Relevant fields in the Options menu				
Field	Meaning			
Session Border Control- ler 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 ac- count, the call routing is handled by the session border con- troller, 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 gate- way 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).			
	• <i>Off</i> : 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 Termina- tion	Determines how RTP sessions are controlled by the system.			
	• Enabled: RTP sessions are terminated on the media gate- way, 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			

Relevant fields in the Options menu

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 ter- mination. The RTP data packets can be routed in complex networks and thus also via other gateways.
	• <i>Disabled</i> (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.

Save configuration	Extensions	SIP Accounts	Call Routing	CLID Translation	Call Translation	Options	
System Management 🛛 👻			-				
Physical Interfaces 🔹 👻	Basic Parameters						
LAN 🔫	Description	6	sterisk				
Routing -	Description	I					
WAN -	Administrative Status	:	Enable				
VPN -	Туре	Т	Trunk 💌				
Firewall 👻	Calling Line	A	ny 🔽				
VoIP 🔺	Calling Address						
Application Level Gateway	Calling Address	I					
Media Gateway	Called Address	11)*				
Local Services 👻	Routing Rule						
Maintenance 👻							
External Reporting 🚽	Trunk Line	6	sterisk 🚩				
Monitoring 👻	Called Address Tran	slation					
		(ок	Cancel)		

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.
Туре	Select <i>Trunk</i> 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 pointto-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.

Save configuration	Extensions	SIP Accounts	Call Routing	CLID Translation	Call Translatio	n Options
System Management 🔹						
Physical Interfaces 🔹	Basic Parameters					
LAN 👻	Description		outgoing_asterisk			
Routing 🗸 🗸	Description		Julgoing_astensk			
WAN -	Administrative Status		Enable			
VPN -	Туре	[External 💌			
Firewall 👻	Calling Line	[asterisk 💌			
VolP 🔺	Calling Address	Г				
Application Level Gateway	Calling Address					
Media Gateway	Called Address		•			
Local Services -	Priority Line	Called Addres	s Translation		Status Ac	tion
Maintenance 👻	1 -				0 1	🕨 💼 🖌
External Reporting 🚽 👻	Add)				
Monitoring 👻	Routing Rule					
	Priority 1					
	Administrative Status 🛛 🗹 Enable					
	Outbound Line bri2-0 💌					
	Called Address Translation					
				pply		
			ок	Cancel)	

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

Relevant fields	s in the	Call	Routing	menu
------------------------	----------	------	---------	------

Field	Meaning
Description	Here you give the entry a name.
Administrative Status	The entry is used with enabled.
Туре	Select <i>External</i> for calls that are to be routed as outgoing ex- ternal 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 di- alled 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.

Save configuration	Extensions	SIP Accounts	Call Routing	CLID Translation	Call Translation	Options	
System Management 🔹 🔻 🔻							
hysical Interfaces 🔹 👻	Basic Parameters						
AN 👻	Description	Description asterisk->ISDN					
touting 👻	Description	Jus					
VAN -	Direction	0	utgoing 💌				
PN 👻	Associated Line	b	ri2-0 🗸				
rewall 👻	Local Address	10	1?				
olP 🔺							
Application Level Gateway	External Address	09	117660069?				
Media Gateway							
.ocal Services 🔹 👻		(ок	Cancel)		

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 sig- nalled 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.

Save configuration	Extensions SIP Ad	counts Call Routing CLID Translation Call Translation Options		
System Management 🔹 🔻				
Physical Interfaces 🔹 👻	Basic Parameters			
LAN 👻	Description	ISDN->asterisk		
Routing 👻	Description			
WAN 👻	Direction	Incoming 💌		
/PN 👻	Associated Line	bri2-0		
irewall 👻	Local Address	10?		
/oIP 🔺				
Application Level Gateway	External Address	7660069?		
Media Gateway				
Local Services 🔹 👻		OK Cancel		

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 sig- nalled 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] 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 canteinvite=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 [1041 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 [1071 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 => OX.,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 => _OX., 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=$(SIP HEADER
(P-Preferred-Identity):5))
                                                           :SRC ADDRESS is filled with P-Prefered-Identity and
                                                           Scolums at the beginning will be removed
exten => R4100, n, SET(DEST ADDRESS=$(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,0,1)));content DEST ADDRESS which begins with a "0" is removed
exten => _R4100,n,SET(SRC_ADDRESS=$(CUT(SRC_ADDRESS,0,1))) ; content of SRC_ADDRESS which begins with "0" will be removed
exten => _R4100, n, SET(CALLERID(num) =$(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,0,1))) :content of SRC_ADDRESS which begins with "0" 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

Field	Menu	Value
Port Usage	Physical Interfaces -> ISDN Ports -> <bri2-0 (te)=""></bri2-0>	Dialup (Euro ISDN)
ISDN Configuration Type	Physical Interfaces -> ISDN Ports -> <bri2-0 (te)=""></bri2-0>	Point-to-point
Port Usage	Physical Interfaces -> ISDN Ports -> <bri2-1 (te)=""></bri2-1>	Dialup (Euro ISDN)
ISDN Configuration Type	Physical Interfaces -> ISDN Ports -> <bri2-1 (te)=""></bri2-1>	Point-to-point

Configuring the external ISDN interface

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. <i>R4100</i>
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 Ad- vanced Settings	e.g. Standard
Echo cancellation	VoIP -> Media Gateway -> SIP Accounts ->New Ad- vanced Settings	Aktiviert
Comfort Noise Generation	VoIP -> Media Gateway -> SIP Accounts ->New Ad- vanced Settings	Aktiviert

Field	Menu	Value
Packet Size	VoIP -> Media Gateway -> SIP Accounts ->New Ad- vanced 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
Туре	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
Туре	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. <i>ISDN->asterisk</i>
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.

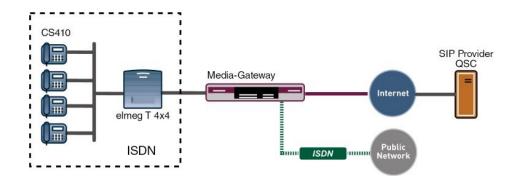


Fig. 27: Example scenario

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 -> <bri>2-0 (TE)

Save configuration		ISDN Configuration MSN Configuration	
System Management	-		
Physical Interfaces	Basic Parameters		
AUX	Port Name	bri2-0 (TE)	
Ethernet Ports			
ISDN Ports	Autoconfiguration on Bootup	Enabled	
LAN	 Result of Autoconfiguration 	Port Usage: Dialup (Euro ISDN), ISDN Configuration Type: Point-to-Poin	
Routing	Port Usage	Dialup (Euro ISDN) 🗸	
WAN	-		
VPN	 ISDN Configuration Type 	O Point-to-Multipoint Point-to-Point	
Firewall	•	Advanced Settings	
VolP	•		
Local Services	•	OK Cancel	

Fig. 28: Physical Interfaces -> ISDN Ports -> ISDN Configuration -> <bri>2-0 (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	JOM	External (factory default setting)
Internal/external switching	J1M	internal
Power supply	JOP	Off (factory default setting)
Power supply	J1P	On
Terminator	JOT	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 *Dia-lup* (*Euro ISDN*) point-to-point (*NT Mode*) on the media gateway.

(1) Go to Physical Interfaces -> ISDN Ports -> ISDN Configuration -> <bri>2-1 (NT)

Save configuration		ISDN Configuration MSN Configuration
System Management 🔹		
Physical Interfaces 🔺	Basic Parameters	
AUX	Port Name	bri2-1 (NT)
Ethernet Ports		
ISDN Ports	Port Usage	Dialup (Euro ISDN) 💌
LAN 👻	ISDN Configuration Type	◯ Point-to-Multipoint . ● Point-to-Point
Routing 👻		
WAN -	Subscriber Number	
VPN -		Advanced Settings
Firewall 🔹 👻		
VolP 🔻		

Fig. 29: Physical Interfaces -> ISDN Ports -> ISDN Configuration -> <bri>2-1 (NT)

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 <i>Dialup</i> (<i>Euro-ISDN</i>).
ISDN Configuration Type	Here, select ISDN access configuration Point-to-Point.

Relevant fields in the ISDN Configuration menu

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.

Save configuration	Extensions SIP Acco	ounts Call Routing CLID Translation Call Translation Options				
/stem Management 💿 👻						
ysical Interfaces 🔹 👻	Basic Parameters					
AN 👻	Description	lasc				
outing 👻						
/AN 👻	Administrative Status	✓ Enabled				
PN 👻	Trunk Mode	○ Off Client Server				
rewall 👻	Registrar	sip.gsc.de				
olP 🔺	Outbound Proxy					
Application Level Gateway						
Media Gateway cal Services 👻	Realm					
aintenance 🔹	Protocol	UDP v Port: 5060				
dernal Reporting 👻	User Name	06227899154				
onitoring 👻	Authentication ID					
	Password	geheim				
	Registration	✓ Enabled				
	Expire Time	600 sec				
	Trunk Settings					
	SIP Header Field(s) for Caller Ad	ddress Display and User Name 👻				
	Advanced Settings					
	Codec Settings					
	Codec Proposal Sequence	◯ Default ◯ Quality ④ Low Bandwidth ◯ High Bandwidth				
	Sort Order	☑ G.711 uLaw ☑ G.711 aLaw ☑ G.729 ☐ G.726-40 ☐ G.726-32 ☐ G.726-24 ☐ G.726-16 ☐ DTMF Outband				
	Voice Quality Settings					
	Echo Cancellation	✓ Enabled				
	Comfort Noise Generation	✓ Enabled				
	Packet Size	30 ms				
		OK Cancel				

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 <i>Client</i> , 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 authentica- tion. You must enter this value here. Maximum number of char- acters: 40.		
Registration	Enables or disables the SIP REGISTER registration mechan- ism.		
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) ad- dress is sent for outgoing calls.		
	Select <i>Display and User Name</i> . 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 Se-	Determine the order in which the codecs are offered for use by
quence	Determine the order in which the codecs are offered for use b

Field	Meaning
	the media gateway. If the first codec cannot be used, the second is tried and so on. Select <i>Low Bandwidth</i> . 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 <i>Enabled</i> is selected, echo feedback is suppressed.
Comfort Noise Genera- tion (CNG)	Specify whether Comfort Noise Generation should be used. The slight comfort noise generation prevents subscribers from think- ing 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 **O**. The status of the VoIP connection is changed by pressing the **I** button or **I** button in the **Action** column.

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

Save configuration	J.	Extensions	SIP Accounts	Call Routing	CLID Translation	Call Tran	nslation	Options	
System Management	-								-
Physical Interfaces	-	Description	Registrar / Outk	ound Proxy	Protocol	Status	Action		T
LAN	-	QSC	sip.gsc.de		UDP	0	1+		
Routing	-			_					
WAN	-				New)				
VPN	-								
Firewall	-								
VolP									
Application Level Gatev	vay								
Media Gateway									



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.

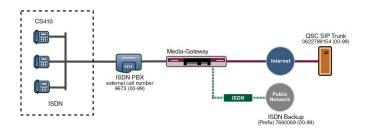


Fig. 32: Extension numbers

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.

Save configuration	Extensions SIP Account	ts Call Routing CLID Translation Call Translation Options
System Management 🔹 👻		
Physical Interfaces 🔹 👻		
LAN 🔫	Basic Parameters	
Routing 🗾 👻	Session Border Controller Mode	Off 👻
WAN -	Media Stream Termination	V Enabled
VPN -	Wedia Stream remination	
Firewall 👻	Default Drop Extension	
VolP 🔺	Dial Latency	5 Seconds
Application Level Gateway		
Media Gateway		Advanced Settings
Local Services 🔹 👻		
Maintenance 🗸 🗸		

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

Relevant fields in the Options menu

Field	Meaning
Session Border Control- ler 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 gate- way 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).
	• <i>Off</i> : 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 pro- vider (account), you must configure a corresponding call rout- ing entry. Internal calls (from internal extension to internal ex- tension) that are only to be routed internally do not require an additional call routing entry.
Media Stream Termina-	Determines how RTP sessions are controlled by the system.
	• <i>Enabled</i> : RTP sessions are terminated on the media gate- way, 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.
	• <i>Disabled</i> (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 dial- ling process (sends the SIP INVITE message). This timeout is reset each time that a button is pressed. Default value: <i>5</i> .

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.

Save configuration System Management	Extensions SIP Accourt	nts Call Routing CLID Translation Call Translation Options
Physical Interfaces 🔹 👻	Basic Parameters	
LAN -	Description	PBX->QSC
Routing • WAN •	Direction	Both
VPN -	Associated Line	QSC 🗸
Firewall 🔹	Local Address	9673??
VoIP Application Level Gateway	External Address	06227899154??
Media Gateway Local Services -		OK Cancel

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

Fig. 34: VoIP -:	Media Gates	way -> Call T	ranslation	-> New
------------------	-------------	---------------	------------	--------

Field	Meaning
Description	Give the number translation a name.
Direction	Here you enter the direction to which the entry is to apply. Select <i>Both</i> 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 trans- lated to the External Address .

Relevant fields in the Call Translation menu

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 sig- nalled 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.

Save configuration	Extensions SIP Acco	unts Call Routing CLID Translation Call Translation Options			
Physical Interfaces 🔹 👻	Basic Parameters				
LAN 👻	Description	PBX<->ISDNBackup			
Routing 👻	Desemption				
WAN 👻	Direction	Both 💌			
VPN -	Associated Line	bri2-0 💌			
Firewall 👻	Local Address	9673??			
VolP 🔺					
Application Level Gateway	External Address	7660069??			
Media Gateway					
Local Services 👻		OK Cancel			

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.

Save configuration	Extensions SIP Accou	Ints Call Routing CLID Translation Call Translation Options			
System Management 🔹 👻					
Physical Interfaces 🔹 👻	Basic Parameters				
LAN 🔫	Description	QSC -> PBX			
Routing 🔹	Description	JUSCK-PPDX			
WAN -	Calling Line	QSC 🔽			
VPN 👻	Called Line	Any 💌			
Firewall 👻	Called Address				
VolP 🔺					
Application Level Gateway	Calling Address Translation	<:0>;			
Media Gateway					
Local Services 👻		OK Cancel			

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.

Save configuration	Extensions SIP Accou	Ints Call Routing CLID Translation Call Translation Options			
System Management 🔹 👻					
Physical Interfaces 🔹 🔻	Basic Parameters				
LAN 👻	Description	ISDN↔PBX			
Routing 👻	Description				
WAN 👻	Calling Line	bri2-0 💌			
/PN 👻	Called Line	Any 💌			
irewall 👻 👻	Called Address				
/oIP 🔺					
Application Level Gateway	Calling Address Translation	<0>;			
Media Gateway					
Local Services 👻		OK Cancel			

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

Relevant fields in the CLID Trai	nslation 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 Transla- tion	Transformation rule to be used on the subscriber number
	The calling party number transmitted by the provider is pre- ceded 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.



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.

Save configuration	Extensions	SIP Accounts	Call Routing	CLID Translation	Call Translation	<u>Options</u>	
System Management 💦 🤊							
Physical Interfaces	Basic Parameters						
.AN 🗖	Description	lis	DN PBX				
Routing	- Description	110					
WAN .	Administrative Status		Enable				
VPN .	Туре	Т	Trunk				
Firewall .	Calling Line	A	Any 👻				
VolP	Calling Address						
Application Level Gateway	Culling Address	I					
Media Gateway	Called Address	*	*				
Local Services	Routing Rule						
Maintenance •			:0.0 				
External Reporting	Trunk Line	b	bri2-3 💌				
Monitoring	Called Address Trans	slation					
		(OK	Cancel)		

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.
Туре	Select <i>Trunk</i> for calls that are routed to a PBX behind the me- dia 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 com- pared 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.

Save configuration	Exten	Extensions SIP Accou		Call Routing	CLID Translation	Call Translati	on Options
System Management 🔹 🔻							
Physical Interfaces 🔹 👻	Basic Parameters						
LAN 👻	Description		[Provider			
Routing 👻	· · · ·		,				
WAN -	Administrativ	e Status	÷	🗹 Enable			
VPN -	Туре		[External 💌			
Firewall 👻	Calling Line		[Any 🔽			
VolP 🔺	Calling Addre		1				
Application Level Gateway	Calling Addre	500	. 1				
Media Gateway	Called Addre	ISS		k .			
Local Services 🔹 🔻	Priority	Line	Called Ac	, Address Translation		Status A	Action
Maintenance 🔹 👻	1	bri2-0					t I 💼 😥
External Reporting 🔹 👻	2	-				0	11 💼 🖉
Monitoring 🔹	Add						
	Routing Rule						
	Priority		ſ	2			
	Administrative Status			✓ Enable			
	Outbound Line		I	OSC 🗸			
	Called Address Translation						
				4	Apply		
				ок	Cancel)	

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

Field	Meaning
Description	Here you give the entry a name.
Administrative Status	The entry is used with Enable.
Туре	Select <i>External</i> for calls that are to be routed as outgoing ex- ternal 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.

