TELES AG
Informationstechnologien

CELLX

Reference Manual

Software version 16.2

CELLX

Reference Manual

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1 About this manual

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Congratulations on the purchase of your new CELLX! This manual is set up to guide you through the step-by-step installation of your CELLX, so that you can follow it through from the front to the back. Quick-installation instructions appear in Chapter 4.7 Startup with Quickstart on page 25.

Make sure you familiarize yourself thoroughly with the safety and security precautions detailed in Chapter 2 Safety and security precautions before you begin to install your CELLX. TELES is not liable for any damage or injury resulting from a failure to follow these safety and security instructions!

1.1 Organization

This manual is organized into the following chapters.

- Chapter 1 About this manual introduces the CELLX Systems Manual and how it is set up.
- Chapter 2 Safety and security precautions contains information about security issues relevant to connection with the IP network.
- Chapter 3 Overview briefly describes the CELLX and its implementation scenarios.
- Chapter 4 Installation contains information on how to connect and configure the system so that it is ready for operation.
- Chapter 5 Configuration files describes the CELLX's individual configuration files and parameters.
- Chapter 7 Mobile configuration options describes mobile configuration entries.
- Chapter 8 Signaling and routing features describes configuration settings in the `route.cfg` used for adjusting PRI signaling and customizing the configuration for specific scenarios.
- Chapter 9 Additional VoIP parameters contains additional configuration entries to fine-tune communication with the VoIP peer.
- Chapter 10 System maintenance and software update describes system messages that are saved in the protocol file, as well as trace options.
- Chapter 12 Troubleshooting contains troubleshooting suggestions.

1.2 Conventions

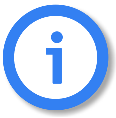
This document uses the following typographic conventions:

- **Bold** – items from the GUI menu.
- Halfbold – items from the GUI and the menu.
- `Code` – file names, variables and constants in configuration files or commands in body text.
- "Conventions" on page 7 – cross-references can be accessed in the PDF files by a single mouse click.

Configuration data or extracts are written in single-column tables with a gray background.

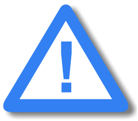
1.3 Safety symbols

The following symbols are used to indicate important information and to describe levels of possible danger.



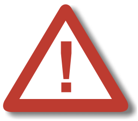
Note

Useful information with no safety implications.



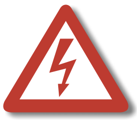
Attention

Information that must be adhered to as it is necessary to ensure that the system functions correctly and to avoid material damage.



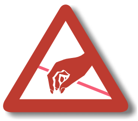
Warning

Danger. Could cause personal injury or damage to the system.



Dangerous voltage

Could cause injury by high voltage and/or damage the system.



Electrostatic discharge

Components at risk of discharge must be grounded before being touched.

2 Safety and security precautions

2 Safety and security precautions

Please be sure and take time to read this section to ensure your personal safety and proper operation of your TELES Infrastructure System.

To avoid personal injury or damage to the system, please follow all safety instructions before you begin working on your TELES Infrastructure System.

TELES Infrastructure Systems are CE certified and fulfill all relevant security requirements. The manufacturer assumes no liability for consequential damages or for damages resulting from unauthorized changes.

This chapter applies for all Access Gateways. Information that applies only for individual Access Gateways specifies the system for which it applies.

2.1 Safety measures

Danger of electric shock - the power supplies run on 230 V. Unplug the TELES Infrastructure System from its power source before working on the power supply or extension socket.
Bear in mind that telephone and WAN lines are also energized and can cause electric shocks.
Do not insert foreign objects into openings in the device. Conductible objects can cause short circuits that result in fire, electric shock or damage to the device.
Do not open the TELES Infrastructure System except to install an additional TELES.Component. Changes in the device are not permitted.

Make sure to install the system near the power source and that the power source is easily accessible. Wire your system using only the cables included in the package contents. Use only proper ISDN and Ethernet cables. Be sure to respect country-specific regulations, standards or guidelines for accident prevention. Failure to follow these guidelines could result in system failure or damage.

2.2 FCC / Industry Canada Notice

The following information applies for CELLX gateways only. Changes or modifications not expressly approved by the party responsible for compliance could void the user's authority to operate the equipment.

This equipment complies with FCC radiation exposure limits set forth for an uncontrolled environment. This equipment must be installed and operated with a minimum distance between the antennas and persons of:

- 58 cm for iGATE and CELLX GSM or
- 20 cm for CELLX CDMA.

The minimum distance between gateway antenna and other antennas must be:

- 262cm for iGATE and CELLX GSM or
- 20 cm for CELLX CDMA.

The CELLX has been tested and found to comply with the limits for a Class B digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference in a residential installation. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instructions, may cause harmful interference to radio communications. However, there is no guarantee that interference will not occur in a particular installation. If this equipment does cause harmful interference to radio or television reception, which can be determined by turning the equipment off and on, the user is encouraged to try to correct the interference by one or more of the following measures:

- Reorient or relocate the receiving antenna.
- Increase the separation between the equipment and receiver.
- Connect the equipment into an outlet on a circuit different from that to which the receiver is connected.
- Consult the dealer or an experienced radio/TV technician for help.

2.3 EMC protection

Use shielded cables.
Do not remove any housing components. They provide EMC protection.

2.4 System security

This section describes all points crucial to the TELES Infrastructure System's system security.

The system's location must support normal operation of TELES Infrastructure Systems according to EN ETS 300 386. Be sure to select the location with the following conditions in mind.

Location: Make sure you install the system horizontally in a clean, dry, dust-free location. If possible, use an air-conditioned site. The site must be free of strong electrical or magnetic fields, which cause disrupted signals and, in extreme cases, system failure.

Temperature: The site must maintain a temperature between 32 and 113°F and provide adequate ventilation. Be sure to guard against temperature fluctuations. Resulting condensation can cause short circuiting. The humidity level may not exceed 80%.
To avoid overheating the system, make sure the site provides adequate ventilation.

Power: The site must contain a central emergency switch for the entire power source.
The site's fuses must be calculated to provide adequate system security. The electrical facilities must comply with applicable regulations.
The operating voltage and frequency may not exceed or fall below what is stated on the label.
Antenna: CELLX contains no provision or protective device against power surges or lightning strikes.
The installation of the antenna must fulfill all necessary safety requirements. Employ the services of a professional antenna installer.

2.5 Servicing the system

Regular servicing ensures that your TELES.System runs trouble-free. Servicing also includes looking after the room in which the system is set up. Ensure that the air-conditioning and its filter system are regularly checked and that the premises are cleaned on a regular basis.

2.5.1 Replacing components

If your system contains any of the following components, replace them according to the following table:

Table 2.1 Component life span

Component	Life span
Filter pads	6 months
Power adapter	5 years
Fan	5 years

2.5.2 Protecting the operating system

Changing configuration data and/or SIM card positions may lead to malfunctions and/or misrouting, as well as possible consequential damage. Make changes at your own risk. TELES is not liable for any possible damage resulting from or in relation to such changes. Please thoroughly check any changes you or a third party have made to your configuration!

To make changes in or perform tests on the database, make sure your hard disk or flash disk contains enough storage space. Downloading the log files and deleting them from the system on a regular basis will ensure your system's reliability.

Be careful when deleting files that you do not delete any files necessary for system operation.To check storage space and/or delete files, use GATE Manager. For more information see the document *TELES.GATE Manager*.

All files with the extension `*.log` can be deleted. To save files before deleting them, use the **Receive File** option in GATE Manager.

The following files, if included, must not be deleted:

Table 2.2 Mandatory files

Mandatory files
Mandatory system files
`boot.rc`
`crypto.vnd`
`crypto5.vnd`
`gbox.tz1`
`gbox5.tz1`
`igate.tz1`
`IMEIs.lst`
`license.key`
`netbsd5fs.vnd`
`netbsd5i`
`netbsd5z`

Table 2.2 Mandatory files *(continued)*

Mandatory files
`netbsdfs.gz`
`netbsdi`
`netbsdz`
`start`
tools.tz0
`xgate.tz1`
`xgate.vnd`
Mandatory configuration files
`ip.cfg`
`pabx.cfg`
`route.cfg`

2.6 CDR files

Call Detail Records are intended for analysis of the system's activity only. They are not designed to be used for billing purposes, as it may occur that the times they record are not exact.

2.7 Network security

Every day hackers develop new ways to break into systems through the Internet. While TELES takes great care to ensure the security of its systems, any system with access through the Internet is only as secure as its user makes it. Therefore, to avoid unwanted security breaches and resulting system malfunctions, you must take the following steps to secure your TELES system if you connect it to the Internet:

- Use an application gateway or a packet firewall.
- To limit access to the system to secure remote devices, delete the default route and add individual secure network segments.
- Access to the system via Telnet, FTP, GUI, or GATE Manager must be password protected. Do not use obvious passwords (anything from `sesame` to your mother-in-laws maiden name). Remember: the password that is easiest to remember is also likely to be easiest to crack.

The firewall must be able to check the following information and only allow trusted users to access the TELES system:

- IP source address
- IP destination address
- Protocol (whether the packet is TCP, UDP, or ICMP)
- TCP or UDP source port
- TCP or UDP destination port
- ICMP message type

For operation and remote administration of your TELES.System, open only the following ports only when the indicated services are used:

Table 2.3 Default ports used for specific services

Service	Protocol	Port
For all systems		
FTP	TCP	21 (default, can be set)
Telnet (for TELES debug access only)	TCP	23 (default, can be set)
SMTP	TCP	25
DNS forward	UDP	53
HTTP	TCP	80 (default, can be set)
SNTP	UDP	123
SNMP	UDP	161 (default, can be set)
H.225 registration, admission, status	UDP	1719 (default, can be set)
H.225 signaling	TCP	1720 (default, can be set)
Radius	UDP	1812 (default, can be set)
Radius accounting	UDP	1813 (default, can be set)
GATE Manager	TCP	4445 (default, can be set)
SIP signaling	UDP / TCP	5060 (default, can be set)
RTP	UDP	29000-29120 (default, can be set)
For NMS		
FTP	TCP	21
Telnet	TCP	23
MySQL database	TCP	3306
NMS protocol	TCP	5000
NMS update	TCP	5001
NMS task	TCP	5002
NMS task	TCP	5003
NMS Listen	TCP	4444
For RoutingManager		
Radius authentication	UDP	1812
Radius accounting	UDP	1813

3 Overview

3 Overview

Mobile phone charges have become an important cost factor for many companies.

The CELLX can help reduce these costs, because calls cost significantly less when they occur between cell phones that share the same plan.

Depending on whether your system includes 4 GSM Boards, CDMA Boards, each CELLX can provide direct access to the GSM, CDMA or UMTS mobile network with up to 32 mobile channels – 4 mobile channels per Mobile Board or up to 8 Mobile Boards per CELLX. The Antenna Splitter Board combines the antennas so that only one or two antennas leave the system.

The CELLX has 2 PRI ports and VoIP functionality for up to 32 channelsCELLXs can be set up in various national or international locations.

3.1 What’s new in version 16.2

- Enhanced TLS stability in bad IP network conditions.
- Initial charge now sent immediately after connect by using the InitialCharge=ON option.
- New payload G726/40 introduced.
- DSS1: Improved handling of call forwarding / partial rerouting. Facility messages with 2 facility info elements are forwarded transparently from ISDN to ISDN; facility info element in Setup message is forwarded transparently from ISDN to ISDN.
- For calls from or to ISDN or VoIP, a second OAD can now be transmitted.
- Improved number manipulation with VoipOad and VoipDad commands.
- New system file tools.tz0 now exists in addition to netbsdfs.gz.
- System files ipv4.vnd, xgate.vnd, and netbsdi discontinued starting version 16.2.

3.2 New Access Gateway product names starting version 16.1

Starting from release 16.1., TELES is dividing it’s access gateway product portfolio into VoIPBox VoIP-ISDN gateways, compact ECOTEL mobile radio gateways and flexibly configurable iGATE mobile radio gateway systems in a 19” chassis. The product names now clearly reflect the three different product lines. The following table lists the old and new access gateway product names.

Table 3.1 New Access Gateway product names

Old AGW Product Names	New AGW Product Names since version 16.1.
VoIPBOX BRI	VoIPBox BRI
VoIPBOX PRI	VoIPBox PRI
VoIPBOX GSM	ECOTEL GSM
VoIPBOX UMTS	ECOTEL 3G
iGATE GSM	iGATE GSM
iGATE 3G	iGATE 3G

3.3 How CELLX works

The CELLX is connected to the PBX and to the mobile network.

- During outgoing calls from the PBX or IP network to mobile, dialed digits are compared with the routing-table entries for various mobile networks. The calls are then routed through the corresponding SIMs in the CELLX and forwarded to the number dialed.
- Only the connection from the SIM in the CELLX to the mobile number in the same mobile network is charged.
- Inbound c alls are forwarded to your PBX
- The CELLX contains SIM cards with your company’s billing plan

3.4 Supported implementation scenarios

In each of the following scenarios, calls are routed through individual gateways into the mobile network:

a) **Failover application**
In case of a landline connectivity failure, the PBX routes outgoing calls to the CELLX which in turn forwards the calls directly to the mobile network.

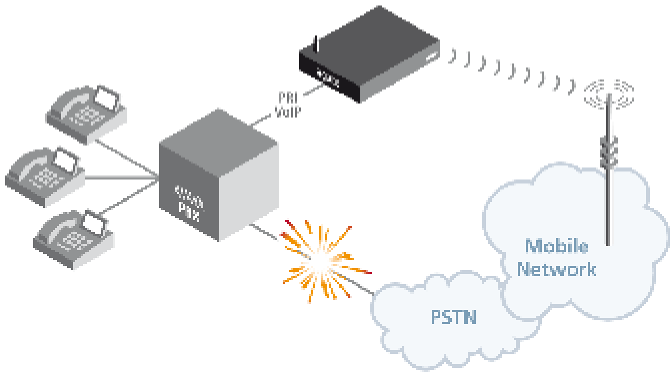


Figure 3.1 Failover application

b) **one-x mobile UC integration**
A company's cell-phone users have smart phones running Avaya's one-X Mobile client. The one-X mobile client requests a callback from the PBX via IP whenever these users make a call. This callback is then routed through the CELLX. As soon as the callback is answered, the PBX calls the B party and connects the call.

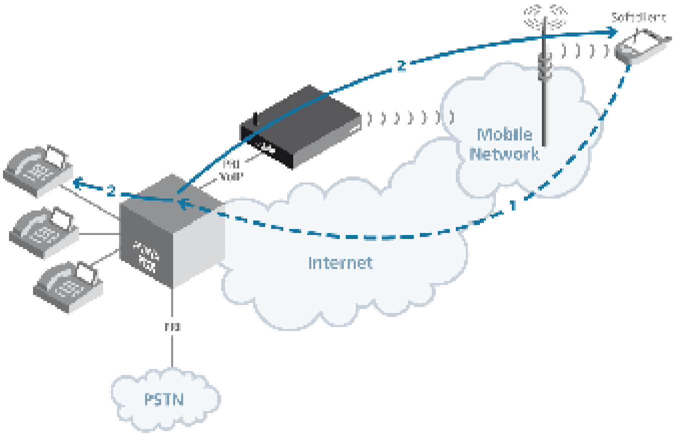


Figure 3.2 one-x mobile UC integration

4 Installation

4 Installation

Follow the easy instructions to set up your CELLX in a matter of minutes. Implementation of individual scenarios requires adjustments to the appropriate interfaces. Tips for basic settings are described here. Links to relevant chapters are provided for more specific configuration changes.

4.1 Checklist

The following checklist provides step-by-step installation instructions.

1. Check the package contents
2. Install the device
3. Connect the Ethernet
4. Connect the T1/E1 trunks (optional)
5. Connect the antennas
6. Using Quickstart, set the configuration (IP address)
7. Check functionality (using the LEDs)
8. Secure the LAN connection
9. Secure connection with the configuration program

4.2 Package contents

Your CELLX package contains the following components. Check the contents to make sure everything is complete and undamaged. Immediately report any visible transport damages to customer service. If damage exists, do not attempt operation without customer-service approval:

- 1 CELLX
- 1 power supply cable
- 1 crossover PRI cable
- 1 RJ-45 LAN cable with gray connectors; 3 meters
- 1 copy of quick installation instructions
- 1 CD containing Quickstart, GATE Manager, system manual and default configuration files
- Mobile antennas (optional)

4.3 Hardware description

Throughout this manual, the following boards will be referred to as Mobile Board, unless otherwise specified:

- 4 GSM Board
- CDMA Board

The CELLX is available in expansion levels from 4 to 32 mobile channels. The following pages describe installation of the CELLX .

Figure 4.1 shows the rear view of a CELLX, which contains the following boards:

From left to right:

- Base Board
- Mobile Board (for mobile channels 1-4)
- Mobile Board (for mobile channels 5-8)
- Mobile Board (for mobile channels 9-12)
- Mobile Board (for mobile channels 13-16)
- Optional Antenna Splitter Board
- Mobile Board (for mobile channels 17-20)
- Optional Mobile Board (for mobile channels 21-24)
- Optional Mobile Board (for mobile channels 25-28)
- Optional Mobile Board (for mobile channels 29-32)

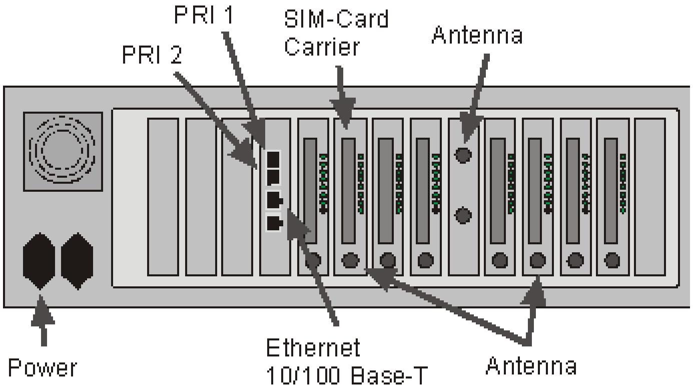


Figure 4.1 4HU CELLX

4.4 Installation requirements

Before installing your CELLX, make sure you have the following connections in place:

- Ethernet connection
- Antenna connection(s)
- Optional ISDN PRI connection to PBX
- Power
- Insert the SIM cards into the SIM card carrier, the SIM card carrier into the Mobile Board.

4.4.1 Ethernet wiring

To connect the CELLX's Ethernet port to your local network, connect the system to an Ethernet switch in your network. Use the three meter cable with gray connectors.

4.4.2 PRI wiring

4.4.2.1 TELES to TBR12

If you are connecting an CELLX to T1/E1 and need to change the assignment of an adapter, assign the pins as follows. Connectors on cables included with the CELLX will be gray for TELES TE and gray for NT on the remote device, blue for TELES NT, and green for TE on the remote device:

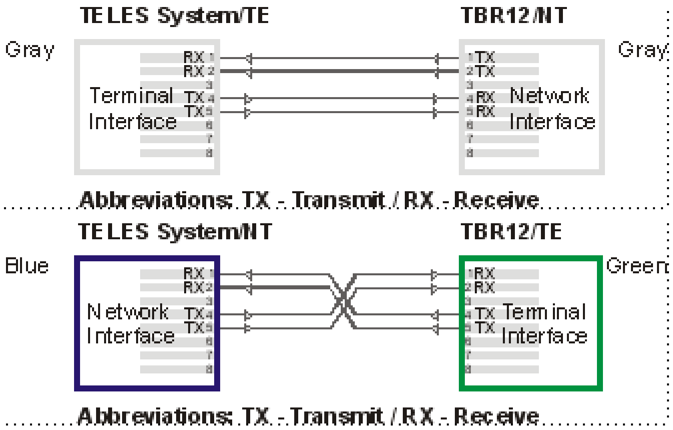


Figure 4.2 TELES to TBR12

4.4.3 Antenna connection

Plug an antenna cable into each of the SMA jacks. If the system contains a Antenna Splitter Board, plug the antenna(s) in there. If not, plug them into the jacks on the Mobile Board.

Antennas connected to the CELLX must be installed by a qulaified technician according to all necessary safety requirements and the antenna's installation specifications. The antenna adaptor does not provide power surge protection.

4.4.4 SIM card assignment

If your gateway is connected to a vGATE, the following information does not apply.

Each gateway has one or more slots for SIM card carriers. The SIM card carrier contains the SIM cards for the individual mobile channels. Insert the SIM cards in the SIM card carrier and then insert the SIM card carrier into the gateway.

You must configure the PINs in the `pabx.cfg` before inserting the SIM card carrier unless the SIM has no PIN or the PIN is 0000.

SIM card carriers are available in two versions, SIM24 and SIM4, whereby the number shows the number of available SIM card positions.

SIM cards are mounted on the front and back of the SIM24 module (Figure 4.3) or the front of the SIM4 module (Figure 4.4). As a guide to help you distinguish top from bottom on the SIM24 module, `SIM0-5` and `SIM12-17` are printed in the upper corner near the module's handle, as can be seen in Figure 4.3. The SIMs on the SIM4 module are numbered from right to left, with one SIM assigned to each mobile channel in ascending order.

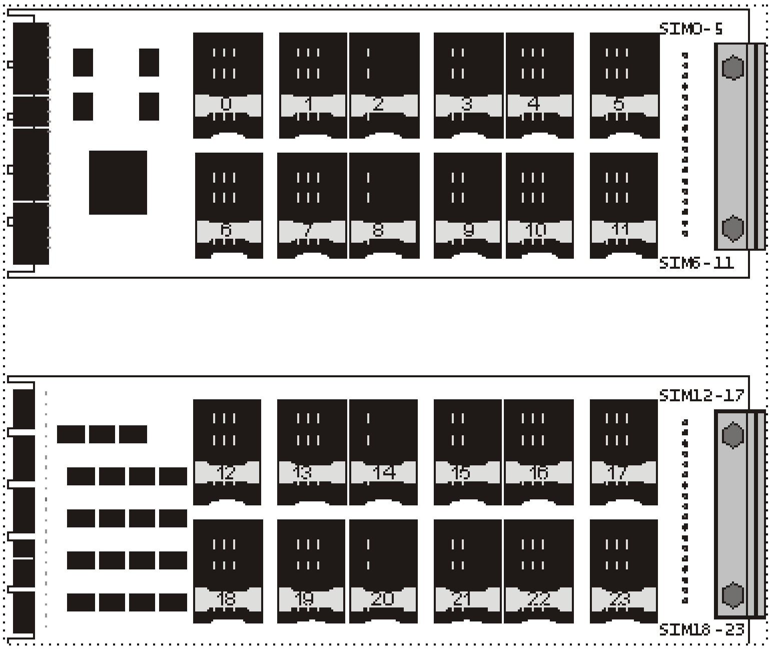


Figure 4.3 SIM24 module: front and rear view

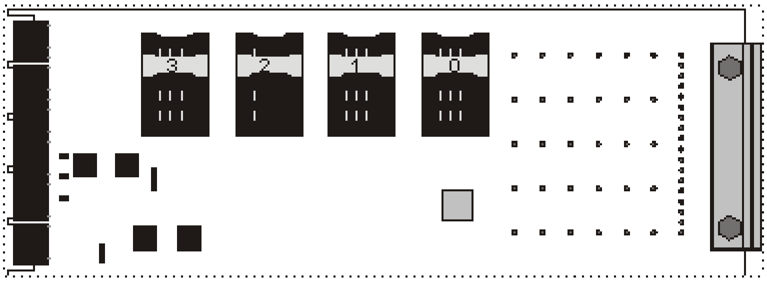


Figure 4.4 SIM4 module

If a SIM24 carrier is used, one out of six SIM cards can be assigned to a mobile controller. To configure that, an index has to be set in the `pabx.cfg` Subscriber line.

