

# innovaphone GW Routing

Gateway routing to manipulate incoming and outgoing numbers

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## 1.2 Setting up the Gateway Interface

The GW interface connects in two directions:

1. EXTERNAL > To the public network (e.g. SIP or ISDN)
2. INTERNAL > To the trunk object in the PBX
  - The IP address of the PBX must be entered (e.g. "localhost").
  - The HW ID and password (e.g. PBX password) of the trunk must be specified.

The screenshot shows the Asterisk Gateway configuration interface. The 'Gateway' tab is active, and the configuration for 'SIP1 Sip Amt' is displayed. The configuration is divided into several sections:

- General:** Name: Sip Amt, Disable:
- 1. Type and Transport:** Type: Provider, Transport: UDP, Without registration:
- Authorization:** Username: qsclogin, Password: [masked], Retype: [masked]
- Media Properties:** Protocol: H.323, STUN Server: [empty], Gatekeeper Address: 127.0.0.1 (primary), Gatekeeper Address: [empty] (secondary), Gatekeeper ID: [empty], Name: Amt\_Master, Number: [empty], Password: [masked], Retype: [masked]

### 1.3 The routes are created automatically

The connection between the EXTERNAL and INTERNAL sides of the GW interface is exclusively via routes. These routes are usually created automatically.

With the setup of the gateway interface, two routes were automatically generated.

1. From public network to PBX (EXTERNAL > INTERNAL)
2. From PBX to the public network (INTERNAL > EXTERNAL)

Hint:

- "SIP1" stands for the first SIP interface
- "RS1" stands for the internal registration of the first SIP interface

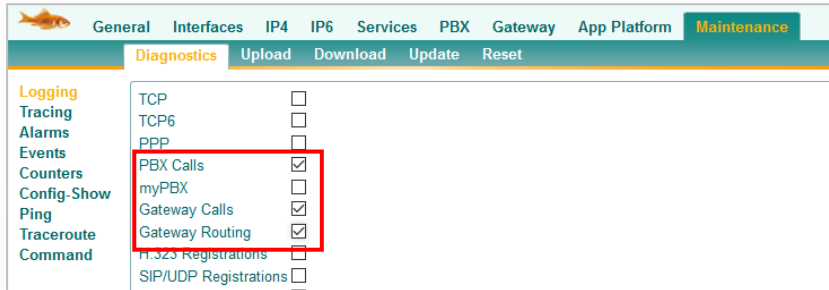
Examples:

- "BRI2" would stand for the second BRI interface
- "RB2" would stand for the internal registration of the second BRI interface
- "PRI3" would stand for the third PRI interface
- "RP3" would stand for the internal registration of the third PRI interface

From	To	Counter	CGPN Maps
SIP1:Sip Amt	RS1:Sip Amt		→
RS1:Sip Amt	TONE	00	→000
	MAP	00	→00049
			→0004930
			→0004930123456
	MAP		000...→00
			00...→0
			0...→030
			→030123456
	SIP1:Sip Amt	b	→

## 2 Call Tracking in Logging

Logging is very well suited for easy call tracking and detection of phone number manipulation, for this the options "PBX Calls", "Gateway Calls" and "Gateway Routing" should be activated.