Relevant fields in the Call Routing menu

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 Gate**way.

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

Save configuration			SIP Proxies	SIP Endpoints			
System Management 🔹 👻							
Physical Interfaces 🔹 👻	Description	Protocol	Port	Low Latency	Status	Action	
LAN 🔫	SIP UDP 5060	UDP	5060	Off	0	+	â 🖉
Routing 🗸 🗸	SIP TCP 5060	TCP	5060	Off	0	1	<u>i</u>
WAN 👻							
VPN 👻		New					
Firewall 👻							
VolP 🔺							
Application Level Gateway							
Media Gateway							

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

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

Go to VoIP -> Application Level Gateway -> <SIP UDP 5060>

Save configuration		SIP Proxies SIP Endpoints
System Management 🔹 👻		
Physical Interfaces 🔹 👻	Basic Parameters	
LAN 🔫	Description	SIP UDP 5060
Routing 👻		
WAN -	Administrative Status	✓ Enabled
VPN -	Protocol	UDP 🗹 Destination Port 5060
Firewall 👻	Session Timeout	7200 sec
VolP 🔺		
Application Level Gateway	Low Latency Transmission	Enabled
Media Gateway		
Local Services 👻		OK Cancel



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 Transmis- sion	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 <i>Enabled</i> , the voice quality is optimised, if <i>Disabled</i> , the voice quality is not optimised.

Relevant fields in the SIP Proxy menu

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.

Save configuration		Controlled Interfaces
Physical Interfaces 🔹		
LAN 🔫	Basic Settings	
Routing 👻	Interface	en1-0 🗸
WAN 🔺	Control Mode	Controlled RTP Streams only
Internet + Dialup	Control Mode	
Leased Line	Maximum Upload Speed	128 kbps
Real Time Jitter Control		
VPN -		OK Cancel
Firewall 👻		

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

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 <i>Controlled RTP Streams only</i> : 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 band-width.

Relevant fields in the Controlled Interfaces menu

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>	bri2-0 (TE)
Autoconfiguration on Bootup	Physical Interfaces -> ISDN Ports -> <bri2-0 (te)=""></bri2-0>	Aktiviert
Result of Autoconfiguration	Physical Interfaces -> ISDN Ports -> <bri2-0 (te)=""></bri2-0>	Port Usage: Dialup (Euro-ISDN), ISDN Configuration Type; Point-to-point
Port Name	Physical interfaces -> ISDN Ports -> <bri2-1 (nt)=""></bri2-1>	bri2-1 (NT)
Port Usage	Physical interfaces -> ISDN Ports -> <bri2-1 (nt)=""></bri2-1>	Dialup (Euro ISDN)
ISDN Configuration Type	Physical interfaces -> ISDN Ports -> <bri2-1 (nt)=""></bri2-1>	Point-to-point

Configuration of SIP accounts

Field	Menu	Value
Description	VoIP -> Media Gateway -> SIP Accounts -> New	e.g. <i>QSC</i>
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. <i>sip.qsc.de</i>
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. 600sec
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-> Ad- vanced Settings	Low Bandwidth
Echo cancellation	VoIP -> Media Gateway -> SIP Accounts -> New-> Ad- vanced Settings	Aktiviert
Comfort Noise Generation	VoIP -> Media Gateway -> SIP Accounts -> New-> Ad- vanced Settings	Aktiviert
Packet Size	VoIP -> Media Gateway -> SIP Accounts -> New-> Ad- vanced 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. <i>PBX<->QSC</i>
Direction	VoIP -> Media Gateway -> Call Translation -> New	e.g. Both
Associated Line	VoIP -> Media Gateway -> Call Translation -> New	e.g. <i>QSC</i>
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. <i>PBX<->ISDNBackup</i>
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. <i>QSC<->PBX</i>
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
Туре	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
Туре	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<br="">5060> 🌠</sip>	e.g. <i>SIP UDP 5060</i>
Administrative Status	VoIP -> Application Level	Aktiviert

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

Field	Menu	Value
	Gateway -> <sip udp<br="">5060> 👔</sip>	
Protocol	VoIP -> Application Level Gateway -> <sip udp<br="">5060> 🍻</sip>	UDP
Destination Port	VoIP -> Application Level Gateway -> <sip udp<br="">5060> 🏹</sip>	5060
Session Timeout	VoIP -> Application Level Gateway -> <sip udp<br="">5060> 🏹</sip>	7200
Low Latency Transmission	VoIP -> Application Level Gateway -> <sip udp<br="">5060> 🌠</sip>	Disabled

Real Time Jitter Control

Field	Menu	Value
Interface	WAN -> Real Time Jitter Control -> Controlled Inter- faces -> New	e.g. en1-0
Control Mode	WAN -> Real Time Jitter Control -> Controlled Inter- faces -> New	Controlled RTP only
Maximum Upload Speed	WAN -> Real Time Jitter Control -> Controlled Inter- faces -> 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.

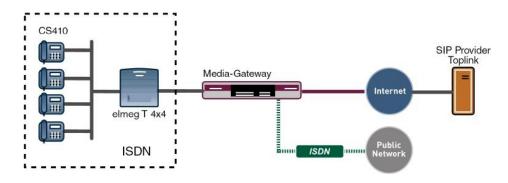


Fig. 43: Example scenario

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) .

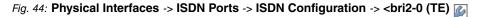
4.2 Configuration

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 -> <bri>2-0 (TE)

Save configuration		ISDN Configuration MSN Configuration	
System Management	-		
Physical Interfaces	Basic Parameters		
AUX Ethernet Ports	Port Name	bri2-0 (TE)	
ISDN Ports	Autoconfiguration on Bootup	✓ Enabled	
LAN	Result of Autoconfiguration	Port Usage: Dialup (Euro ISDN), ISDN Configuration Type: Point-to-Point	
Routing	Port Usage	Dialup (Euro ISDN) 🐱	
WAN VPN	ISDN Configuration Type	O Point-to-Multipoint O Point-to-Point	
Firewall		Advanced Settings	
VolP			
Local Services	•		



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	JOM	External (factory default setting)
Internal/external switching	J1M	internal
Power supply	J0P	Off (factory default setting)
Power supply	J1P	On
Terminator	JOT	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 *Dia-lup* (*Euro ISDN*) point-to-point (NT Mode) on the media gateway.

(1) Go to Physical Interfaces -> ISDN Ports -> ISDN Configuration -> <bri>2-1 (NT)

Save configuration		ISDN Configuration MSN Configuration		
System Management 🔹				
Physical Interfaces	Basic Parameters			
AUX	Port Name	bri2-1 (NT)		
Ethernet Ports				
ISDN Ports	PortUsage	Dialup (Euro ISDN) 💌		
LAN 🔫	ISDN Configuration Type	○ Point-to-Multipoint ⑧ Point-to-Point		
Routing 👻	Subscriber Number			
WAN -	Subscriber Number			
VPN -		Advanced Settings		
Firewall 🔹				
VoIP -		OK Cancel		

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

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.

Relevant fields in the ISDN Configuration menu

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.

Save configuration	Extensions	SIP Accounts	Call Routin	g <u>CLID Tran</u>	islation C	Call Translation	<u>Options</u>
System Management 🔹							
Physical Interfaces 🔹	Basic Parameters						
.AN 🔻	Description	T	oplink				
Routing 👻	Administrative Status	2	Enabled				
WAN -		- 7.5	1004000000000	A			
VPN -	Trunk Mode		Off Client				
Firewall 🔹	Registrar	to	plink-voice.de				
VoIP Application Level Gateway	Outbound Proxy	Г					
Media Gateway	Realm	Г.					
ocal Services 👻	Realm	J					
Maintenance 🗸 🗸	Protocol	L	DP Port: 5	060			
External Reporting 🗾 👻	User Name		1093941000				
Monitoring 👻	Authentication ID	Г					
	Password	g	geheim				
	Registration		✓ Enabled				
	Expire Time	6	00	sec			
	Trunk Settings						
	SIP Header Field(s) for Caller Address P-Preferred						
	Advanced Settings						
	Codec Settings						
	Codec Proposal Sec	juence 🤇	⊃Default ○Qu	ality 💿 Low Ba	ndwidth 🔿 H	ligh Bandwidth	
	Sort Order	[🗹 G.711 uLaw	🗹 G.711 aLaw	🗹 G.729	G.726-40	
		[G.726-32	G.726-24	G.726-1	6 🔲 DTMF Outban	d
	Voice Quality Settings						
	Echo Cancellation	Echo Cancellation					
	Comfort Noise Gene	ration	Enabled				
	Packet Size	3	30 ms				
	OK Cancel						

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 <i>Client</i> , 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 pro- vider 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 authentica- tion. You must enter this value here. Maximum number of char- acters: 40.
Registration	Enables or disables the SIP REGISTER registration mechan- ism.
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) ad- dress is sent for outgoing calls.
	Select <i>P</i> - <i>Preferred</i> . 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 Se- quence	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 <i>Low Bandwidth</i> . 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 <i>Enabled</i> is selected, echo feedback is suppressed.
Comfort Noise Genera- tion (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.

Save configuration		Extensions	SIP Accounts	Call Routing	CLID Translation	Call Tran	slation	Options	
System Management	•								
Physical Interfaces	-	Description	Registrar / Outk	ound Proxy	Protocol	Status	Action		T
LAN	-	Toplink	toplink-voice.c	le	UDP	0	1+	Ê	
Routing	-								-
WAN				<u> </u>	New)				
VPN	+								
Firewall	-								
VolP	-								
Application Level Gatew	ay								
Media Gateway									

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.

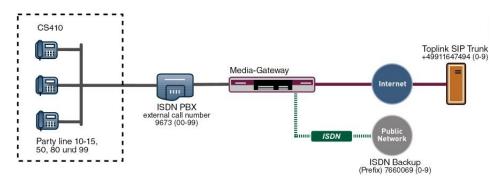


Fig. 48: Extension numbers

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.

Save configuration	Extensions SIP Accou	unts Call Routing CLID Translation Call Translation Options
System Management 🛛 👻		
Physical Interfaces 🔹 👻	Basic Parameters	
LAN 👻	Description	1?<->Toplink
Routing 👻	Description	
WAN -	Direction	Both 🕑
VPN 🔫	Associated Line	Toplink 💌
Firewall 👻	Local Address	96731?
VolP 🔺		
Application Level Gateway	External Address	+49911647494?
Media Gateway		
Local Services 👻		OK Cancel

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 <i>Both</i> 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 sig- nalled 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.

Save configuration	Extensions SIP Ac	counts Call Routing CLID Translation Call Translation Options
Physical Interfaces 🔹 👻	Basic Parameters	
LAN 🔫	Description	1?<->ISDNBackup
Routing 🔹	Description	
WAN 👻	Direction	Both 💌
VPN 👻	Associated Line	bri2-0 💌
Firewall 🔹 👻	Local Address	96731?
VolP 🔺		
Application Level Gateway	External Address	7660069?
Media Gateway Local Services 👻		OK Cancel

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.

<u>Save configuration</u> System Management →	Extensions SIP Accou	ants Call Routing CLID Translation Call Translation Options
Physical Interfaces 🔹 👻	Basic Parameters	
LAN 👻	Description	50<->Toplink
Routing 🔹		
WAN -	Direction	Both 💌
VPN -	Associated Line	Toplink 💌
Firewall 🔹	Local Address	967350
VolP 🔺	E terre et é délace e	400110474040
Application Level Gateway	External Address	+499116474946
Media Gateway		
Local Services 🔹 👻		OK Cancel