The following examples show how a SIM card at a certain position on the SIM24 carrier is assigned to a defined mobile controller and what index it needs for assignment.

Table 4.1 shows the assignment to the mobile controllers 00 to 03. The 24 available SIM card positions on the SIM24 carrier are listed in the body rows. The 6 different SIM cards which are available for each mobile controller are indexed in the heading row. This index needs to be entered in the `pabx.cfg` Subscriber line behind the SMSC entry. Example 4.1 shows that the SIM card at position 0 on the SIM24 carrier is assigned to mobile controller 00 by giving it the index 1. The SIM card which is at position 1 on the SIM24 carrier is assigned to the mobile controller 01 by also giving it the index 1. SIM card number 10 on the SIM24 carrier is assigned to mobile controller 02 with the index 3. Analogously, SIM card number 11 is assigned to mobile controller 03 with the index 3.

Table 4.1 SIM card assignment to mobile controllers 1 (SIM24 carrier)

Index in Subscriber line	**1**	**2**	**3**	**4**	**5**	**6**
Controller/Subscriber00	**0**	4	8	12	16	20
Controller/Subscriber01	**1**	5	9	13	17	21
Controller/Subscriber02	2	6	**10**	14	18	22
Controller/Subscriber03	3	7	**11**	15	19	23

Example 4.1 SIM card assignment to mobile controllers 1 (SIM24 carrier)

```
Subscriber00=TRANSPARENT GSM[0000,00000,+491770610000,1,1,1,SIM24] ALARM
Subscriber01=TRANSPARENT GSM[0000,00000,+491770610000,1,1,1,SIM24] ALARM
Subscriber02=TRANSPARENT GSM[0000,26202,+491770610000,3,1,1,SIM24] ALARM
Subscriber03=TRANSPARENT GSM[0000,00000,+491770610000,3,1,1,SIM24] ALARM
```

The following corresponding example shows how SIM cards at position 4, 5, 6, and 7 on the SIM24 carrier are assigned to the mobile controller 08 to 11. They all get the index 2 in the Subscriber line.

Table 4.2 SIM card assignment to mobile controllers 2 (SIM24 carrier)

Index in Subscriber line	**1**	**2**	**3**	**4**	**5**	**6**
Controller/Subscriber08	0	**4**	8	12	16	20
Controller/Subscriber09	1	**5**	9	13	17	21
Controller/Subscriber10	2	**6**	10	14	18	22
Controller/Subscriber11	3	**7**	11	15	19	23

Example 4.2 SIM card assignment to mobile controllers 2 (SIM24 carrier)

```
Subscriber08=TRANSPARENT GSM[0000,00000,+491770610000,2,1,1,SIM24] ALARM
Subscriber09=TRANSPARENT GSM[0000,00000,+491770610000,2,1,1,SIM24] ALARM
Subscriber10=TRANSPARENT GSM[0000,26202,+491770610000,2,1,1,SIM24] ALARM
Subscriber11=TRANSPARENT GSM[0000,00000,+491770610000,2,1,1,SIM24] ALARM
```

In a SIM4 carrier, each SIM card corresponds with one mobile controller. In the `pabx.cfg` Subscriber line each SIM card always gets the index 1, as shown in the example below.

Example 4.3 SIM card assignment to mobile controllers (SIM4 carrier)

```
Subscriber00=TRANSPARENT GSM[0000,00000,+491770610000,1,1,1,SIM4] ALARM
Subscriber01=TRANSPARENT GSM[0000,00000,+491770610000,1,1,1,SIM4] ALARM
Subscriber02=TRANSPARENT GSM[0000,26202,+491770610000,1,1,1,SIM4] ALARM
Subscriber03=TRANSPARENT GSM[0000,00000,+491770610000,1,1,1,SIM4] ALARM
```

4.5 Preparing for installation

Each computer that is to communicate with the CELLX requires a network connection. DHCP can be used to automatically assign an IP address and the netmask. If you don't use DHCP, please have the following information for connection to your network available:

- IP address in the local network for the CELLX to be configured
- Netmask for the CELLX to be configured
- Default gateway for CELLX to be configured

Bear in mind that the preconfigured CELLX's default IP address is 192.168.1.2. If it is already being used in your local network, you must run Quickstart without a connection to your local network. This can occur using a back-to-back Ethernet connection from your computer to the CELLX. If the desired IP address for the CELLX is not in your network, you must assign your computer a temporary IP address from this range.

4.6 Hardware connection

- Connect your computer with the local network
- Connect the CELLX with the local network
- If you choose to connect the CELLX to ISDN, use the ISDN connection cables included in the package contents to connect the CELLX with your PBX and/or the PSTN according to the required port configuration.
- Connect the CELLX to the power supply.

4.7 Startup with Quickstart

Quickstart is an application that helps you to configure the IP settings of your CELLX quickly and conveniently without changing any network settings on your computer.

Quickstart can be installed on any of the following operating systems:

- Windows 2000
- Windows XP
- Windows Vista
- Windows 7

If you are using any of these operating systems, please follow the instructions in this chapter.

4.7.1 Installing Quickstart

Make sure the GATE Manager is not running on your computer. To install Quickstart on your computer, insert the CD and select Quickstart from the menu.

When asked if you want to install components on your machine, click **Install**.

Click **Next** in the introduction window to begin installation of the Quickstart.

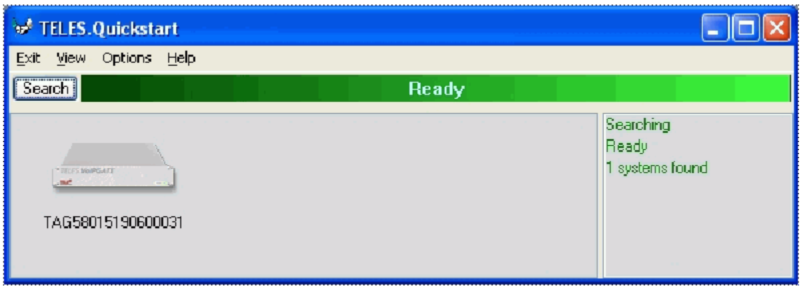
Once installation begins, click **Next** to install Quickstart in the predefined folder. To install it in another location, click **Browse** and select a folder from the browser that appears. Then click **Next**.

Click **Close** to exit when installation is complete.

4.7.2 Configuration with Quickstart

Now you can use Quickstart, to set up your CELLX's IP configuration. Open Quickstart`.exe`. The program will automatically search for your CELLX in the local network. For Quickstart, the source UDP port is 57445. It might be necessary to change the firewall rules on your system.

Click **Search** if you would like to restart the search. When the program has found your CELLX, it will appear in the window. As soon as it appears, you can end the search by clicking **Stop**.



Searching
Ready
1 systems found

Figure 4.5 Quickstart

The system's icon will appear in gray if it is unconfigured. Once it has been configured, it will appear in green. The serial number appears as the system's name. The CELLX is partially pre-configured. The configuration files `pabx.cfg` and `route.cfg` are already on the system. Only the system's IP-related entries must be set. Individual port adjustments are to be made manually later. Port properties can be changed and parameters can be assigned then.

To change the appearance of the window, click **Large Icons**, **Small Icons** or **Details** in the **View** menu. In the following description, we will use the Details View, which contains the following columns:

Table 4.3 Quickstart details view columns

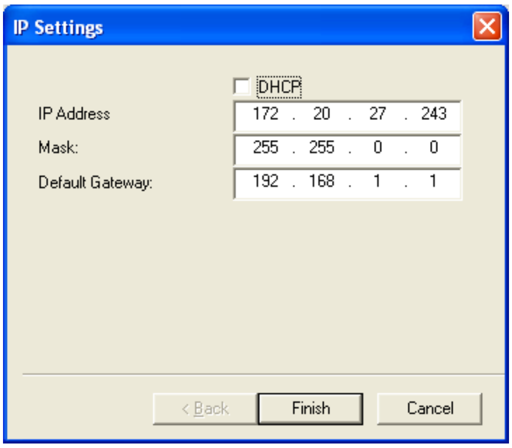
Heading	Definition
Identifier	This column lists the CELLX's serial number.
IP Address	This column lists the CELLX's IP address.
Configured	An X means the CELLX contains the configuration files.
# of VoIP Ctrls	This column lists the number of VoIP Modules installed in the CELLX. Each VoIP Module represents one VoIP controller.
VoIP Channels	This column shows the number of VoIP channels per VoIP Module.
Type	Lists the type of the system.

Table 4.3 Quickstart details view columns *(continued)*

Heading	Definition
Box	An X means the system is a VoIPBox BRI.
CF Mounted	An X means the CELLX contains a compact flash disk.

In the **Options** menu, you can suppress or activate ICMP ping to test the Internet connection.

To perform the initial configuration of the system, double-click the icon or right-click and select **Configure**. The **IP Settings** dialog will appear. If you want the gateway to use a dynamic IP address, activate the checkbox **DHCP**. This will deactivate the next three lines. Your DHCP server will automatically provide all of the other necessary information. If you do not have a DHCP server, leave the **DHCP** checkbox empty. The default IP address appears in the **IP Address** box. Enter a new IP address. If the address you enter already exists in the network, you will be notified to choose another address at the end of the configuration process. Enter the system's netmask in the **Mask** dialog box. Enter the IP address for the **Default Gateway.** Click **Finish**.



IP Settings

□ DHCP

IP Address: 172 . 20 . 27 . 243

Mask: 255 . 255 . 0 . 0

Default Gateway: 192 . 168 . 1 . 1

Figure 4.6 Quickstart configuration: IP settings

There is no internal time generation for the system when the power is interrupted. That means the default time is used when the system is restarted or rebooted! Therefore it is important to set the system time with an NTP server.

If the system is connected via ISDN, a clock may come from the network connected to the corresponding port. Enter `TIME` in the `pabx.cfg`'s `Subscriber` line for the TE port to retrieve the time from the port.

Now the IP settings are configured; all other processes run automatically. First the system's IP address will be changed and then the system will start with the new IP address.

If you right-click the system's icon in the main window and choose **Temporarily Configure IP Address**, only the IP address for the system's first Ethernet interface and the netmask will be temporary changed. This can be helpful if you want to set up local remote access to the system and use other IP settings on the remote device than the system's IP configuration in the network. Bear in mind that the functions on the system's first Ethernet interface work with the new settings.

Now you can complete the system's configuration using the GUI (please see Chapter 4.8 Startup via GUI on page 28).

4.8 Startup via GUI

System configuration can occur via the GUI.

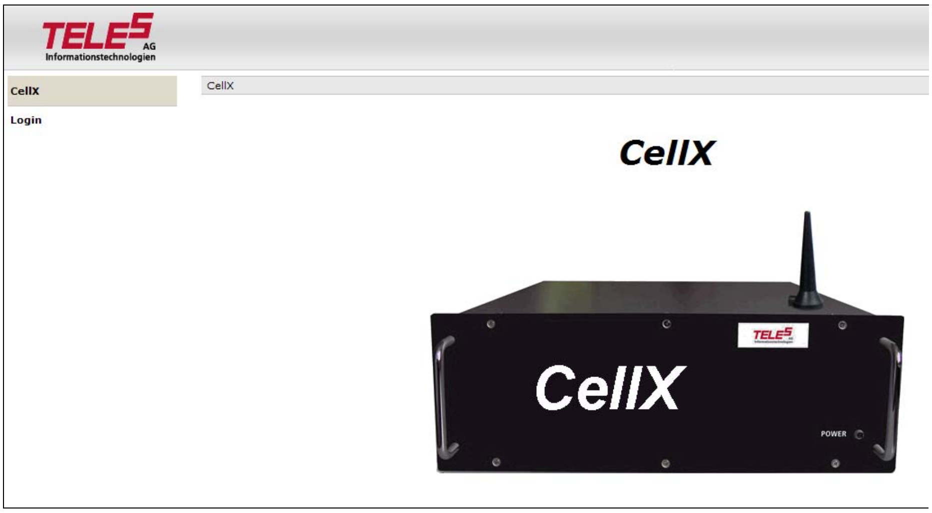
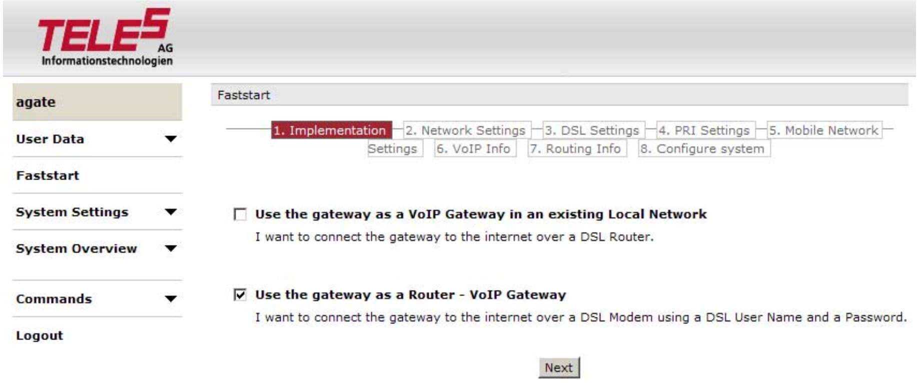


Figure 4.7 GUI

We recommend you use Internet Explorer 6/7/8. Simply open a browser, enter the system's IP address in the address bar, and click **Login** in the navigation menu on the left. Enter the username **teles-carrier** and the password **tcs-carrier** to access the system.



□ **Use the gateway as a VoIP Gateway in an existing Local Network**
I want to connect the gateway to the internet over a DSL Router.

☑ **Use the gateway as a Router - VoIP Gateway**
I want to connect the gateway to the internet over a DSL Modem using a DSL User Name and a Password.

Figure 4.8 GUI faststart

Using the navigation menu on the left, click **Faststart** to configure the system. Follow the steps as they appear.

To edit the default configuration, follow the directions in Chapter 5 Configuration files. Upload the configuration files into the `/boot` directory.

4.9 LED functionality

4.9.1 Ethernet port LEDs

Each ethernet port has 2 LEDs to show its status. The left LED blinks to indicate data traffic. The right LED is currently not used.

4.9.2 Base Board PRI port LEDs

Each PRI port has one red and one green LED to show the port's status.

The red LED displays the status of the bypass relay that connects the ports with each other when the relay between the PRI ports is off. That means when the system is connected between a PBX and the PSTN, it is transparent when the LED is red.

The green LED displays whether or not layer 1 is active on the PRI port's connected cable.

Table 4.4 Base Board PRI Port LEDs

LED	Description
Red ON	The system and bypass relay are inactive (normally during the startup phase).
Red OFF	The system has started and the bypass relay is active.
Green ON	Layer 1 is active.
Green OFF	Layer 1 is inactive.

4.9.3 Mobile Board SIM card LEDs

On the spine of the Mobile Board, to the right of the SIM card module, two columns of green LEDs display the status of each mobile channel.

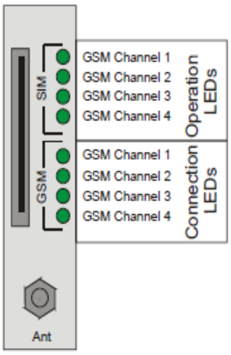


Figure 4.9 Mobile Board SIM card LEDs

The LEDs in the upper column show the general operational status of the SIM cards, while the status of the mobile channels is displayed in the lower column.

Table 4.5 contains a description of the LEDs and what they mean:

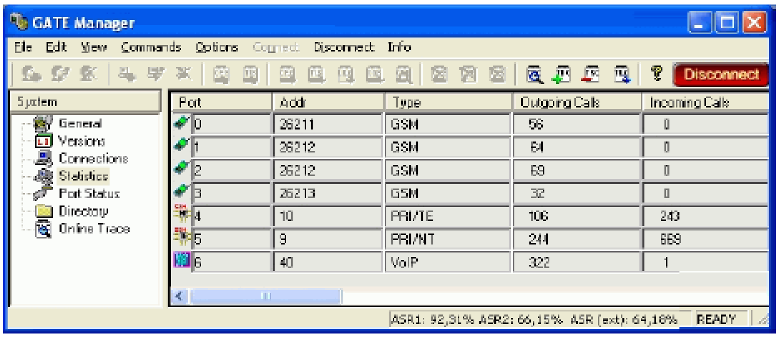
Table 4.5 Mobile Board LEDs

Operational Status	Connection Status	Definition
OFF	OFF	The mobile channel is not operational because: ▪ No external power supply ▪ SIM module slot is empty ▪ No SIM card
OFF	Blinking slowly	Not possible
OFF	Blinking quickly	Not possible
OFF	ON	Not possible
Blinking slowly	OFF	The SIM card is attached, but the mobile channel is not operational because: ▪ Mobile channel is in logon phase ▪ Mobile channel's status is unknown
Blinking slowly	Blinking slowly	Not possible
Blinking slowly	ON	Not possible
Blinking quickly	OFF	The mobile channel is not operational because: ▪ SIM card has been blocked ▪ Reception field strength below limit
Blinking quickly	Blinking slowly	Not possible
Blinking quickly	Blinking quickly	Status during initializing phase (system start up). Display changes when status of mobile changes.
Blinking quickly	ON	Not possible
ON	OFF	The mobile channel is operational, the SIM card has logged on.
ON	Blinking slowly	Not possible
ON	Blinking quickly	The mobile channel is operational, the SIM card has logged on, a connection is being set up on this channel
ON	ON	The mobile channel is operational, the SIM card has logged on, a connection has been set up on this channel

4.10 Remote access and access security

After the system has been configured and all cables are connected, remote administration and maintenance can occur with the GATE Manager (Chapter 4.10.1), the GUI (Chapter 4.10.2)or via FTP (Chapter 4.10.3).

4.10.1 GATE Manager



Port	Addr	Type	Outgoing Calls	Incoming Calls
0	26211	GSM	56	0
1	26212	GSM	64	0
2	26212	GSM	69	0
3	26213	GSM	32	0
4	10	PRI/TE	106	243
5	9	PRI/NT	244	669
6	40	VoIP	322	1

ASR1: 92,31% ASR2: 66,15% ASR (ext): 64,16% READY

Figure 4.10 GATE Manager

The GATE Manager administration and maintenance software offers a broad range of functions. The GATE Manager is user friendly and can be customized to suit your needs.

The following maintenance functions are possible:

- Display system information and network element status.
- Retrieve and display configuration files.
- Restart network elements.
- Use of a trace option for checking functions and fault diagnosis. Option to use an external tool, for example to display and break down trace data.
- Update the system software and configuration tables.
- Retrieve CDRs (Call Detail Records).
- Display the current connections (status).
- Display statistical information for network elements and interfaces.
- Display the status of the interfaces.

Use the CD enclosed in your package contents to install the GATE Manager. For a detailed description of installation and implementation of the GATE Manager, please refer to the GATE Manager and Utilities Programs Manual.

GATE Manager remote access can occur via IP or ISDN. GATE Manager access via IP uses port 4444 as source TCP port and port 4445 as destination port. You can change the port in the `pabx.cfg` file using the following parameter: MoipPort=4567.

Bear in mind that the same port must be configured in the GATE Manager. The TCP port can be specified behind the IP address and a colon: `IP:172.20.25.5:4567.`

In the default configuration, ISDN remote access is disabled. To configure the system so that certain data calls are received as remote administration calls, make the following changes in the `pabx.cfg`:

```
RemoteCode=BBB
```

Add the following mapping to the `route.cfg`:

```
MapAll<direct>=BBB DATA
```

Make the following entries in the `route.cfg` if the system is to handle all ISDN data calls as remote-administration calls:

```
MapAll?=BBB DATA
```

4.10.2 Graphical user interface (GUI)

Remote access can occur via the GUI. Even users with little experience can easily configure standard system settings with this interface. Simply open a browser and enter the system's IP address in the address bar.

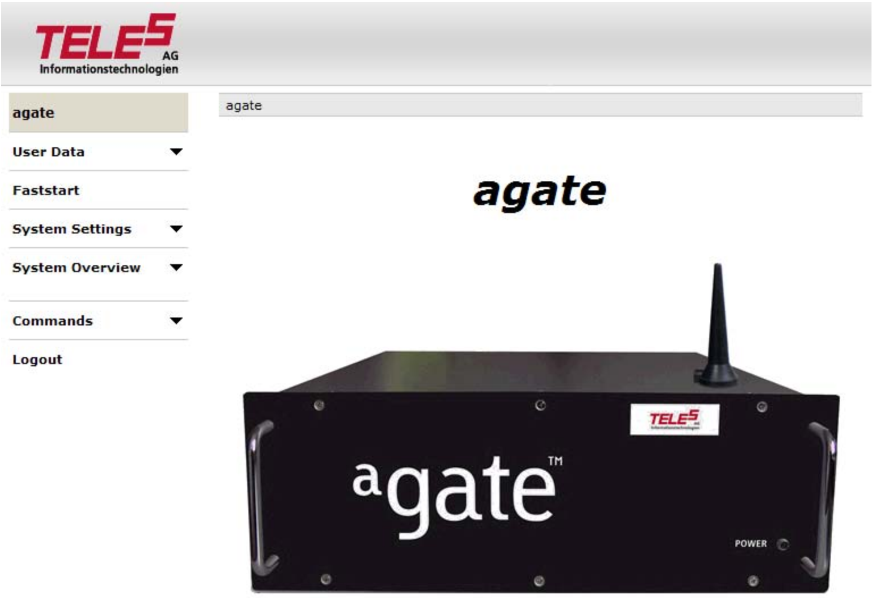


Figure 4.11 GUI

The following administrative levels apply:

Carrier mode (full access)

User: `teles-carrier`

Password: `tcs-carrier`

All configuration pages can be accessed in this mode.

Example 4.4 Carrier mode (full access)

```
[httpd]
PwdUser=k24X0sdc.uMcM
PwdAdmin=k2UMj19qtovzI
PwdCarrier=k2jryo6Xd5vN6
```

Never copy these entries from one system to another, as the encryption is unique for each system.

Administrator Mode

User: `teles-admin`

Password: `tcs-admin`

This access level is for the user network's administrator. All IP and routing entries, with the exception of VoIP carrier entries, can be set here.

Read-Only Mode

User: `teles-user`

Password: `tcs-user`

No configuration changes can be made at this level. Only status and statistics can be retrieved.