In the logging itself, it is recommended to look for the entries with "INTERFACE MAP". Here you can see how the phone numbers are manipulated at the interfaces, and which interfaces are used at all. The route map can usually be derived from this.

```

Syslog
20191115-140645 CALL 8 Alloc
20191115-140645 CALL 8 A:Call -> / RS1::>:*:
20191115-140645 ROUTE 8 INTERFACE MAP if=RS1:'Deutsche_Telek' CGPN-In 1421->1421, CDPN-In 00491603333333->00491603333333, DGPN-In ->
20191115-140645 ROUTE 8 EVAL ROUTE route=RT7
20191115-140645 ROUTE 8 EVAL MAP route=RT7 map=1 dest='TONE' in=''->out=''
20191115-140645 ROUTE 8 MAP(CDPN-MATCH OK) route=RT7 map=1 dest='TONE' in=''->out=''
20191115-140645 ROUTE 8 APPLY CDPN-MAP in='00491603333333'->out='00491603333333'
20191115-140645 ROUTE 8 SUCCESS route=RT7 map=1 dest='TONE' in=''->out=''
20191115-140645 ROUTE 8 INTERFACE MAP if=TONE CGPN-Out 1421->1421, CDPN-Out 00491603333333->00491603333333, DGPN-Out ->
20191115-140645 CALL 8 B:Call 1421:test@test.de->00491603333333 / RS1:1421:_trunk->TONE:00491603333333:
20191115-140645 CALL 8 B:Rel 1421:test@test.de->00491603333333 / RS1:1421:_trunk->TONE:00491603333333: Cause: Resources unavailable, unspec
20191115-140645 ROUTE 8 EVAL ROUTE route=RT7
20191115-140645 ROUTE 8 EVAL MAP route=RT7 map=1 dest='TONE' in=''->out=''
20191115-140645 ROUTE 8 MAP(CDPN-MATCH OK) route=RT7 map=1 dest='TONE' in=''->out=''
20191115-140645 ROUTE 8 CONTINUE TO NEXT MAP route=RT7 map=1 dest='TONE' in=''->out='' reason='retry>0' found=false
20191115-140645 ROUTE 8 EVAL MAP route=RT7 map=2:'Routing Abgehende CLIP' dest='MAP' in=''->out=''
20191115-140645 ROUTE 8 MAP(CDPN-MATCH OK) route=RT7 map=2:'Routing Abgehende CLIP' dest='MAP' in=''->out=''
20191115-140645 ROUTE 8 EVAL CGPN-MAP cgpn=1421 verify=false in='1406'->out='0030555511111'
20191115-140645 ROUTE 8 EVAL CGPN-MAP cgpn=1421 verify=false in='1405'->out='0030555511111'
20191115-140645 ROUTE 8 EVAL CGPN-MAP cgpn=1421 verify=false in='1404'->out='0030555511111'
20191115-140645 ROUTE 8 EVAL CGPN-MAP cgpn=1421 verify=false in='1403'->out='0030555511111'
20191115-140645 ROUTE 8 EVAL CGPN-MAP cgpn=1421 verify=false in='1402'->out='0030555511111'
20191115-140645 ROUTE 8 EVAL CGPN-MAP cgpn=1421 verify=false in='1401'->out='0030555511111'
20191115-140645 ROUTE 8 EVAL CGPN-MAP cgpn=1421 verify=false in='1408'->out='0030555511111'
20191115-140645 ROUTE 8 CONTINUE TO NEXT MAP route=RT7 map=2:'Routing Abgehende CLIP' dest='MAP' in=''->out='' reason='processed MAP interface'
20191115-140645 ROUTE 8 EVAL MAP route=RT7 map=3 dest='MAP' in='00'->out='00'
20191115-140645 ROUTE 8 MAP(CDPN-MATCH OK) route=RT7 map=3 dest='MAP' in='00'->out='00'
20191115-140645 ROUTE 8 EVAL CGPN-MAP cgpn=1421 verify=false in='000...'->out='000'
20191115-140645 ROUTE 8 EVAL CGPN-MAP cgpn=1421 verify=false in='00...'->out='00049'
20191115-140645 ROUTE 8 EVAL CGPN-MAP cgpn=1421 verify=false in='0...'->out='000492871'
20191115-140645 ROUTE 8 EVAL CGPN-MAP cgpn=1421 verify=false in=''->out='00049305555'
20191115-140645 ROUTE 8 EVAL CGPN-MAP SUCCESS in=''->out='00049305555'
20191115-140645 ROUTE 8 APPLY CGPN-MAP in='1421'->out='000493055551421'
20191115-140645 ROUTE 8 CONTINUE TO NEXT MAP route=RT7 map=3 dest='MAP' in='00'->out='00' reason='processed MAP interface'
20191115-140645 ROUTE 8 EVAL MAP route=RT7 map=4 dest='MAP' in=''->out=''
20191115-140645 ROUTE 8 MAP(CDPN-MATCH OK) route=RT7 map=4 dest='MAP' in=''->out=''
20191115-140645 ROUTE 8 EVAL CGPN-MAP cgpn=000493055551421 verify=false in='000...'->out='00'
20191115-140645 ROUTE 8 EVAL CGPN-MAP SUCCESS in='000...'->out='00'
20191115-140645 ROUTE 8 APPLY CGPN-MAP in='000493055551421'->out='000493055551421'
20191115-140645 ROUTE 8 CONTINUE TO NEXT MAP route=RT7 map=4 dest='MAP' in=''->out='' reason='processed MAP interface'
20191115-140645 ROUTE 8 EVAL MAP route=RT7 map=5 dest='SIP1' in=''->out=''
20191115-140645 ROUTE 8 MAP(CDPN-MATCH OK) route=RT7 map=5 dest='SIP1' in=''->out=''
20191115-140645 ROUTE 8 APPLY CDPN-MAP in='00491603333333'->out='00491603333333'
20191115-140645 ROUTE 8 SUCCESS route=RT7 map=5 dest='SIP1' in=''->out=''
20191115-140645 ROUTE 8 INTERFACE MAP if=SIP1:'Deutsche_Telek' CGPN-Out 00493055551421->I493055551421, CDPN-Out 00491603333333->00491603333333,
20191115-140645 CALL 8 B:Call 00493055551421:test@test.de->00491603333333 / RS1:1421:_trunk->SIP1:00491603333333:
20191115-140645 CALL 8 B:Rel 00493055551421:test@test.de->00491603333333 / RS1:1421:_trunk->SIP1:00491603333333:
20191115-140645 CALL 8 A:Rel 00493055551421:test@test.de->00491603333333 / RS1:1421:_trunk->SIP1:00491603333333:
20191115-140645 CALL 8 Free
20191115-140807 CALL 13 Alloc
20191115-140807 CALL 13 A:Call -> / RS1::>:*:

```

### 3 Easy routing and number manipulation

#### 3.1 Routing Ways

##### Outgoing call (red):

- The call goes from the end device **(1.)** via the trunk object to the gateway **(2.)**.
- That's where it comes down to the INTERNAL registration **(3.)**
- After routing, the call goes via the EXTERNAL registration (4th) to the **GW interface** (5th) **and from there to the office.**

##### Incoming call (purple):

- The call comes from the office **(1.)** via the **GW interface** (2.) **into the gateway.**
- There it depends on the EXTERNAL registration **(3.)**
- After routing, the call goes via the INTERNAL registration **(4.)** to the trunk object **(5.)** and from there to the end device.

**CGPN = CallinG Party Number**  
**CDPN = CalleD Party Number**

The screenshot displays the Asterisk PBX configuration interface. It includes several tables and configuration panels:

- Registrations Table:**

Long Name	Name	No.	HW-ID	Node	IP
Amt_Master	Amt_Master	0	Amt_Master	root	127.0.0.1*
Amt_Slave1	Amt_Slave1	#01.0	Amt_Slave1		127.0.0.1
Amt_Slave2	Amt_Slave2	#02.0	Amt_Slave2		
Conference1	Conference1	81	Conference1		
Conference2	Conference2	82	Conference2		
DECT Master					
DECT Slave1					
extern-web	extern-web		extern-web		
- Routes Table:**

From	To
SIP1:Sip Amt	RS1:Sip Amt
RS1:Sip Amt	TONE
	MAP
00-00	
	MAP
	SIP1:Sip Amt
- Counter CGPN Maps Table:**

000...	→000
00...	→00049
0...	→0004930
	→0004930123456
000...	→00
00...	→0
0...	→030
	→030123456
- Interface Table:**

Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out	State	Alias	Registrations
SIP1 Sip Amt	1-00	0301234560-666	00-0	00-0			
		030123456	0-0	0-0			
SIP2	+						
SIP3	+						
SIP4	+						
SIP5	+						

Red arrows (1-5) trace the path of an outgoing call from a PSTN cloud through the gateway and internal registrations. Purple arrows (1-5) trace the path of an incoming call from the gateway through external registrations and internal registrations to the end device.



### 3.2 Outbound Manipulation Example 1

The extension "10" dials the official number "0041 111222333"

The screenshot shows the Asterisk PBX configuration interface with several key sections:

- 1. User Configuration:** Shows a user named 'Amt\_Master' with a 'Long Name' of 'Amt\_Master' and a 'No' of '0'. This is linked to the initial dialing sequence.
- 2. Gateway Configuration:** Shows a gateway configuration for 'Amt\_Master' with a 'Long Name' of 'Amt\_Master' and a 'No' of '0'. This is linked to the initial dialing sequence.
- 3. Routing Rule 1:** A routing rule that matches 'SIP1:Sip Amt' and 'RS1:Sip Amt' and sends the call to 'RS1:Sip Amt' with a 'Counter' of '00-00' and a 'CGPN Map' of 'MAP'. This is linked to the first routing entry in the list.
- 4. Routing Rule 2:** A routing rule that matches 'SIP1:Sip Amt' and 'RS1:Sip Amt' and sends the call to 'RS1:Sip Amt' with a 'Counter' of '00-00' and a 'CGPN Map' of 'MAP'. This is linked to the second routing entry in the list.
- 5. Routing Rule 3:** A routing rule that matches 'SIP1:Sip Amt' and 'RS1:Sip Amt' and sends the call to 'RS1:Sip Amt' with a 'Counter' of '00-00' and a 'CGPN Map' of 'MAP'. This is linked to the third routing entry in the list.
- 6. Routing Rule 4:** A routing rule that matches 'SIP1:Sip Amt' and 'RS1:Sip Amt' and sends the call to 'RS1:Sip Amt' with a 'Counter' of '00-00' and a 'CGPN Map' of 'MAP'. This is linked to the fourth routing entry in the list.
- 7. Interface Configuration:** Shows the 'SIP1 Sip Amt' interface with a 'CGPN-In' of 'i->00' and a 'CDPN-In' of '0301234560->666'. This is linked to the interface configuration table.