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.
```

Save configuration	- F	Extensions	SIP Accounts	Call Routing	CLID Translation	Call Translation	Options
System Management							
Physical Interfaces	Basic	Parameters					
LAN	Desc	ription	50)<->ISDNbackup			
Routing	-	anpuon					
WAN	Direc	tion	B	oth 💌			
VPN	- Asso	ciated Line	b	ri2-0 💌			
Firewall	Local	Address	96	67350			
VolP	•						_
Application Level Gateway	Extern	nal Address	76	6600696			
Media Gateway							
Local Services	-		(ок	Cancel)	

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.

Save configuration	Extensions	SIP Accounts	Call Routing	CLID Translation	Call Translation	Options	
System Management 🔹 👻			-				
Physical Interfaces 🔹 👻	Basic Parameters						
LAN 👻	Description	80)<->Toplink				
Routing 👻	Description						
WAN 👻	Direction	B	Both 💌				
VPN 👻	Associated Line	Т	Toplink 💌				
Firewall 🔹 👻	Local Address	98	67380				
VolP 🔺		/				_	
Application Level Gateway	External Address	+4	499116474947				
Media Gateway							
Local Services 🔹 👻		(ок	Cancel)		

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.

Save configuration	Extensions SIP Accourt	nts Call Routing CLID Translation Call Translation Options				
System Management 🔹 👻						
Physical Interfaces 🔹 👻	Basic Parameters					
LAN -	Description	80<->ISDNBackup				
Routing 👻	Description					
WAN -	Direction	Both 💌				
VPN -	Associated Line	bri2-0 💌				
Firewall 👻	Local Address	967380				
VolP 🔺						
Application Level Gateway	External Address	76600697				
Media Gateway						
Local Services 👻		OK Cancel				

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.

Save configuration System Management	Extensions	SIP Accounts	Call Routing	CLID Translation	Call Translation	<u>Options</u>	
Physical Interfaces 👻	Basic Parameters						
LAN 🔫	Description	90	I <-> Toplink				
Routing 🔹	· · · · · · · · · · · · · · · · · · ·						
WAN -	Direction	В	Both 💌				
VPN -	Associated Line	Т	oplink 💌				
Firewall 🔹	Local Address	98	7399				
VoIP Application Level Gateway	External Address	+4	199116474948				
Media Gateway Local Services 🗸	-	(ок	Cancel)		

Fig. 55: VolP -> 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.

Save configuration	Extensions	SIP Accounts	Call Routing	CLID Translation	Call Translation	Options
System Management 🛛 👻						
Physical Interfaces 🔹 👻	Basic Parameters					
LAN 🔫	Description		->ISDNBackup			
Routing 👻	Description	195				
WAN +	Direction	В	oth 💌			
VPN 👻	Associated Line	b	ri2-0 🔽			
Firewall 👻	Local Address	98	57399			
VolP 🔺		/				_
Application Level Gateway	External Address	78	6600698			
Media Gateway						
Local Services 👻		(OK	Cancel)	

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

The complete configuration then looks like this:

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

Save configuration	Extensions	SIP Accounts	Call Routing	CLID Tran	slation Call	Translation	Options	
System Management 🔹 👻								
Physical Interfaces 🔹 👻	Description	Local Addre	ess Externa	al Address	Direction	Associated Lin	e	Т
.AN 👻	1?<->Toplink	96731?	+4991	1647494?	Both	Toplink	1	1
touting 👻	1?<->ISDNBackup	96731?	76600	69?	Both	bri2-0	1	1
VAN -	50<->Toplink	967350	+4991	16474946	Both	Toplink	1	1
/PN 👻	50<->ISDNbackup	967350	76600	696	Both	bri2-0	1	1
irewall 👻	80<->Toplink	967380	+4991	16474947	Both	Toplink	â	
/oIP	80<->ISDNBackup	967380	76600	697	Both	bri2-0	Ê	1
Application Level Gateway	99≺->Toplink	967399	+4991	16474948	Both	Toplink	1	1
Media Gateway	99<->ISDNBackup	967399	76600	698	Both	bri2-0	Ô	1
.ocal Services 👻								
Maintenance 👻				New)				

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.

Save configuration System Management	Extensions SIP Accou	nts Call Routing CLID Translation Call Translation Options
Physical Interfaces 🔹 👻	Basic Parameters	
LAN 👻	Description	Toplink->PBX
Routing 👻		
WAN 🔫	Calling Line	Toplink 💌
VPN 🔻	Called Line	Any 💌
Firewall 👻	Called Address	
VolP 🔺		
Application Level Gateway	Calling Address Translation	(<0);
Media Gateway Local Services -		OK Cancel

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.

Save configuration	Extensions SIP Acco	unts Call Routing CLID Translation Call Translation Options
System Management 🛛 👻		
Physical Interfaces 🔹 👻	Basic Parameters	
LAN 👻	Description	ISDN->PBX
Routing 👻	Description	ISDN-9PBA
WAN 👻	Calling Line	bri2-0 💌
VPN 👻	Called Line	Any 💌
Firewall 🔹 👻	Called Address	
VolP 🔺		
Application Level Gateway	Calling Address Translation	<:0>;
Media Gateway		·
Local Services 👻		OK Cancel

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

Relevant fields in the CL	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 pre- ceded 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.

Save configuration	Extensions	SIP Accounts	Call Routing	CLID Translation	Call Translation	Options
System Management 🔹 🔻						
Physical Interfaces 🔹 🔻	Basic Parameters					
LAN 👻	Description	lis	DN_PBX			
Routing 🔹						
WAN 👻	Administrative Status		Enable			
VPN 👻	Туре	Т	runk 💌			
Firewall 🗸 👻	Calling Line	A	ny 🔽			
VolP 🔺	Calling Address					
Application Level Gateway	Caning Address	I				
Media Gateway	Called Address	96	673*			
Local Services 🔹 👻	Routing Rule	r Routing Rule				
Maintenance 🔹 🔻			:0 0 v			
External Reporting 🗾 👻	Trunk Line	b	ri2-3 💌			
Monitoring 🗸 🗸	Called Address Tran	slation				
		(ок	Cancel)	

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.
Туре	Select <i>Trunk</i> for calls that are routed to a PBX behind the me- dia 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 com- pared 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.

Save configuration	E	tensions	SIP Account	s Call Routing	CLID Translation	Call Translation	options
System Management 🔹 👻							
Physical Interfaces 🔹 👻	Basic Parameters						
LAN 🔫	Descrip	ation		Provider			
Routing 🗾 👻							
WAN -	Adminis	strative Status	3	🗹 Enable			
VPN -	Туре			External 🚩			
Firewall 👻	Calling	Line		Any 🔽			
VoIP 🔺	Calling Address						
Application Level Gateway	Canny	Audress					
Media Gateway	Called Address			*			
Local Services 🔹 🔻	Priority	Line	Called A	ddress Translation		Status Act	ion
Maintenance 👻	1	bri2-0				0 1	J î 🖉
External Reporting 🗾 👻	2	-				0 1	
Monitoring 👻	Add						
	Routing F	Rule					
	Priority			2			
	Adminis	strative Status	3	🗹 Enable			
	Outbound Line			bri2-0 💌			
	Called	Address Tran	slation				
					Apply		
				ОК	Cancel)	



Field	Meaning
Description	Here you give the entry a name.
Administrative Status	The entry is used with Enable.
Туре	Select <i>External</i> for calls that are to be routed as outgoing ex- ternal 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.

Relevant fields in the Call Routing menu

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.

Field	Meaning
Priority	Determines the order of the filter rules, starting with $ \ensuremath{ \ansuremath{$
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.

Relevant fields in the Routing Rule menu

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.

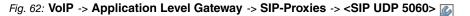
In our example the media gateway is connected to the internet over an ADSL path. For security reasons **Network Address Translation** and the **Stateful Inspection Firewall** have been enabled. The **Application Level Gateway** must be enabled to prevent any negative interference by the firewall to VoIP 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> [2].

Save configuration		SIP Proxies SIP Endpoints
Physical Interfaces 🔹 👻	Basic Parameters	
LAN 👻	Description	SIP UDP 5060
Routing - WAN -	Administrative Status	✓ Enabled
/PN v	Protocol	UDP V Destination Port 5060
irewall 👻	Session Timeout	7200 sec
Application Level Gateway	Low Latency Transmission	Enabled
Media Gateway Local Services -		OK Cancel



Relevant fields in the SIP Proxies menu

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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 Transmis- sion	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 connec- tion.
	If <i>Enabled</i> , the voice quality is optimised, if <i>Disabled</i> , 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.

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ote

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.

Save configuration		Controlled Interfaces
System Management 🔹 👻		
Physical Interfaces 🔹 👻		
LAN 👻	Basic Settings	
Routing 🗾 👻	Interface	en1-0
WAN 🔺	Control Mode	Controlled RTP Streams only
Internet + Dialup		
Leased Line	Maximum Upload Speed	128 kbps
Real Time Jitter Control		
VPN -		OK Cancel



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 <i>Controlled RTP Streams only</i> : By means of the data routed through the media gateway, the system recognises VoIP data traffic and optimises the voice transmission. This set- ting 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.

Relevant fields in the Controlled Interfaces menu

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>	bri2-0 (TE)
Autoconfiguration on Bootup	Physical Interfaces -> ISDN Ports -> <bri2-0 (te)=""></bri2-0>	Aktiviert
Result of Autoconfiguration	Physical Interfaces -> ISDN Ports -> <bri2-0 (te)=""></bri2-0>	Port Usage: Dialup (Euro-ISDN), ISDN Configuration Type; Point-to-point
Port Name	Physical interfaces -> ISDN Ports -> <bri2-1 (nt)=""></bri2-1>	bri2-1 (NT)
Port Usage	Physical interfaces -> ISDN Ports -> <bri2-1 (nt)=""></bri2-1>	Dialup (Euro ISDN)
ISDN Configuration Type	Physical interfaces -> ISDN Ports -> <bri2-1 (nt)=""></bri2-1>	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. <i>D1093941000</i>
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 Ad- vanced Settings	Low Bandwidth
Echo cancellation	VoIP -> Media Gateway -> SIP Accounts ->New Ad- vanced Settings	Aktiviert
Comfort Noise Generation	VoIP -> Media Gateway -> SIP Accounts ->New Ad- vanced Settings	Aktiviert
Packet Size	VoIP -> Media Gateway -> SIP Accounts ->New Ad- vanced 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?

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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
	lilond	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>;

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

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
Туре	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
Туре	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<br="">5060> 🌠</sip>	e.g. <i>SIP UDP 5060</i>
Administrative Status	VoIP -> Application Level Gateway -> <sip udp<br="">5060> 🌠</sip>	Aktiviert
Protocol	VoIP -> Application Level Gateway -> <sip udp<br="">5060> 🌠</sip>	UDP
Destination Port	VoIP -> Application Level Gateway -> <sip udp<br="">5060> 🌠</sip>	5060
Session Timeout	VoIP -> Application Level Gateway -> <sip udp<br="">5060> 🌠</sip>	7200
Low Latency Transmission	VoIP -> Application Level Gateway -> <sip udp<br="">5060> 🌠</sip>	Disabled

Real Time Jitter Control

Field	Menu	Value
Interface	WAN -> Real Time Jitter Control -> Controlled Inter- faces -> New	e.g. <i>en1-0</i>
Control Mode	WAN -> Real Time Jitter Control -> Controlled Inter- faces -> New	Controlled RTP only

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

Field	Menu	Value
	WAN -> Real Time Jitter Control -> Controlled Inter- faces -> 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.

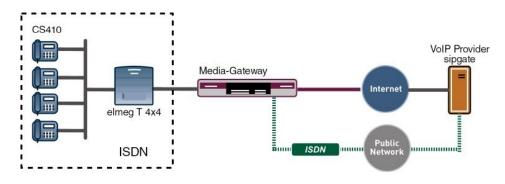


Fig. 64: Example scenario

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:

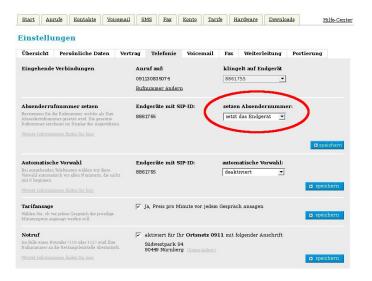


Fig. 65: Settings

5.1.2 Configuring the ISDN interfaces

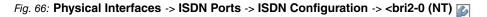
The external ISDN 50 port on the PBX (point-to-multipoint connection) is connected to a BRI port on the media gateway. The ISDN mode for this BRI port must be changed to NT *Mode*. To do this, the housing on the media gateway must be opened. The link plugs for the ISDN-0 and ISDN-1 interfaces can be 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 typecan be set on Dialup (Euro-ISDN) Point-
to-Multipoint.
```

(1) Go to Physical Interfaces -> ISDN Ports -> ISDN Configuration -> <bri>2-0 (NT)

Save configuration		ISDN Configuration MSN Configuration
Physical Interfaces	Basic Parameters	
Ethernet Ports	Port Name	bri2-0 (NT)
ISDN Ports	Autoconfiguration on Bootup	Enabled
LAN 👻	Port Usage	Dialup (Euro ISDN) 🔽
Routing 👻	ISDN Configuration Type	Point-to-Multipoint O Point-to-Point
WAN -	Tobre conliguration type	22 Point-to-Multipoint C Point-to-Point
VPN -		Advanced Settings
Firewall 👻		
VolP -		Un Canver



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 the ISDN access configuration <i>Point-to-Multipoint</i> .

Relevant fields in the ISDN Configuration menu

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 -> <bri>2-1 (TE)

Save configuration		ISDN Configuration MSN Configuration
System Management 🔹 👻		
Physical Interfaces	Basic Parameters	
AUX	Port Name	bri2-1 (TE)
Ethernet Ports		
ISDN Ports	Autoconfiguration on Bootup	✓ Enabled
AN 👻	Result of Autoconfiguration	Port Usage: Dialup (Euro ISDN), ISDN Configuration Type: Point-to-Multipoint
outing 👻	Port Usage	Dialup (Euro ISDN) 👻
/AN 👻		
PN 👻	ISDN Configuration Type	Point-to-Multipoint Point-to-Point
rewall 👻		Advanced Settings
olP 👻		
ocal Services 🔹 👻		

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 automatic- ally recognised.
Result of Autoconfigura- tion	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 <i>Point-</i> <i>to-Multipoint</i> .

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.

Save configuration	Extensions	SIP	Call	CLID	Call	ISDN	Options		
System Management 🔹 👻	Extensions	Accounts	Routing	Translation	Translation	<u>Trunks</u>			
Physical Interfaces 🔹 👻									
LAN 👻	Basic Paramete	ers							
Routing 👻	Description		sipgate	sipgate					
WAN -	Administrative	e Status	Fnabled						
VPN -	Trunk Mode		0.000						
Firewall 🔹				○ Off [®] Client [©] Server [©] gw-trunk					
VolP 🔺	Registrar		sipgate.de	sipgate.de					
Application Level Gateway Media Gateway	Outbound Pro	эху							
Local Services 👻	Realm								
Maintenance 👻	Protocol			ort: 5060					
External Reporting Monitoring	User Name		8861755						
	Authentication	Authentication ID							
	Password	Password							
	Registration	Registration							
	Expire Time	Expire Time		sec					
	Trunk Settings	Trunk Settings							
	SIP Header F	SIP Header Field(s) for Caller Address Disabled							
	Advanced Settings								
	Codec Settings								
	Codec Propo	Codec Proposal Sequence							
	Sort Order		G.711 U	JLaw	G.729 G.7 G.726-16 DTM		.38 Fax		
	Voice Quality S	Voice Quality Settings							
	Echo Cancellation		🗹 Enabled	✓ Enabled					
	Comfort Noise Generation		🗹 Enabled	✓ Enabled					
	Packet Size 40 ms								
			0	K Car	ncel				

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 ${\it client},$ the media gateway is run as a SIP client.
Registrar	Enter the IP address of the remote SIP terminal (client or serv- er) 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 pro- vider 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 authentica- tion. You must enter this value here. Maximum number of char- acters: 40.
Registration	Enables or disables the SIP REGISTER registration mechan- ism.
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) ad- dress is sent for outgoing calls.
	Select <i>Display Only</i> . 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 Se- quence	Determine the order in which the codecs are offered for use by the media gateway. If the first codec cannot be applied, an at- tempt is made to use the second codec, and so on. Set Codec Proposal Sequence to <i>default</i> . The codec in the first posi- tion 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 co- decs 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 <i>Enabled</i> is selected, echo feedback is suppressed.

Field	Meaning
Comfort Noise Genera- tion	Specify whether Comfort Noise Generation should be used. The slight comfort noise generation prevents subscribers from think- ing 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

 \bigcirc . The status of the VoIP connection is changed by pressing the \frown button or \bigcirc button in the **Action** column.

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

Save configuration	Extensions	SIP	Call	9	LID	Call	ISDN	Ont	ione	
System Management 🔹 🔻	Extensions	Accounts	Routing	<u>Translation</u>		Translation	Trunks	opt	Options	
Physical Interfaces 🔹 👻										
LAN 👻	Description	Regist	rar / Outbound Prox	/	Protocol	Status	Action			
Routing -	sipgate	sipga	te.de		UDP	0	1	â		
WAN -				<u> </u>	lew					
VPN 🔻										
Firewall 🔹										
VolP 🔺										
Application Level Gateway										
Media Gateway										

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.

Save configuration	Extensions	SIP	Call	CLID	Call	ISDN	Option		
System Management	*	Accounts	Routing	Translation	<u>Translation</u>	Trunks	option		
Physical Interfaces	-								
LAN	 Basic Paramet 	ers							
Routing	 Description 								
WAN	- Extension (1	Extension / User Name		76600690					
VPN	- Extension/ C	Serivaine	1	1.0000000					
Firewall	 Interface Typ 	e	⊖sip⊙i	⊖ SIP ® ISDN					
VolP	Select ISDN	interface	bri2-0 💌						
Application Level Gateway									
Media Gateway			A	dvanced Setting	<u>gs</u>				
Local Services	*		0	K Car	ncel				
Maintenance	-								



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 <i>ISDN</i> 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.

Relevant fields in the Extensions menu

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.

Save configuration System Management	Extensions	<u>SIP</u> Accounts	<u>Call</u> Routing	<u>CLID</u> Translation	<u>Call</u> <u>Translation</u>	<u>ISDN</u> Trunks	Opt	ions
Physical Interfaces 🔹 👻								
LAN 👻	Description	Exter	nsion	Туре	Interface	Status		
Routing 🗸 👻		766	0690	ISDN	bri2-0	0	盦	ø
WAN -		766	00691	ISDN	bri2-0	0	â	
		766	0692	ISDN	bri2-0	0	窗	
VPN -		766	00693	ISDN	bri2-0	0	窗	
Firewall 🔹		766	00694	ISDN	bri2-0	0	面	_
VolP 🔺		766	0695	ISDN	bri2-0	0	<u></u>	
Application Level Gateway			0696	ISDN	bri2-0	0		
Media Gateway			00697	ISDN	bri2-0	0		_
Local Services 🔹 🔻			0097			0		
Maintenance 🗸 👻				ISDN	bri2-0	-	â	_
External Reporting 🗾 👻		766	0699	ISDN	bri2-0	0	盦	
Monitoring 🗸 🗸				New)			

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.

Save configuration	Extensions	<u>SIP</u> Accounts	<u>Call</u> Routing	<u>CLID</u> <u>Translation</u>	<u>Call</u> <u>Translation</u>	ISDN Trunks	Option		
Physical Interfaces 🔹 👻									
_AN 👻									
Routing 👻	Basic Paramete	rs							
WAN 👻	Session Bord	ler Controller Mode	Off 💌	Off 💌					
VPN -	Media Stream	n Termination	🗹 Enabled	Enabled					
Firewall 👻	Default Drop	Extension	76600691	76600691					
/oIP 🔺	Boldan Brop	Extension	1.0000001						
Application Level Gateway	Dial Latency		5	Seconds					
Media Gateway									
Local Services 🗾 👻		Advanced Settings							
Maintenance 👻									
External Reporting 🚽 👻									

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

Field	Meaning
Session Border Control- ler 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 exten- sions. 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 exten- sion) that are only to be routed internally do not require an addi- tional call routing entry.
Media Stream Termina- tion	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-

Relevant fields in the Options menu

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.