Of course, these configuration levels correspond with the most important scenarios. The passwords are saved in the `ip.cfg` in encrypted form:

```
PwdCarrier=<crypt>
PwdAdmin=<crypt>
PwdUser=<crypt>
```

The user interface is divided into the following main sections:

Table 4.6 GUI: sections

Section	Description
User Data	Here you can change the user passwords and the language for the GUI.
Faststart	Faststart helps you to configure the system settings of your quickly and conveniently.
System Settings	IP Settings: Settings for the Ethernet interfaces and related services. Port Settings: Settings for the ECOTEL GSM ports. VoIP Settings: VoIP settings for the SIP or H.323 carrier. Telephony Routing:Routings for telephone numbers.
System Overview	Overview of system information and drivers.
Commands	Here you can activate a configuration or restart the system.

All of the user interface's pages contain **Help** buttons and links to the online help, which provides a detailed description of all of the individual configuration settings.

4.10.3 FTP

Remote access can also occur via FTP. You can use FTP to transfer configuration files. You can also carry out functions and traces with raw commands. Use the username `teles` and the defined password to connect to the system with FTP.

The following entries in the pabx.cfg ensure the security of your FTP access:

Table 4.7 FTP security entries

FTP Security
`FtpdPort=<port>` Defines the FTP access port (default 21).
`RemotePassword=<password>` Defines the password for FTP and GATE Manager access. Please refer to Chapter 4.10.4 for instructions on how to enter an encrypted password in the pabx.cfg. If you do not define a password, access to the system via GATE Manager occurs without a password, and FTP access occurs with the default password tcs-ag.

Once you have access to the system, you will be in the folder /home/teles. To upload or download configuration files change to the directory /boot. To download log files, change to the directory /data if the system contains a flash disk. Otherwise change to the directory /boot.

The following commands can be carried out via FTP access:

Table 4.8 FTP commands

Command	Function
site xgboot	Boots the entire system.
site xgact	Activates the configuration.
site xgact 1-19	Activates the Night section corresponding with the number 1-19.
site xgtrace 0	Deactivates trace.
site xgtrace 1	Activates layer 2 trace.
site xgtrace 2	Activates layer 3 trace.

If your FTP client does not support the site command, try “literal site” instead.

4.10.4 Setting a password for remote access

The following entry ensures the security of your remote access. Use the **mkpwd.exe** tool to generate the password. You will find it on the enclosed CD in the directory pwd.

Start the program in a command window with the entry mkpwd <password>. The output shows the encrypted password. Enter the encrypted password in the configuration file pabx.cfg’s parameter line as follows.

Example 4.5 Password for remote access

```
RemotePassword=<crypt>
```

When the file has been transferred to the system and the configuration has been activated, access to the system can occur only with the password. Don't forget to memorize the password!

If you do not define a password, access to the system via GATE Manager occurs without a password, and FTP access occurs with the default password `tcs-ag`.

5 Configuration files

5 Configuration files

This chapter describes the basic setup and the most commonly used entries for the configuration files. Configuration of CELLXs is managed in the following three files:

Table 5.1 Configuration files

File	Function
`ip.cfg`	This file is for the basic configuration of the Ethernet interfaces.
`pabx.cfg`	This file is for system-specific and port-specific settings.
`route.cfg`	This file is for call-routing entries.

Changing configuration data and/or SIM card positions may lead to malfunctions and/or misrouting, as well as possible consequential damage. All changes are made at own risk. TELES is not liable for any possible damage out of or in relation with such changes. Please do therefore thoroughly check any changes you or a third party have made to your configuration.

The default configuration with the IP address 192.168.1.2 is active when the files are not on the system. You can configure the files using GATE Manager or via FTP (user teles, password tcs-ag). If you use the GUI to make configuration changes, the files will be adjusted automatically.

Make sure you secure the system with new passwords following configuration and remember to memorize the passwords!

These configuration files contain all system-specific settings and are used when the system starts. Comments included in these files must begin with a semicolon. They do not need to be at the beginning of a line. Configuration files must end with an empty line.

Please save a backup of the files `pabx.cfg` and `route.cfg` before starting configuration.

The configuration files follow these conventions: Individual files are divided into sections. These sections always begin with a line entry in square brackets. The basic required sections are in these files:

Table 5.2 Required configuration file sections

Section	File	Function

`[System]`	`pabx.cfg`
	`route.cfg`
	`ip.cfg`
		This section contains the system's basic settings.

`[Night<num>]`
EXAMPLE:
`[Night1]`
`[Night2]`
	`pabx.cfg`
	`route.cfg`
		This section contains time dependent entries that only apply for limited times.

`[emac0]`	`ip.cfg`	This section contains the IP configuration for the first Ethernet interface.

5 Configuration files

5.1 Configuration file ip.cfg

The basic settings for the two Ethernet interfaces are entered here. One interface usually suffices. The second interface can be used for special requirements, for example as a hub port, DSL router or vLAN interface. Generally, these settings are entered once and then left unchanged.

This file contains the following sections, which must appear in the order given:

Table 5.3 Sections in the `ip.cfg` file

Section	Function
`[System]` (required)	This section contains entries that define the default gateway and/or special routing entries.
`[emac0]` (required) `[emac1]` (optional)	The Ethernet Media Access Controller section(s) define the physical Ethernet interface(s).
`[nat]` (optional)	This section includes settings for Network Address Translation.
`[bridge0]` (optional)	These section(s) contain settings for the second Ethernet controller in bridge mode.
`[pppoe<x>]` (optional)	These sections contain settings for direct connection between the system and the DSLAM when the PPPoE protocol is used. `<x>` can be `0` or `1`.
`[firewall]` (optional)	This section contains settings for activating the system's firewall.
`[altqd]` (optional)	This section enables prioritization of VoIP packets in the CELLX through an IP network using bandwidth control.
`[dhcpd]` (optional)	This sections contains a list of parameters and settings for the DHCP server in the system. It is divided into global settings for the server and parameters for the DHCP subnet.
`[xppp<x>]` (optional)	This section contains settings for point-to-point dial-up setup via ISDN.
`[vlan<x>]` (optional)	These section(s) contain settings for the virtual networks. <x> can be anything from 0 to 9.

5.1.1 System section configuration

The [System] section contains entries that define the default gateway and/or special routing entries.

To define the standard gateway, use the following entry to set the IP address:

```
DefaultGw=<ip addr>
```

Example 5.1 System section configuration 1

```
[System]
DefaultGw=192.168.1.254
```

If you must route specific net ranges to gateways other than what is defined in the default route, make the following entries in the [System] section:

`Route=<target range> -netmask <ip mask> <ip gateway>`

Example 5.2 System section configuration 2

```
[System]
DefaultGw=192.168.1.254
Route=10.0.0.0 -netmask 255.0.0.0 192.168.1.1
```

If only certain routes apply, leave the line `DefaultGw` empty.

5.1.2 Ethernet interface configuration

The system includes two Ethernet interfaces (emac0 and emac1). Only the first is active in the default configuration. Therefore, make sure you plug the cable into the right controller. The second Ethernet interface can be configured as needed.

The following settings are possible for the sections [emac0] (matched to the first Ethernet controller) and [emac1] (matched to the second Ethernet controller):

`IpAddress=<ip addr>/<netmask>`

The IP address is entered in decimal notation, followed by a slash (/) and the netmask in bit notation.

Example 5.3 Ethernet interface configuration

```
IpAddress=192.168.1.2/24
```

The following entry is used to allocate an IP address via DHCP:

`IpAddress=dhcp`

The following entry is used in the `[emac1]` section if operation of the system occurs in bridge mode.

`IpAddress=up`

5.1.3 GUI settings

The following parameter is used to change the GUI port in the section [httpd] (default 80):

`GuiPort=<num>`

Bear in mind that the passwords for different access levels are not set here. The encrypted passwords are stored here and can only be changed via GUI (see Chapter 4.10.2 on page 32).

Example 5.4 GUI settings

```
[httpd]
GuiPort=80
PwdUser=k24X0sdc.uMcM
PwdAdmin=k2UMj19qtovzI
PwdCarrier=k2jryo6Xd5vN6
```

5.1.4 Bridge configuration

A bridge can connect two networks with each other. A bridge works like a hub, forwarding traffic from one interface to another. Multicast and broadcast packets are always forwarded to all interfaces that are part of the bridge. This can occur on the Ethernet or VLAN level:

`BrConfig=add <interface-x> add <interface-y> up`

Activating another Ethernet interface in this way is useful, for example, when the Ethernet switch does not have any more ports available for connection of the system. You can simply unplug a cable and plug it into the system's second Ethernet interface.

Example 5.5 Bridge configuration

```
[bridge0]
BrConfig=add emac0 add emac1 up
```

5.1.5 NAT configuration

The NAT (Network Address Translation) module translates IP addresses from the local network to an IP address or range on a public interface. All rules are defined in the `[nat]` section:

Table 5.4 NAT configuration

map=<interface> <local network address/mask> -> <public network address/mask> <optional entries>	
This parameter maps the IP address in the local network to the IP address in the public network.	
`<interface>`	Defines the translated interface or protocol: `emac1` The system's second Ethernet interface `pppoe0` Protocol used for DSL connections `xppp<0>` Protocol used for ISDN and CDMA dial-up connections
`<local network address/mask>`	The IP address is entered in decimal notation, followed by a slash (/) and the netmask in bit notation. The entire local network range is configured.
`<public network address/mask>`	Defines the public network range, with network address and mask (usually exactly one address), into which the local IP addresses are to be translated. The IP address is entered in decimal notation, followed by a slash (/) and the netmask in bit notation.
`<optional entries>`	Special rules can be defined for some services or protocols. The system can serve as a proxy for FTP: proxy port ftp ftp/tcp Special ports for the public address(es) can be assigned for the protocols TCP and UDP. The range is defined by the start and end ports: portmap tcp/udp <start port>:<end port> If no optional entry is defined, all other addresses will be translated without special rules.
rdr=<interface> <public network address/mask> port <port> -> <local network address/mask> port <port_number> <protocol>	
This parameter sends packets from one port and IP address to another.	

Table 5.4 NAT configuration *(continued)*

`<interface>`	Defines the translated interface or protocol: emac1 The system's second Ethernet interface pppoe0 Protocol used for DSL connections Protocol used for ISDN and CDMA dial-up connections
`<public network address/mask>`	Defines the public network range, with network address and mask (usually exactly one address), into which the local IP addresses are to be translated. The IP address is entered in decimal notation, followed by a slash (/) and the netmask in bit notation.
`<port>`	Defines the port number.
`<local network address/mask>`	The IP address is entered in decimal notation, followed by a slash (/) and the netmask in bit notation. The entire local network range is configured.
`<protocol>`	Defines the protocol. `tcp` and `udp` are possible.
watch=<interface 1> <interface 2> ... <interface n>	
Enter all interfaces that you have configured. If an interface is activated, the NAT table is resetted to ensure correct IP address translation.	

The following NAT settings are for a system in which PPPoE (DSL) is used toward the Internet. The local network range 192.168.1.0 Class C is translated with the following rules:

- The proxy mode is used for FTP.
- All other TCP and UDP packets are mapped to the external ports 40000 to 60000.
- There are no special rules for any other services.
- Incoming requests to port 80 and 443 in the public IP address 192.168.1.100 are redirected to ports 80 and 443 in the local IP address 192.168.1.100.

Example 5.6 NAT configuration

```
[nat]
map=emac1 192.168.1.0/24 -> 0/32 proxy port ftp ftp/tcp
map=emac1 192.168.1.0/24 -> 0/32 portmap tcp/udp 40000:60000
map=emac1 192.168.1.0/24 -> 0/32
rdr=emac1 0/0 port 80 -> 192.168.1.100 port 80 tcp
rdr=emac1 0/0 port 443 -> 192.168.1.100 port 443 tcp
```

5.1.6 PPPoE configuration

The protocol Point-to-Point over Ethernet is used for DSL communication with the DSLAM. That means the system can connect directly with a DSL modem.

All necessary information for setup of the PPPoE connection is defined in the `[pppoe<x>]` section. That means username, password and authentication protocol are set here. The Ethernet interface is emac1 and the gateway can also be defined. The parameter `PppoeIf` defines the physical Ethernet interface used (always emac1). The settings are entered as follows: Bear in mind that configuration of the firewall, the NAT module and prioritization of the VoIP packets must be considered when routing voice and data through the DSL line.

The following entry will create the interface `pppoe0`, with the username `user` and the password `pwd`. The PAP authentication protocol is used. The default route occurs via DSL.

Example 5.7 PPPoE configuration

```
[pppoe0]
PppoeIf=emac1
User=user
Pwd=pwd
AuthProto=pap
Route=0.0.0.0
```

5.1.7 Firewall settings

The firewall settings provide options for limiting or denying access to and from the system. If you do not configure this section, the firewall is inactive and access is unlimited. Make sure you configure the firewall rules carefully. The rules are processed from top to bottom. If you use the option `quick`, you will break the sequence. We recomend that you put the most restrictive rule at the end of the configuration.

In the following example, only port 4445 allows incoming connections from the IP address 192.168.1.10. All others will be blocked.

Example 5.8 Firewall settings 1

```
[firewall]
fw=pass in quick on emac0 proto tcp from 192.168.1.10/32 to any port
eq 4445 flags S keepstate keep frags
fw=block in log quick on emac0 all
```

Table 5.5 Settings in the `[firewall]` section of the `ip.cfg`

[firewall] **fw=<mode> <direction> <list>**	
`<mode>`	Two modes are possible for permitting or denying access: `pass` permit access `block` deny access
`<direction>`	Possible directions are in and out: `in` external to internal `out` internal to external
`<list>`	All other entries specify the other settings for the corresponding firewall rules and are optional. The order in the line is as listed below:
log Records non-matching packets.	

Table 5.5 Settings in the `[firewall]` section of the `ip.cfg` *(continued)*

[firewall] **fw=<mode> <direction> <list>**
`quick` Allows short-cut rules in order to speed up the filter or override later rules. If a packet matches a filter rule that is marked as quick, this rule will be the last rule checked, allowing a short-circuit path to avoid processing later rules for this packet. If this option is missing, the rule is taken to be a "fall-through rule, meaning that the result of the match (block/pass) is saved and that processing will continue to see if there are any more matches.
`on <interface>` The firewall rule is used only for the defined interface (for example emac0, pppoe0).

`from <networkaddress/mask>`
`to <networkaddress/mask>`
from defines the source IP-address range for incoming packets. to defines the target IP-address range for outgoing packets. The IP address appears in decimal notation, followed by a slash (/) and the netmask in bit notation. any stands for all IP addresses (example: to any).
NOTE: If you use the rule pass in/out in combination with the option from <ip> to <ip>, you must specify a protocol number with proto and a port number. If you not specify the port, the system may not be reachable.
EXAMPLE:
fw=pass in quick on pppoe0 proto tcp from any to any port eq 4445

`proto <protocol>` defines the protocol, for which the rule is valid (example: proto tcp, proto udp, proto icmp).
`port eq <num>` <num> defines the port as number (example: port eq 4445).
`keep state` Ensures that the firewall checks packets from the beginning to the end of a session. This is necessary, as the firewall cannot process when a session begins or ends.
`flags S` Only syn. packets are accepted and recorded in the state table. In conjunction with keep state, packets from sessions that have been inactive will also be routed. The advantage of this entry is that random packets will not be accepted.
`keep frags` Fragmented packets are also routed.

Example 5.9 Firewall settings 2

```
[firewall]
; loopback
fw=pass in quick on emac0 all
fw=pass out quick on emac0 all

; traffic to outgoing
fw=pass out quick on pppoe0 proto tcp all flags S keep state keep frags
fw=pass out quick on pppoe0 proto udp all keep state keep frags
fw=pass out quick on pppoe0 proto icmp all keep state keep frags

; incoming traffic
fw=pass in quick on pppoe0 proto tcp from 10.4.0.0/16 to any port eq 21 flags S
keep state keep frags
fw=pass in quick on pppoe0 proto tcp from 10.4.0.0/16 to any port eq 23 flags S
keep state keep frags
fw=pass in quick on pppoe0 proto tcp from 10.4.0.0/16 to any port eq 4445 keep
state

; icmp traffic
fw=pass in quick on pppoe0 proto icmp all keep state

; other will be blocked
fw=block in log quick on pppoe0 all
fw=block out log quick on pppoe0 all
```

5.1.8 Bandwidth control

In many implementation scenarios, the CELLX in router mode (for example as DSL router) sends voice and data traffic through a connection with limited bandwidth. This can lead to lost voice packets that arrive too late to be used in the voice stream. To avoid lost packets, this QOS setting prioritizes packet transmission. You must set the priority for voice signaling and for the voice packets. That means you must prioritize SIP/H.323, RTP and RTCP. You will find the ports used in Table 5.14, in the following entries:

`H225Port`

`SipPort`

`VoipRtp Port`

`VoipRtpPortSpacing`

Different ports are used for RTP and RTCP, depending on the configuration.

The parameter VoipRtpPort shows the first RTP port used. The corresponding RTCP port is the next one up. The parameter VoipRtpPortSpacing shows the next RTP port (RTP port + port spacing).

Table 5.6 Settings in the [altqd] section of the `ip.cfg`

Interface=<interface> bandwidth <bw> priq	
Defines the interface for which the rule applies.	
`<interface>`	Sets the interface for which prioritization applies (e.e. pppoe0).
`<bw>`	Sets the bandwidth that is available on the interface in Kbit/s (for example 256K).
`priq`	Priority qeueing. A higher priority class is always served first.
classPrio=<interface> <class> root priority <prio>	

Table 5.6 Settings in the [altqd] section of the `ip.cfg` *(continued)*

Interface=<interface> bandwidth <bw> priq	
Defines the priority of the filter entries.	
<class>	Two types can be set: ▪ realtime_class (VoIP packets) ▪ regular_class (data packets)
<prio>	Enter a value between 0 and 15. The higher the value (for example 15), the higher the priority.
Filter=<interface> <class> <values>	
Defines the individual rules for the class.	
<values>	The individual values are divided into the following entries. A 0 can be entered as a wildcard, in which case all values are possible: ▪ <dest_addr> (can be followed by netmask <mask>) ▪ <dest_port> ▪ <src_addr> (can be followed by netmask <mask>) ▪ <src_port> ▪ <protocol tos value>: 6 for TCP 17 for UDP

In the following example, prioritization is set for a thirty-channel VoIP connection. The SIP signaling port 5060 and the RTP/RTCP ports 29000 to 29059 are prioritized at level 7. All other services are set at level 0.

Example 5.10 Bandwidth control

```
[altqd]
interface pppoe0 bandwidth 512K priq
class priq pppoe0 realtime_class root priority 7
  filter pppoe0 realtime_class 0 5060 0 0 0
  filter pppoe0 realtime_class 0 0 0 5060 0
  filter pppoe0 realtime_class 0 29000 0 0 17
  filter pppoe0 realtime_class 0 0 0 29000 17
  filter pppoe0 realtime_class 0 29001 0 0 17
  filter pppoe0 realtime_class 0 0 0 29001 17
  ....
  filter pppoe0 realtime_class 0 29058 0 0 17
  filter pppoe0 realtime_class 0 0 0 29058 17
  filter pppoe0 realtime_class 0 29059 0 0 17
  filter pppoe0 realtime_class 0 0 0 29059 17
class priq pppoe0 regular_class root priority 0 default
```

5.1.9 DHCP server settings

The DHCP (Dynamic Host Configuration Protocol) server provides a mechanism for allocation of IP addresses to client hosts. The `[dhcpd]` section contains a list of parameters and settings for the DHCP server in the system. It is divided into global settings for the server and parameters for the DHCP subnet.

Table 5.7 Settings in the [dhcpd] section of the `ip.cfg`

[dhcpd]
; Global dhcpd parameters
`allow unknown-clients;` All DHCP queries are accepted and the configured settings are transmitted to the clients.
`ddns-update-style none;` Deactivates dynamic update of the domain name system as per RFC 2136.
; Parameters for the Subnet
subnet <network address> netmask <mask for network range> {<list>}
In `<list>` you can enter any of the following specific network settings activated by the DHCP server. Each oprion must begin in a new line and end with a semicolon (`;`).
`range <start IP address> <end IP address>`; The DHCP network range is defined by the first and last address in the range. Client assignment begins with the last address.
`option broadcast-address <IP address>;` Defines the broadcast address for the clients in the subnet.
`option domain-name "<string>";` Defines the domain name used in the network.
`option domain-name-servers <IP address>`; Defines the DNS-server address to be assigned (as per RFC 1035) All of the following optional entries defining server addresses are also transmitted as per RFC 1035. Separate multiple addresses per server with a comma: `… <IP address>, <IP address>;` (this also applies for all other optional entries with IP addresses).
`option netbios-name-servers <IP address>` Defines the WINS-server address to be assigned.
`option ntp-servers <ip address>;` Defines the NTP-server address to be assigned.
`option time-servers <ip address>;` Defines the time-server address to be assigned (RFC 868).
`option routers <IP address>;` Defines the router address to be assigned.
`option subnet-mask <net mask>;` Defines the netmask to be assigned (as per RFC 950).

`option tftp-server-name "<link>";`
Defines the TFTP server name (option 66), as per RFC 2132.
EXAMPLE: option tftp-server-name "http://192.168.0.9";

Example 5.11 DHCP server settings

```
[dhcpd]
; Global dhcp parameters
allow unknown-clients;
ddns-update-style none;

; Parameter for the Subnet
subnet 192.168.1.0 netmask 255.255.255.0 {
 range 192.168.1.3 192.168.1.20;
 option broadcast-address 192.168.1.255;
 option domain-name "company.de";
 option domain-name-servers 192.168.1.100;
 option routers 192.168.1.2;
 option subnet-mask 255.255.255.0;
}
```

5.1.10 DNSmasq settings

Dnsmasq is an easy to configure DNS forwarder. It is designed to provide DNS to a small network.

Table 5.8 Settings in the `[dnsmasq]` section of the `ip.cfg`

[dnsmasq]
bogus-priv Bogus private reverse lookups. All reverse lookups for private IP ranges (ie 192.168.x.x, etc) which are not found in /etc/hosts or the DHCP leases file are answered with "no such domain" rather than being forwarded upstream.
filterwin2k Later versions of windows make periodic DNS requests which don't get sensible answers from the public DNS and can cause problems by triggering dial-on-demand links. This flag turns on an option to filter such requests. The requests blocked are for records of types SOA and SRV, and type ANY where the requested name has underscores, to catch LDAP requests.
user=<username> Specify the userid to which dnsmasq will change after startup. Dnsmasq must normally be started as root, but it will drop root privileges after startup by changing id to another user. Normally this user is "nobody" but that can be over-ridden with this switch.
cache-size=<cachesize> Set the size of dnsmasq's cache. The default is 150 names. Setting the cache size to zero disables caching.
clear-on-reload Whenever /etc/resolv.conf is re-read, clear the DNS cache. This is useful when new nameservers may have different data than that held in cache.