Additional annotations and tables:

- Initial Dialing:** CGPN-Out: 10 | CDPN-Out: 0-00 41 111222333
- After Gateway:** CGPN-Out: 10 | CDPN-Out: 00 41 111222333
- After Routing Rule 1:** CGPN-Out: 10 | CDPN-Out: 00-00 41 111222333
- After Routing Rule 2:** CGPN-Out: 00 49 30 123456-10 | CDPN-Out: 00 41 111222333
- After Routing Rule 3:** CGPN-Out: 00049 30 123456-10 | CDPN-Out: 0041 111222333
- After Routing Rule 4:** CGPN-Out: 0049 30 123456-10 | CDPN-Out: 0041 111222333
- Interface Mappings:**

Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out	State	Alias	Registrat
SIP1 Sip Amt	i->00	0301234560->666	00->i	00->i			
		030123456->	0->0	0->0			
SIP2	+						
SIP3	+						
SIP4	+						
SIP5	+						
- PSTN Cloud:** CGPN-Out: i49 30 123456-10 | CDPN-Out: i41 111222333

- The extension sends its single extension and must also dial the dial "0".  
**CGPN-Out: 10 | CDPN-Out: 0-0041 111222333**
- Once the call leaves the PBX and reaches the route, the "0" dial is automatically removed. You can't prevent that.  
**CGPN-Out: 10 | CDPN-Out: 0-0041 111222333 > 0041 111222333**
- The first routing entry only generates the official tone and has no other effect  
**CGPN-Out: 10 | CDPN-Out: 0041 111222333**
- The second routing entry checks if the CDPN starts with "00" and replaces it with "00" (left of the arrow truncates, right adds). So no change to the CDPN here.  
BUT: Since the CDPN starts with the "00", this route and thus the CGPN mapping applies.  
**CGPN-Out: 10 > 00049 30 123456-10 | CDPN-Out: 0041 111222333**
- The third routing entry always applies, so does this CGPN mapping.  
**CGPN-Out: 00049 30 123456-10 > 0049 30 123456-10 | CDPN-Out: 0041 111222333**
- The last entry routes to the SIP interface without manipulation  
**CGPN-Out: 0049 30 123456-10 | CDPN-Out: 0041 111222333**
- In the SIP interface, the interface mappings are still effective.  
**CGPN-Out: i49 30 123456-10 | CDPN-Out: i41 111222333**

### 3.3 Outgoing Manipulation Example 2

The extension "666" dials the official number "4473" (in the same city)

**1.** CGPN-Out: 666  
CDPN-Out: 0-4473

**2.** CGPN-Out: 666  
CDPN-Out: 4473

**3.** X

**4.**

**5.**

**6.** CGPN-Out: 030 123456-666  
CDPN-Out: 4473

**7.**

PSTN  
CGPN-Out: 030 123456-666  
CDPN-Out: 030 4473

Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out	State	Alias	Registrati
SIP1 Sip Amt	i→00	0301234560→666	00→i	00→i			
		030123456→	0→0	0→0			
SIP2	+						
SIP3	+						
SIP4	+						
SIP5	+						

1. The extension sends its single extension and must also dial the dial "0".  
**CGPN-Out: 666 | CDPN-Out: 0-4473**
2. Once the call leaves the PBX and reaches the route, the "0" dial is automatically removed. You can't prevent that.  
**CGPN-Out: 666 | CDPN-Out: 0-4473 > 4473**
3. The first routing entry only generates the official tone and has no other effect  
**CGPN-Out: 666 | CDPN-Out: 4473**
4. The second routing entry checks if the CDPN starts with "00" and replaces it with "00" (left of the arrow truncates, right adds). Because the CDPN does not start with the "00" in this example, this route does not apply  
**CGPN-Out: 666 | CDPN-Out: 4473**
5. The third routing entry always applies, so does this CGPN mapping.  
**CGPN-Out: 666 > 030 123456-666 | CDPN-Out: 4473**
6. The last entry routes to the SIP interface without manipulation  
**CGPN-Out: 030 123456-666 | CDPN-Out: 4473**
7. In the SIP interface, the interface mappings are still effective.  
**CGPN-Out: 030 123456-666 | CDPN-Out: 4473 > 030 4473**

### 3.4 Example of incoming manipulation 1

An incoming call with the number 0041 111222333 to extension "0"

The screenshot illustrates the configuration and call flow for an incoming call. It is divided into four numbered steps:

- 1. PSTN:** The call originates from a PSTN cloud with **CGPN-In: i41 111222333** and **CDPN-In: 030 123456-0**. This information is reflected in the SIP interface configuration table.
- 2. SIP Interface:** The call enters through the SIP interface. The configuration table shows mappings for SIP1 Sip Amt, where the incoming CDPN (030123456) is mapped to an outgoing CDPN (666).
- 3. Routing:** The call is routed through the 'Routes' configuration. The 'From' field is 'RS1:Sip Amt' and the 'To' field is 'RS1:Sip Amt'. The routing process does not manipulate the numbers at this stage.
- 4. Trunk Object:** The trunk object automatically adds its own prefix '0-'. The final call details show **CGPN-In: 0-0041 111222333** and **CDPN-In: 666**.