Save configuration	Extensions	SIP	Call	CLID	Call	ISDN	Options		
System Management 🔹 👻		<u>Accounts</u>	Routing	Translation	Translation	Trunks			
Physical Interfaces 🔹 👻									
LAN 👻	Basic Paramete	rs							
Routing 🗾	Description		sipgate						
WAN -	Administrativ	e Status	Enable						
VPN -	-								
Firewall 🔹	Туре		External						
VoIP	Calling Line		bri2-0 💙						
Application Level Gateway	Calling Address								
Media Gateway			1						
Local Services 🔹	Called Addre	SS	00*						
Maintenance 🔹 👻	Priority	Line Calle	d Address Translation	ı	Status	Action			
External Reporting 🚽 👻	1	-			0	1	`````````````````````````````````````		
Monitoring 👻	Add								
	Routing Rule								
	Priority		1						
	Administrativ	e Status	🗹 Enable						
	Outbound Line		sipgate 💌						
	Called Addre	ss Translation							
		Apply							
			0	K Car	ncel				

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

Field	Meaning
Description	Here, enter the name of the call routing entry.
Administrative Status	The entry is used with enabled.
Туре	Select <i>External</i> 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.

Relevant fields in the Call Routing menu

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.

Save configuration	Extensions	SIP	Call	CLID	Call	ISDN	Options	
System Management 🔹 🔻	Extensions	<u>Accounts</u>	Routing	<u>Translation</u>	Translation	<u>Trunks</u>		
Physical Interfaces 🔹 👻								
LAN 👻	Basic Paramete	ers						
Routing 👻	Description		ISDN					
WAN -	Administrative	e Status	🗹 Enable					
VPN -	-			1				
Firewall 🔹	Туре		External N					
VolP 🔺	Calling Line		bri2-0 💌					
Application Level Gateway Media Gateway	Calling Address							
Local Services 🔹 👻	Called Addre	ss	*					
Maintenance 🔹	Priority	Line Called	Address Translation		Status	Action		
External Reporting 🔹 🔻	1	-			0	†	1	
Monitoring 🔹 👻	Add							
	Routing Rule							
	Priority		1					
	Administrative	e Status	🗹 Enable	✓ Enable				
	Outbound Lin	ie	bri2-1 💌]				
	Called Addre	ss Translation						
	Apply							
		OK Cancel						

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 .

Save configuration)	Extensions	SIP	Call		CLID		Call		ISDN	0	otion
System Management	-	Extensions	<u>Accounts</u>	Routing	I	ranslation	I	ranslation		<u>Trunks</u>	-	non
Physical Interfaces	-											
LAN	-	Description	Calling Line	Calling Address		Called Address		Туре	Stat	us Action		
Routing	-	sipgate	bri2-0			00*		External	0	1	Í	1 🖉
WAN	-	ISDN	bri2-0			*		External	0	1	Í	1 🖌
VPN	-				\subset	New						
Firewall	-										_	
VolP	-											
Application Level Gatew	ay											
Media Gateway												

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.

Save configuration	Extensions	SIP	Call	CLID	Call	ISDN	Ontion
System Management 🔹 👻	Extensions	<u>Accounts</u>	Routing	<u>Translation</u>	Translation	<u>Trunks</u>	Option
Physical Interfaces 🔹 👻							
LAN 🔫	Basic Paramete	rs					
Routing 👻	Description		sipgate				
WAN -	Direction		Outgoing	~			
VPN - Firewall -	Associated Li	ine	sipgate 🗸]			
VolP 🔺	Local Address		7660069?	7660069?			
Application Level Gateway Media Gateway	External Addr	ess	499117660	069?			
Local Services - Maintenance -	OK Cancel						

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

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

Relevant fields in the Call Translation menu

Select Outgoing for outgoing calls.

is to apply.

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 trans- lated 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 sig- nalled 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 Configura- tion -> <bri2-0 (nt)<="" th=""><td>Dialup (Euro ISDN)</td></bri2-0>	Dialup (Euro ISDN)
ISDN Configuration Type	Physical Interfaces -> ISDN Ports -> ISDN Configura- tion -> <bri2-0 (nt)<="" th=""><td>Point-to-multipoint</td></bri2-0>	Point-to-multipoint

Configuring the second ISDN interface

Field	Menu	Value
Autoconfiguration on Bootup	Physical Interfaces -> ISDN Ports -> ISDN Configura- tion -> <bri2-1 (te)<="" td=""><td>Aktiviert</td></bri2-1>	Aktiviert
Result of Autoconfiguration	Physical Interfaces -> ISDN Ports -> ISDN Configura- tion -> <bri2-1 (te)<="" th=""><th>Port Usage: Dialup (Euro ISDN), ISDN Configuration Type: Point-to-multipoint</th></bri2-1>	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 . <i>sipgate.de</i>
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-> Ad- vanced Settings	Default
Echo cancellation	VoIP -> Media Gateway -> SIP Accounts -> New-> Ad- vanced Settings	Aktiviert
Comfort Noise Generation (CNG)	VoIP -> Media Gateway -> SIP Accounts -> New-> Ad- vanced Settings	Aktiviert
Packet Size	VoIP -> Media Gateway -> SIP Accounts -> New-> Ad- vanced 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		Malue
Field	Menu	Value
Description	VoIP -> Media Gateway -> Call Routing -> New	e.g. <i>sipgate</i>
Administrative Status	VoIP -> Media Gateway -> Call Routing -> New	Enable
Туре	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. <i>sipgate</i>
Description	VoIP -> Media Gateway -> Call Routing -> New	e.g. ISDN

Field	Menu	Value
Administrative Status	VoIP -> Media Gateway -> Call Routing -> New	Enable
Туре	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. <i>sipgate</i>
Direction	VoIP -> Media Gateway -> Call Translation -> New	Outgoing
Associated Line	VoIP -> Media Gateway -> Call Translation -> New	e.g. <i>sipgate</i>
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.

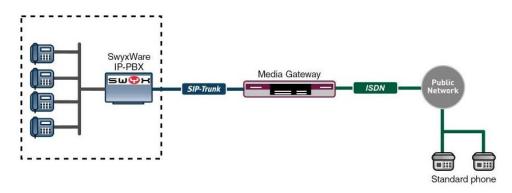


Fig. 77: Example scenario

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....

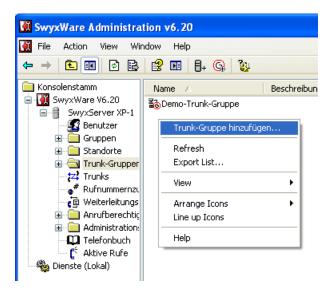


Fig. 78: Create new trunk group

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.

Assistent zum Hinzufügen ein	er Trunk-Gruppe	×
Name und Beschreibung der Geben Sie den Namen der Tr	Trunk-Gruppe unk-Gruppe und die Beschreibung ein.	
anderweitig z.B. als Name für Telefonbucheintrag verwende	Namen für die Trunk-Gruppe ein. Dieser darf nicht einen Trunk, einen Benutzer, eine Gruppe oder als et werden. schreibung ein, mit der Sie diese Trunk-Gruppe später	
Name der Trunk-Gruppe:	Demo-Trunk-Gruppe	
Beschreibung:		
	< Back Next > C	ancel

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.

Assistent zum Hinzufügen einer Trunk	c-Gruppe 🛛 🔀
Art der Trunk-Gruppe Geben Sie die Art der Trunk-Gruppe an u	nd wählen Sie das geeignete Profil aus.
verwendende Profil aus. Wenn Sie Inforn Installation erforderlich ist, schauen Sie in nach. Wenn Sie eine Trunk-Gruppe für einen h	Benutzerdefiniert' aus. Damit können Sie in
Art der Trunk-Gruppe:	SIP-Gateway
Profil:	SwyxConnect
	< Back Next > Cancel

- 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.

Assistent zum Hinzufügen einer Trunk-Gruppe
Definition der Weiterleitung Geben Sie an, für welche Rufe diese Trunk-Gruppe verwendet werden soll.
Je nach Auswahl werden Weiterleitungseinträge erstellt. Üffentliche Rufnummern sollten im kanonischen Format (z.B. '+4930123456') eingegeben werden, Sie können den Platzhalter ™ verwenden.
Trunks dieser Trunk-Gruppe verwenden
Fuir alle externen Rufe
nur für externe Rufe an folgende Zielrufnummer oder SIP-URI:
G für alle externen Rufe und alle nicht zugewiesenen internen Rufnummern
C für folgende interne Rufnummern:
< Back Next > Cancel

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.

Assistent zum Hinzufügen einer Trunk-Gruppe	<
Anrufberechtigung Wählen Sie eine Anrufberechtigung für diese Trunk-Gruppe aus.	ĺ
Mit der Anrufberechtigung einer Trunk-Gruppe wird festgelegt, wohin kommende Rufe dieser Trunk-Gruppe weitergeleitet werden können. Bitte wählen Sie eine der aufgeführten Anrufberechtigungen aus, die dieser Trunk-Gruppe zugewiesen werden soll. Anrufberechtigung: Keine Rufbeschränkung	
Standardprofil, das Rufe zu allen Zielen zulässt < Back]

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.

Standortprofil Wählen Sie das entsprechende Standortprofil für diese Trunk-Gruppe aus.
Ein SwywWare-Standort definiert alle ortsspezifischen Einstellungen, wie Zeitzone, Amtsholung, Länder- und Ortskennzahl. Bitte wählen Sie einen der aufgeführten Standorte aus, der dieser Trunk-Gruppe zugewiesen werden soll.
Standort: Nuemberg
- Beschreibung
< Back Next > Cancel

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....

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

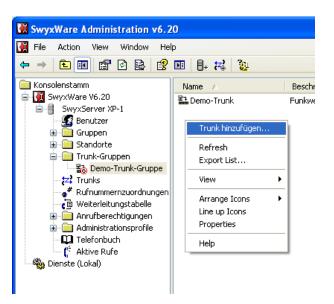


Fig. 86: Add trunk

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.

Assistent zum Hinzufügen ei	ines Trunks	\mathbf{X}
Name des Trunks Wählen Sie einen eindeutige	en Namen für den neuen Trunk.	
Name einer Trunk-Gruppe, G verwendet werden.	n Trunk-Namen ein. Dieser darf nicht anderweitig z.B. als Gruppe, einen Benutzer, oder Telefonbucheintrag eschreibung ein, mit der Sie diesen Trunk später eindeutig	
Name des Trunks:	Demo-Trunk	
Beschreibung:	Funkwerk Media Gateway	
	< Back Next > Canc	el

- 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.

Assistent zum Hinzufügen eines	Trunks
SIP-Konto Geben Sie das SIP-Konto für die	sen SIP-Gateway-Trunk an.
Geben Sie die Parameter des SIF diesen Trunk am SwyxServer an	P-Kontos an, mit dem sich das SIP-Gateway über meldet.
In der Gerätekonfiguration des S werden.	IP-Gateways müssen dieselben Parameter verwendet
Benutzer-ID:	bintec elmeg
Authentifizierungs-Methode:	Immer authentifizieren
Benutzername:	bintec elmeg
Kennwort:	password
	< Back Next > Cancel

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.

Assistent zum H	inzufügen ei	nes Trunks	
Rufnummern Geben Sie d	lie Rufnummern (ein.	
Bei nicht zu: und geben S Wenn diese alle Felder le Hinweis: La vorgegeben	sammenhängend Sie die anderen N r Trunk keine öff ser und klicken S ndes- und Ortske	lummern dann in den Eige entlichen Rufnummern zur ie auf Weiter'. nnzahl sind durch den Sta	s hier nur die erste Nummer ein schaften des Trunks an. n System hinzufügt, lassen Sie ndort der Trunk-Gruppe
Landes- kennzahl	Orts- kennzahl	Erste Rufnummer	Letzte Rufnummer
49	911	6898924	- 6898927
		< Back	K Next > Cancel

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.

Assistent zum Hinzufügen eines Trunks	×
Codecs Wählen Sie die Codecs für die Datenübertragung aus.	
Durch die Auswahl des Codecs wird die Kompressionsart für Rufe über diesen Trunk festgelegt. Die Auswahl eines Codecs wirkt sich somit auf die benötigte Bandbreite und die Sprachqualität aus.	
Codecs ☑ G.711 (ca. 84 kBit/s pro Ruf) ☑ G.729 (ca. 24 kBit/s pro Ruf) □ Fax over IP (T.38, ca. 20 kBit/s pro Ruf)	
< Back Next > Cancel	



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 t_{WO} .

Assistent zum Hinzufügen eines Trunks
Anzahl der Kanäle Wählen Sie die Anzahl der Kanäle aus, die von diesem Trunk verwendet werden.
Die Anzahl der Rufe, die gleichzeitig über einen Trunk geleitet werden können, wird normalerweise durch die Art des Trunks, die verfügbare Bandbreite oder eine Beschränkung des Dienstanbieters begrenzt.
Außerdem kann die Anzahl der gleichzeitigen Rufe eingeschränkt werden, um zusätzlich Kanäle (z.B. ISDN) oder Bandbreite für andere Anwendungen zu reservieren.
Über einen ISDN-Basisanschluss (S0) werden normalerweise max. zwei Rufe gleichzeitig unterstützt, bei einem Primärmultiplexanschluss (S2m) bis zu 30.
Anzahl der gleichzeitigen Rufe auf diesem Trunk:
< Back Next > Cancel

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.

Assistent zum Hinzufügen	eines Trunks
Computername Geben Sie den Namen de	es Computers an, auf dem der Trunk verwaltet wird.
lst dies der Fall, geben Sie den vorgegebenen Wert.	n anderen Computer als dem SwyxServer gehostet werden. e den Computernamen hier ein. Andernfalls verwenden Sie namen ein, wie er in den Systemeigenschaften von Windows
Computer:	SwyxWare
	< Back Finish Cancel

Fig. 93: Computer name

- (1) Under Computer, enter the computer name of the SwyxWare server.
- (2) This concludes trunk configuration. Click **Finish**.

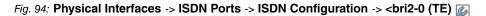
6.2.3 Configuration of the bintec media gateway

ISDN interface configuration

The ISDN port *ISDN-0* of the media gateway is connected to the NTBA of the pointto-multipoint. Configuration of the ISDN interface is already wired in ISDN TE mode exworks, 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 ->
dri2-0 (TE) .

Save configuration		ISDN Configuration MSN Configuration
System Management 🔹 🔻		
Physical Interfaces 🔹 🔺	Basic Parameters	
AUX Ethernet Ports	Port Name	bri2-0 (TE)
ISDN Ports	Autoconfiguration on Bootup	✓ Enabled
AN 🔫	Result of Autoconfiguration	Port Usage: Dialup (Euro ISDN), ISDN Configuration Type: Point-to-Multipoint
outing 🗸 👻	Port Usage	Dialup (Euro ISDN) 🗸
/AN 👻		
PN 🔻	ISDN Configuration Type	Point-to-Multipoint O Point-to-Point
irewall 🗸 🗸		Advanced Settings
olP 🔻		
.ocal Services 🔹 👻		



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.

Save configuration System Management	<u>Extensions</u>	SIP Accounts	<u>Call</u> Routing	<u>CLID</u> Translation	<u>Call</u> Translation	ISDN Trunks	Options		
Physical Interfaces 🛛 👻									
LAN 🔫	Basic Paramete	rs							
Routing 🗸 👻	Description		SwyxWare						
WAN -	Administrativ	e Status	Enabled						
VPN -	Trunk Mode			lient OServer 🖲 gw					
Firewall 👻	Trunk Wode				-trunk				
/oIP 🔺	Registrar		192.168.0.2	211					
Application Level Gateway Media Gateway	Outbound Pro	жy							
Local Services 👻	Realm								
Maintenance 🗸 👻	Protocol	Protocol User Name		UDP VPort: 5060					
External Reporting 🗾 👻	User Name			bintec elmeg					
wonitoring •	Authentication	Authentication ID							
	Password	Password		geheim					
	Registration	Registration		✓ Enabled					
	Expire Time		120	sec					
	Trunk Settings								
	SIP Header F	SIP Header Field(s) for Caller Address P-Preferred							
			A	dvanced Setting	<u>as</u>				
			0		ncel				

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

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 g_{W} -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 pro- vider 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 authentica- tion. You must enter this value here. Maximum number of char-

Field	Description
	acters: 40.
Registration	Enables or disables the SIP REGISTER registration mechan- ism.
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. <i>120</i> 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 num- ber is indicated via the SIP header field <i>P-Preferred</i> (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.