Example 5.12 DNSmasq settings

```
bogus-priv
filterwin2k
user=teles
cache-size=150
cler-on-reload
```

5.1.11 PPP configuration for mobile and ISDN dial-up

The point-to-point protocol is used for dial-up connections via GPRS/3G or CMDA, or via ISDN lines. The system can set up an mobile Internet connection for the companies' local users or an ISDN data link between subsidiaries of the company.

The mobile internet access can be used as regular internet access for small companies or as an internet back up solution.

The ISDN dial-up can be used to transmit VoIP calls.

The advantages of VoIP over ISDN can be seen especially in corporate implementation. For example, it is useful when a very high number of connections occurs between subsidiaries and one subsidiary does not have a broadband Internet connection. An ISDN B-channel can be connected to the Internet and up to six voice calls can occur simultaniously over one ISDN line.

All necessary information for setup of the PPP connection is defined in the section [xppp<num>].

Table 5.9 Settings in the `[xppp]` section of the `ip.cfg`

[xppp<num>]
`Dad=<num>` Enter the dial-up number. Only digits can be defined here. Any required special characters (* or #) can be set in the mapping entry.
`User=<username>` Enter a username.
`Pwd=<password>` Enter a password.
`Route=<ip-addr>` Enter the target IP address range, for example 0.0.0.0 (default route).
`AuthProto=<protocol>` Enter `chap` or `pap` for the protocol used for authentication.
`AutoUp=<int>` Defines if the PPP interface is activated automatically after system start. The following values are possible: 0 = No automatic PPP activation (default) 1 = Automatic PPP activation
`IdleTO=<sec>` Enter the number of seconds without traffic before the interface tears down the connection.
`MTU=<int>` Maximum Transfer Unit. We recommend the following default values: 1500 for ISDN dial-up and 120 for CDMA dial-up.
`Rfc1662=<val>` Framing to be used: 0 for ISDN or 1 for CDMA
`LcpTO=<msec>` Allows you to change the value of the LCP timeout. The timeout-value must be specified in milliseconds (default 1000).
`StartDelay=<sec>` Time in seconds the system will wait to start the ppp process.

Table 5.9 Settings in the `[xppp]` section of the `ip.cfg` *(continued)*

[xppp<num>]
`DNS=<bitmask>` Enter here to which of the carrier's DNS server the gateway shall send the DNS request. The following values are possible: 1 = primary DNS server 2 = secondary DNS server 3 = both servers
`OwnIP=<IP address>` A temporay IP address assigned to the interface (such as 0.0.0.0). This address is valid until an IP address has been assigned to the interface by the carrier. Not needed for the xppp0 interface.
`PeerIP=<IP address>` The IP address that is configured for the peer (such as 0.0.0.1). Not needed for the xxxp0 interface, each other interface has to have a different peer IP address.

Example 5.13 PPP configuration for ISDN and CDMA dial-up

```
[xppp0]
Dad=12345
User=user
Pwd=pwd
Route=0.0.0.0
AuthProto=chap
IdleTO=60
MTU=1500
Rfc1662=0
LcpTO=500
StartDelay=10
AutoUp=1
```

Make sure you configure the firewall and NAT options accordingly.

5.1.12 VLAN configuration

A VLAN (Virtual Local Area Network) is a virtual LAN within a physical network. Each VLAN is assigned a unique number (VLAN ID) and defined in the `[vlan<x>]` section with

Tag: value between 1 and 4095

Priority: value between 0 and 7 (0 is lowest and 7 is the highest priority)

```
[vlan0]
IfConfig=vlan <tag>,<priority> vlanif <interface>
```

The following entry will create the interface vlan1, with VLAN tag 10 and priority 7, on the Ethernet interface emac0. Following this configuration, IP addresses (and/or other protocols) can be assigned to the vlan1 interface.

Example 5.14 VLAN configuration

```
[vlan1]
IfConfig=vlan 10,7 vlanif emac0
IpAddress=192.168.199.1
```

5.1.13 Examples

5.1.13.1 Default configuration

In the following example, the system’s IP address is 192.168.1.1, the netmask is 255.255.255.0, and the standard gateway is 192.168.1.254.

Example 5.15 Default configuration

```
[System]
DefaultGw=192.168.1.254

[emac0]
IpAddress=192.168.1.1/24
```

5.1.13.2 Active ethernet bridge

In the following example a two-port Ethernet bridge is configured. The system’s IP address is 192.168.1.1, the netmask is 255.255.255.0, and the standard gateway is 192.168.1.254,

The emac1 interface is active and both Ethernet interfaces are set to bridge mode in the `[bridge0]` section.

Example 5.16 Active ethernet bridge

```
[System]
DefaultGw=192.168.1.254

[emac0]
IpAddress=192.168.1.1/24

[emac1]
IpAddress=up

[bridge0]
BrConfig=add emac0 add emac1 up
```

5.1.13.3 Integrated DSL-router scenario for VoIP

In the following example, the system is connected to the local IP network through emac0. The DSL modem is connected to the emac1 interface, which enables the system to connect directly to the Internet without an additional router when the connection is used only for VoIP data. A DHCP server is used for dynamic IP-address allocation.

Example 5.17 Integrated DSL-router scenario for VoIP traffic with an active DHCP server and firewall

```
[System]

[emac0]
IpAddress=192.168.0.2/24

[emac1]
IpAddress=up

[pppoe0]
PppoeIf=emac1
User=usertelekom
Pwd=pwd
AuthProto=chap
Route=default

[nat]
map=pppoe0 192.168.0.0/24 -> 0/32 proxy port ftp ftp/tcp
map=pppoe0 192.168.0.0/24 -> 0/32 portmap tcp/udp 40000:60000
map=pppoe0 192.168.0.0/24 -> 0/32

[firewall]
; loopback
fw=pass in quick on emac0 all
fw=pass out quick on emac0 all

; traffic to outgoing
fw=pass out quick on pppoe0 proto tcp all flags S keep state keep frags
fw=pass out quick on pppoe0 proto udp all keep state keep frags
fw=pass out quick on pppoe0 proto icmp all keep state keep frags

; incoming traffic
fw=pass in quick on pppoe0 proto tcp from 10.4.0.0/16 to any port eq 21 flags S
keep state keep frags
fw=pass in quick on pppoe0 proto tcp from 10.4.0.0/16 to any port eq 23 flags S
keep state keep frags
fw=pass in quick on pppoe0 proto tcp from 10.4.0.0/16 to any port eq 4445 keep
state

; icmp traffic
fw=pass in quick on pppoe0 proto icmp all keep state

; other will be blocked
fw=block in log quick on pppoe0 all
fw=block out log quick on pppoe0 all

[dhcpd]
; Global dhcp parameters
allow unknown-clients;
ddns-update-style none;
; Parameter for the Subnet
subnet 192.168.1.0 netmask 255.255.255.0 {
 range 192.168.1.3 192.168.1.20;
 option broadcast-address 192.168.1.255;
 option domain-name "company.de";
 option domain-name-servers 192.168.1.100;
 option routers 192.168.1.2;
 option subnet-mask 255.255.255.0;
```

5.1.13.4 VLAN scenario

In the following example, the system is connected to the IP backbone through emac0. One Computer is connected to the emac1 interface. You can separate voice and data traffic with two different VLANs (vlan0 with tag 10 for voice, vlan1 with tag 11 for data). All traffic coming from emac1 will be sent to vlan1. Voice and data will not be mixed.

Example 5.18 VLAN scenario

```
[System]
[emac0]
IpAddress=192.168.1.12/16

[emac1]
IpAddress=up

[vlan0]
IfConfig=vlan 10,7 vlanif emac0
IpAddress=10.0.1.2/24

[vlan1]
IfConfig=vlan 11,1 vlanif emac0
IpAddress=172.16.4.5/16

[bridge0]
BrConfig=add vlan1 add emac1 up
```

5.2 Configuration file pabx.cfg

The `pabx.cfg` file contains system-specific settings and the port configuration. It is divided into the [System] and [Night<num>] sections.

5.2.1 System settings

The [System] section is divided into several categories to ensure clarity.

- Global settings
- Log files
- Controllers
- Subscribers
- IP configuration

The following subchapters contain a detailed description of these categories.

5.2.1.1 Global Settings

The entry in this category is responsible for the life-line (bypass) functionality of the PRI port’s relay when the system is on. When the system is off, both PRI ports are connected to each other, which means that it provides a transparent connection between the PBX and the PSTN if the system is installed between the PBX and the PSTN. When the system is on, all routing algorithms are active.

`Bypass=ON/OFF`

`ON`: PRI relay is on (system controls both PRI ports).

`OFF`: PRI relay is off (both PRI ports are connected to each other, regardless of whether or not the system is running).

To ensure bypass functionality, make sure this parameter is always set to `ON`.

Use the following parameter to configure the system for µ-law coded voice data. Make sure this parameter is set to Yes if you connect your gateway to a T1 line in the U.S.A: `Mulaw=yes.`

5.2.1.2 Log files

CDRs, unconnected calls, system events, trace output and statistics can be saved into files. The following entries are necessary to generate log files:

Table 5.10 `pabx.cfg`: log file entries

Entry	Description
ActionLog=/data/protocol.log	System events
Log=/data/cdr.log	CDR entries
failedlog=/data/failed.log	Unconnected calls
TraceLog=/data/trace.log	System trace
MsgLog=/data/msg.log	Incoming SMS and USSD messages

You can define how the log files are to be divided. There are two possibilities for saving entries into a new file:

The available internal memory is approximately 8 MB if the does not contain optional memory expansion. Make sure you monitor the available memory.

- In increments of time (twice-daily, daily, weekly, monthly)
- Depending on the size of the file

You can also define a maximum number of up to 35 of the most recent files.

A dash (-) appears in place of information that is to be ignored.

Table 5.11 `pabx.cfg`: log parameters

Log=/data/<file.log> <saved> <size> <number>	
`<file>`	The name of the log file is generated as follows: [file]yymmdd[0-9|A-Z].log.
`<saved>`	Refers to the frequency with which the file is saved. The following options are possible: `halfdaily` Every day at 11:59 and 23:59 `daily` Every day at 23:59 `weekly` Sunday at 23:59 `monthly` The last day of the month at 23:59
`<size>`	Regardless of the value entered in <saved>, the file will be saved when the <file size> has been reached (in kB). NOTE: We recommend a file size of a multiple of 60kB.
`<number>`	Refers to the number of files that will be saved in the system (between 5 and 35) before the first file is overwritten. This setting is useful not only for limited file size, but also for files that store events. Normally size can be limited for these files, for example 5 files of 1MB each. If the fifth file is full, the first one will automatically be overwritten.

In the following entry, the files `cdr.log` and `failed.log` are renamed every day or when the file reaches 180kB, whichever comes first. Up to 7 CDR files will be saved on the system. If the file size reaches 180kB on one day, the second file will have the same date. Only the running number will be increased.

Example 5.19 Log files renamed 1

```
Log=/data/cdr.log daily 180 7
failedlog=/data/failed.log daily 180 7
```

In the following entry, the file protocol.log is renamed every day or when the file reaches 60 kB. Up to 21 failed files will be saved on the system.

Example 5.20 Log files renamed 2

```
ActionLog=/data/protocol.log daily 60 21
```

In the following entry, the file `trace.log` is renamed every day when the file has reached 600kB. Up to seven log files will be saved on the system.

Example 5.21 Log files renamed 3

```
TraceLog=/data/trace.log daily 600 7
```

In the following entry, the statistic values are reset daily at 12:00 midnight and saved in the asr.log.

Example 5.22 Log files statistic values reset

```
StatisticTime=/data/asr.log 00:00 11111111
```

Please remember to keep track of how much memory is available on the system.

5.2.1.3 Night configuration

The sections for the time-dependent configuration changes and time-controlled routings are defined here.

A maximum of 19 additional daily configuration zones are possible (Night1 to Night19). The entry NightResetTime reactivates the original configuration contained in the System section.

The entry will have the following syntax:

Table 5.12 pabx.cfg: night parameters

Night<num>=<time> <day>	
<num>	Enter a value between 1 and 19 to define which configuration is to be loaded.
<time>	If there is a time set with the format hh:mm after this entry, this configuration is loaded at that time on the defined day.
<day>	Use a bitmask to set the weekdays on which the configuration applies here. The daymap appears in the following order: HoSaFrThWeTuMoSu.

In the following example, the configuration section is activated Fridays, Wednesdays and Mondays at noon unless the day in question is a holiday.

Example 5.23 Night configuration 1

```
Night2=12:00 00101010
```

In the following example, the configuration section switches back to the default configuration (System section) every day at 8:00 p.m.

Example 5.24 Night configuration 2

```
NightResetTime=20:00 11111111
```

Holidays

Up to 50 different dates can be set for night sections used by holiday. The variable dd.mm sets the day and month in which the night section is activated when the 8th bit is set in the bitmask (see Table 5.12).

Any defined Night sections must be set in the files pabx.cfg and route.cfg. If there are no changes in these sections, you must copy them from the System section. The complete Subscriber section must appear in the Night section of the pabx.cfg (see Chapter 5.2.5 on page 68). The active route(s) (MapAll, Restrict and Redirect entries) must appear in the Night section of the route.cfg (see Chapter 5.3 on page 70).

5.2.1.4 Controllers

This category defines the parameters that apply to the ports. The order of the ports is defined as follows: The CELLX contains integrated Mobile Boards, each of which contain four mobile modules. Each Mobile Board's mobile channels are configured as additional controllers. That means four controllers are configured for each board. Beginning with 0, these controllers are defined as the first controllers in the section. Next the PRI controllers are defined, followed by the VoIP controllers. All controllers are defined in ascending order.

Table 5.13 describes the order for additional boards.

Table 5.13 Configuration order: controller parameters

Function	Number of Controllers
Mobile BoardS	Up to 32 (optional)
Base Board (PRI)	2
Base Board (VoIP)	Up to 4 (optional)
DTMF (virtual)	Up to 1 (optional)

Table 5.14 shows only the maximum number of controllers for each individual interface. Any possible combinations will depend on the system's specifications.

The individual ports are defined with the following parameters.

Table 5.14 `pabx.cfg`: controller parameters

Controller<port>=<address> <type> <mode> <line_type> ADR:<hardware address> IRQ:<interrupt> UNIT:<unit> VALUE:<value>	
`<port>`	Defines the running (physical) port number.
`<address>`	Defines the configured (virtual) port address. In the default configuration, PRI TE ports are 9 and PRI NT ports are 10. VoIP ports are 40.
`<type>`	Defines the connection type: TES2M PRI external (terminal endpoint) NTS2M PRI internal (network termination) VOIP VoIP module GSM GSM port CDMA CDMA port UMTS UMTS port TE BRI external (if you change from NT to TE or vice versa, you must change the DIP switches for the respective port on the 4BRI Board) NT BRI internal DTMF virtual controller for activating DTMF tone detection
`<mode>`	Defines the protocol variation for PRI and BRI lines: DSS1 CASR2 (only for PRI lines)

`<line_type>`	Switches CRC4 mode for PRI lines on or off:
	CRC4 CRC4 on
	DF double frame: CRC4 off
	Additional entry for T1 only:
	T1 US Defines this controller as T1. Bear in mind that if one controller is defined as T1, all controllers must be thus defined. If you configure T1, you must also enter CHMAX[23] in the corresponding Subscriber lines.
	T1 EXAMPLE:
	`MULAW=Yes`
	`Controller00=20 TES2M DSS1 T1 US`
	`Controller01=21 NTS2M DSS1 T1 US`
	`...`
	`Subscriber00 = TRANSPARENT ROUTER CHMAX[23]`
	`Subscriber01 = TRANSPARENT ROUTER CHMAX[23]`

`ADR:<hardware address>`	(Optional) Defines the hardware address used for the first controller on an additional Mobile Board. These entries are preconfigured and cannot be changed.
`IRQ:<interrupt>`	(Optional) Defines the interrupt used for the first controller on an additional Mobile Board. These entries are preconfigured and cannot be changed.

Table 5.14 `pabx.cfg`: controller parameters *(continued)*

Controller<port>=<address> <type> <mode> <line_type> ADR:<hardware address> IRQ:<interrupt> UNIT:<unit> VALUE:<value>	

`UNIT:<unit>`	(Optional) Defines the currency for the charges (default EUR). Special charge generation is possible. Special charge generation is possible for:
	France UNIT:&F
	Spain UNIT:&SP
	Portugal UNIT:&P
	Greece UNIT:&G
	Switzerland
	UNIT:&CH
	Netherlands
	UNIT:&NL
	Italy UNIT:&I
	NOTE: The <line_type> must be configured for these entries to work.
	EXAMPLE:
	`Controller02=10 NT DSS1 PMP UNIT:€ VALUE:0.010`
	`Controller03=10 NT DSS1 PMP UNIT:€ VALUE:0.010`

`VALUE:<value>`	(Optional) Defines the charges that accumulate by unit (default 12).

Ports set to the same type can have the same address. In this case they will form a trunk group. If you change this parameter in the configuration, you must restart the system.

Each Mobile Board contains 4 controllers. The hardware address and the interrupt are defined behind the first controllers, which are defined in the configuration before the Base Board.

In the following example, the system contains four Mobile Boards. One PRI controller is configured for TE and one for NT. The protocol used is DSS1, and CRC4 is active. One VoIP Module is attached.

Example 5.25 Controller settings 1

```
Controller00=20 GSM ADR:D800 IRQ:5
Controller01=20 GSM
Controller02=20 GSM
Controller03=20 GSM
Controller04=20 GSM ADR:D900 IRQ:7
Controller05=20 GSM
Controller06=20 GSM
Controller07=20 GSM
Controller08=20 GSM ADR:DA00 IRQ:5
Controller09=20 GSM
Controller10=20 GSM
Controller11=20 GSM
Controller12=20 GSM ADR:DB00 IRQ:7
Controller13=20 GSM
Controller14=20 GSM
Controller15=20 GSM
Controller16=9 TES2M DSS1 US DF
Controller17=10 NTS2M DSS1 US DF
Controller18=40 VoIP
```

5.2.1.5 Subscribers

Various functions for individual interfaces (ISDN or VOIP) are defined in each controller's `Subscriber` line. The order of the subscriber lines is the same as the order of the controller lines (see Chapter 5.2.1.4 on page 56). Most changes become active following a restart. If it suffices to activate the configuration, this is noted in the parameter description.

Additional parameters for mobile controllers are described in Table 5.16 and Table 5.17. The parameters listed in Table 5.16 are required for mobile controllers and those listed in Table 5.17 are optional, depending on the implementation scenario.

Table 5.15 `pabx.cfg`: subscriber parameters

Subscriber<port>=<list>	
`<port>`	Refers to the running (physical) port number.
The <list> variable may contain one or more of the following keywords:	
`DEFAULT`	The standard configuration will be used. No other parameters in this table are set.
`TRANSPARENT`	Only the number is sent as caller ID (without the virtual port address). Activate configuration suffices to activate changes. If TRANSPARENT is not set, the address of the incoming port is added to the A party number as a prefix.
`CASR2[<name>]`	Activates the profile defined in the corresponding [CASR2] section.
`ALARM`	Activates the monitoring mode for the respective port. If a relevant error occurs at the port, the error is written in the protocol.log file. Depending on the configuration, a remote connection to the number defined for AlarmCallback is established and/or an SNMP trap is generated. Activate configuration suffices to activate changes.
`SWITCH`	Changes internal port handling. In the default configuration, the VoIP controller is set to NT. You can use this parameter to change it from NT to TE. Restart the system to activate the changes.
`CHMAX[xx]`	Defines the number of channels per controller, for example 5 for the virtual DTMF controller. A maximum of five concurrent channels are possible for DTMF detection.

`CHSTART[xx]`
`CHSTART[xx,<mode>]`
	Defines the channel where the search for availabe B-channels for outgoing calls starts. Three optional modes are possible for the mode of search: CYC, LIN, INV:
	CYC A round-robin search for the next free B channel occurs. The first call receives the first B channel, the second call the second B channel and so on. When the respective B channel is occupied, then the next one is selected.
	LIN The search for the next free B channel occurs linearly. The search always begins at the specified B channel. When that B channel is busy, the next one is selected until a free B channel is found.
	INV The search for the next free B channel occurs inversely and always begins at the specified B channel. When that B channel is busy, the previous one is selected.

`DTMF[<sec>,/<dir>/<file>]`	Please refer to Chapter 11.2.1 Announcements.

Example 5.26 Subscriber settings 1

```
Subscriber00=TRANSPARENT ROUTER ALARM
Subscriber01=TRANSPARENT ROUTER ALARM
Subscriber02=TRANSPARENT ROUTER SWITCH CHMAX[16] ALARM
```

Required mobile parameters

Specific settings for each mobile interface appear in square brackets behind the keywords **GSM**, **UMTS** or **CDMA**. These parameters are separated with a comma.

Table 5.16 Required mobile parameters in `pabx.cfg`

Subscriber<port>=<type> [<pin>,<lain>,<SMSC>,<sim>,<loudGSM>,<loudPCM>,SIM<x>,...]	
`<port>`	Refers to the running (physical) port number.
`<type>`	Enter `GSM`, `CDMA` or `UMTS` depending on your hardware configuration.
`<pin>`	Defines the SIM card's PIN. The PIN is always four digits. If no PIN is defined for a SIM card, the PIN 0000 must be used. NOTE: An error message appears in the protocol.log file when a PIN is incorrectly configured.
`<lain>`	Defines the LAIN (**L**ocal **A**rea **I**dentification **N**umber) – the mobile network to be used. The LAIN consists of the MCC (Mobile Country Code) and the MNC (Mobile Network Code). Setting this parameter prevents roaming into another mobile network. If the LAIN is set at 00000, roaming will not be prevented. The LAIN configuration prevents accidental logon of the SIM card with another network and the use of false SIM cards.
`<SMSC>`	Defines the SMS center's access number. The number must always begin with + and the country code.
`<SIM>`	Defines the SIM card to be used. You can enter the values 1, 2, 3, 4, 5, 6 (optional when using the 24 SIM card carrier). Default 1. Do not change the default entry if your gateway is equipped with SIM4 carriers . Activate configuration suffices to activate changes. NOTE: Please see the example following Table 5.17 for information on numbering SIM cards.
`<loudGSM>`	Defines the volume level for the mobile line. The values 0 to 3 are possible. 0 is loudest and 3 is the least loud. Activating echo cancellation (for GSM modules only): Depending on the base station (BTS) one of three algorithms will work for this feature. The algorithms must be tested during activation to determine which one fits the base station type. The following values are added to the volume setting: - 16 -> algorithm 1 - 32 -> algorithm 3 - 48 -> algorithm 6 EXAMPLE 1: If the volume level is set at 1, and algorithm 1 is used for echo cancellation, the configuration for <loudGSM> is 17: Subscriber00=… GSM[0000,00000,+000000,1,17,1,SIM4] … EXAMPLE 2: If the volume level is set at 2, and algorithm 6 is used for echo cancellation, the configuration for <loudGSM> is 50: Subscriber00=… GSM[0000,00000,+000000,1,50,1,SIM4] …

Table 5.16 Required mobile parameters in `pabx.cfg` *(continued)*

Subscriber<port>=<type> [<pin>,<lain>,<SMSC>,<sim>,<loudGSM>,<loudPCM>,SIM<x>,...]	
`<loudPCM>`	Defines the volume level to the fixed network. The values 0 to 7 are possible. 7 is loudest and 0 is the least loud.
`SIM4`	Indicates that the gateway is equipped with SIM4 carriers.