Additional configuration details shown include the 'PBX' and 'Gateway' sections, and a table of interface mappings:

Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out	State	Alias	Registration
SIP1 Sip Amt	i→00	0301234560→666	00→i	00→i			
		030123456→	0→0	0→0			
SIP2	+						
SIP3	+						
SIP4	+						
SIP5	+						

1. The call arrives with the "i" instead of 00.  
**CGPN-In: i41 111222333 | CDPN-In 030 123456-0**
2. In the SIP interface, the interface mappings come into play first.  
**CGPN-In: i41 111222333 > 0041 111222333 | CDPN-In 030 123456-0 > 666**
3. There is no manipulation in the incoming routing.  
**CGPN-In: 0041 111222333 | CDPN-In 666**
4. The trunk object automatically adds its own prefix. You can't prevent that.  
**CGPN-In: 0-0041 111222333 | CDPN-In 666**

### 3.5 Example of incoming manipulation 2

An incoming call to 030 4473 to extension "10"

The screenshot illustrates the configuration and call flow for an incoming call. It shows the SIP interface configuration, routing rules, and trunk object settings. The call flow is annotated with four steps:

- 1.** The call arrives in the normal national format. **CGPN-In: 030 4473 | CDPN-In 030 123456-10**
- 2.** In the SIP interface, the interface mappings come into play first. **CGPN-In: 030 4473 | CDPN-In 030 123456-10 > 10**
- 3.** There is no manipulation in the incoming routing. **CGPN-In: 030 4473 | CDPN-In 10**
- 4.** The trunk object automatically adds its own prefix. You can't prevent that. **CGPN-In: 0-030 4473 | CDPN-In 10**

1. The call arrives in the normal national format.  
**CGPN-In: 030 4473 | CDPN-In 030 123456-10**
2. In the SIP interface, the interface mappings come into play first.  
**CGPN-In: 030 4473 | CDPN-In 030 123456-10 > 10**
3. There is no manipulation in the incoming routing.  
**CGPN-In: 030 4473 | CDPN-In 10**
4. The trunk object automatically adds its own prefix. You can't prevent that.  
**CGPN-In: 0-030 4473 | CDPN-In 10**

### 3.6 Summary Example Call Forwarding

The extension "666" has a diversion to the 004111222333

**1.** CGPN-In: 030 4473 | CDPN-In 030 123456-10  
CGPN-Out: i49 30 4473 | CDPN-Out: i41 111222333

**2.** CGPN-In: 0-030 4473 | CDPN-In: 10  
CGPN-Out: 0-030 4473 | CDPN-Out: 0-00 41 111222333

**3.** CGPN-Out: 0-030 4473 | CDPN-Out: 0-00 41 111222333  
CGPN-In: 030 4473 | CDPN-In: 10

**4.** CGPN-Out: 0-030 4473 | CDPN-Out: 00 41 111222333  
CGPN-In: 030 4473 | CDPN-In: 10

**5.** TONE

**6.** MAP

**7.** MAP

**8.** CGPN-In: 030 4473 | CDPN-In: 10  
CGPN-Out: 00 49 30 4473 | CDPN-Out: 00 41 111222333

**9.** CGPN-In: 030 123456-10 | CDPN-In: 030 123456-10  
CGPN-Out: i49 30 4473 | CDPN-Out: i41 111222333

Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out	State	Alias	Registrati
SIP1 Sip Amt	i→00	0301234560→666	00→i	00→i			
		030123456→	0→0	0→0			
SIP2	+						
SIP3	+						
SIP4	+						
SIP5	+						

1. The call arrives in national format and is customized in the interface mapping.  
**CGPN-In: 030 4473 | CDPN-In 030 123456-10**
2. In the PBX, the sender phone number arrives with the official prefix.  
**CGPN-In: 0-030 4473 | CDPN-In 10**
3. The outgoing phone number during the redirection is the same as the original number including the "prefix"  
**CGPN-Out: 0-0304473 | CDPN-Out: 0-0041 111222333**
4. Once the call leaves the PBX, only the CDPN prefix "0" is removed. (NOT the CGPN).  
**CGPN-Out: 0-0304473 | CDPN-Out: 0041 111222333**
5. The first routing entry only generates the official tone and has no other effect
6. The second routing entry checks if the CDPN starts with "00" and replaces it with "00". Since the CDPN matches, the CGPN mapping is done.  
**CGPN-Out: 00049 304473 | CDPN-Out: 0041 111222333**
7. The third routing entry always applies, so does this CGPN mapping.  
**CGPN-Out: 0049 304473 | CDPN-Out: 0041 111222333**
8. The last entry routes to the SIP interface without manipulation  
**CGPN-Out: 0049 304473 | CDPN-Out: 0041 111222333**
9. In the SIP interface, the interface mappings are still effective.  
**CGPN-Out: i49 304473 | CDPN-Out: i41 111222333**



## 4 Special Route Options

In the routes themselves, various special settings can be made or routing destinations can be set.