Save configuration	Extensions	<u>SIP</u> Accoun	<u>its</u>	Call Routing	CLID Translation	<u>Call</u> Translation	ISDN Trunks	Options	
Physical Interfaces 🔹						1			
LAN 👻	Basic Paramete	rs							
Routing 🔹	Description		Swyx->ISDN						
WAN 👻	Administrative	a Statue		Enable					
VPN 👻		5 010103			3				
Firewall 🔹	Туре			External N	<u> </u>				
VoIP	Calling Line			SwyxWare	*				
Application Level Gateway Media Gateway	Calling Address								
Local Services 🔹	Called Address			*					
Maintenance 🔹 👻	Priority	Line	Called Add	ress Translation		Status	Action		
External Reporting 🔹 👻	1	-				0	1 ↓	â 🖉	
Monitoring 🗾 👻	Add								
	Routing Rule								
	Priority			1					
	Administrative	e Status		✓ Enable					
	Outbound Lin	e		bri2-0 💌					
	Called Addre	ss Translatio	on						
					Apply				
				0	K Car	ncel			

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.
Туре	Select <i>External</i> 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 <i>SwyxWare</i> , 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 ex- ample, * means that at the end of a character string any num- ber 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.

Save configuration	Extensions	SIP	Call	CLID	Call	ISDN	Options			
System Management 🔹 🔻	Extensions	<u>Accounts</u>	Routing	Translation	<u>Translation</u>	Trunks	options			
Physical Interfaces 🔹										
LAN 🔫	Basic Paramete	Basic Parameters								
Routing 👻	Description		ISDN->Sw	ух						
WAN 👻	Administrative Status		🗹 Enable							
VPN -										
Firewall 👻	Туре		I runk	Trunk						
VolP 🔺	Calling Line		bri2-0 💌	bri2-0 💌						
Application Level Gateway	Calling Addre	ss								
Media Gateway										
Local Services 🔹	Called Addre	ss	*							
Maintenance 👻	Routing Rule									
External Reporting 🚽 👻	Trunk Line		SwyxWare	e 🔽						
Monitoring -	Called Addre	ss Translation								
			0	K Car	icel					

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

Field	Description
Description	Here, enter the name of the call routing entry.
Administrative Status	The entry is used with enabled.
Туре	Select <i>Trunk</i> for calls that are routed to a PBX behind the me- dia 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 ex- ample, * means that at the end of a character string any num- ber 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 .

Save configuration System Management Physical Interfaces	Extensions	<u>SIP</u> Accounts	Call Routing	<u>CLID</u> Translation	<u>Call</u> Translation		<u>SDN</u> unks	Option	<u>ns</u>
LAN	Description	Calling Line	Calling Address	Called Address	Туре	Status	Action		
Routing	Swyx->ISDN	SwyxWare		*	External	0	+	<u></u>	7
WAN -	ISDN->Swyx	bri2-0		*	Trunk	0	+		
VPN -			(New					
Firewall 👻									
VoIP									
Application Level Gateway									
Media Gateway									



6.3 Overview of configuration steps

Add trunk group

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

Add trunk

Field	Menu	Value
SwyxWare Adminis-	SwyxWare -> Swyx Server -> Trunk	e.g. Demo Trunk
tration	Groups -> Demo Trunk Group -> Add	
	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 Ac- count	e.g.bintec elmeg
Authentication meth- od	Assistant for adding a trunk ->SIP Ac- count	e.g. Always authen- ticate
User Name	Assistant for adding a trunk ->SIP Ac- count	e.g. bintec elmeg
Password	Assistant for adding a trunk ->SIP Ac- count	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)<="" th=""><th>Aktiviert</th></bri2-0>	Aktiviert
Result of Autoconfig- uration	Physical Interfaces -> ISDN Ports -> ISDN Configuration -> <bri2-0 (te)<="" th=""><th>Port Usage: Dialup (Euro ISDN), ISDN Configuration Type: Point- to-multipoint</th></bri2-0>	Port Usage: Dialup (Euro ISDN), ISDN Configuration Type: Point- to-multipoint

Configuration of SIP accounts

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

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

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
Туре	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
Туре	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.

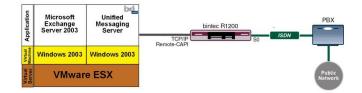


Fig. 99: Example scenario

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**.

tem Management 🔹 🔺					
itatus	Automatic	Automatic Refresh Interval 60 Seconds Apply			
Global Settings	Warning	Warning: System Password not changed!			
nterface Mode / Bridge	System In		SMOLUTION	nungeu.	
Groups	Uptime	ion notion		19 Day(s) 16 Hour(s) 52 Minute(s)	
Administrative Access		1970 142		and the second sec	
	System I	Date		Thu Apr 28 23:34:47 2005	
vsical Interfaces 🔹 👻	Serial N	umber		R1E180006500018	
N 👻	BOSS Ve	ersion		V.7.8 Rev. 2 IPSec from 2009/02/04 00:00:00	
reless LAN 👻	Ressourc	e Information			
uting 👻	CPU Us	age		0%	
AN 👻	Memory	Usade		21.4/31.9 MB (67%)	
N 👻		age External		0 / 4B Channels	
ewall 👻		27 <u>-</u> 2	TD -t-		
IP 👻		ssions (SIF, F	<ip, etc)<="" td=""><td>0</td><td></td></ip,>	0	
cal Services 👻	Active IP	Active IPSec Tunnels		0/1	
intenance •	Modules				
1 (1997) (1997) (1997) (1997)	DSP Mo	dule		4 Channel VINETIC	
ernal Reporting 🔹 👻	Physical In	Physical Interface		Interface Specifics	Link
nitoring 👻	en1-0			10.0.0.1947 255.255.255.0	0
	en1-4			Not configured / Not configured	0
	WLAN1			Access Point / Channel Auto / 0 Clients / FW: 2.17.4.0.i.9d8/2.17.2.0.i.1	0
	com0-8			Not configured	0
	bri2-0			Not configured	0
	bri2-1			Not configured	0
	Recent Sy:	stem Logs			
	Time	Level	Subsystem	Message	
	02:27:21	Information	IPSec	P1: peer 1 (IPSec Test) sa 0 (-): reactivated	
	02:27:11	Information	IPSec	P1: peer 1 (IPSec Test) sa 0 (-): Remote IP address lookup: t	imeout
	02:27:11	Information	IPSec	P1: peer 1 (IPSec Test) sa 0 (-): blocked for 10 seconds	
		Information	INET	dialup if 100001 prot 1 10.0.0.194:2048->10.10.0.1:59015	
		Information	IPSec	IPSEC CB - CB mode of Peer "IPSec Test" changed -> reset	IsdnCBNextMode
	05:47:29	Information	INET	sshd: pid 67 - listening on 0.0.0.0 port 22.	
		Information	USB	usb6-0-2: unplugged	
		Information	USB	usb6-0-1: unplugged	
	04:46:38	Error	TTY	UMTS Ctl umtsctl_ttyiftrap(): mib_get failed	

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.

Save configuration		ISDN Configuration MSN Configuration	
System Management 👘 🔻			
hysical Interfaces 🔹			
AUX	Port	ISDN Switch Type	
Ethernet Ports	bri2-0 (TE)	None , Point-to-Multipoint	P
ISDN Ports	bri2-1 (TE)	None , Point-to-Multipoint	
	white if (if they		

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.

Save configuration		ISDN Configuration MSN Configuration
System Management 🔹 👻		
Physical Interfaces		
AUX	Basic Parameters	
Ethernet Ports ISDN Ports	ISDN Port	bri2-0 💌
LAN -	Service	ISDN Login 🕑
Wireless LAN 👻	MSN	999999
Routing 🔹	MONED	
WAN -	MSN Recognition	
VPN -	Bearer Service	Itata + Voice O Data O Voice
Firewall 🗸 🗸		
VolP -		(OK) (Cancel)

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

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.

Save configuration		User Options		
System Management 🛛 👻				
Physical Interfaces 🔹				
LAN 👻	Basic Parameters	Basic Parameters		
Wireless LAN 👻	Enable server	Enabled		
Routing -				
WAN -	CAPI Server TCP Port	2662		
VPN -		OK Cancel		
Firewall 🗸				
VoIP -				
Local Services				
DIIS	1			
DynDNS Client				
DHCP Server				
Web Filter				
CAPI Server				

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

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.

Save configuration		User Options		
System Management				
Physical Interfaces				
LAN	Basic Parameters	Basic Parameters		
Wireless LAN .	User Name	Servonic		
Routing				
WAN	Password			
VPN	Access	✓ Enabled		
Firewall .				
VolP		OK Cancel		
Local Services	•			
DHS				
DynDNS Client				
DHCP Server				
Web Filter				
CAPI Server				

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 <i>Enabled</i> .	

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.

Remote Clients Configuration					
	CAPI Configuration				
Remote CAPI	Advanced				
Device IP addr 192.168.0.254	ess or host name:	TCP port of Remote CAPI Server: 2662			
User: Servonic		Password			
	Use these values	More Devices (CAPI2032.DLL)			
Info 32-bit CAPI:	You are using the multi device v Please press 'More Devices (CA				
ОК	Cancel	Help			

Fig. 105: Remote Clients Configuration

- To log in the remote CAPI client, the Device IP address or host name of the bintec R1200must 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.

Remote Multi CAPI Client Configuration		? 🛛
Remote Multi CAPI Clie	nt Configuration	
Device / Controller a (192.168.0.254, Port: 2662)	Local Controller	Add Device
🔷 Controller: 1	1	<u>R</u> emove Device
		<u>C</u> ontroller
Profile Information B3-layer protocol support: + Transparent + IS0 R06 (x25 DTE DTE) + X25 DCE + 1.30 Icr fax group 3 + 1.30 Icr fax group 3 with extensions	cordance with T.30 App	
OK Cancel		Help

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 .

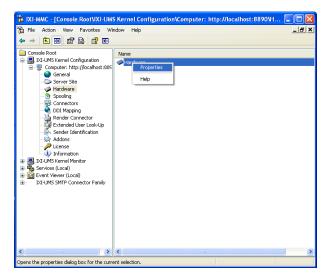


Fig. 107: IXI-UMS Kernel Configuration

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

Hardware (http://localhost:8890)	X
Description	
Device:	
Service:	
SISDN All Purpose (Funkwerk Enterprise Communications GmbH (RMCC)	Add
Fax SMS	Edit
Voice	Luk
Alert	Delete
<	
Hardware detection	Apply

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).

Add ISDN device	
General Advanced Access Routing	1
Controller	Add
■ 1. Controller(2 B-channels)	Edit Delete
Used controllers count 1 Start with controller 1 Channels exclusively reserved for receiving 0	
Total controllers: 1 Total channels: 2	
(<u>C</u> ancel

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.

http://www.comsole.com/active/	Kernel Monito	r\Computer: http://locall	host:8890\Channels	1 🗖 🗖 🚺
🚡 Eile Action View Favorites Wir	ndow <u>H</u> elp			_ & ×
← → 🗈 🖬 🔮 🖬				
Concole Root Co	Status Status © Channel 2 Channel 2	tp://tocalhost.8890) Description Wahing for incoming call Ready		
< · · · · · · · · · · · · · · · · · · ·	<			>

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.

🚡 IXI-MMC - [Console Root\IXI-UM	5 Kernel Monitor\Computer: http://localhost:8890\Queue]	
🚡 File Action View Favorites Wi	ndow Help	_ _ _ / ×
Console Root	Queue (http://localhost:8890)	
😑 🔜 IXI-UMS Kernel Monitor	≡ 🖻 🖻 × #∗ 🗹	
Computer: http://localhost:889		
Queue	Market IXI-UMS Kernel	
😥 🦓 Services (Local)		
Event Viewer (Local) IXI-UMS SMTP Connector Family	Fax/SMS Recipient number 6863828	
	ISDN Originator Address	
	Message type FAX 💌	
<pre></pre>	Send	Cancel

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 ->Op- tions	Aktiviert
CAPI Server TCP Port	Local Services ->CAPI Server ->Op- tions	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.

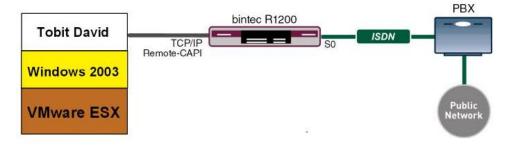


Fig. 112: Example scenario

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.

Management 🔺					
	Automatic	Refresh Interv	0.01	Seconds Apply	
Settings					
ce Mode / Bridge		Warning: System Password not changed:			
\$		formation			
istrative Access	Uptime			19 Day(s) 16 Hour(s) 52 Minute(s)	
e Authentication	System (Date		Thu Apr 28 23:34:47 2005	
l Interfaces 🔹 🔻	Serial N	umber		R1E180006500018	
-	BOSS Ve	rsion		V.7.8 Rev. 2 IPSec from 2009/02/04 00:00:00	
s LAN 👻	Ressourc	e Information			
-	CPU Us	ane		0%	
-	Memory	- - -		21.4/31.9 MB (67%)	
-		age External		0 / 4B Channels	
		25.4			
÷		Active Sessions (SIF, RTP, etc)		0	
ervices 👻	Active IP	Sec Tunnels		0/1	
ance 🔻	Modules				
Cox 10.	DSP Mod	DSP Module		4 Channel VINETIC	
Reporting 👻	Physical Inf	Physical Interface		Interface Specifics	Link
ing 🔻	en1-0			10.0.0.194/ 255.255.255.0	0
	en1-4			Not configured / Not configured	0
	WLAN1	WLAN1		Access Point/ Channel Auto / 0 Clients / FW: 2.17.4.0.i.9d8/2.17.2.0.i.1	0
	com0-8			Not configured	0
	bri2-0			Not configured	0
	bri2-1			Not configured	0
	Recent Sys	stern Logs			
	Time	Level	Subsystem	Message	
	02:27:21	Information	IPSec	P1: peer 1 (IPSec Test) sa 0 (-): reactivated	
	02:27:11	Information	IPSec	P1: peer 1 (IPSec Test) sa 0 (-): Remote IP address lookup: til	meout
	02:27:11	Information	IPSec	P1: peer 1 (IPSec Test) sa 0 (-): blocked for 10 seconds	
	02:26:42	Information	INET	dialup if 100001 prot 1 10.0.0.194:2048->10.10.0.1:59015	
	02:26:42	Information	IPSec	IPSEC CB - CB mode of Peer "IPSec Test" changed -> reset is	sdnCBNextMode
	05:47:29	Information	INET	sshd: pid 67 - listening on 0.0.0.0 port 22.	
	04:46:38	Information	USB	usb6-0-2: unplugged	
04:		Information	USB	usb6-0-1: unplugged	
	04:46:38	Error	TTY	UMTS Ctl umtsctl_ttyiftrap(): mib_get failed	
	04:46:38	Error	TTY	UMTS Ctl umtsctl ttyiftrap(): mib_get failed	

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.

Save configuration		ISDN Configuration MSN Configuration	
System Management 🔹 🧃			
Physical Interfaces	•		
AUX	Port	ISDN Switch Type	
Ethernet Ports	bri2-0 (TE)	None, Point-to-Multipoint	\$
ISDN Ports	bri2-1 (TE)	None , Point-to-Multipoint	
LAN			

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.

Save configuration		ISDN Configuration MSN Configuration
System Management 🛛 👻		
Physical Interfaces 🔹		
AUX	Basic Parameters	
Ethernet Ports		
ISDN Ports	ISDN Port	bri2-0 💌
LAN -	Service	ISDN Login
Wireless LAN 🗸	MSN	999999
Routing 🗸 🗸		
WAN -	MSN Recognition	Ight to Left ○ Left to Right (DDI)
VPN -	Bearer Service	
Firewall 🗸 🗸		
VolP -		OK Cancel

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

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.

Relevant fields in the MSN Configuration menu

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.

Save configuration		User Options
System Management	-	
Physical Interfaces	-	
LAN	 Basic Parameters 	
Wireless LAN	 Enable server 	Enabled
Routing	CAPI Server TCP Port	2662
WAN		12002
VPN	•	OK Cancel
Firewall	-	
VolP	•	
Local Services	•	
DHS		
DynDNS Client		
DHCP Server		
Web Filter		
CAPI Server		

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

Field	Description
Enable server	The function is activated by selecting <i>Enabled</i> .
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.