Optional mobile parameters

In addition to the usual parameters, you can enter the following optional mobile parameters. Separate each parameter with a comma.

Table 5.17 Optional mobile parameters in `pabx.cfg`

Parameter	Description
`BAND(<int>)`	Defines the GSM frequency band and (<int>) can have the following values: 1 = Mono-band mode 850MHz (Q24CL001 modules only) 2 = Mono-band mode 900MHz (Q24CL001 modules only) 3 = Mono-band mode 1800MHz (Q24CL001 modules only) 4 = Mono-band mode 1900MHz (Q24CL001 modules only) 5 = Dual-band mode 850/1900MHz (Q24CL001 and GE864-QUAD modules) 6 = Dual-band mode 900/1800MHz (Q24CL001 and GE864-QUAD modules) 7 = Dual-band mode 900/1900MHz (Q24CL001 and GE864-QUAD modules) 8 = Dual-band mode 850/1800MHz (GE864-QUAD modules only) After changing the band settings, you must restart the system to activate the changes. NOTE: The BAND parameter can only be used with quad-band GSM module-type Q24CL001. These quad-band GSM modules are available as of hardware revision 1.61 (May, 2007). There is no default band setting! If there is no BAND configuration in the `pabx.cfg` when the system is started, the last band stored on the module will be used. This can cause the system to attempt to register the SIM with the wrong GSM band.
`BNDS<int>`	**For** UMTS Board**s with module type UC864-G only:** Selects the UMTS or GSM or auto network `0` = auto (default) 1 = GSM 2 = UMTS
`BNDU(<int>)`	**For** UMTS Board**s with module type UC864-G only:** Configures BAND selection in the UMTS network `0` = 850/1900/2100 MHz (default) 1 = 850 MHz 2 = 1900 MHz 3 = 2100 MHz

5.2.1.6 Global settings

For a detailed description of the configuration of the Mobile Board, including the keywords `CHADDR`, `NEXT`, `LIMIT` and `CONTINUE`, please refer to Chapter 7 Mobile configuration options.

This category contains the following system parameters:

Table 5.18 `pabx.cfg`: global settings

System Parameters
`VoipGlobalMaxChan=<count>` Max. number of VoIP channels for the entire system.
`VoipMaxChanOut=<count>` Limits outgoing calls through this profile. You have to define the number of VoIP channels to be used.
`VoipSuppressRtcp=<mode>` Suppresses (`Yes`) or allows (`No`) the sending of RTCP packets.
`VoipAnnounce=<filename>,NOCONN` Only for outgoing calls which are sent through the VoIP profile where this parameter is included. `VoipAnnounce` defines an audio file which is played to the caller. The entire file is played, even if the called person picks up the call before the end of the announcement. In the default setting, a connect is sent to the caller straight away so that the caller can hear the announcement. Set the option `NOCONN` to suppress this connect.
`VoipStopAnnounceOnConnect=<mode>` If `Yes` is set and the call changes to the state CONNECTED, the announcement configured with VoipAnnounce is stopped.
`VoipStopAnnounceOnAlert=<mode>` If `Yes` is set and the call changes to the state ALERTING, the announcement configured with VoipAnnounce is stopped.
`VoipFaxVolume=<volume>` Defines the volume of fax and CID tones. Range: 0 (-21 dB) to 15 (-6 dB) Default: 9 (-12 db)
`VoipCEDTransferMode=<int>` Defines whether CED (a tone initially sent by a fax device) is sent via T.38 or RTP: 0 = T.38 (default) 1 = RTP
`VoipCngRelayEnable=<int>` Defines whether CNG (a tone initially sent by a fax device) is transmitted by means of T.38 or RTP: 0 = RTP (default) 1 = T.38
`VoipSendUpdate=<mode>` Allows (`Yes`) sending of UPDATE messages or not (`No`). `Yes` is the default value.
`VoipRtpPort=<port>` Defines the starting UDP port used to transmit RTP and RTCP packets (default 29000).
`VoipRtpPortSpacing=<count>` Defines the space between the ports used for individual RTP streams (default 2). Minimum value is 2 (default). Each connection requires two ports (one for RTP and one for RTCP).

Table 5.18 `pabx.cfg`: global settings *(continued)*

System Parameters
`H225Port=<port>` Endpoint-to-endpoint port (default 1720).
`SipPort=<port>` SIP signaling port (default 5060). A different port for sending can be set in the `route.cfg`'s VoIP profile. Otherwise the port set here will be used for sending and receiving.
`VoipMaximumBandwidth=<int>` Defines an upper limit for available bandwidth for the VoIP profiles to be configured (see Voip-BandwidthRestriction in Table 9.6) if traffic shaping is active for the corresponding VoIP profile. Individual codecs are assigned the following values in kBit/s: g711a, f711u, trp: 8 g72632, t38: 4 g72624 3 g72616, gsm 2 Other 1 You must define the list of codecs to be used in the VoIP profiles, whereby the codec with the highest priority must be defined first. Calls will be set up using the codec with the highest priority as long as the sum of the values for individual calls remains lower than defined here. If the sum is greater, the next call will be set up with, and existing calls will be switched to, a higher compression rate. Bear in mind that the VoIP peer must support this feature.
`VoipStrictRfc3261=<mode>` If yes is set, the SIP transaction/dialog matching will occur strictly as per RFC3261. You must disable this feature for peers that use RFC2543 (from and to name). Default is yes.
`VoipLinger=<sec>` After an H.323 release complete has been sent, the TCP session will remain open for the number of seconds entered if there is no response from TCP message sent so that the system can resend the TCP packet.
`StunServerAddress=<ip addr>` When this parameter is active, the CELLX looks for a (NAT) firewall in the network and figures out how to bypass it without requiring changes. All ports for signaling, RTP and RTCP are checked. The parameter VoipGlobalMaxChan defines the number of ports for RTP and RTCP. NOTE: This is not a solution for all firewall types.
`StunServerPollInterval=<sec>` Interval (in seconds) for the stun request at each port (default 600).
`Radius=<mode>` On (default) activates the Radius service. If you change Off to On, you must restart the system.
`RadiusAuthPort=<num>` Port used for Radius authentication (default 1812).
`RadiusAcctPort=<num>` Port used for Radius accounting (default 1813).
`NameServer=<ip addr>` IP-address configuration for the DNS server. Enter your network or ISP's DNS server. If you don't know it, you can also enter another DNS server. If you have more than one address, enter this parameter up to three times on different lines.
`Timezone=<continent/city>` Defines the time difference between the CELLX's time zone and time zone 0 (Greenwich Mean Time). Enter the continent and a large city (usually the capital) in the time zone.

Table 5.18 `pabx.cfg`: global settings *(continued)*

System Parameters

`NtpServer=<ip addr>`
Sets the SNTP server's IP address for standard-time queries. The query occurs every four hours.
NOTE: If your system is not attached to an NTP server, you can enter the following configuration to retrieve the time from an ISDN TE port:
Subscriber=...TIME

`Clockmaster=<type>` Enter S0 to take the system clock from the BRI port if the system has an additional BRI board and special firmware installed on which at least one controller is connected to the PSTN in TE mode. This parameter only makes sense if the system does not have a PRI port connected to the PSTN.
`S2MLongHaul=<mode>` This option increases the sensitivity on PRI receiving side to support Long Haul applications. The default value is No (Short Haul).
`MoipPort=<port>` Defines the GATE Manager access port (default 4445).
`FtpdPort=<port>` Defines the FTP access port (default 21).
`TelnetdPort=<port>` Defines the TELNET access port (default 23).
`TftpdPort=<port>` Defines the TFTP access port (default 69).
`Ftpd=<mode>` Activates (on) or deactivates (off) FTP access. Default on.
`Telnetd=<mode>` Activates (on) or deactivates (off) TELNET access. Default on.
`Tftpd=<mode>` Activates (on) or deactivates (off) TFTP access. Default off.
`RemotePassword=<password>` Defines the password for FTP and GATE Manager access. Please refer to Chapter 4.10.4 on page 34 for instructions on how to enter an encrypted password in the `pabx.cfg`. If you do not define a password, access to the system via GATE Manager occurs without a password, and FTP access occurs with the default password tcs-ag.
`DialTone=<country>` If the system is used in a corporate settings and attached through a PBX to the PSTN, it may be necessary to generate the carrier's dial tone. It depends on whether the system sends the dialed digits to the PSTN or whether it waits for a routing entry to take the call. The following values can be entered: GE, DE, IR, UK, US, FR, IT

Example 5.27 Global settings

```
VoipGlobalMaxChan=60
H225Port=1720
SipPort=5060
VoipRtpPort=29000
VoipRtpPortSpacing=2
NameServer=192.168.0.254
Timezone=Europe/Berlin
NtpServer=192.168.0.254
DialTone=GE
```

There is no internal time generation for the system when the power is interrupted. That means the default time is used when the system is restarted or rebooted! Therefore it is important to set the system time with an NTP server.

Alternatively, if the system is connected via BRI or PRI, a clock may come from the network connected to the corresponding port. Enter `TIME` in the `pabx.cfg`'s `subscriber` line of the TE port and then activate the configuration to activate this clock.

5.2.2 SMTP-client configuration

The following entries in the `pabx.cfg`'s [Mail] section are used to send e-mail messages from the CELLX. The connection to the SMTP server can be used to send CDR files, incoming SMS to an e-mail account or alarm messages.

You must restart the system after making changes to activate the settings.

The following features are possible:

- Sending SMS via e-mail
- Receiving SMS in an e-mail, SMS or in a file
- Sending and receiving USSD text messages
- Displaying incoming calls via e-mail
- Setting up connections using e-mail
- Sending automatic SMS for unconnected calls
- Sending CDRs via e-mail
- Sending alarm messages via e-mail

Table 5.19 SMTP client configuration

SMTP parameters
`SmtpServer=<ip addr>` In `<ip addr>`, enter the IP address of the destination SMTP server that is to receive the e-mail messages.
`MailUserIn=<username>` Enter a username for incoming e-mail authentication.
`MailUserOut=<username>` Enter a username for outgoing e-mail authentication.
`MailPwdIn=<password>` Enter a password for incoming e-mail authentication.
`MailPwdOut=<password>` Enter a password for outgoing e-mail authentication.
`MailPortIn=<num>` Enter a TCP port for incoming email (default 25).
`MailPortOut=<num>` Enter a TCP port for outgoing email (default 25).
`MailAuthEncr=<type>` Enter an encryption method for e-mail authentication (default base64).

Table 5.19 SMTP client configuration *(continued)*

SMTP parameters
`MailRcpt=<domain>` In <domain>, enter the destination domain, the destination address and an @ sign. If the destination address is already complete (with an @ sign), <domain> is not added.

`MailFrom=<URL>`
Enter here the URL that will be sent in the e-mail's FROM field in the following possible formats:
MailFrom=domain to send OAD@domain (or user@domain, if no OAD available).
MailFrom=ipaddress to send OAD@ipaddress (or user@ipaddress, if no OAD available).
MailFrom=123@ to send 123@<IP address of gateway>.
MailFrom=123@teles.de to send 123@teles.de.
MailFrom=123@ipaddress to send 123@ipaddress.

`MailRcvMax=<count>` Maximum number of incoming e-mails queued for transmission via SMS or USSD.
`MailRcptMax=<count>` Number of "RCPT TO" entries in e-mails that come from the LAN (a message is sent to the LCR for each "RCPT TO" entry in each incoming e-mail).
`MaxMailsToHost=<count>` Maximum number of e-mail messages sent to the LCR simultaneously. Default 4
`MailToHostRetries=<count>` Number of retries when SMS transmission is not successful. When the limit entered is reached, an error message is sent to the e-mail sender (default 3).
`MailSendRetries=<count>` Number of times an attempt is made to send an e-mail. Default 10.
`MailMaxIncomingClients=<count>` Defines the maximum number of clients that can access the system simultaneously. If 0 is entered, the SMTP port (25) will be blocked for incoming sessions. Default 100.
`MailTcpRcvTimeout=<sec>` Defines the number of seconds after which a session will be terminated following a possible receiving error in the data stream. Default 0 (immediately).
`MailTcpSndTimeout=<sec>` Defines the number of seconds after which a session will be terminated following a possible transmission error in the data stream. Default 0 (immediately).
`MailAllowedPeers=<ip addr>` Defines IP addresses from which incoming SMTP connections will be accepted. Separate IP addresses with a space. If a dash (-) is entered, the SMTP port (25) will be blocked for incoming sessions. If this parameter is left empty (default), incoming connections will be accepted from all IP addresses.
`MailPropPort=<num>` Enter the port number for a TELES proprietary mail protocol.

Sending alarm messages via e-mail

With the appropriate configuration, you can send e-mails containing alarm messages that are written into the log file. The sender is given as `alarm` and the system's name appears in the subject box. The text box contains the alarm message.

The following entry in the configuration file activates this function.

Example 5.28 Sending alarm messages via e-mai

```
...
ActionLog=/data/protocol.log daily 1000 5 @<e-mail account>
...
```

5.2.3 Number portability settings

The [NumberPortability] section includes the parameters necessary for communication with the database server. For a description of the functionality and configuration of this feature, please see Chapter 11.5 Ported number screening

You must restart the system after making changes to activate the settings.

Table 5.20 Number portability settings

Number portability parameters
`MNPQAddress=<ip addr>` For iMNP or direct queries to Enquire: Enter the IP address to which the number portability query is to be sent.
`MNPQPort=<port>` For iMNP or direct queries to Enquire: Enter the tcp port to which the number portability query is to be sent.
`MNPQAddress2=<ip addr>` For iMNP or direct queries to Enquire: Enter the IP address to which the second number portability query is to be sent when ! appears in the mapping entry. A second database will then be queried, for example if the first one is not online.
MNPQPort2=<port> For iMNP or direct queries to Enquire: Enter the tcp port to which the second number portability query is to be sent.
`MNPQSum=<mode>` For iMNP: This parameter must be activated (Yes) if an iMNP is used.
`E2EMRSAddress=<ip addr>` For direct queries to End2End: Enter the IP address to which the number portability query is to be sent.
`E2EMRSPort=<port>` For direct queries to End2End: Enter the udp port to which the number portability query is to be sent.

5.2.4 SNMP settings

The Simple Network Management Protocol facilitates network management and monitoring of CELLX network devices and their functions. For a detailed description of SNMP configuration, please refer to Chapter 10.4 SNMP agent.

You must restart the system after making changes to activate the settings.

5.2.5 Time-controlled configuration settings

The [Night<num>] section is reserved for prospective time-controlled configuration changes. In the `pabx.cfg` file, the `Night` sections contain all of the system's `Subscriber` entries.

5.2.6 .CASR2 settings

If you are working with Channel Associated Signaling, you must activate a CAS profile in the relevant `Controller` and `Subscriber` entries and define a profile for each `Subscriber` entry in a separate `[CASR2:<name>]` section.

Generally you will need to set only the country code 55 for Brazil. The default country code is 0, which sets the ITU-T standard.

Table 5.21 CASR2 settings

CAS profile parameters
`CountryCode=<num>` Defines set of pre-configured R2 parameters according to the E.164 country code (for Brazil 55, etc.) When not defined or set at 0, the ITU-T standard set of parameters will be used. Default 0.
`CDbit=<int>` Specifies the default setting of the C and D bits when the port transmits line signals. The default setting for C is 0 and for D is 1. Generally they will not need to be changed.
`Iabcd=<int>` Specifies whether any of the A B C or D bits are inverted. Default is 0. Generally it will not need to be changed.
`BlockBeforeIdle=<int>` Set to 1 when backward blocking is required (before going to idle) after the clear forward has been received (default 0).
`AnsTone=<int>` Defines the Group A tone used to respond to incoming calls and switch to Group B tones (default 3).
`ANIReqCatTone=<int>` Defines the group A tone used to request the calling-party category before the ANI digits. The default value is 5 or 6, depending on the country code.
`ANIReqAniTone=<int>` Defines the Group A tone used to request the next ANI digit. The default value is 1, 2, 4, 5 or 9, depending on the country code.

Table 5.21 CASR2 settings *(continued)*

CAS profile parameters
`RepeatBeginning=<int>` Defines the group A tone used to restart sending the DNIS from the first digit. The default value is 0, 2, 9 or 10, depending on the country code.
`SendMinus1=<int>` Defines the Group A tone used to request repetition of the last DNIS digit. The default value is 0, 2, 8 or 9, depending on the country code.
`SendMinus2=<int>` Group A tone used to request repetition of the second to the last DNIS digit. The default value is 0, 7 or 9, depending on the country code.
`SendMinus3=<int>` Group A tone used to request repetition of the third to the last DNIS digit. The default value is 0 or 8, depending on the country code.
`GrBIdlTone=<int>` Defines the Group B signal to indicate that the called line is available and charges will be tabulated if the called party answers. Default 1, 6 or 5, depending on the country code.
`GrBIdlNoChargeTone=<int>` Defines the Group B signal to indicate that the called line is available and charges will not be tabulated if the called party answers.
`GrBBusyTone=<int>` Defines the Group B busy signal, which specifies the Group B backward tone to be sent by the incoming R2 register to indicate the busy condition.
`GrBCongTone=<int>` Defines the Group B congestion tone, which specifies the Group B backward tone to be sent by the incoming R2 register to indicate the congestion condition (default 4).
`GrBOutOfOrdTone=<int>` Defines the Group B line out of order tone, which specifies the Group B backward tone to be sent by the incoming R2 register to indicate that the line is out of order (default 8).
`GrBUnallocNumbTone=<int>` Defines the Group B unallocated number tone, which specifies the Group B backward tone to be sent by the incoming R2 register to indicate that the called number does not exist.
`EndDIDTone=<int>` Defines the Group I tone that indicates the end of the DID. When 0 is set, this tone will not be used and T3 timeout and A-3 pulse from the incoming side will indicate the end of the DID.
`ANICallingCat=<int>` Defines the index of possible categories for the country that defines the Group II tone.
`ANIRequest=<int>` Specifies whether or not ANI is requested. When 0 is set, AniMaxRxDigits is automatically reset to 0 (default 1).
`AniMaxRxDigits=<int>` Defines the maximum number of ANI digits for incoming calls (default 32).
`DnisMaxRxDigits=<int>` Defines the maximum numberof DNIS digits for incoming telephony calls. It is important to set this parameter accurately for national variants that do not handle variable-length dialed digit strings well. For other variants, a large number can be used.
`GetAniAfterDnis=<int>` Enter 0 to get the ANI after the first DNIS digit. Enter 1 to get the ANI digits after the DNIS is complete.

Example 5.29 CASR2 settings

```
Controller00=9 TES2M CASR2
...
Subscriber00 = TRANSPARENT ROUTER CASR2[BRAZIL1] ALARM
Subscriber01 = TRANSPARENT ROUTER CASR2[BRAZIL2] ALARM
...
[CASR2:BRAZIL1]
CountryCode=55

[CASR2:BRAZIL2]
CountryCode=55
```

You must restart the system after making changes to activate the settings.

5.3 Configuration file route.cfg

The system’s routing information is saved in the `route.cfg`. The file contains the following sections:

Table 5.22 Sections in the `route.cfg` file

Section	Function
`[System]`	Contains all routing entries (MapAll, Restrict, Redirect) that are to be active when the default configuration is used.
`[Night<num>]`	Contains all routing entries (MapAll, Restrict, Redirect), and VoIP, gatekeeper and registrar profiles that are to be active with the defined time configuration. Bear in mind that you must also copy all routing and profile settings that may already appear in the das System section or in the individual profile sections, even if they do not change!
`[VoIP:<name>]`	Contains all settings necessary for communication with the VoIP peer.
`[GateKeeper:<name>]`	Contains all settings for the gatekeeper. This profile is then assigned to the VoIP profiles.
`[Registrar:<name>]`	Contains all settings to register with the registrar.

5.3.1 Entries in the [System] section

The[System]section contains the following entries.

5.3.1.1 Restrict

`Restrict` entries are used to handle calls in a mapping based on the controller / controller group where the calls originate. A `Restrict` entry can be used, for instance, to route all calls coming from PSTN directly to the PBX. If no called party number (DAD) is transmitted, `Restrict` can also be used to make the call mappable, for instance for calls coming from GSM.

The `Restrict` parameter adds a prefix to a DAD before the DAD is mapped. `Restrict` parameters are always handled before the `MapAll` parameters and always require a matching `MapAll` parameter.

The left side of the equals sign in the `Restrict` parameter contains the controller number plus an optional trunk number or a specific calling number (OAD). The special symbol `?` may be used as a wildcard to represent any character. The right side contains the prefix that is to be put in front of the DAD and an optional service indicator.

In the `route.cfg`, the list of `Restrict` parameters is searched from bottom to top for a matching controller plus optional trunk number / OAD. Because the search is done bottom up, place the more specific `Restrict` entries **below** the more general ones. Once a match has been found, the DAD is prefixed with the contents of the `<pl>` variable. Then the call is mapped.

Table 5.23 `route.cfg`: restrict parameters

Restrict<ns>[R][T]=<pl> <sin>			
Parameter	Description	No. Digits	Optional
`<ns>`	Contains the controller number plus an optional trunk number or a specific calling number (OAD). The special symbol `?` may be used as a wildcard to represent any character.	59	
[R]	For calls that are redirected with `Redirect3`, the original OAD can be changed again using `RestrictR`. Only in combination with service indicator 15.		X
[T]	For calls that are redirected with `Redirect2`, the original OAD can be changed again using `RestrictT`. Only in combination with service indicator 15.		X
`<pl>`	Stands for a virtual placeholder. The DAD is prefixed with the contents of this variable.	59	
`<sin>`	The service indicator variable `sin` restricts the command to a service. Without a `sin`, the Restrict command is valid for all services. Possible service indicator values are: `00` All services `01` Telephony `02` Analog services `03` X.21-services `04` Telefax group 4 `05` 64 kbps videotext or TELES-specific SMS services `06` TELES-specific USSD services `07` Data transfer 64 kbps `08` X.25-services `09` Teletext 64 `10` Mixed mode `15` Used internally for calling-party (OAD) manipulation. `16` Video telephone	2	

All calls from PRI controller `9` (PSTN) are sent to PRI controller `10` (PBX). First, the Restrict command adds the prefix “pl” to the DAD of the received call for all calls coming from PSTN. Then the call is mapped. In the mapping, everything that is prefixed with “pl” is sent to controller 10 by removing the prefix and adding the controller to the number.