A description of all options can be found in [Help]. The main options for routing manipulations are as follows:

### Route "WENN":

This route only applies if the initial digits of the CDPN match this entry. Otherwise, this route will be skipped. If these initial digits are to be retained, they must be entered 1 to 1 in the "Replace field".

### Verify CGPN:

This route only works if the initial digits of the CGPN match an entry from the "CGPN Mapping"

### Route target MAP:

A "MAP" is actually a real goal. There is only a number mapping, after which the route continues normally. This entry is used when you want to manipulate the phone number in the route.

### Route target TONE:

Only one of the TONES is played, after which the route continues normally.

### Route target DISC:

Only a DISCONNECT is sent, after which the route aborts.





## 5.2 Gateway interface with dedicated internal registration

You can also set up a gateway interface without an internal registration. This then only connects to the SIP provider EXTERNALLY:

1. EXTERNAL > To the public network (e.g. SIP or ISDN)
2. INTERNAL > Off

The screenshot shows the Asterisk Gateway configuration interface. The top navigation bar includes 'General', 'Interfaces', 'IP4', 'IP6', 'Services', 'PBX', 'Gateway', and 'Maintenance'. The 'Gateway' tab is active, and the 'SIP' sub-tab is selected. Below the navigation is a table of SIP interfaces:

Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out	State	Alias	Registration
SIP1 Sip Amt	i→00	030123456→00→i	00→i	00→i			Amt_Master
SIP2	+						
SIP3	+						
SIP4							
SIP5							

The configuration form for SIP4 is expanded and highlighted with a red box. It contains the following fields:

- Name:** Sip Amt
- Disable:**
- Type:** Provider
- Transport:** UDP  Without registration
- AOR:** [ ] @ [ ]
- Local Hostname:** [ ]
- Local Port:** [ ]
- Proxy:** [ ]
- STUN Server:** [ ]
- Authorization:**
  - Username:** qsclogin
  - Password:** [ ] Retype: [ ]
- Media Properties:**
  - General Codec Preference:** G711A
  - Framesize (ms):** 20
  - Silence Compress:** [ ]
- Internal Registration:**
  - Protocol:** None

The internal registration of the trunk object in the PBX is then done via its own GW interface. This is then set up at the "GK-Interfaces".

The screenshot shows the Asterisk Manager GUI configuration for a Gateway (GW2). The configuration is as follows:

Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out	Alias	Registration
GW1 Amt +	000→00	00→0	**10→**331	**20→**331	Amt	→ 192.168.198.136
			**30→**331			

Configuration details for GW2:

- Name: Amt
- Disable:
- Protocol: H.323/TLS
- Mode: Register as Gateway
- Address: 192.168.198.136
- Address (alternate): (NULL)
- Gatekeeper Identifier:
- Local Signaling Port:
- Authorization:
  - Password:
  - Retype:
- Alias List:
 

Name	Number
Amt	<input type="text"/>
- Media Properties:
  - General Codex Preference: G711A, Framesize [ms]: 20, Silence Compress:
  - Local Network Codex: G711A, Framesize [ms]: 20, Silence Compress:
  - Enable T.38:  No DTMF Detection:  Media-Relay: Off, Video:
  - SRTP Cipher: AES128/32, SRTP Key Exchange: SDES-DTLS, No ICE:  No RTCP:
  - Record to (URL):
- H.323 Interop Tweaks:
  - No Faststart:  No H.245 Tunneling:

- Now there are two independent registries that are not yet connected to each other. The connection is then made in the routes.



## 6 Advanced Routing Examples

### 6.1 Advanced Inbound Routing 1

Incoming call to the central "0" of SIP number 1

**5.** CGPN-In: 0-030 4473  
CDPN-In: \*\*331

*Hinweis: Bei der GW Registrierung zur PBX müssen IN und OUT umgedreht betrachtet werden!*

**4.** CGPN-In: 030 4473  
CDPN-In: \*\*10

**3.** CGPN-In: 030 4473  
CDPN-In: \*\*10

**2.** CGPN-In: 030 4473  
CDPN-In: i49 30 445566-0

**1.** CGPN-In: 030 4473  
CDPN-In: i49 30 445566-0

Interface	CGPN-In	CDPN-In	CGPN-Out	CDPN-Out	State
SIP1 M_net	i→00	i4930445566→**1	00→i	00→i	
		004930445566→**10	0→i49	0→i49	
			→i4930445566		
SIP2 EWE	i→00	i495114473→	00→i	00→i	
		00495114473→	0→i49	0→i49	
			→i495114473		
SIP3 Telekom	i→00	i4940506070→**3	00→00	00→00	
		004940506070→**30	0→0	0→0	
			→004940506070		