Save configuration		User Options
System Management	•	
Physical Interfaces	•	
LAN	Basic Parameters	
Wireless LAN	User Name	Tobit
Routing	•	
WAN	 Password 	******
VPN	 Access 	Enabled
Firewall	•	'
VolP	-	OK Cancel
Local Services	•	
DNS		
DynDNS Client		
DHCP Server		
Web Filter		
CAPI Server		

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 <i>Enabled</i> .

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.

🐮 Remote Clier	ts Configuration	? 🛛
*	CAPI Configu	iration
Remote CAPI	Advanced	
	ess or host name:	TCP port of Remote CAPI Server:
192.168.0.254	<u> </u>	2662
User: Tobit		Password
TIODA		1
	Use these values	More Devices (CAPI2032.DLL)
Info 32-bit CAPI:	You are using the multi device Please press More Devices (C	version of the CAPI2032 DLL API2032 DLL) to get more info.
OK	Cancel	Help

Fig. 118: Remote Clients Configuration

- To log in the remote CAPI client, the Device IP address or host name of the bintec R1200must 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.

Remote Multi CAPI Client Configuration		? 🛛
Remote Multi CAPI Clier	nt Configuration	
Device / Controller	Local Controller	<u>A</u> dd Device
 (192.168.0.254, Port: 2662) Controller: 1 	1	Bemove Device
		Device
		Controller
		<u>⊺</u> est
Profile Information		
B3-layer protocol support: + Transparent + 150 NL, with compatibility to T.70 NL in acc + 150 2018 (>250 DTE DTE) + X 250 DE + T.30 tot fax group 3 + T.30 tot fax group 3 with extensions	ordance with T.90 App	endix II
OK Cancel		Help

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.

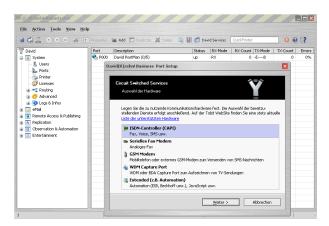


Fig. 120: XP-2 - David Administrator

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.

Circuit Switched Servio Auswahl der Dienste	ces		Ý
Die gewählte Hardware Dienste, die für den akt auswählbaren Punkte w	uellen Port verwendet	werden sollen. Die nic	
Fax-Gruppe 3			
Voice Mail (Anrufbea	antworter)		
🔲 TMail (Synthetische	Sprache)		
🔲 SMS (Short Messagi	ng Service)		
TAPI (ECT)			
🔲 eMail (IHS via ISDN)			

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.

Geben Sie als "Beschreibung" einen Namen an, unter dem der aktuelle Port bei David geführt werden soll. Trogen Sie als "Faskennung" die Paxnummer ein, die an die Gegenstele übermittelt wird. Beschreibung binker: R1200 Absenderkennung	Erweiterte Hardwarek	ices onfiguration		Y
bintec R1200	bei David geführt were die an die Gegenstelle	den soll. Tragen Sie als "Faxk	er dem der aktuelk ennung" die Faxm	e Port Immer ein,
	-			

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.

ircuit Switched Se	vices		
Erweiterte Hardware	konfiguration		
		ktuellen Port (senden u en, geben Sie die "Vorv	nd/oder empfangen). rahl für die Amtsholung
Betriebsmodus senden und empfar	igen (TX/RX)	*	
		~	

Fig. 124: Advanced hardware configuration

David(R)zehn! Business Port Setup	×
Circuit Switched Services Erweitete Hardwarekonfiguration	
Geben Sie an, ob Sie einen ISDN Anlagenanschluss oder Mehrgeräteanschluss verwenden.	
[<u>ISON-Me</u> hrgeräteanschluss (Punkt-zu-Mehrpunkt Verbindung) [ISON-Anlogenanschluss (Punkt-zu-Punkt Verbindung)	
< Zurück Weiter > Abbrech	ien

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.

	tched Services		\sim	
Erweiter	e Hardwarekonfigur	ation		
Karte gel	e die MSN (Multi Sub ten soll. Wird keine ellen Port angenom	scriber Number) an, die für Auswahl getroffen, werder nen.	r die ISDN- n alle Anrufe	
MSN 0	6898925	MSN 5		
MSN 1		MSN 6		
MSN 2		MSN 7		
MSN 3		MSN 8		
MSN 4		MSN 9		

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....

i 🕼 🚠 💿 💿 👬 🗄	Properties	🐚 Add 🔭 I	Dupl	cate 💢 Delete 🏻 🎱	🛃 💋 Da	vid Services	QuickFinder		00	16?
7 David	Port	Description	_		Status	RX-Mode	RX-Count	TX-Mode	TX-Count	Error
- 💿 System	📤 P000			(5)	up	RX		-E8	0	0%
- 🚨 Users	JP P001	Funkwerk bir	-	Add Port	up	RX	1	FV-X-	0	0%
- 🌽 Ports 			2	Duplicate						
- Icenses			×	Delete						
Routing Advanced			đ	Status Monitor						
😟 😟 Logs & Infos			۲	Stop						
Remote Access & Publishing	-		۲	Restart						
Replication			۲	Startup type 🔹 🕨						
Observation & Automation Entertainment			5	Properties						
	-									

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

DRT 001	- Funkwerk	bintec R1200	<u>~ ? ×</u>
General	Services	ISDN DDI Advanced IHS	
Routi	ng prefix e to	testuser	•
	ile Name Edit script	\\xp-2\david\tld\common\tld.dcc	
	an company		
		OK (Close

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.

atei <u>B</u> earbeite	n Aktionen Ansicht Optionen Werkzeuge Netzwerk Eenster Hilfe	Tobil:Soft	twar
🕄 Neu 🖕 💽 Form	ulare 🖡 🖅 🐓 Senden/Empf. 🖕 😋 💥 🔍 😒 Arkwarten 🖏 Alen arkwarten 🥥 Westarleiten 🕐 📅 🕵 😨		
🗈 🧼 🖨 🛛 david	ØEingang → 🕨 Go Ocogle 🛛 🗞 🛊 🖓 🕸 ♥ Duldifinder		
voiten ×	Navigator X F 🖾 🖉 🗸 Ven Betreff	Datum	
*	David InfoCenter		
Neu	*TESTFAX - David InfoCenter		
Q	Datei Bearbeiten Ansicht Optionen Einfügen Eormat Hilfe		
Heute	🖳 Senden 👩 🖤 y 🎭 🤊 💡 Automatisch 🐨 🔛 🛷 🔍 🛞 🕼 y 🗐 y 🌝 🃎 y 🐚 Variabien y		
6			
Unverteilt	at Normal v War Tahoma v 36 v F X U A u E ± ≣ H H H H		
Engang	Si An 6893829 ··· Antwort erwartet innerhalb		
3	si⊈c ⊤ von Ohne ⊤ um 17:00 ⊤		
Versand	Betreff TESTFAX *		
-			
Ausgang			
192	TECTEAN		ß
	TESTFAX		
RSS-Feeds Co			
KlickDown: Anlege			
24 7: David MalGa			
KlickDown: Kalend 24/7: So setzen Si			
KlickDown: Via Kat			
As distributions in the second			

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 ->Op- tions	Aktiviert
CAPI Server TCP Port	Local Services ->CAPI Server ->Op- tions	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 re- ceiving (TX/RX)
ISDN point- to-multipoint connec- tion	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.



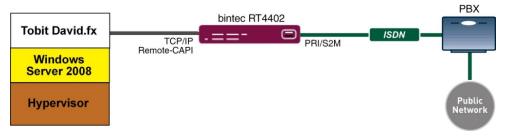


Fig. 130: Example scenario

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**.

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

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

Save configuration		System	Passwords	Date and Time	System Licences			
Assistants 👻					-,			
System Management 🔹 🔺								-
Status	System Licence ID: RN3BCC010210053							_
Global Settings	Installed Software Options							
Interface Mode / Bridge	Bridging, CAPI, IP (builtin), OSPF, PIM-	SM - Protoco	I Independent N	fulticast (Sparse Mod	e), Data Encryption Acceleration	on, IPSec (0/	10), V	PN
Groups	PPTP, PPTP, Fax, BRRP							
Administrative Access	Description			Licence Typ	e Licence Serial Number	Status		
Remote Authentication	IPSec			Software	RN3IPSFRFactory	OK	窗	
Certificates	PIM-SM - Protocol Independent Multica	ast (Sparse M	lode)	Software	RN3PIMFTFactory	OK	龠	
Physical Interfaces 🔹	PPTP			Software	RN3PPTFRFactory	OK	朣	
AN .	Data Encryption Acceleration			Software	RN3DEA00Factory	ок	朣	
Vireless LAN Controller 🛛 🔻	BRRP			Software	RN3RRP00Factory	ок	窗	
Networking 👻	Fax			Software	RNZFAX00	ок	窗	
Routing Protocols 🚽		(New	Default Lice	acad			
Multicast 🗸			NCW	Denault Lice	1009			

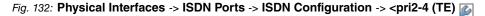


Configure the ISDN Primary Rate interface

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) is a second second

Save configuration		ISDN Configuration MSN Configuration					
Assistants 👻							
System Management 🔹 👻							
Physical Interfaces	Basic Parameters						
AUX	Port Name	pri2-4 (TE)					
Ethernet Ports	Port Usage	EURO ISDN S2M (TE)					
ISDII Ports	Tonosage						
ADSL Modem	ISDN Line Framing	CRC4 (Standard) 💌					
LAN	P-P Base Number						
Wireless LAN Controller 🛛 🔻							
Networking 👻	Channel Selection	\odot Any Channel \bigcirc No channel identification \bigcirc Submit preferred channel					
Routing Protocols 🔹 👻							
Multicast 👻		OK Cancel					



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.

Save configuration			ISDN Configuration MSN Configuration
Assistants System Management	• •		
Physical Interfaces	•	Basic Parameters	
AUX Ethernet Ports		ISDN Port	pri2-4 🗸
ISDN Ports		Service	ISDN Login 🔽
ADSL Modem		MSN	999999
LAN	-		
Wireless LAN Controller	•	MSN Recognition	Right to Left C Left to Right (DDI)
Networking	•	Bearer Service	In the second secon
Routing Protocols	•		
Multicast	•		(OK) (Cancel)

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 ${\tt Right\ to\ Left}.$

- (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.

Save configuration	User Options					
System Management	•					
Physical Interfaces	▼ Basic Parameters					
LAN	Enable server Enabled					
Wireless LAN Controller	Faxheader Enabled					
Networking	▼					
Routing Protocols	CAPI Server TCP Port 2662					
Multicast	• OK	Cancel				
WAN	•					
VPN	•					
Firewall	•					
VolP	•					
Local Services	▲					
DNS						
HTTPS						
DynDNS Client						
DHCP Server	_					
Web Filter	_					
CAPI Server						

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>

Save configuration)	User Options				
Assistants	-					
System Management	-					
Physical Interfaces	-	Basic Parameters				
LAN	-	User Name	capi			
Wireless LAN Controller	-	Password				
Networking	-					
Routing Protocols	-	Access	✓ Enabled			
Multicast	-		OK Cancel			
WAN	-					
VPN	-					
Firewall	-					
VolP	-					
Local Services	•					
DNS						
HTTPS						
DynDNS Client DHCP Server	_					
Web Filter						
CAPI Server						

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.

🖫 Remote Clients Configuration 🛛 💽 🔀								
CAPI Configuration								
Remote CAPI Advanced	Remote CAPI Advanced							
Device IP address or host name:	TCP port of Remote CAPI Server: 2662							
User:	Password:							
capi	****							
Use these values	More Devices (CAPI2032.DLL)							
Info 32-bit CAPI: You are using the multi device version of the CAPI2032.DLL. Please press 'More Devices (CAPI2032.DLL)' to get more info.								
OK Cancel	Help							

Fig. 136: Remote Clients Configuration

- 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.

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

Remote Multi CAPI Client Configuration					
Remote Multi CAPI I	Client Configuration				
Device / Controller	Local Controller	Add Device			
 (192.168.10.60, Port: 2662) Controller: 3 	1	<u>R</u> emove Device			
		Device			
		<u>C</u> ontroller			
		<u>I</u> est			
Profile Information					
B3-layer protocol support: + Transparent + T.90NL with compatibility to T.70NL in + IS0 8208 (X.25 DTE-DTE) + X.25 DCE + T.30 for fax group 3 + T.30 for fax group 3 with extensions	accordance with T.90 App	endix II			
OK Cancel		Help			

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**.

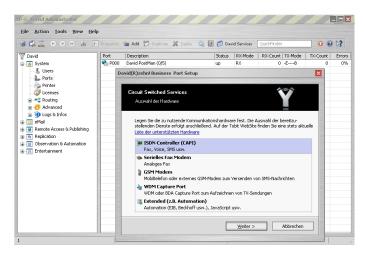


Fig. 138: David.fx Administrator

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

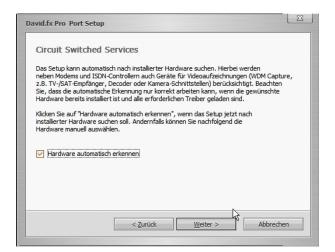
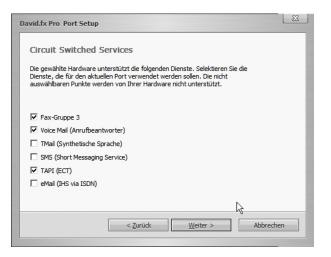


Fig. 139: Hardware detection

(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.

David.fx Pro Port Setup	23
Circuit Switched Services	
Wählen Sie den "Betriebsmodus" für den aktuellen Port (senden und/oder empfangen). Wenn Sie eine Nebenstellenanlage einsetzen, geben Sie die "Vorwahl für die Amtsholung ein (z.B. 0).	-
Betriebsmodus	
senden und empfangen (TX/RX)	
Vorwahl für Amtsholung	
N	
< <u>Z</u> urück <u>W</u> eiter > Abbrech	en

Fig. 141: Advanced hardware configuration

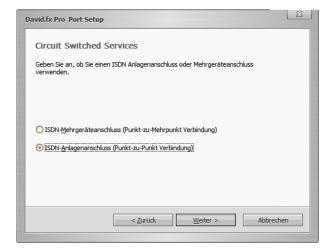


Fig. 142: Advanced hardware configuration

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.

David.fx Pro Port Setup	23
Circuit Switched Services Geben Sie den Nummernblock ('DDI-Range'') und die Länge ('DDI-Length'') an, für die die Anrufbeantworterfunktion gelten soll.	
DDI-Range (z.B. 150 - 255)	
DDI-Length (z.B. 3) 8	
< <u>Z</u> urück <u>F</u> ertig stellen Abbrev	chen

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.

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

PORT 001 - Funkwerk Enterprise Communicat	
Allgemein Dienste ISDN DDI Erweitert IHS	
Präfix	
Länge 8	
Offset -1	
Bereich 0	
Ausgehende	
Ruf annehmen wenn DDI-Länge erreicht	
	C
	ĥ
OK Schließen	

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....

🖾 🖉 💿 🗿 👬 💈	Properties	h Add 🔭	Dupl	icate 样 Delete 🔍	🛃 📁 Dav	rid Services	QuickFinder		00	6?
David	Port	Description	. (0	(m)	Status	RX-Mode	RX-Count		TX-Count	Erro 0'
- 5ystem - 8 Users	P000	David PostM Funkwerk bir		Add Port	up up	RX RX		-EB FV-X-	0	0
			0	Duplicate						
www.Licenses			×	Delete						
 The second second			al i	Status Monitor						
😐 🗓 Logs & Infos	-		۲	Stop						
eMail Remote Access & Publishing				Restart						
Replication	1		T	Startup type 🔹						
Observation & Automation			5	Properties						
Entertainment										

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.