Example 5.30 Restrict 1

```
Restrict9=pl
MapAllpl=10
```

Calls from mobile controllers with the LAIN 26212 are sent to PRI controller 10 (PBX), extension 0. This is imperative because the caller cannot dial an extension directly with mobile.

Example 5.31 Restrict 2

```
Restrict26212=100
```

For a detailed description, please see Chapter 7.2 Incoming voice calls from mobile.

5.3.1.2 MapAll

Mapping entries are necessary for routing calls. The prefix or telephone number (DAD) for which the mapping applies is searched and the call routed according to the matching mapping entry.

Mapping entries begin with the keyword `MapAll`. They work as follows: anything on the left of the equals sign is removed from the prefix / telephone number (DAD) that has come in and replaced with what is on the right of the equals sign.

If, for example, the incoming DAD is 12345678 with 123456 being the trunk number and 78 the extension, `MapAll123456=9123456` means that 123456 is cut off the number and 9123456 is added to it. The called number is 912345678 with 9 being the port. `MapAll123456=9` means that 123456 is cut off and 9 added. The called number is 978 with 9 being the port.

Mappings are searched from top to bottom. Place the more specific entries **above** the more general ones.

Table 5.24 `route.cfg`: map parameters

MapAll<direct>=<num> <mode>			
Parameter	Description	No. Digits	Optional
`<direct>`	Defines the prefix or telephone number for which the entry applies.	11	
`<num>`	Defines the routing for a call in the order given: ▪ Destination port's controller number ▪ Optional VoIP profile name followed by a colon if the call is terminated via VoIP ▪ Optional prefix ▪ Part of the number on the left that is transmitted The special symbol `?` may be used as a wildcard to represent any character. The special symbol `.` may be used as a wildcard to represent any digit.	59	
`<mode>`	`VOICE` Applies for calls with the service indicator **voice** (default). `DATA` Applies for calls with the service indicator **data**.	4 or 5	X

All mobile calls with the prefix 01555 are transmitted to the mobile controllers `(20)`. All international calls are sent to the VoIP carrier (`40`) with the profile name `DF`. All national calls are sent to the PRI controller with the number `9`. It is important that the mapping for international calls is placed above the mapping for national calls. If you change the order of both mappings, international calls would be sent to controller 9 instead of 40.

Example 5.32 MapAll 1

```
MapAll01555=|2001555<<14
MapAll00=40DF:00
MapAll0=90
```

If `CHADDR` appears in the mobile port's `Subscriber` lines, the entry will look like this:

`MapAll<num>=<lain><num>`

All calls with the prefixes 01555 and 01556 are sent to the mobile controllers with the LAIN 26212. All calls with the prefixes 01444 and 01445 are sent to the mobile controllers with the LAIN 26213. Digit collection is activated.

Example 5.33 MapAll 2

```
MapAll01555=|2621201555<<17
MapAll01556=|2621201556<<17
MapAll01444=|2621301444<<17
MapAll01445=|2621301445<<17
```

Make sure that the numbers for the carriers are routed to the correct ports! For detailed information on digit collection and enblock/overlap receiving, please see Chapter 8.4 Digit collection (enblock/overlap receiving).

5.3.1.3 Redirect

This entry facilitates alternative routing when the first destination cannot be reached or is busy. A placeholder appears to the right of the equal sign. The routing entry (MapAll) can be defined for the redirect using the placeholder entered.

Table 5.25 `route.cfg`: redirect parameters

Redirect<type><num>=<redirect> <sin> <time>			
Parameter	Description	No. Digits	Optional
`<type>`	Enter 2 or 3 to set the following types: 2 call forwarding no answer 3 call forwarding when busy	1	
`<num>`	Defines the number for which calls will be redirected.	59	
`<redirect>`	Defines the placeholder used in the two-target routing entry and the number to which calls to <x> will be redirected.	59	
`<sin>`	The service indicator variable sin restricts the command to a service. Without a sin, the Redirect command is valid for all services. Possible service indicator values are: `01` Telephony `02` Analog services `03` X.21-services `04` Telefax group 4 `05` Videotext (64 kbps) `07` Data transfer 64 kbps `08` X.25-services `09` Teletext 64 `10` Mixed mode `15` Videotext (new standard) `16` Video telephone NOTE: Fax forwarding must be set for analog and telephony services because incoming fax calls from the analog network may arrive with either telephony or analog service indicators.	2	(X) (Only optional if <time> is not set. If <time> is set and <sin> is not needed please select 00 for sin.)
`<time>`	For type 2 redirect entries, a timer (in seconds) can be defined after the service indicator entry.	255	X

In the following example, all mobile calls with the prefix 01555 are transmitted to the mobile carrier with the LAIN 26212. Digit collection is activated. If the carrier cannot be reached or is busy, the redirect command activates the second target mapping with the placeholder `A` and the call is automatically sent to PRI controller `9`.

Example 5.34 Redirect 1

```
MapAll01555=|2621201555<<17
Redirect326212=A
MapAllA=9
```

In the following example, calls to 26212 that remain unanswered for 12 seconds and calls to 26213 that remain unanswered for 20 seconds are redirected through the PRI port.

Example 5.35 Redirect 2

```
MapAll01555=|2621201555<<17
MapAll01556=|2621301556<<17
Redirect226212=A 01 12
Redirect226213=A 01 20
MapAllA=9
```

Excluding busy calls or specific cause values from redirect

Defines a hexadecimal cause value according to DSS1. When connections to the destination are rejected because of the reason defined by the cause value, the CELLX sends a busy signal to the attached PBX. Alternative routing is not carried out.

To avoid second-choice routings when the called-party number is busy, set the following parameter in the first-choice port's Subscriber line in the `pabx.cfg`:

`BUSY[<cause>]`

Defines a hexadecimal cause value according to DSS1. When connections to the destination are rejected because of the reason defined by the cause value, the CELLX sends a busy signal to the attached PBX. Alternative routing is not carried out. You can also define a range of consecutive cause values:

`BUSY[<cause>,<cause>]`

An exclamation point (`!`) in front of a cause value means all cause values except the one listed. For example, BUSY[!95], means all cause values except 95 will be rejected with a busy signal.

In the following example, all outgoing calls over controller 04 are rejected with the cause value 91 when the called party is busy. Alternative routing is not carried out.

Example 5.36 Called party is busy

```
Subscriber04=....BUSY[91]
```

5.3.1.4 Setting the time-controlled sections

If you use a time-configured route on the system, please see Chapter 5.2.1.3 for a definition of individual configuration zones. The active route is configured in the `route.cfg` file.

The following example contains three sections ([System], [Night1] and [Night2]), in which the route changes. All international calls are sent to the VoIP carrier DF in the default configuration. Digit collection is actived. In the time span for [Night1], these international calls are routed to VoIP carrier Ni, and in the time span for [Night2] they are routed through the PRI controller to the carrier with the prefix 010xx. National calls are always sent to VoIP carrier DF and local calls are routed to the outside line.

Example 5.37 Setting the time-controlled sections

```
[System]
MapAll00=|40DF:00<<24
MapAll0=|40DF:0<<24
MapAll?=9?

[Night1]
MapAll00=|40Ni:00<<24
MapAll0=|40DF:0<<24
MapAll?=9?

[Night2]
MapAll00=9010xx00
MapAll0=|40DF:0<<24
MapAll?=9?
```

Any defined Night configurations must be set in the files `pabx.cfg` and `route.cfg`. If there are no changes in these sections, you must copy them from the System section. The complete Subscriber section must appear in the Night section of the `pabx.cfg` (see Chapter 5.2.5 on page 68). The active route must appear in the `route.cfg` (see Chapter 5.3 on page 70).

5.3.2 VoIP profiles

This section includes all of the most important parameters for communication with the VoIP peer.

Basic parameters

Table 5.26 `route.cfg`: VoIP basic parameters

VoIP Basic Parameters
`[Voip:<name>]` Name of the routing profile. The name must begin with a letter. Use a short and meaningful name.
`VoipDirection=<mode>` Defines the direction in which VoIP calls can be set up. Possible options: In, Out, IO, None).
`VoipPeerAddress=<ip addr> or <name>` The peer's IP address or name. Default is 0 (if it is not set, please set the parameter VoipIpMask to 0x00000000).
`VoipIpMask=<ip mask>` The subnetmask is used to determine the size of the IP address range for incoming traffic. The syntax is 0x followed by the mask in hexadecimal notation. Example of a Class C mask entry: 0xffffff00. Default is 0xffffffff (only incoming traffic is accepted from the defined peer address).

Table 5.26 `route.cfg`: VoIP basic parameters *(continued)*

VoIP Basic Parameters

`VoipSignalling=<int>`

Determines the profile's signaling protocol for outgoing VoIP calls. In the case of incoming calls, auto detection ensures that each call from the peer is accepted, regardless of the protocol: 0=H.323 (default), 1=SIP udp, 2=SIP tcp, 3=tls.

NOTE: TLS requires the following additional software: crypto.vnd and the key files key.pem and cert.pem.

`VoipCompression=<list>`

The compression to be used, in order of preference. At least one matching codec with the peer must be defined.

Voice:

g729, g729a, g729b, g729ab

These codecs have a bit rate of 8 kbit/s (compression ratio 1:8). A stands for annex a and b for annex b.

g72616, g72624, g72632, g72640

These ADPCM codecs have various bit rates: g72616 = 16kBit/s (compression ratio 1:4), g72624 = 24kBit/s, g72632 = 32kBit/s (compression ratio 1:2), g72640 = 40kBit/s.

NOTE: G726 32kBit/s can also be signaled as G.721 by using the entry g721.

g728

The Codec has a bit rate of 16kBit/s (compression ratio 1:4).

g711a, g711u

These PCM codecs have a bit rate of 64kBit/s. No voice compression occurs. a stands for a-law and u for µ-law.

g723, g723L

These codecs work with 30ms data frames. g723.1 uses a bit rate of 6.3 kbit/s, and g723L uses a bit rate of 5.3 kbit/s to send RTP packets.

NOTE: This has no influence on the compression ratio of incoming RTP packets. Both sides must be able to receive both ratios.

gsm

GSM-FR (full rate) has a bit rate of 13 kbit/s.

The following codecs are also possible: g721 (SIP only)

Fax: t38

T.38 (fax over IP) allows the transfer of fax documents in real time between 2 fax machines over IP. Following fax detection during a call, the voice codec will switch to T.38.

Data: trp

Transparent or clear mode (RFC 4040). Transparent relay of 64 kbit/s data streams.

gnx64:

Clear channel codec

ccd:

Clear channel data (as per RFC3108)

Define a special profile for data call origination or destination numbers. Bear in mind that echo cancelation in this VoIP profile might be switched off (VoipECE=no).

`VoipMaxChan=<count>`

Maximum number of channels that can be used with the profile. If this parameter is not defined (default), there will be no limit.

NOTE: For versions 13.0c or lower, we recommend that you also set the parameter `VoipDelayDisc` to `Yes` to improve the ASR.

Table 5.26 `route.cfg`: VoIP basic parameters *(continued)*

VoIP Basic Parameters
`VoipSilenceSuppression=<mode>` Yes activates silence suppression, CNG (comfort noise generation) and VAD (voice activity detection). No (default) deactivates silence suppression. NOTE: In SIP signaling, silence suppression is negotiated as per RFC3555.
`VoipTxM=<num> or <list> fix` The multiplication factor (1-12) for the frame size for transmission of RTP packets (default is 4). 10ms is the default frame size (20ms for CELLX32). A list can be defined if different frame sizes are to be used for different codecs in the VoIP profile. The list must correspond with the list in the parameter VoipCompression. Normally the peer's frame size will be used if it is smaller than the one defined. If you enter fix, the configured factor will always be used.

Please refer to Chapter 9 Additional VoIP parameters for information on other possible entries.

Management parameters

Table 5.27 `route.cfg`: VoIP management parameters

VoIP Management Parameters
`VoipGk=<list>` Name of the assigned gatekeeper profile. You can assign a profile to several gatekeepers to define backup gatekeepers for a VoIP profile. In this case, the next gatekeeper will be used if the previous one fails.
`VoipProxy=<ip addr>` Enter the IP address of the SIP server.
`VoipUser=<username>` Define the username for the remote device if authentication is required (SIP only).
`VoipPwd=<password>` Define the password for the remote device if authentication is required (SIP only).
`VoipRegistrar=<name>` Enter the name of a registrar to be used for the VoIP profile.
`VoipRadiusAuthenticate=<name>` Enter the name of the Radius server to activate user authentication.
`VoipRadiusAccounting=<name>` Enter the name of the Radius server to activate accounting. If the call is sent to the mobile network, the SIM card's IMSI is transferred in the Vendor attribute: Vendor-2170-Attr-1 = in hexadecimal notation in the following format: IMSI=<IMSI>
`VoipRadiusIMSINegotiation=<mode>` If yes is entered, the <system> will negotiate the IMSI to be used in the Radius access request. It will transmit the IMSI in the RADIUS attribute: Filter-Id (11). Default no. NOTE: The call number of the dial attempts must begin with the LAIN or the port number if CHADDR is not used. The following routing entry must be added: MapAll*??01:<LAIN>=<LAIN>*??01:

Table 5.27 `route.cfg`: VoIP management parameters *(continued)*

VoIP Management Parameters
`VoipIpLogging=<mode>` Enter Yes to activate recording IP addresses in the CDRs (default is No). The first IP address is the signaling address and the second is the RTP address, followed by the the codec and the frame size used. The IMSI appears after the IP addresses if the keyword IMSI is defined in the `pabx.cfg`. Example of a CDR entry: 21.08.07-11:01:42,21.08.07- 11:01:58,40,912345,192.168.0.2:192.168.0.2,G729,10,0101,16,10,0 Example of a failed log entry: 21.08.07-11:11:30,40,91234,192.168.0.2:192.168.0.2,G729,10,0101,ff,2,1
`VoipStatLogging=<mode>` When Yes is entered, statistic values (for example fraction lost, round trip time, and so on) for the VoIP profile are saved into the protocol.log file every ten minutes. This is helpful during problem analysis when IP issues occur (default = No).
`VoipHold=<mode>` Determines the behavior of the HOLD feature if a PBX sends an Re-INVITE message for HOLD to the gateway. `transparent`: HOLD is sent to PSTN `notify`: HOLD is sent as notification to the telephone exchange `ignore`: HOLD is not sent to the telephone exchange - the gateway suppresses the messages to PSTN
`VoipSelectProfilesBySignalling=<Yes/No>` When Yes is entered, VoipSignalling=<int> is taken as the criterion for selecting a voip-profile for an incom-ing voip call, i.e. the signalling for this voip call must match the VoipSignalling parameter in this profile. Default No.

5.3.3 Gatekeeper profiles

Gatekeeper profiles are used to connect the CELLX to several systems by using a gatekeeper if the protocol is H.323. It is possible to configure different gatekeepers for different destinations and to define backup gatekeepers. These gatekeeper profiles are then assigned to the VoIP profiles.

Table 5.28 `route.cfg`: gatekeeper parameters

Gatekeeper Parameters
`[Gatekeeper:<name>]` Name of the gatekeeper profile.
`RasPort=<port>` Indicates the port the gatekeeper uses (default 1719) for registration, admission and status.
`OwnRasPort=<port>` Indicates the port the system uses (default 1719) for registration, admission and status.
`RasPrefix=<list>` CELLX's defined prefix(es). Use a space to separate entries.
`RasId=<name>` The alias used for gatekeeper registration.
`GkId=<name>` The gatekeeper's alias.

Table 5.28 `route.cfg`: gatekeeper parameters *(continued)*

Gatekeeper Parameters
`GkPwd=<name>` Password to log onto the gatekeeper. If you do not use authentication, leave this entry blank.
`GkAdd=<ip addr>` The gatekeeper's IP address.
`GkTtl=<sec>` Gatekeeper time to live (default 0 means infinite).
`GkMaxChan=<count>` Max. number of channels used for this gatekeeper. If this parameter is not defined (default), there will be no limit.
`GkDynMaxChan=<mode>` The static number of available channels in the gatekeeper profile (GkMaxChan=<count>) is replaced with a dynamic number of active mobile ports (up to the number entered in GkMaxChan) when Yes is entered here. Default is No.
`GkUseStun=<mode>` Enter yes (default) to use the STUN values for the GK profile.
`GkTerminalAliasWithPrefix=<mode>` Some gatekeepers may require that prefixes are listed in the Terminal Alias section. Enter Yes to activate this function; default value is No).
`GkTerminalTypeWithPrefix=<mode>` Enter no to deactivate sending the Dialed Prefix Information in the Registration Request (default yes).
`GkDynRai=<mode>` When yes is entered, the GK receives an RAI (resource availability indication) when a status change occurs on the available mobile channels. When no is entered, the RAI is sent with each ARQ (admission request) and DRQ (disengaged request). Default is no.
`GkNoResourceAvailableIndication=<mode>` With this parameter the <system> will not send RAI indications to the Gatekeeper. Default No.

5.3.4 Registrar profiles

Registrar profiles are used to register the CELLX with a SIP registrar. It is possible to configure different registrars for different destinations and to define backup registrars. These registrar profiles are then assigned to the VoIP profiles.

Table 5.29 `route.cfg`: registrar parameters

Registrar Parameters
`[Registrar:<name>]` The name of the registrar profile.
`RegId=<name or ip addr>` Host name or IP address used in the register's request header. Bear in mind that the DNS service must be active if you enter the host name.
`RegOwnId=<name@ip addr/domain>` Typically a host name or telephone number followed by an `@` sign and a domain name or IP address. The entry used in the `From:` field. The default setting is `RegUser@RegId`.

Table 5.29 `route.cfg`: registrar parameters *(continued)*

Registrar Parameters
`RegSameCallID=<mode>` When Yes is set (default), the same caller ID is always used for SIP registration. Set No to change the caller ID for each SIP registration.
`RegContact=<name or ip addr>` Used in the `Contact:` field.
`RegContactParam=<string>` Sets additional header-parameters in the contact field (for example the q-value: RegContactParam=q=1.0).
`RegUser=<name>` Enter a username for authorization.
`RegPwd=<password>` Enter a password for authorization.
`RegProxy=<ip addr>` Enter an alternative IP address if you want the request to be sent to an address other than the one entered in `RegId`.
`RegExpires=<sec>` Enter the number of seconds registration is to be valid. Default `0` means infinite.
`RegPing=<sec>` Interval (in seconds) for the registrar ping. The CELLX sends an empty UDP packet to the registrar's IP address. The packet is essentially an alive packet to avoid possible firewall problems.
`RegSignalling=<int>` Determines the profile's signaling protocol for registration with the SIP registrar. 1=SIP udp (default) 2=SIP tcp 3=SIP tls
`RegUseReceived=<mode>` Enter Yes when an CELLX appears behind a NAT and STUN cannot be used. Default No..
`RegSameCallID=<mode>` The same call ID is used for SIP registration Enter No to change the call ID for every reregistration. (default Yes).

5.3.5 Radius profiles

Radius profiles are used to connect the CELLX to a Radius server. You can use a Radius server for different destinations and for access and/or accounting. These Radius profiles are then assigned to the VoIP profiles.

Table 5.30 `route.cfg`: radius parameters

Radius Parameters
`[Radius:<name>]` The name of the Radius server profile assigned to one or more VoIP profiles.
`Host=<name or ip addr>` Radius server's host name or IP address. Bear in mind that the DNS service must be active if you enter the host name.

Table 5.30 `route.cfg`: radius parameters *(continued)*

Radius Parameters
`User=<name>` Enter a username for authorization.
`Password=<password>` Enter a password for authorization.
`Secret=<secret>` Enter the shared secret.
`OwnId=<name or ip addr>` Host name or IP address used in the NAS identifier or NAS IP address (Cisco VSA gateway ID).
`ServiceType=<num>` As defined in RFC 2865, Chapter 5.6.
`RequestTimeout=<sec>` Number of seconds during which the request is repeated if the Radius server does not respond.
`RequestRetries=<count>` Number of packet retries sent at one time.
`StopOnly=<mode>` When **`yes`** is entered, only Accounting Request Messages with the status type **`stop`** are transmitted to the Radius server.
`AlwaysConnected=<mode>` Enter **`No`** (default) to set the value for the field **`ConnectedTime`** to that of the field **`DisconnectedTime`** in accounting-stop messages when the call was not connected.
`CallingStationId=<num>` This parameter is used to set the calling station ID. The default setting is the OAD, but you can define any calling station ID. To define a partial calling station ID, enter a **`?`** for each digit. For example, **`CallingStationId=???`** will consist of the first three digits of the OAD.
`CallType=<int>` Enter one of the following to define the call type: 3 = VoIP and telephony 2 = VoIP only 1 = Telephony only
`FramedProtocol=<int>` Enter one of the following to define the framed protocol (see RFC 2865, Chapter 5.7): 1 = PPP 2 = SLIP 3 = AppleTalk Remote Access Protocol (ARAP) 4 = Gandalf proprietary SingleLink/MultiLink protocol 5 = Xylogics proprietary IPX/SLIP 6 = X.75 Synchronous
`NasId=<string>` The string entered is used as network access server identifier attribute in access requests. If no string is entered, the attribute will not be set (default).

6 Routing examples

6.1 CELLX integration in an H.323 carrier network

In the following example, an CELLX32 is integrated in a carrier network via H.323. The system contains six SIM cards for each mobile channel, and the SIM 24 Carrier is used. All calls coming from VoIP are routed to the mobile network. Four VoIP Modules with 16 media channels each are attached in the system. H.323 is used as the signaling protocol and a gatekeeper is used in the VoIP network. Because the gatekeeper assigns and authorizes the peer, only one VoIP profile is necessary. Since the peers may use various compression algorithms, you can define several if you so choose. The codec with the highest priority is G.729. If the peer does not support it, G.726 32Bit/sec, G.711a, G.711u are also possible. Silence suppression is active. The gatekeeper's IP address is 192.168.0.10. This gatekeeper profile can handle up to 30 simultaneous VoIP calls. This value is dynamic and changes depending on the number of active SIM cards. The CELLX's alias is iGATE01. The prefix list is 01555 01556 01444 01445. The gatekeeper's alias is GK1 and no password is used. Calls with the prefixes 01555 and 01556 are sent to the carrier with the LAIN 26212 at controllers 0-15. Calls with the prefixes 01444 and 01445 are sent to the carrier with the LAIN 26313 (controllers 16-31). Digit collection is activated, so that incoming calls with overlap dialing are not transmitted until the number is complete or a wait timer (5 seconds) has run out. The `NEXT` parameter makes sure that calls are distributed evenly to the individual mobile channels in the trunk group. The parameter `CHADDR` ensures that calls are not misrouted, since the controller definition changes to the SIM card's LAIN when a SIM card is mistakenly used for another mobile controller. Problems can occur when SMS messages are also sent, as service center numbers are definitively configured. The parameter `LIMIT` is set so that the system automatically switches to the mobile controllers' SIM cards when the active SIM card has been used for 3600 seconds. The parameter `CONTINUE` makes sure the mobile channel switches to the first SIM card after the limit has been reached on the last SIM card. The SIM card will not switch until currently active calls have been disconnected.