From	To	Counter	CGPN Maps
GW1:Amt	→ TONE	000...	→ 00
	→ MAP	00...	→ 0
	→ SIP1:M_net	bv	**1 →
	→ SIP2:EWE	bv	**2 →
	→ SIP3:Telekom	bv	**3 →
	→ SIP2:EWE	b	→
	→ SIP1:M_net		→ GW1:Amt
	→ SIP2:EWE		→ GW1:Amt

1. The call comes in via the 1st SIP trunk.  
**CGPN-In: 030 4473 | CDPN-In i49 30 445566-0**
2. In the mapping, the root number is truncated and replaced by \*\*1.  
**CGPN-In: remains 030 4473 | CDPN-In i49 30 445566-0 > \*\*1-0**
3. There is no manipulation in the routing for SIP1 and the call is routed to GW 1.  
**CGPN-In: 030 4473 | CDPN-In \*\*1-0**
4. In interface mapping, the \*\*1-0 is routed to the common control center \*\*3-31.  
**CGPN-In: 030 4473 | CDPN-In \*\*3-31**
5. The trunk object automatically adds its own prefix to the CGPN.  
**CGPN-In: 0-030 4473 | CDPN-In \*\*3-31**

## 6.2 Advanced Inbound Routing 2

Incoming call to an extension from SIP number 2

**5.** CGPN-In: 0-0041 111222333  
CDPN-In: \*\*230

*Hinweis: Bei der GW Registrierung zur PBX müssen IN und OUT umgedreht betrachtet werden!*

**4.** CGPN-In: 0041 111222333  
CDPN-In: \*\*230

**2.** CGPN-In: i41 111222333  
CDPN-In: i49 511 4473 30

**3.** CGPN-In: 0041 111222333  
CDPN-In: 30

**1.** CGPN-In: i41 111222333  
CDPN-In: i49 511 4473 30

PSTN

1. The call comes in via the 2nd SIP trunk.  
**CGPN-In: i41 111222333 | CDPN-In i49 511 4473 30**
2. In the mapping, the CDPN root number is truncated and the CGPN "i" is replaced.  
**CGPN-In: i41 111222333 > 0041 111222333 | CDPN-In i49 511 4473 30 > 30**
3. In the routing for SIP2, \*\*2 is added and the call is routed to GW 1.  
**CGPN-In: 0041 111222333 | CDPN-In 30 > \*\*230**
4. There is no match in the interface mapping.  
**CGPN-In: 0041 111222333 | CDPN-In \*\*230**
5. The trunk object automatically adds its own prefix to the CGPN.  
**CGPN-In: 0-0041 111222333 | CDPN-In \*\*230**

### 6.3 Advanced Outbound Routing 1

The extension "30" of node "\*\*1" dials the official number "0041 111222333"

**1.** CGPN-Out: **\*\*1 30**  
CDPN-Out: **0-00 41 111222333**

*Hinweis: Bei der GW Registrierung zur PBX müssen IN und OUT umgedreht betrachtet werden!*

**2.** Interface CGPN-In CDPN-In CGPN-Out CDPN-Out

**3.** Alias Registration

CGPN-Out: **\*\*1 30**  
CDPN-Out: **00 41 111222333**

**4.** GW1:Amt

**5.** TONE

**6.** MAP

**7.** SIP1:M\_net

CGPN-Out: **30 (SIP1)**  
CDPN-Out: **00 41 111222333**

PSTN

CGPN-Out: **i49 30445566 30 (SIP1)**  
CDPN-Out: **0041 111222333**

- 1.** The extension transmits the complete phone number as CGPN and has to dial the dial "0".  
**CGPN-Out: \*\*1 30 | CDPN-Out: 0-0041 111222333**
- 2.** As soon as the call leaves the PBX, the "0" dial is automatically removed.  
**CGPN-Out: \*\*1 30 | CDPN-Out: 0-0041 111222333 > 0041 111222333**
- 3.** There is no match in the interface map of GW1.  
**CGPN-Out: \*\*1 30 | CDPN-Out: 0041 111222333**
- 4.** The first routing entry only generates the official tone.
- 5.** In the second routing entry, there is a CGPN mapping, but there is no match because the sender number does not start with "00" or "000".
- 6.** The third routing entry takes effect because "Verify CGPN" is set. In addition, the \*\*1 will be removed from the CGPN.  
**CGPN-Out: \*\*1 30 > 30 | CDPN-Out: 0041 111222333**
- 7.** In the SIP interface, the interface mappings are still effective.  
**CGPN-Out: 30 > i4930 445566 30 | CDPN-Out: 0041 111222333**