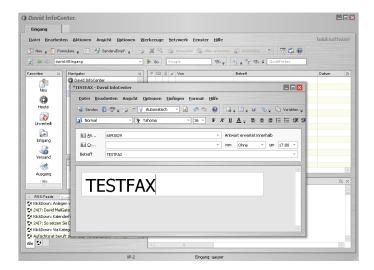


Fig. 146: Test fax

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)<="" th=""><th>EURO ISDN S3M (TE)</th></pri2-4>	EURO ISDN S3M (TE)
ISDN Line Framing	Physical Interfaces -> ISDN Ports -> ISDN Configuration -> <pri2-4 (te)<="" th=""><th>CRC4 (Standard)</th></pri2-4>	CRC4 (Standard)
Channel Selection	Physical Interfaces -> ISDN Ports -> ISDN Configuration -> <pri2-4 (te)<="" th=""><th>Any channel</th></pri2-4>	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 ->Op- tions	Enabled
User Name	Local Services ->CAPI Server ->User-> New.	e.g. <i>capi</i> .
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 . <i>capi</i> .
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 re- ceiving (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

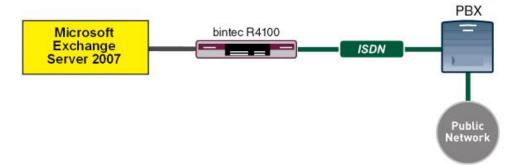


Fig. 147: Example scenario

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 adminis**tration console :

ile Action Yew Help					
1					
Microsoft Exchange	🚱 Unified Mess	aging		objects 🚺	ctions
Organization Configuration All Malbox	UM Dial Plans UM IP	Sateways UM Mailbox Po	icies UM Auto Attendants	u	Inified Messaging
Client Access	UM Dial Plan	# Digits	Associated LM Serve	rs	😤 New UM Dial Plan
- 👸 Hub Transport	DP-nbg	3	EXCHANGE07		New UM IP Gateway
 Unified Messaging Server Configuration 	DP-peine	3	EXCHANGE07		New UM Mailbox Policy
- to Malbox					
- Dient Access					New UM Auto Attendant
Hub Transport					🗼 Export List
- 🔚 Unified Messaging					View
- A Malbox					है। Refresh
- 🥂 Distribution Group					💡 Help
- 🤤 Mail Contact A Disconnected Mailbox				118	B. Helb
- and Toobox					

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...

New UM Dial Plan Completion	New UM Dial Plan This wizard helps you create a UM dial plan for use by Microsoft Exchange Unified Messaging. A dial plan is a grouping of unique telephone extension numbers.
	Name
	demo_dialplan
	Number of digits in extension numbers:
	3
	U <u>R</u> I type:
	Telephone Extension
	VolP security: Unsecured
	After you create a new dial plan, the dial plan must be added to one or more UM servers before it will be used.

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.

New UM Dial Plan Completion	Completion The wizard completed successfully. Click Finish to close this wizard. Elapsed time: 00:00:01 Summary: 1 #em(s).1 succeeded, 0 failed.	
	📄 demo_dialplan 🕜 Completed	3
	Exchange Management Shell command completed: new-UMDiaPlan Name 'demo_diablan' NumberOlDigitsInExtension '3'-URIType 'TeExtri-YoUPSecurity 'Unsecured'	
	Elapsed Time: 00:00:01	

Fig. 150: New UM dial plan

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

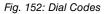
After the wizard is closed, dial plan properties must be edited.

Settings	Dialing Rule Groups	Dialin	g Restrictions
General	Subscriber Access	Dial Codes	Features
Welcome Greeti Welcome greeti	-		
Use default gre	-		Modify
Informational an	nouncement		
-	nouncement is disabled		Modify
Enter the teleph	scriber Access Numbers		
Enter the teleph	one number to associate:		
Enter the teleph	one number to associate:		
Enter the teleph	one number to associate:		

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*.

demo_dialplan Prop	erties		2
Settings General	Dialing Rule Groups Subscriber Access	Dialing R Dial Codes	estrictions Features
<u>C</u> ountry/Region c	is code: ss code: the United States) refix: nnce, 1 for the United States)	0	
Incoming Configur In-country/region (Example: 142555 International numb (Example: 4420xx)	number format: 50198) er format:	0	
	OK Cancel	Apply	Help



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 codefield, 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.

emo_dialplan Properties	
General Subscriber Access Settings Dialing Rule	
Dial by name primary method:	Last First
Dial by name secondary method:	SMTP Address
Audio <u>c</u> odec:	G.711 💌
Operator extension:	810
Logon failures before disconnect:	3
Timeouts and Retries	
Maximum call duration (min):	30
Maximum recording duration (min):	20
Recording idle time-out (sec):	5
Input idle timeout (sec):	5
Input retries:	3
Input failures before disconnect:	3
Language Settings	
Default language:	English (United States)
OK	Cancel Apply Help

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).

General	Subscriber Acc	ess	Dial Code	s	Features
Settings	Dialing Ru		1		estrictions
n-Country/Regio					
Name	N	umber Mask		Dia	aled Numbe
national	0*			0*	
▲ International Rule	e Groups				P
nternational Rule	Edi <u>t</u> 🗡	umber Mask		Di	
nternational Rule	Edi <u>t</u> 🗡	umber Mask)*		Dia	aled Numbe
nternational Rule	Edi <u>t</u> 🗙				aled Numbe

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.

lemo_dialplan Prop	erties					×
General	Subscriber Acc	ess	Dia	l Codes	Features	1
Settings	Dialing Ru	le Groups		Dialing	Restrictions	
🔽 Allow calls to j	users within the s	ame dial pla	n			
Allow calls to g	extensions					
Select allowed in-	country/region rul	e groups fro	m dial	plan:		
砕 A <u>d</u> d 🗡						
national						
Select allowed inte	ernational rule gro	ups from di	al plan			
Add 🗙						
International						
1						
	ОК	Cance	- 1	Apply	Help	1

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.

- _ 🗆 🗙 Elle Action View ↔ → 1 € 10 2 10 Constitution of the transmission of the t St Mic 🌆 Unified Me: Unified Mess Export List Name A EXCHANGE07 Role Hub Transport, Client Acc... Version Version 8.1 (Build 240.6) View Refrest General | System Settings | UM Settings | 12 Help Associated Dial Plans EXCHANGE07 Recipient Configuration Malbox Malbox Malbox Mal Contact Mal Contact Mal Contact Toolbox Select Dial P - 🗆 🗵 👍 Agd.... 🗙 Ele Name DP-nbg DP-peine Find Now Search Name ^ - and i DP-nbg DP-nbg.vitualnet.funkwerl DP-peine.vitualnet.funkwe . Miscellaneous Confin Prompt languages Maxi Maximum co
- (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.

New UM IP Gateway	New UM IP Gateway
Completion	This wizard helps you create a UM IP gateway for use by Microsoft Exchange Unitied Messaging, UM IP gateways represent the connection between a physical gateway or IP PBX and Unitied Messaging.
	Name:
	demo_UM-GW
	192.168.10.222
	Example: 192.168.10.10
	C Fully qualified domain name (FODN):
	Example: smatthost.company.com
	Dial plan
	demo_diaplan
	If a dial plan is selected, a default hunt group will be created to associate this new UP IP gateway to the specified dial plan. If no dial plan is selected, a hunt group must be created manually.

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 administra-tion console**.

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

New UM Hunt Group Completion	New UM Hunt Group This wizard helps you create a UM hunt group for use by Microsoft Excl Messaging. A hunt group represents a connection between a UM IP ga dial plan, and associates the dial plan with the pilot identifier specified b	iteway and a UM
	Agsociated UM IP gateway: Idemo UM-GW	
	Name:	
	name: maibox_demo	
	Dial plan:	
	demo_dialplan	Blowse.
	Pilot identifier:	
	500	

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.

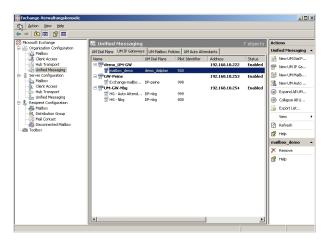


Fig. 159: UM IP Gateways

Configuration of a UM Mailbox Policy

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

mo_dialplan Default P	olicy Properties	
àeneral Message Text	PIN Policies Dialing Restric	tions
damo dialolar	n Default Policy	
	T Default T blicy	
Associated UM dial pla	n: demo_dialplan	
Modified:	Montag, 25. Mai 2009 15	5:07:20
Maximum greeting durat	ion (minutes):	5
Allow missed call no	difications	
Allow missed call no	ouncations	

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).

no_dia	plan Default Policy Properties	
ieneral	Message Text PIN Policies Dialing	Restrictions
<u>F</u> ax ide	entity:	
Micro	soft Exchange	
Texts	ent when a <u>U</u> M mailbox is enabled:	
Willko	mmen bei Microsoft Exchange UM	A
Texts	ent when a PIN is <u>r</u> eset:	
Ihre F	N wurde zurückgesetzt!	×
Text in	cluded with a <u>v</u> oice message:	
neue	Sprachnachricht!	×
Textin	cluded with a fax message:	
neues	FAXI	A V
	OK Cance	el <u>A</u> pply Help

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.

demo_dialplan Defa	ult Policy Prop	oerties		×
General Message T	ext PIN Policie	es Dialing Rest	rictions	
Minimum PIN lengt	h:			
🔲 PIN lifetime (da	ays):			
Number of previou	s PINs to disallo	AI:	5	
Allow common	patterns in PIN			
Failed Logons				
Number of inc reset:	orrect <u>P</u> IN entrie	s before PIN is a	utomatically 5	
Number of inclusion	orrect PIN entrie	s before UM mail	box is 15	5
	OK	Cancel	Apply	Help

Fig. 162: PIN Policies

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

demo_dialplan Default Policy Properties	×
General Message Text PIN Policies Dialing Restrictions	_,
Allow calls to users within the same dial plan	
Allow calls to extensions	
Select allowed in-country/region rule groups from dial plan:	
♣ Add ×	
national	
Select allowed international rule groups from dial plan:	
🖶 Add 🗡	
_	
🖶 Add 🗡	

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 **Pi-Iot 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.

The selected mailbox will be enabled for Unified Messaging. Upon completion, an e-ma
message will be sent to the malbox notifying the user that they have been enabled for Unified Messaging. The message will include the PIN and the number to dial to gain ac
to their mailbox. By default, an extension number and PIN are automatically generated. can also manually specify an extension number and PIN.
Unified Messaging Mailbox Policy:
demo_dialplan Default Policy Browse
PIN Settings
C Automatically generate PIN to access Dutlook Voice Access
Manualy specify PIN:
Require user to reset EIN at first telephone logon
[3] Unified Messaging is a premium feature and requires an Exchange Enterprise Client Access License (CAL) to enable it for the malbox.

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.

Introduction	Extension Configuration	
Evenian Corriguation Corriguation Corriguation Corriguation Corriguation Corriguation Corriguation Completion	 C. Activative generated anabox externion C. Marculay entered makes externion: SIP Resource learning For a SIP URI day lays. The into SIP address of the user target material days in the intervention of the art EAM of parts the intervention of the art EAM of parts the intervention of the art EAM of parts the intervention of the art EAM of the intervention of the	720 In (normalie The E TLEA address of the user

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 autorecognition provides detection of a point-to-point or point-to-multipoint connector.

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

Save configuration		ISDN Configuration MSN Configuration		
ystem Management 👘 👻				
hysical Interfaces 🔹 🔺	Basic Parameters			
AUX	Port Name	bri2-0 (TE)		
Ethernet Ports	I OILINAIIIE	Mi2-0 (12)		
ISDN Ports	Autoconfiguration on Bootup	✓ Enabled		
AN 👻	Result of Autoconfiguration	Port Usage: Dialup (Euro ISDN), ISDN Configuration Type: Point-to-Multipoint		
outing 👻	Port Usage	Dialup (Euro ISDN)		
AN 👻				
PN 👻	ISDN Configuration Type	Point-to-Multipoint Point-to-Point		
rewall 👻		Advanced Settings		
olP 🔻				
.ocal Services 👻	OK Cancel			

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 Autoconfigura- tion	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.

Save configuration	Extensions	<u>SIP</u> Accounts	<u>Call</u> Routing	<u>CLID</u> Translation	Call Translat		
Physical Interfaces 🔹					1		-
AN -	Basic Paramete	rs					
Routing 👻	Description		Mailbox				
VAN -			Production and a second				
PN 👻	Extension / U	ser Name	500				
irewall 👻	Interface Type	12					
oIP	Registration		Enabled				
Application Level Gateway Media Gateway	SIP Endpoint	IP Address	192.168.10	101			
ocal Services 🔹 👻	Authentication	n ID					
aintenance 🔹	Password						
dernal Reporting 🔹 👻							
onitoring 👻	Protocol		TCP 💌				
	Port		5060				
			Ac	vanced Settin	gs		
	Codec Settings						
	Codec Propo	sal Sequence	💿 Default	Oquality O Lowest	⊖ Highest		
	Sort Order		G.711 u	Law 🗹 G.711 aLaw	🗹 G.729	G.726-40	🗹 T.38 Fax
	Surronder		G.726-3	2 🔲 G.726-24	G.726-16	DTMF Outband	
	Voice Quality S	ettings					
	Echo Cancell	ation	Enabled				
	Comfort Nois	e Generation	Enabled				
	Packet Size		30	ms			
			0	< Ca	ncel		

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.

Save configuration	Extensions	<u>SIP</u> Accounts	Call Routing	<u>CLID</u> Translation	<u>Call</u> Translation	<u>ISDN</u> Trunks	Options
System Management 🔹 👻		Accounts	Routing	mansiation	mansiación	ITUIKS	
Physical Interfaces 🔹 👻							
LAN -	Basic Paramete	ers					
Routing 👻	Description		to_isdn				
WAN 👻	Administrativ	e Status	🗹 Enable				
VPN 👻				=			
Firewall 🔹	Туре		External	*			
VolP 🔺	Calling Line		Any 🚩				
Application Level Gateway	Calling Addre	155					
Media Gateway			1				
Local Services 🔹 🔻	Called Addre	ss	*				
Maintenance 👻	Priority		Called Address Trans	lation	Status	Action	
External Reporting 🗾 👻	1	bri2-0			0	1	1
Monitoring 🗾 👻	Add						
	Routing Rule						
	Priority		1				
	Administrativ	e Status	🗹 Enable				
	Outbound Lir	10	bri2-0 💌				
	Called Addre	ss Translation					
				Apply			
				K Car	ncel		

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.
Туре	Specify how calls are to be routed. With <i>External</i> 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 num- ber) 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 be- fore 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> Di- aling Rule Groups	national,0*,0*
International Rule Groups	Organization Configuration -> Unified Messaging -> New UM Dial Plan> Di- aling Rule Groups	international,00*, 00*
Allow calls to uses within the same dial plan	Organization Configuration -> Unified Messaging -> New UM Dial Plan> Di- aling Restrictions	Aktiviert
Allow calls to exten- sions	Organization Configuration -> Unified Messaging -> New UM Dial Plan> Di- aling 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	Organization Configuration -> Unified	e.g. demo_UM-GW
gateway	Messaging -> New UM Hunt Group	

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 en- abled	Organization Configuration -> Unified Messaging -> New UM Mailbox Policy - > Message Text	e.g. Welcome to Mi- crosoft 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 mes- sage!
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 exten- sions	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 -> Recipi- ent Configuration -> Mailbox	e.g. demo_dialplan Default Policy
Manually specify PIN	Organization Configuration -> Recipi- ent Configuration -> Mailbox	Your PIN
Manually entered mailbox extension	Organization Configuration -> Recipi- ent Configuration -> Mailbox	e.g. 720

ISDN Configuration

Field	Menu	Value
Autoconfiguration on	Physical Interfaces -> ISDN Ports ->	Aktiviert
Bootup	<bri2-0 (te)=""> 👔.</bri2-0>	

SIP extension configuration

Field	Menu	Value
Description	VoIP -> Media Gateway -> Subscriber - > New	e.g. <i>Mailbox</i>
Extension / User Name	VoIP -> Media Gateway -> Subscriber - > New	500
SIP Endpoint IP Ad- dress	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 Out- band

Configuration of call routing

Field	Menu	Value
Description	VoIP -> Media Gateway -> Call Routing	e.g. to_isdn

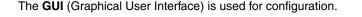
10 Media Gateway - bintec R1200 VoIP/R4100 VoIP as Unified Messaging Gateway for Microsoft Exchange Serv-

Field	Menu	Value	
	-> New		
Туре	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.



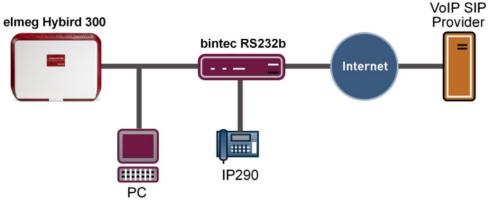


Fig. 169:

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 **el-meg hybird 300**.

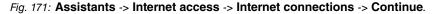
11.2.2 Setting up the Internet access at the bintec RS232b gateway

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.