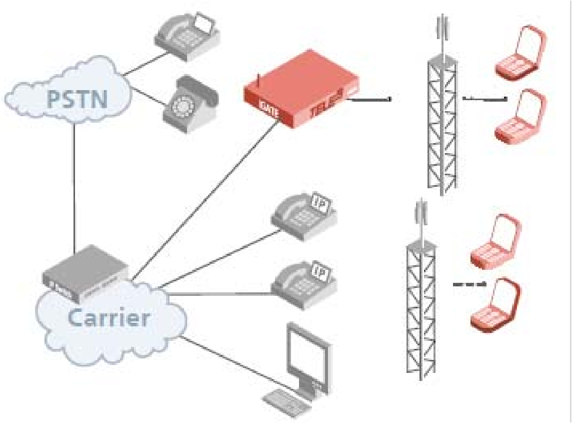


Figure 6.1 CELLX integration with SIM card switching in an H.323 carrier network

Example 6.1 CELLX integration with SIM card switching in an H.323 carrier network (`pabx.cfg`)

```
Subscriber00 = TRANSPARENT ROUTER GSM[0000,00000,+00000,1,1,1,SIM24] CHADDR
LIMIT[3600,3600,3600,3600,3600,3600] CONTINUE ALARM NEXT
Subscriber01 = TRANSPARENT ROUTER GSM[0000,00000,+00000,1,1,1,SIM24] CHADDR
LIMIT[3600,3600,3600,3600,3600,3600] CONTINUE ALARM
....
Subscriber34 = TRANSPARENT ROUTER SWITCH CHMAX[16] ALARM
Subscriber35 = TRANSPARENT ROUTER SWITCH CHMAX[16] ALARM
ChargeUnitGenerate=1
LimitWODisc=ON
```

Example 6.2 CELLX integration with SIM card switching in an H.323 carrier network (`route.cfg`)

```
[System]
DTMFWaitDial=5

MapAll01555=|2621201555<<17
MapAll01556=|2621201556<<17

MapAll01444=|2621301444<<17
MapAll01445=|2621301445<<17

[Voip:DF]
VoipDirection=In
VoipPeerAddress=10.0.0.0
VoipIpMask=0xffff0000
VoipSignalling=0
VoipCompression=g729 g72632 g711a g711u
VoipSilenceSuppression=Yes
VoipMaxChan=30
VoipTxM=2
VoipGk=GK1

[Gatekeeper:GK1]
RasPort=1719
OwnRasPort=1719
RasId=iGATE01
RasPrefix=01555 01556 01444 01445
GkId=GK
GkAdd=192.168.0.10
GkPwd=
GkTtl=300
GkMaxChan=30
GkDynMaxChan=Yes
```

6.2 CELLX as a second-generation LCR with VoIP

In the following example of a PBX connection, all mobile calls are terminated through the mobile channels. Eight mobile channels form a group for one mobile network. One SIM card is available on each mobile channel. Digit collection is activated, so that incoming calls with overlap dialing are not transmitted until the number is complete or a wait timer (5 seconds) has run out. The `NEXT` parameter makes sure that calls are distributed evenly to the individual mobile channels in the trunk group. If all of a carrier's SIM cards are busy, rerouting (`redirect3`) via PSTN is automatically initiated. All international calls are terminated to VoIP (40). The system contains two VoIP Modules, for a total of 32 media channels. The VoIP carrier profile `DF` and the SIP protocol are used. National calls are routed through the carrier with the prefix 010xx. All other calls are sent to the PSTN unchanged. All calls from the PSTN or from a VoIP carrier are sent directly to the NT controller, to which the PBX is attached. All incoming calls from the mobile networks are routed to the PBX's central number (001). For the

VoIP profile `DF`, the system uses the registrar `reg` and registers with `user@sip-carrier.de`, username `user` and password `pwd`. SIP UDP is used for signaling. A maximum of 30 media channels with the G.729 codec can be used. The Peer is `sip-carrier.de`.

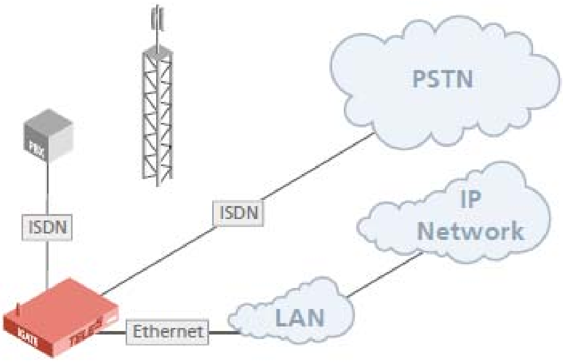


Figure 6.2 CELLX as a second-generation LCR with VoIP

Example 6.3 CELLX as a second-generation LCR with VoIP (`pabx.cfg`)

```
Subscriber00 = TRANSPARENT ROUTER GSM[0000,00000,+00000,1,1,1,SIM4] CHADDR ALARM
NEXT
Subscriber01 = TRANSPARENT ROUTER GSM[0000,00000,+00000,1,1,1,SIM4] CHADDR ALARM
....
Subscriber08 = TRANSPARENT ROUTER GSM[0000,00000,+00000,1,1,1,SIM4] CHADDR ALARM
NEXT
Subscriber09 = TRANSPARENT ROUTER GSM[0000,00000,+00000,1,1,1,SIM4] CHADDR ALARM
....
Subscriber16 = TRANSPARENT ROUTER ALARM
Subscriber17 = TRANSPARENT ROUTER ALARM
Subscriber18 = TRANSPARENT ROUTER SWITCH CHMAX[16] ALARM
Subscriber19 = TRANSPARENT ROUTER SWITCH CHMAX[16] ALARM
```

7 Mobile configuration options

7 Mobile configuration options

7.1 Network-specific mobile routing

7.1.1 Routing decisions for calls to the mobile network

Internal routing decisions

Usually, routing decisions are made internally by the gateway. Calls to the mobile network are routed based on the first digits of the called destination number (mobile network access number) via a port or LAIN. The mapping looks like this:

`MapAll<mobile network access number>=<port/LAIN><mobile network access number>`

In the example shown below, calls to the mobile network access number 0172 are routed via port 20.

Example 7.1 Internal routing decision 1

```
MapAll0172=200172
```

In the example below, calls to 0176 are routed via the LAIN 26227. To route via LAIN, the keyword `CHADDR` needs to be added to the mobile port's `Subscriber` configuration.

Example 7.2 Internal routing decision 2

```
MapAll0176=262270176
```

External routing decisions

In some cases, the routing decision needs to be made before the call arrives on the gateway. This can be done by adding a technical prefix to the called number.

Each prefix represents a national mobile network. For instance 7777# could be the prefix to route calls to the LAIN 26222, 8888# could route calls to the LAIN 26227.

If a call to the mobile network arrives on the gateway containing a technical prefix, this prefix needs to be removed in the `MapAll` parameter and replaced by the port or LAIN. The configuration is:

`MapAll<technical prefix>=<port/LAIN>`

The mapping can be extended to contain the mobile network access number.

`MapAll<technical prefix><mobile network access number>=<port/LAIN><mobile network access number>`

In the following example, calls containing the prefix 7777# are routed via port 20. Calls containing the prefix 8888# are routed via port 21.

Example 7.3 External routing 1

```
MapAll7777#=20
MapAll8888#=21
```

In the following example, calls which come in with the prefix 9999# and the mobile network access number 0176 are routed via the LAIN 26227 to 0176.

Example 7.4 External routing 2

```
MapAll9999#0176=262270176
```

7.1.2 Using the LAIN as the mobile port address

Use the LAIN as controller with the CHADDR parameter to prevent logging onto the wrong SIM card. This will ensure that routing is network specific. The following example is based on the German country code. One carrier's LAIN is 26212 and the other carrier's LAIN is 26213.

Example 7.5 Using the LAIN as the mobile port address (`pabx.cfg`)

```
...
Controller00=20 GSM
Controller01=20 GSM
Controller02=20 GSM
Controller03=20 GSM

Controller04=20 GSM
Controller05=20 GSM
Controller06=20 GSM
Controller07=20 GSM
...
Subscriber00=TRANSPARENT ROUTER GSM[0000,00000,+49556,1,1,1,SIM4] CHADDR ALARM
Subscriber01=TRANSPARENT ROUTER GSM[0000,00000,+49556,1,1,1,SIM4] CHADDR ALARM
Subscriber02=TRANSPARENT ROUTER GSM[0000,00000,+49556,1,1,1,SIM4] CHADDR ALARM
Subscriber03=TRANSPARENT ROUTER GSM[0000,00000,+49556,1,1,1,SIM4] CHADDR ALARM

Subscriber04=TRANSPARENT ROUTER GSM[0000,00000,+49555,1,1,1,SIM4] CHADDR ALARM
Subscriber05=TRANSPARENT ROUTER GSM[0000,00000,+49555,1,1,1,SIM4] CHADDR ALARM
Subscriber06=TRANSPARENT ROUTER GSM[0000,00000,+49555,1,1,1,SIM4] CHADDR ALARM
Subscriber07=TRANSPARENT ROUTER GSM[0000,00000,+49555,1,1,1,SIM4] CHADDR ALARM
...
```

If you remove the keyword `CHADDR` from the `pabx.cfg`, you must restart the system. Controllers belonging to the same trunk group must have the same address. You must delete all routing entries based on port addresses when using the LAIN as controller.

Example 7.6 Using the LAIN as the mobile port address (`route.cfg`)

```
...
MapAll01555=2621201555
...
MapAll01556=2621301556
...
```

7.1.3 Fixed LAIN for a mobile port

Enter `CHADDR[<addr>]` to remove a mobile controller belonging to an LAIN group from the standard routing process (for example for specific routes or only for SMS transmission). The port address can be set to `<addr>`.

Example 7.7 Fixed LAIN for a mobile port

```
Subscriber05=TRANSPARENT ROUTER GSM[0000,00000,+49555,1,1,1,SIM4] CHADDR[444]
ALARM

MapAllSMS=444 DATA
```

The value entered for <addr> may not exceed 6 digits.

7.2 Incoming voice calls from mobile

Incoming mobile calls (service indicator 01 represents voice calls) can be routed to a specified number. This enables each mobile controller to receive a unique identifier. It will then be mapped to a number.

Example 7.8 Incoming voice calls from mobile 1

```
Restrict20=90123 01
Restrict21=91234 01
```

The mobile controllers can also have the same identifier, so that all voice calls (service indicator 01) from controller 20 are sent to number 1111 at port 9. This number could, for example, serve a call center.

Example 7.9 Incoming voice calls from mobile 2

```
Restrict20=91111 01
```

7.3 Blocking ports

This function allows you to block a port, so that the corresponding mobile channel is omitted from the distribution of calls. The function is particularly useful when mobile channels fail or SIM cards cannot be immediately replaced.

To disable the port for outgoing calls enter the keyword `CHINC[01,01]` in the `Subscriber` line and restart the system.

In the following example, port 04 is blocked for outgoing calls.

To enable the port for outgoing calls, remove the entry and enter Activate Configuration.

Example 7.10 Blocking ports

```
...
Subscriber04=TRANSPARENT ROUTER [0000,00000,<SMSC>,1,1,1,SIM4] CHADDR ALARM
CHINC[01,01]
...
```

Incoming calls are always possible. The status is not displayed in the GATE Manager.

7.4 Mobile user PBX callback

When the CELLX is implemented in a corporate network and connected to a PBX or between a PBX and the outside line, the following configuration entry activates a feature, that uses a mobile caller’s OAD to connect with the last PBX extension the caller unsuccessfully dialed.

`DialBack=<hours>`

The callback list is active for the number of hours entered.

`DialBackMinutes=<minutes>`

The callback is active for the number of minutes entered. After that time, the callback record is deleted in the system.

`DialBackStopOnConnect=<mode>`

Deletes (`On`) the dialback record once a dialback call is connected. `Off` deactivates this behavior.

`DialBackCompare=<digits>`

Shows the number of digits at the end of a phone number that are compared during dialback handling. The default value is 10. Comparison of the end part of numbers is necessary when the same number is displayed differently (when, for instance, one version includes the country code, but the other doesn’t).

For calls coming from VoIP, add the following parameter to the used VoIP profile:

`VoipOadIn=<name of profile used for callback>:`

Enter the name of the VoIP profile that you use to call back the extension.

In the following example, the callback list is active for the previous five hours. The German country code is used for the LAINs. All calls with the prefixes 1111 and 2222 are terminated through the carrier with the LAIN 26212. Calls with the prefix 3333 are terminated through the carrier with the LAIN 26213.

Example 7.11 Mobile user PBX callback

```
DialBack=5

MapAll1111=262121111
MapAll2222=262122222
MapAll3333=262133333
```

Make sure that no `Restrict` entries are configured for these mobile controllers.

7.5 Optional mobile quality parameters

The following parameters can be set for specific carriers if SIM cards for different mobile networks are used and different configurations must be set.

For all controllers without a LAIN, the following default setting applies: `GSM=`

Bear in mind that for iGATE UMTS you must use the keyword UMTS= instead of GSM=! The following syntax applies: `UMTS1=<Lain1> ALFI[<val1/val0>]...`

Up to four additional carrier-specific entries may be set:

```
GSM1=<Lain1> ALFI[<val1/val0>]...
GSM2=<Lain2> ...
GSM3=<Lain3> ...
GSM4=<Lain4> ...
```

Up to four LAIN-specific configurations are possible. The user must enter the index (1 to 4). The LAIN appears as the first entry behind the equal sign. Everything else follows.

There is no automatic default behavior for this feature! All options must always be entered. We recommend that you enter GSM= settings, as all controllers without a LAIN or with a default address use these options!

The following table describes specific signaling and quality parameters for configuration of the mobile interface.

Table 7.1 Optional mobile parameters

GSM=... (for iGATE GSM) or UMTS= ... (for iGATE UMTS)	

Enter any of the following parameters after the equal sign for the following functions. Entries may appear in any order, but all entries must appear in the same line and in double-digit notation as follows:
`GSM=RSSI[10] STOP[18,08] ANNOUNCE[00,08] FAX[a2] ASR[20,35]`
or
`UMTS=RSSI[10] STOP[18,08] ANNOUNCE[00,08] FAX[a2] ASR[20,35]`	

`ALERT[<sec>]`	Set this parameter to generate an alert signal in the D channel immediately after the dial-end signal. If you enter optional square brackets containing a number of seconds, the alert signal will occur when the number entered has passed following setup.

Table 7.1 Optional mobile parameters *(continued)*

GSM=... (for iGATE GSM) or UMTS= ... (for iGATE UMTS)	

`ANNOUNCE`	Set this parameter to define what happens when a recorded announcement is detected:
	No `ANNOUNCE` entry (default)
	A D-channel PROGRESS message stating Inband Information Available will be generated
	`ANNOUNCE[<cause>]`
	The connection will be rejected with the defined ISDN cause value. Do not enter the cause value ff!
	`ANNOUNCE[00,<sec>]`
	A timeout for voice detection is defined in seconds (default value: 120 seconds). After the interval entered has passed, the connection is torn down.
	`ANNOUNCE[ff]`
	The call will be connected as soon as an announcement is detected.

`FORWARD[<cause>]`	**For GE864-QUAD modules only** The connection will be rejected with the defined ISDN cause value when call forwarding is detected. NOTE: The GSM network must support this feature for it to work.
`ASAL`	Announcement Stop during ALert: Set this parameter to release calls if the GSM network sends an alert message, after which voice is detected (for example the B-party rejects the call, which is then forwarded to the B-party's voicemail following the ring state). The parameter ANNOUNCE[<cause>] must be set to activate this parameter.
`MINV[<count>]`	Minimum Voice detections: Parameter used with ASAL to define the number of voice detections in a row before the call is released (for example GSM-network background noise detected as voice). The default value is 1, but 2 is recommended. The parameters ANNOUNCE[<cause>] and ASAL must be set to activate this parameter.
`ASR[<limit>, <calls>]`	Allows you to change the default value (40 calls at 30% ASR). For a definition of ASR, see Chapter 11.4.1 Generating and retrieving CDRs.
`CDC[<seconds>, <count>]`	Call duration check. When more than (<count>) connected calls in a row occur with a call duration lower than <seconds>, an alarm is generated and the port is restarted.
`FAX[<cause>]`	This entry allows you to reject fax calls with the defined cause value.
`RSSI[<limit>]`	Configure this parameter to set a limit for the reception field strength. When the reception field strength falls below this limit, the mobile channel will be blocked. If the field strength is above the limit, the mobile channel will log on with the mobile carrier again. The values used are 0 to 31, which represent the following field strengths: -113dBm to -51 dBm. An error is generated in the `protocol log`. The result must be divided by 2. EXAMPLE: To define a field strength of -95 dBm, subtract -95 from-113 and divide the result by 2: - 113dB - (-95dB) = -18dB / 2 = 9 Enter `RSSI[9]`

Table 7.1 Optional mobile parameters *(continued)*

GSM=... (for iGATE GSM) or UMTS= ... (for iGATE UMTS)	
`STOP[<val1>, <val2>]`	This entry allows you to define a maximum number of connection setups that always result in a recorded message (<val1>) without an alert detection or call-connected signal. The second value (<val2>) counts calls that connect immediately without an Alert detection. The mobile port is blocked when the defined value is reached and an entry is recorded in the log file (...Err: Voice). In this way inactive SIM cards that are forwarded to a recording (with or without a connect from the mobile carrier) can be detected and blocked so that they are removed from the routing process. The default status of this function is off.
`NOCP`	When this option is configured and the call is from ISDN to GSM, the Call Proceeding signaling message will be eliminated from signaling. This may be necessary if the ISDN peer does not support Call Proceeding. Bear in mind that the peer's Setup Ack Timer is usually set at 5 seconds, which means that an Alert must be generated as follows: `GSM=ALERT[5]` (for iGATE GSM) `UMTS=ALERT[5]` (for iGATE UMTS)
`FMIN[xxx] FMAX[yyy]`	Sets the frequency range for ringtone detection (default 400-444 Hz). EXAMPLE: `GSM=FMIN[400] FMAX[444]` (for iGATE GSM) `UMTS=FMIN[400] FMAX[444]` (for iGATE UMTS)
`ALFI[<val1/val0>]`	GSM module GE864-QUAD only. Switches the digital filter on (val=1) the Telit module to suppress frequencies below 250 Hz per LAIN during alerting. (switch off val=0) Please use the parameter to enable alert-tone detection for dual tone LAINs only (please see Chapter 7.5 Optional mobile quality parameters).
`VOICE[<sec>]`	The voice channel in the switching matrix is connected after <sec> (default immediately). The voice channel is activated no later than the point at which the call is connected.
`WND9`	Voice detection is activated in the GSM module if the GSM network signals WND9 (voice). If this parameter is not set, voice detection is activated immediately when the GSM call is initialized. Module Q24xx only.
`WND2A`	Generates Alert when WIND:2 is received. Module Q24xx only.
`WND2N`	Suppresses voice detection after WIND:2 is received. Module Q24xx only.
`MINA[<count>]`	Defines the number of alert tones that are required before an Alert is detected.

Table 7.1 Optional mobile parameters *(continued)*

GSM=... (for iGATE GSM) or UMTS= ... (for iGATE UMTS)	

`REDIAL[<percent>,`
`<waitmin>,`
`<waitmax>,`
`<callingmin>,`
`<callingmax>,`
`<connectmin>,`
`<connectmax>,`
`<cfg>]`
	Repeats a call to the same B-party number as soon as the A party hangs up. This parameter is useful for improving call statistics:
	▪ **Percent**: Percentage of calls (including alert only) through the gateway that are redialed.
	▪ **Waitmin**: Minimum number of seconds the gateway waits before redial.
	▪ **Waitmax**: Maximum number of seconds the gateway waits before redial.
	▪ **Callingmin**: Minimum number of seconds the gateway waits before call setup for redial. We recommend no more than 8 seconds.
	▪ **Callingmax**: Maximum number of seconds the gateway waits before call setup for redial. We recommend no less than 15 seconds.
	▪ **Connectmin**: Minimum connect time (in seconds) if call is connected.
	▪ **Connectmax**: Maximum connect time (in seconds) before call is disconnected.
	▪ Cfg: Enter a value 1 – 4; valid for the `1=` line to `4=` line for the iGATE GSM, or for the `1=` line to `4=` line for the iGATE UMTS.

`CPSA[0|1]`	When `CPSA` ist set to 0, the GSM port is forced to register the SIM card to it's home network, which is identified by the first 5 or 6 digits of the IMSI. This may result in a faster registration process. `CPSA[1]` is the default value.

7.6 Deactivating mobile rerouting

Use the following entries in the `pabx.cfg` to deactivate rerouting for rejected calls.

Table 7.2 Deactivating mobile rerouting

Parameters
AllClassNext=<mode> Enter AllClassNext=Off to deactivate rerouting for all rejected calls, regardless of the cause value (default not configured). NOTE: You cannot configure both AllClassNext and Class2Next simultaneously!
`Class2Next=off` Enter `Class2Next=Off` to deactivate rerouting for calls rejected with a class 2 cause value (default not configured). NOTE: You cannot configure both AllClassNext and Class2Next simultaneously!

`SelfNext=<int>`
Set this parameter to reroute a failed call through the same port. <int> defines the number of call attempts. Recommended values for <int> are 2 or 3. Bear in mind that this parameter must be used in conjunction with the parameter `AllClassNext=Off.` Default not configured.
EXAMPLE:
`SelfNext=3`
`AllClassesNext=Off`

7.7 Setting autodial

When the following configuration is set, the gateway calls a defined list of phone numbers automatically. Save the script in a text file with the extension *.ad.

The autodial process starts automatically when the script file has been copied to `/boot` or `/data`. The file extension is renamed *.ax and is deleted when process has ended. A log file *.al is generated in the course of the process. The file contains whether the call was successful, the number called, the call duration for successful calls, and the cause value for failed calls. To abort the process, simply delete the *.ax file.

The following settings are possible:

- `OPTION CALLDURATION`: Enter an interval in seconds to define the length of the call.
- `OPTION CALLRETRIES`: Enter the number of call attempts for each number.
- `OPTION CONFIGURATION`: Refers to the number of the = configuration.
- `OPTION NOALARM`: Set this option to suppress sending alarms when connected to a vGATE.
- `D<sec>`: Enter a value to define the duration for calls to this number.
- `P<sec>`: Enter a value to define the pause between calls.

Example 7.12 Setting autodial

```
OPTION CALLDURATION 155-170
OPTION CALLRETRIES 2
OPTION CONFIGURATION 1
OPTION NOALARM
017222921 D10 P30
017222922 D10 P30
017222923 D10 P30
017222924 D10 P30
```

7.8 Disconnecting calls after ring

When the following parameter is configured, the call will be disconnected once the number of configured seconds have passed after the mobile network has sent the first ring:

`AWSTime=<sec>`

7.9 Checking ports/mobile channels

Monitoring ASR for mobile ports

ASR monitoring of the last 40 calls occurs for all mobile ports. If the ASR (ASR2) is lower than 30 percent, an alarm is generated at the corresponding port and the port is blocked. The port is then restarted and a corresponding entry appears in the protocol.log file (ASR). The port is then unblocked.