## 6.5 Advanced Outbound Routing 2b

The extension "\*\*\*2 30" is redirected to "0-0041 111222333"

**1. CGPN-Out: 001525 3600777 (Rufumleitung)**  
**CDPN-Out: 0-00 41 111222333**

*Hinweis: Bei der GW Registrierung zur PBX müssen IN und OUT umgedreht betrachtet werden!*

**2.** Interface CGPN-In CDPN-In CGPN-Out CDPN-Out

**3.** Alias Registration

**4.** TONE

**5.** SIP1:M\_net SIP3:Telekom

**6.** SIP1 M\_net

**CGPN-Out: 01525 3600777**  
**CDPN-Out: 00 41 111222333**

**CGPN-Out: 01525 3600777 (SIP2)**  
**CDPN-Out: 00 41 111222333**

**CGPN-Out: i49 1525 3600777**  
**CDPN-Out: 0041 111222333**

PSTN

➤ *Note: The difference from the previous example is that the removal of the leading "0" on the mobile number is now done in the mapping of the GW interface. This example is only intended to illustrate once again that there are often several good solutions.*

- 1.** The mobile phone number is transmitted with the leading "Amt-0" as CGPN, just as it was signalled when it arrived.  
**CGPN-Out: 001525 3600777 | CDPN-Out: 0-0041 111222333**
- 2.** As soon as the call leaves the PBX, the "0" dial is automatically removed.  
**CGPN-Out: 001525 3600777 | CDPN-Out: 0-0041 111222333 > 0041 111222333**
- 3.** In the interface map of GW1, the CDPN mapping applies.  
**CGPN-Out: 01525 3600777 | CDPN-Out: 0041 111222333**
- 4.** The first routing entry only generates the official tone.
- 5.** Only the last routing entry takes effect again, because no "Verify CGPN" fits before. There is no more manipulation here.  
**CGPN-Out: 01525 3600777 | CDPN-Out: 0041 111222333**
- 6.** In the SIP interface, the interface mappings are still effective.  
**CGPN-Out: i49 1525 3600777 | CDPN-Out: 0041 111222333**



## 7 PBX: "Trunk Line" Object vs. "Gateway" Object

For most connections from external connections, a "Trunk Line" object is better suited, because here the drop targets can be easily declared, and special trunk settings are stored.

Most "gateway" objects are better suited for networking to other PBXs, because they allow you to make phone number plan settings.

However, there are crucial differences when it comes to routing and number manipulation:

- **Outgoing:** The "Gateway" object submits its own prefix to the routing.
  - **Incoming:** The "Gateway" object does NOT add its own prefix to the call.
- As a reminder, the trunk object always has the office prefix added when it arrives and is automatically removed when the outgoing dial is made. In other words, exactly the opposite of the gateway. However, if you check the "Prefix" box for the gateway, it behaves like a trunk object again.

The image shows two overlapping configuration windows from a PBX management system. The top window is for a 'Trunk Line' object, and the bottom window is for a 'Gateway' object. Both windows have a 'General' tab selected.

**Trunk Line Configuration:**

- Type: Trunk Line
- Description: [Empty]
- Long Name: Amt
- Name: Amt
- E-Mail: Amt
- Password: [Masked]
- Node: root
- PBX: berlin
- Send Number: [Empty]
- Max Calls: [Empty]
- Hide Connected Endpoint:
- UC:
- Reporting:
- Voicemail:
- Devices: [Table with Hardware Id and Name columns]
- Loopback:
- Incomplete:
- Invalid:
- Busy:
- Rejected:
- No Answer:
- Reroute supported:
- Set Calling=Diverting No:
- Set Diverting=Calling No:
- Discard received diverting No:
- Outgoing Calls restricted:
- Automatic Hangup:
- Outgoing Calls CGPN:
- Outgoing Calls No Name:
- No Presence/Dialog Subscribe:
- Fake Connect on inc. Call:
- Filter: [Empty]
- Name as Number:

**Gateway Configuration:**

- Type: Gateway
- Description: [Empty]
- Long Name: Amt
- Name: Amt
- E-Mail: Amt
- Prefix:  (highlighted with a red box)
- Don't add if CGPN matches escape:
- Domain: [Empty]
- Loop Detect:
- International Match: [Empty]
- National Match: [Empty]
- Subscriber Match: [Empty]
- Set incoming call UUI: [Empty]
- Set outgoing call UUI: [Empty]
- Internal Destination:
- Outgoing Calls No Name:
- Outgoing Calls No URL:
- No Presence/Dialog Subscribe:
- Dialtone on Incoming calls:
- No Inband Disconnect:
- Fax License:
- Filter: [Empty]

## Contact

Do you have any questions about the content, interest in my service or any other concerns?

I look forward to your message.

Tobias Rust

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