Save configuration		Internet	Connections
Assistants 🖌			
First steps			
Internet Access			ISP Data for Internal
VPN	Description	ADSL_Provider	ADSL/SHDSL Modem
Wireless LAN	Select your Internet Se	rvice Provider (ISP) from the list:	
VoIP PBX in LAN	Internet Service		For Internet access you must set up a
System Management 🔹	Provider	Germany - T-Home	connection to your Internet Service Provider (ISP).
Physical Interfaces	Enter the authenticatio	n data for your Internet account:	Follow your provider's instructions!
LAN	- User Name	user#0001@t-online.de	Description
Wireless LAN 🗖	Password		Enter a description for the Internet connection.
Routing	Passwurd		
	Select the connection	mode:	You can select one of the predefined ISPs or define a custom Internet connection. Different
WAN -	Always active	Enabled	settings are required depending on the choice
VPN •	•		you make for the ISP or the user-defined
Firewall			connection protocol.
VolP			Internet Service Provider:
Local Services			Select your ISP or define a customized provider by choosing User-defined via the
Local Services			required connection protocol PPPoE (PPP
Maintenance •	1		over Ethernet), PPPoA (PPP over ATM),
External Reporting			ETHoA (Ethernet over ATM) or IPoA (IP over ATM).
Monitoring -			A110).
			When establishing an Internet connection, you are normally prompted for authentication by the ISP. A user name and a password are normally used for authentication. You can
	(OK Cancel	



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.

Save configuration		VolP PBX in LA	4
ssistants 🔺		VOIP PBA III EA	•
First steps			
Internet Access	Enter the settings of the VolP PBX within	a your LAN	VoIP PBX
VPN	-	-	
Wireless LAN	WAN interface for VoIP priorisation	ADSL_Provider 👻	Select the settings for the required WAN
VoIP PBX in LAN	Maximum Upload Speed	1024 kbps	WAN interface for VoIP priorisation
ystem Management 🛛 👻			Select the interface via which the VoIP PBX
nysical Interfaces 🔹 👻	DSCP filter for priorisation	DSCP Binary Value 🔽 101110	within your LAN reaches the Internet and, for
N +		IP Address	which QoS prioritisation and the activation of
ireless LAN 👻	IP Address of VoIP PBX within	192 168 1 100	"Full Cone NAT" are configured. Maximum Upload Speed
outing -	your LAN		Enter the maximum upload speed for the
-		Add	WAN connection (outgoing) in kbps. Possible
AN 👻			values are 0 to 1000000.
'N 🔻			DSCP filter for priorisation
ewall 👻			If you configured DSCP values (Differentiated
IP +			Services Code Point) for signalling the priority of VoIP packets on your VoIP PBX, configure
			the DSCP/TOS filter. This value must equal
			the one you configured on your VoIP PBX.
intenance 👻			Please consider possible requirements of your provider!
ternal Reporting 🛛 👻			The default value is Expedited Forwarding
nitoring 🗸 👻			(DSCP=46/TOS =184).
			IP Address of the VoIP PBX within your LAN
			Enter the LAN IP address of your VoIP PBX.
			NAT modus for the entered IP address is set
	ОК	Cancel	

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.

Save configuration		Ba	ic Setup	
Assistants				
First steps				^
System Management 🔹 👻	Enter the basic system settings:		Basic Setup	
Physical Interfaces 🔹 🔻	System Name	hybird_300	Here you can configure all of the settings	
VolP 👻	oystern Name		required to integrate your device into your	
Numbering 👻	Location	Data center	Local Area Network (LAN).	
Call Routing 🔹	Contact	admin@bintec-elmeg.com	The following parameters are only used to	
Applications 🔹	Enter the System Admin Password:		describe your device.	
LAN 👻	System Admin Password		System Name: System name is indicated as a login prompt	
Routing 🔹			or configuration interface header when you	
Firewall 🔹	Confirm Admin Password		access the device.	
Local Services 👻	Select the physical Ethernet port that		Location where the device is installed.	
Maintenance 🗸	Physical Ethernet Port (LAN)	ETH1 💟	Contact:	
External Reporting 🗸 👻	Enter the LAN IP Configuration:		Should contain the person who is responsible for the device (e-mail address is	
Monitoring -	Logical Ethernet/Bridge Interface	en1-0	recommended).	
monitoring	Address Mode	Static ○ DHCP Client		
	IP Address	192.168.1.100	To protect your device against unauthorized access it is urgently recommended that you	
	Netmask	255.255.255.0	configure a system password for your device. In the ex works state, the system password is	
	Default Gateway IP Address	192.168.1.254	funkwerk . System Admin Password:	
	DNS Server 1	192.168.1.254	Enter a password. Confirm Admin Password:	~
	DNS Server 2	0.0.0.0		Ĩ
	Warning! Configuration connection Address! Click OK and login again	ection may be lost when changing the n to proceed!	IP	
	Is this device used as DHCP Server?			
	Use this device as DHCP server	Enabled		
	Adva	anced Settings		
	ОК	Cancel		

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.

Save configuration		SIP Provider Locations Codec Profiles Options
Assistants 👻		
System Management 🔹 👻		
Physical Interfaces 🔹 👻	Basic Settings	
VoIP 🔺	Description	sipgate
Settings Numbering -	Parent Location	None 💌
Call Routing -	Туре	● Addresses ○ Interfaces
Applications + LAN + Routing +	Addresses	P Address/DNS Name Netmask [sipgate.de [255.255.255 Image: Control of the second
Firewall 👻	Upstream Bandwidth Limitation	🗹 Enabled
Local Services - Maintenance -	Maximum Upstream Bandwidth	1024 kbps
External Reporting 🚽 👻	Downstream Bandwidth Limitation	Enabled
Monitoring 🗸 👻	Maximum Downstream Bandwidth	16000 kbps
		Advanced Settings
	DSCP Settings for rtp Traffic	DSCP Binary Value V 101110

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.

Save configuration		SIP Provider Locations Codec Profiles Options
ssistants 👻		
ystem Management 💦 👻		
nysical Interfaces 🔹 👻	Basic Parameters	
olP 🔺	Description	sipgate
Settings	Provider Status	
all Routing 🔹	Access Type	◉ Single Number(s) ○ Direct Dial-In
oplications 👻	Authentication ID	userid
AN 👻	Password	•••••
outing 👻	User Name	
rewall 🔹	Domain	
aintenance •	Registrar	
xternal Reporting 🔹	Registrar	sipgate.de
ionitoring +		
onikoring •	Registrar Port	5060
	Transport Protocol	© UDP ○ TCP
	STUN	
	STUN server	
	Port STUN server	3478
	Timer	
	Registration Timer	60 Seconds
		Advanced Settings
		OK Cancel

Fig. 175: VolP -> 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.

Save configuration			QoS		
Assistants	-				
System Management	-				
Physical Interfaces	- QoS				
AN	✓ Interface	QoS Queue	Send	Dropped	Queued
Wireless LAN	ADSL_Provider				
Routing	-	High Priority	2497	0	0
		unpriorized	8527	0	0
WAN	•				
VoIP Local Services Maintenance External Reporting	• • •				
Monitoring					
Internal Log					
PSec					
ISDN/Modem					
Interfaces					
WLAN					
HotSpot Gateway					
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 Set- tings	192.168.1.100
gateway	Dime Manager -> IP Set- tings	192.168.1.254

Select connection type

Field	Menu	Value
Connection type	Assistants-> Internet Ac- cess -> Internet Connec- tions -> New.	e.g. internal ADSL mo- dem

Setting up the Internet connection

Field	Menu	Value
Description	Assistants -> Internet Ac- cess -> Internet Connec- tions -> Continue.	e.g. <i>ADSL_Provider</i>
Internet Service Provider	Assistants -> Internet Ac- cess -> Internet Connec- tions -> Continue.	e.g. <i>Germany-T-Home</i>
Username	Assistants -> Internet Ac- cess -> Internet Connec- tions -> Continue.	e.g. user#0001@t-online.de
Password	Assistants -> Internet Ac- cess -> Internet Connec- tions -> Continue.	e.g. supersecretge- heimkey

Adaptation to the Internet gateway

Field	Menu	Value
WAN interface for VoIP prior- itisation	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. <i>DSCP Binary Value</i> and <i>101110</i>
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 . <i>supersecretge-</i> <i>heimkey</i>
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 -> Loca- tions -> New.	e.g. <i>sipgate</i>
Addresses	VoIP -> Settings -> Loca- tions -> New.	e.g . <i>sipgate.de</i>
Maximum Upstream Band- width	VoIP -> Settings -> Loca- tions -> New.	e.g. 1024
Maximum Downstream Bandwidth	VoIP -> Settings -> Loca- tions -> New.	e.g. 16000
DSCP Settings for RTP Data	VoIP -> Settings -> Loca- tions -> New-> Advanced Settings	e.g. <i>DSCP Binary Value</i> and 101110

Storing SIP provider's login data

Field	Menu	Value
Description	VoIP -> Settings -> SIP Pro- vider -> New.	e.g. <i>sipgate</i>
Authentication ID	VoIP -> Settings -> SIP Pro- vider -> New.	e.g. userid
User name	VoIP -> Settings -> SIP Pro- vider -> New.	e.g. userid
Registrar	VoIP -> Settings -> SIP Pro- vider -> New.	e.g . <i>sipgate.de</i>
STUN Server	VoIP -> Settings -> SIP Pro- vider -> 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.

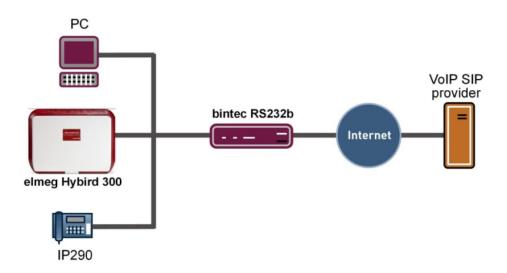


Fig. 177: Example scenario

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.

Save configuration			Options
Assistants	-		
System Management	-		
Physical Interfaces	-	Basic Parameters	
LAN	-	SIP Proxy	Enabled
Wireless LAN	-	SIP Port	5060
Networking	-	Prioritize SIP Calls	
Routing Protocols	-	Prioritize SIP Calls	Enabled Enabled
Multicast	-		OK Cancel
WAN	-		
VPN	-		
Firewall	-		
VolP	-		
SIP			
RTSP			

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.

sistants 🔺			
irst steps			
stem Management 🔹 👻	Enter the basic system settings:		Basic Setup
vsical Interfaces 🔹 🔻	System Name	hybird_300	Here you can configure all of the settings
•	Location		required to integrate your device into your Local Area Network (LAN).
ering 👻	Location		LUCAI Alea Network (LAN).
outing 👻	Contact	TELDAT	The following parameters are only used to
tions 👻	Enter the System Admin Password:		describe your device. System Name:
-	System Admin Password	•••••	System name is indicated as a login prompt
, ,	Confirm Admin Password		or configuration interface header when you access the device.
I	Select the physical Ethernet port that i	in used to connect to the Lith	access the device.
ervices 👻		ETH1 V	Location where the device is installed.
ance 👻	Physical Ethernet Port (LAN)		Contact
Reporting 🔹	Enter the LAN IP Configuration:		Should contain the person who is responsible for the device (e-mail address is
•	Logical Ethernet/Bridge Interface		recommended).
	Address Mode	Static ○ DHCP Client	To protect your device against unauthorized
	IP Address	192.168.0.250	access it is urgently recommended that you
	Netmask	255.255.255.0	configure a system password for your device. In the ex works state, the system password is
		192.168.0.254	funkwerk .
	Default Gateway IP Address		System Admin Password:
	DNS Server 1	192.168.0.254	Enter a password. Confirm Admin Password:
	DNS Server 2	0.0.0.0	
	Warning! Configuration conne Address! Click OK and login again	ction may be lost when changing the to proceed!	IP
	Is this device used as DHCP Server?		
	Use this device as DHCP server	Enabled	
	Adva	anced 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.

Save configuration	System	Passwords Date and Time Timer System Licences
Assistants 👻		
System Management 🔹 🔺		
Status	Basic Settings	
Global Settings	System Name	hybird_300
Access Codes Administrative Access	System Name	iysiid_300
hysical Interfaces 🔹 👻	Location	
olp •	Contact	TELDAT
umbering 👻	Maximum Number of Syslog Entries	50
all Routing 🗾 👻	Maximum Message Level of Syslog Entries	Information V
pplications 👻	Maximum Number of Accounting Log Entries	20
AN 👻	System Settings	
outing 👻		
irewall 👻	Transfer to busy extension	⊙ Off ○ With Ringing Tone ○ With Music On Hold
ocal Services 🔹 👻	Rerouting to Number	None - Busy Tone 💌
laintenance 🔹	Interconnect external calls	✓ Enabled
xternal Reporting 🔹 👻	Country Settings	
lonitoring 🗸 👻	Country Profile	Deutschland 💌
	Display Language	Deutsch 💌
	International Prefix / Country Code	00 / 49
	National Prefix / City Code	0 / 5171
		Advanced Settings
		OK Cancel

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.

Save configuration		SIP Provider Locations Codec Profiles Options			
Assistants 👻					
System Management 🔹 👻					
Physical Interfaces 🔹 👻	Basic Parameters				
/oIP	Description	sipgate			
Settings	Provider Status	● Active ○ Inactive			
lumbering 👻	Access Type				
Applications -	Authentication ID	36			
AN -					
Routing -	Password				
irewall 👻	User Name	36			
.ocal Services 🔹	Domain				
laintenance 🔹	Registrar				
xternal Reporting 🔹 👻	Registrar	sipgate.de			
lonitoring 👻	Registrar Port	5060			
	Transport Protocol				
	STUN				
	STUN server				
	Port STUN server	3478			
	Timer	Timer			
	Registration Timer	60 Seconds			
		Advanced Settings			
		OK Cancel			

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).

Save configuration	-			SIP Provide	Locations	Codec Profiles	Options			
Assistants	-									
System Management	-									
Physical Interfaces	-	View	20 per page <	🔊 Filter in 🛛 None 💌	equal 💌	Go				
VolP	-	No.	Description	Registrar	Access Type		Status	Action		
Settings		1	sipgate	sipgate.de	Single Number(s)	۲	+	窗	P
Numbering	-	Page: 1	1, ttems: 1 - 1, Max item	ns: 25						-
Call Routing	-				New					
Applications	122				INEW					

Fig. 182: VoIP -> Settings -> SIP Providers .

You change the status of VoIP configuration by pressing the full 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.

Save configuration		SIP Provider	Locations	Codec Profiles	Options
Assistants	•				
System Management	•				
Physical Interfaces	Basic Parameters				
/oIP	Description	sipgate			
Settings lumbering	Provider Status				
all Routing	+ Access Type	◉ Single Number(s) ○ Direct Dial-In			
pplications	Authentication ID	36			
AN	Password				
louting	User Name	36			
irewall		100			
ocal Services	Domain				
laintenance	← Registrar				
external Reporting	▼ Registrar	sipgate.de			
Aonitoring	Registrar Port	5060			
	Proxy Proxy Port	5060			
	Transport Protocol				
	Further Settings				
	Further Settings From Domain	[
		No Limitation			
	From Domain	No Limitation	~		
	From Domain Number of allowed simultaneous Calls		v v		
	From Domain Number of allowed simultaneous Calls Location	Any Location	 ✓ ✓ It ✓ 		
	From Domain Number of allowed simultaneous Calls Location Codec Profiles	Any Location System Defau	 ✓ ✓ It ✓ 		
	From Domain Number of allowed simultaneous Calls Location Codec Profiles Dial End Monitoring Time	Any Location System Defau 5 Seco	 ✓ ✓ It ✓ 		
	From Domain Number of allowed simultaneous Calls Location Codec Profiles Dial End Monitoring Time Call Hold inside the PBX system	Any Location System Defau 5 Seco Enabled	 ✓ ✓ It ✓ 		

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 ad- dress	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 Pro- viders -> New.	e.g. <i>sipgate</i>
Authentication ID	VoIP -> Settings -> SIP Pro- viders -> New.	e.g. 36
User Name	VoIP -> Settings -> SIP Pro- viders -> New.	e.g. 36
Registrar	VoIP -> Settings -> SIP Pro- viders -> New.	e.g . <i>sipgate.de</i>

Automatically generate prefix

Field	Menu	Value
Create inland call number	VoIP -> Settings -> SIP Pro- viders ->New -> Advanced Settings	Activated