The following entry in the `pabx.cfg` causes the mobile port to block automatically when this error occurs three times in a row:

`ASRBlock=On`

When ASRBlock=Off is used, the port will be restarted and will remain open.

The following parameter in the `pabx.cfg` file allows you to change the default value (30% for 40 calls):
`GSM=ASR[<percent>,<number of calls>]` (for iGATE GSM)

`UMTS=ASR[<percent>,<number of calls>]` (for iGATE UMTS)

Example 7.13 Monitoring ASR for mobile ports (iGATE GSM)

```
GSM=ASR[20,35]
```

Example 7.14 Monitoring ASR for mobile ports (iGATE UMTS)

```
UMTS=ASR[20,35]
```

7.10 Defining special characters for voice calls

In cases in which the called number includes special characters (for example * or #), it may be necessary to define the call type used in the mobile network (command or voice call). Calls to GSM or CDMA that begin with * or #, are sent as command calls by default. For voice calls beginning with * or #, you must define the call type voice in the mapping entry with a > sign.

The routing entry will look like this:

`MapAll<num>=<LAIN>><num>`

Example 7.15 Defining special characters for voice calls

```
MapAll222=11111>*222
```

8 Signaling and routing features

8.1 Least cost routing

CELLXs are connected between the customer’s private branch exchange (PBX) and the public telephone network (ISDN) and/or VoIP. The customer saves connection charges and can effortlessly and automatically connect to the corporate network as needed using one of six routing methods:

- Carrier selection
- Dedicated lines
- Direct line access with subaddressing
- Direct line access with DTMF
- Callback with subaddressing
- Callback with DTMF

This manual contains information only on carrier selection. If you would like to configure any other variation, please contact TELES or refer to the TELES Infrastructure Systems Manual Version 4.5, Chapter 3.

Calls are routed transparently for the PBX and its users. CELLXs can generate charges and route calls using alternate settings in case of network failures. The provider can access the system via ISDN for routine maintenance and monitoring.

The following additional services are supported by this feature package:

- Generation of charges
- Time-controlled configuration
- Alternative routing

8.1.1 Carrier selection

Carrier selection is currently one of the most commonly used routing methods supported by the CELLX. In the CELLX, this routing process also includes direct calls into the mobile network or through a VoIP network. That means the system is a full-fledged second generation LCR.

8.1.1.1 Routing entries

Use the MapAll command to route calls using Carrier Selection.

Use the following syntax for connections routed via the provider:

```
MapAll<AreaCode>=9<CarrierSelection><AreaCode>.
```

where `<AreaCode>` is the number or number range to be routed and `<CarrierSelection>` is the access number required to reach the provider’s network.

For unrouted connections (placed via the public telephone network), use:

```
MapAll<AreaCode>=9<AreaCode>.
```

To block undesired carrier selection prefixes use:

```
MapAll<CarrierSelection>=&91;(Busy signal)
```

In the following example, calls to international destinations are terminated through the VoIP interface. The profile names iG1 and iG2 in the routing entries refer to different VoIP carriers. Calls to the mobile network (01555 and 01556) are routed directly through SIM cards for the corresponding mobile carriers (LAIN 26212 and 26213). All other national long distance and local calls are routed through an alternative carrier (01019). All calls from the PSTN to the PBX are put through transparently.

Example 8.1 Carrier selection routing entries

```
MapAll001=40iG1:001
MapAll0044=40iG2:0044
...
MapAll01555=2621201555
MapAll01556=2621301556
...
MapAll01=90101901
MapAll02=90101902
...
MapAll09=90101909

MapAll1=9010191
MapAll2=9010192
...
MapAll9=9010199

Restrict9=10
```

Be sure to enter phone numbers in the routing file in ascending order.

8.1.2 Alternative routing settings

Alternative routing refers to the ability to establish connections using a different (alternative) network in case of provider failure (for example all mobile controllers are in use). Alternative routing ensures uninterrupted operation of the attached PBX. In such cases, connections are often made via the public network using the Redirect command.

```
MapAll<num>=<port><num>
Redirect3<port><num>=<placeholder>
MapAll<placeholder>=<alt port><num>
```

Example 8.2 Alternative routing settings

```
MapAll01555=2621201555
Redirect32621201555=A
MapAllA=901555
```

8.1.3 Charge models

CELLXs can either generate charge information or transmit received charges from the public or corporate networks to the attached PBX. Charge simulation is achieved using variables, which ensure a great degree of flexibility for the implementation of many different charge models including:

- Charge units per time unit
- Flat rate (initial charge without time interval)
- Initial charge plus time interval
- Initial charge plus time interval after delay
- Time interval and/or flat rate plus received charges
- Received charges only or no charge information
- Initial toll-free period with retroactive charge generation afterwards
- Price-per-minute (with whole second accuracy)

In this chapter, **unit** means that charge information is transmitted as a whole-numbered value, and **currency** means that the charge information is sent as a currency amount (for example EUR 3.45). The charge impulse generation options can be set for each mapping by adding charge-specific arguments to the `MapAll` commands as shown below. The use of each variable is explained in Table 8.1.

```
MapAllsrc=dst mode time start/wait.
```

Table 8.1 Charge variables

Variable	Purpose
`time`	Determines the length of each time interval (how long each unit lasts). The value is entered in seconds and hundredths or thousandths of a second (the maximum value accepted is 655.35 seconds, 65.535 if thousandths are entered). If `time` is set to zero or not present no charges are generated, external charge information is passed through if received.
`start`	Optional. Sets the initial unit level. Enter a value between 0 and 127 whole units. If you want to use a flat rate, set the desired number of units here and set the wait to 255 to turn off the time interval.
`wait`	Optional. Determines the delay after which charge generation begins. Once this time has elapsed, charge impulses are sent in the interval determined with time. Enter a value between 0 and 254 seconds. 255 deactivates the charge pulse. In this case, the time variable is ignored.

Any external charges can be added to the generated charges by adding 128 to the *start* value. (The value range for the initial unit level is still set from 0 to 127). The maximum supported number of units per connection is 32767 units.

Additional adjustments may be made to allow for the implementation of new charge models.

- When charge information is sent as Currency, values can be expressed in thousandths for greater precision in charge calculation.
  For the internal Layer 3 protocols, charges can be specified to the third decimal place (thousandth) using the /Value option (Example: /Value:1.056). In this fashion, charges can be generated for units of currency requiring accuracy to the third decimal place or

for fractions such as tenths of a cent. This allows for greater flexibility in the transmission of charges to terminal devices. In order to make use of this option, connected devices must support “AOC-D Currency”. In the current version, this option is only available for the DSS1 protocol.

- A multiplication factor can be specified for received or generated charges.
  During the charge generation process, each charge unit is multiplied by a preset factor. This factor appears in the mapping entry after the time and start/wait variables (`MapAllsrc=dst mode time start/wait*factor`).
  Each unit, for example, can be converted to 12 cents. The following example illustrates the use of this feature.

In this example, all received charge units are multiplied by 12 and passed on. If AOC-Currency is set on the internal port, each unit appears as 12 cents. The multiplication factor is also used to implement two new charge models:

- If the factor value exceeds 128, this marks the use of an initial toll-free phase followed by retroactive charge generation.
- If the multiplication factor is set to 255, a “minute price” is used in place of the time variable.

Example 8.3 Charge model

```
...
MapAll1=91 1 128/255*12
...
```

8.2 Online traffic monitor

The Online Traffic Monitor allows you to collect and monitor statistics and call detail records (CDRs). The following functions are possible with this feature package:

- ASR calculation
- Generation of CDRs
- Generation of online CDRs using e-mail

8.2.1 ASR calculation and resetting statistic values and counters

When these functions are configured in the `pabx.cfg` file, the following statistic values are calculated for the entire system at a defined time and are copied into a file in the following order:

- Day and time of entry
- System name
- Number of connected calls followed by the number of total calls in parentheses
- Number of minutes termintated
- ASR1: ratio of total calls to connected calls disconnected by the A party
- ASR(ext): external ASR for the traffic source
- ASR(int): internal ASR for the CELLX
- ACD: average call duration

The following example shows how the statistics appear in the file into which they are copied.

Example 8.4 Statistics file

```
26.10.04-00:00:00,iGATE810000: Calls: 19351 (29716) - Minutes: 46647 - ASR1:
65.12% -  ASR(ext): 65.12% - ASR(int): 65.30% - ACD: 144.63s
```

Set the time using the following syntax: `<hh:mm>`. If ?? appears instead of a specified hour (for example `??:mm`), the ASR is written into the asr.log file once every hour.

You can set `<day>` to apply for days of the week or for one specific day of the month as follows:

Use a bitmask to set the weekdays on which the configuration applies here. The daymap appears in the following order: HoSaFrThWeTuMoSu. For example, enter 00000001 if the configuration is to apply every Sunday.

To set the configuration to apply on one specific day every month, enter the day of the month followed by a full stop. For example, enter 15. if the configuration is to apply on the 15th of every month.

If the configuration is to apply every day, do not set `<day>`.

Do not configure both StatisticTimeReset and StatisticTime or StatisticTimeReset and StatisticCounter together.

8.2.1.1 Saving and sending statistics

The statistics are copied into a specified file. This information can also be sent to an e-mail or SMS recipient. The following syntax must be used:

```
StatisticTime=/data/asr.log <hh:mm> <day>
StatisticTime=/data/asr.log <hh:mm> <day> @<email address>
StatisticTime=/data/asr.log <hh:mm> <day> @SMS<mobile number>
```

In the following example, the system's statistic values are saved every Wednesday into the file `asr.log` and sent to an e-mail account.

Example 8.5 Saving and sending statistics 1

```
StatisticTime=/data/asr.log 00:00 00001000 @info@teles.de
```

In the following example, the system's statistic values are saved at midnight on the first of every month into the file `asr.log` and sent to an SMS recipient.

Example 8.6 Saving and sending statistics 2

```
StatisticTime=/data/asr.log 00:00 01. @SMS01234567890
```

Do not configure both `StatisticTimeReset` and `StatisticTime` together.

8.2.1.2 Saving statistics

The statistics are copied into a specified file and the counters (A-F) are reset. The following syntax must be used:

StatisticTimeReset=/data/asr.log <hh:mm> <day>

In the following example, statistics will be saved every hour on the hour of every day and the statistic counters will be reset to 0.

Example 8.7 Saving statistics 1

```
StatisticTimeReset=/data/asr.log ??:00
```

In the following example, the system's statistic values are saved at noon on the 15th of every month into the file asr.log.

Example 8.8 Saving statistics 2

```
StatisticTimeReset=/data/asr.log 12:00 15.
```

ⓘ Do not configure both StatisticTimeReset and StatisticTime or StatisticTimeReset and StatisticCounter together.

8.2.1.3 Resetting statistic counters

The following setting in the pabx.cfg resets the statistic counters and unblocks the controller group. The following syntax must be used:

ResetCounter=<port> <hh:mm> <day> or

ResetCounter=<LAIN> <hh:mm> <day>

Up to five entries for as many groups and/or times are possible. Bear in mind that StatisticTimeReset must be inactive to avoid repetition. The mapping to a LAIN requires that in the pabx.cfg file, the CHADDR parameter is added to the Subscriber line, to change the controller definition to the SIM card's LAIN.

In the following example the counters will reset for LAIN group 26211 every day at midnight, for the group 26212 only once on the first day of the month, and for 26213 each Saturday at 8:00pm.

Example 8.9 Resetting statistic counters

```
ResetCounter = 26211  00:00 11111111
ResetCounter = 26212  00:00 01.
ResetCounter = 26213  20:00 01000000
```

Do not configure both StatisticTimeReset and StatisticCounter together.

8.2.2 Generating and retrieving CDRs

With the `Log` and `failedlog` commands, you save CDRs and unconnected calls in the CELLX.

For these parameters (`Log` and `failedlog`), a folder and file name must always be specified after the equal sign. The function is not active (no data is recorded) until a file name is specified.

Example 8.10 Generating and retrieving CDRs

```
Log=/data/cdr.log
failedlog=/data/failed.log
```

With recording of files, system maintenance increases. You have to be sure to download or delete files and ensure that there is enough disk space left on the hard drive.

The service indicator listed in the call log and missed calls list describes the type of connection as a four digit hexadecimal number. The coding is conducted according to the 1TR6 standard. A few frequently used values are listed below.

Table 8.2 Service indicator list

Service indicator	Description
0101	ISDN-telephony 3.1 kHz
0102	analog telephony
0103	ISDN-telephony 7 kHz
0200	Fax group 2
0202	Fax group 3
0203	Data via modem
0400	Telefax group 4
0500	SMS or BTX (64 kbps)
0700	Data transfer 64 kbps
1001	Video telephone – audio 3.1 kHz
1002	Video telephone – audio 7 kHz
1003	Video telephone – video

For detailed information on how to automatically divide the files (for example on a daily basis), please refer to the Chapter 5.2.1.2 Log files.

8.2.2.1 Call log

The following entry in the `pabx.cfg` configuration file activates the capability to generate CDRs in the CELLX:

```
Log=/data/cdr.log
```

The cdr.log file is stored in the data directory. New entries are always added to the end of the file. The file is open only during editing.

Each line represents an outgoing call with the following information separated by commas:

Table 8.3 Call log entries

Column	Description
0	Version
1	Start time (format DD.MM.YY-hh.mm.ss)
2	End time (format DD.MM.YY-hh.mm.ss)
3	Source. The following format applies: [node number:automatically set internal channel number]
4	Destination. The following format applies: [node number:automatically set internal channel number]
5	IMSI
6	IP logging signaling: RTP
7	Audio codec used
8	Frame size
9	Service indicator (please see Chapter 8.2.2 Generating and retrieving CDRs)
10	Call duration
11	Cause values
12	Charge from the public line (in units)
13	Charge generated from the system (in units)
14	Cell ID
15	RSSI

Sample log file

The example below shows a sample log file.

Example 8.11 Sample log file

```
V1,25.11.09-10:16:20,25.11.09-
10:16:27,[0000:01]9,[0006:01]111,123456789123451,,,,0102,7,1f,0,,3663,10
V1,25.11.09-10:35:16,25.11.09-
10:35:26,[0000:01]9,[0004:01]111,123456789123452,,,,0102,10,1f,0,3,38922,14
V1,25.11.09-10:38:30,25.11.09-
10:38:41,[0000:01]9,[0004:01]111,123456789123453,,,,0102,11,90,0,3,38922,14
```

Differentiating between ports in the same trunk group

To differentiate between ports with the same number in the CDRs, a specific node number must be defined. You can expand the subscriber configuration line with the keyword `NODE[<num>]` for this purpose. `<num>` can be a string of between 1 and 15 characters:

`Subscriber<xx>=... NODE[<num>]`

In the below formula, `<num>` consists of a four-digit number that is included in the CDR.

Example 8.12 Differentiating between ports in the same trunk group (CDR entry)

```
V1,25.11.09-10:16:20,25.11.09-
10:16:27,[0000:01]9,[0006:01]111,123456789123451,,,,0102,7,1f,0,,3663,10
```

The following example shows the `pabx.cfg` configuration file changed according to the formula.

Example 8.13 Differentiating between ports in the same trunk group (`pabx.cfg` entry)

```
...
Subscriber00=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0001]
Subscriber01=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0002]
Subscriber02=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0003]
Subscriber03=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0004]
Subscriber04=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0005]
Subscriber05=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0006]
Subscriber06=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0007]
Subscriber07=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0008]
Subscriber08=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0009]
Subscriber09=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0010]
Subscriber10=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0011]
Subscriber11=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0012]
Subscriber12=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0013]
Subscriber13=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0014]
Subscriber14=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0015]
Subscriber15=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0016]
...
```

Differentiating between SIM cards

The CDR can contain the IMSI (International Mobile Subscriber Identity), which identifies each SIM card used.

Example 8.14 Differentiating between SIM cards (CDR entry)

```
V1,25.11.09-10:35:16,25.11.09-
10:35:26,[0000:01]9,[0004:01]111,123456789123451,,,,0102,10,1f,0,3,38922,14
```

The following example shows the pabx.cfg configuration file changed according to the formula.

Example 8.15 Differentiating between SIM cards (pabx.cfg entry)

```
...
Subscriber00=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,IMSI,SIM24] CHADDR
ALARM NEXT
Subscriber01=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,IMSI,SIM24] CHADDR
ALARM NEXT
Subscriber02=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,IMSI,SIM24] CHADDR
ALARM NEXT
Subscriber03=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,IMSI,SIM24] CHADDR
ALARM NEXT
Subscriber04=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,IMSI,SIM24] CHADDR
ALARM NEXT
Subscriber05=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,IMSI,SIM24] CHADDR
ALARM NEXT
Subscriber06=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,IMSI,SIM24] CHADDR
ALARM NEXT
Subscriber07=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,IMSI,SIM24] CHADDR
ALARM NEXT
Subscriber08=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,IMSI,SIM24] CHADDR
ALARM NEXT
Subscriber09=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,IMSI,SIM24] CHADDR
ALARM NEXT
Subscriber10=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,IMSI,SIM24] CHADDR
ALARM NEXT
Subscriber11=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,IMSI,SIM24] CHADDR
ALARM NEXT
Subscriber12=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,IMSI,SIM24] CHADDR
ALARM NEXT
Subscriber13=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,IMSI,SIM24] CHADDR
ALARM NEXT
Subscriber14=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,IMSI,SIM24] CHADDR
ALARM NEXT
...
```

If you remove the keyword IMSI from the pabx.cfg, you must restart the system.

Activating peer data for VoIP calls

To generate a VoIP-call CDR entry that includes IP addresses for the remote device's signaling and voice data, audio codec and frame size, the entry VoipIpLogging=Yes must be included in the VoIP profile. If the entry also contains the mobile controller's IMSI, it will appear before the IP addresses.

The following entry shows the `route.cfg` configuration file changed according to the formula.

Example 8.16 Activating peer data for VoIP calls (`route.cfg entry`)

```
[Voip:DF]
VoipDirection=IO
VoipPeerAddress=192.168.0.2
VoipIpMask=0xffffffff
VoipCompression=g729 t38
VoipMaxChan=30
VoipSilenceSuppression=Yes
VoipSignalling=0
VoipTxM=4
VoipIPLogging=Yes
```

The following CDR entry includes IP addresses for signaling and voice data, audio codec and frame size.

Example 8.17 Activating peer data for VoIP calls (`CDR entry`)

```
V1,24.11.09-16:52:20,24.11.09-
16:52:22,[0008:01]401419,[0006:01]IN777,123456789123451,172.20.25.103:172.20.25.103,G711a,20,0101,2,10,0,,341
93,11
```

CDRs for callback and two stage calls

In the case of CDR entries for Two stage dialing/Callback calls, the beginning and ending times for the first call leg is always used as the call time. The call time in seconds appears first for the first leg, followed by a slash and the connection time for the second leg.

Example 8.18 **CDR entry for callback and two stage calls**

```
V1,24.11.09-17:15:29,24.11.09-
17:15:57,[0002:01]CB,[0008:01]DLA,,172.20.25.103:172.20.25.103,G711a,20,0102,28/
3,90,0,,,
```

Specific cause values

The CELLX will generate cause values that represent an event, such as exceeding a defined limit.

Table 8.4 Specific cause values

Cause value	Description
0a	One of the following limits has been reached: Unit limit defined in the port’s subscriber line
0b	Maximum call duration assigned via vGATE has been reached.
0c	Maximum call duration based on mapping entry settings has been reached.

Table 8.4 Specific cause values

Cause value	Description
0d	Maximum number of minutes per SIM assignment via vGATE has been reached.
10, 1f	Normal call termination (disconnected by calling party).

To avoid sending these values as the reason for call teardown, translate the cause values to standard values Chapter 8.14 Changing cause values.

8.2.2.2 Missed calls list

All incoming calls that are not connected can be recorded in a list to facilitate return calls. Recording is activated using the failedlog=<name> entry in the `pabx.cfg`. Specify a file name, for example failedlog=failed.log. Once this setting is made, recording begins at once.

Each line represents an unaccepted incoming call with the following information separated by commas.

Table 8.5 Failed log entries

Column	Description
0	Version
1	Start time (format DD.MM.YY-hh.mm.ss)
2	Source. The following format applies: [node number:automatically set internal channel number]
3	Destination. The following format applies: [node number:automatically set internal channel number]
4	IMSI
5	IP logging signaling: RTP
6	Audio codec used
7	Frame size
8	Service indicator (please see Chapter 8.2.2 Generating and retrieving CDRs)
9	Cause values
10	Call duration (if the call does not result in an Alerting, the entry will be -1)
11	Number of call attempts
12	Cell ID
13	RSSI

Example 8.19 `Failed.log` file

```
V1,24.11.09-16:13:08,[0006:01]IN,[0008:01]GSM,123456789123456,,,,0101,92,-
1,1,34193,9
V1,24.11.09-16:33:34,[0006:01]IN,[0008:01]GSM,123456789123456,,,,0101,92,-
1,1,34193,12
V1,24.11.09-16:35:19,[0006:01]IN,[0008:01]GSM,123456789123456,,,,0101,92,-
1,1,34193,11
V1,24.11.09-16:35:59,[0006:01]IN,[0008:01]GSM,123456789123456,,,,0101,92,-
1,1,34193,11
V1,24.11.09-16:37:29,[0006:01]IN,[0008:01]GSM,123456789123456,,,,0101,92,-
1,1,34193,11
V1,24.11.09-
16:39:17,[0006:01]IN,[0008:01]GSM,123456789123456,,,,0101,ff,7,1,34193,11
```

The reason the connection could not be established is specified using DSS1 codes:

- 91 – (user busy)
- ff – call not answered (disconnected by calling party)

When callback with DTMF is configured and no connection is established to the B subscriber, an entry recording the A subscriber’s connection time is generated in the failed.log file.

Example 8.20 `Failed.log` entry 1

```
V1,24.11.09-
16:39:17,[0006:01]IN,[0008:01]GSM,123456789123456,,,,0101,ff,7,1,34193,11
```

The CDR contains the IP addresses for signaling and voice data. The first IP address is the signaling address and the second one is the RTP address.The IMSI is written behind the IP addresses if the keyword IMSI is defined in the `pabx.cfg`.

Example 8.21 `Failed.log` entry 2

```
V1,24.11.09-16:52:20,24.11.09-
16:52:22,[0008:01]401419,[0006:01]IN777,262032441017556,172.20.25.103:172.20.25.
103,G711a,20,0101,2,10,0,,34193,11
```

In the case of missed-call entries for Two stage dialing/Callback calls, dur is the connection time for the first leg.

Example 8.22 `Failed.log` entry 3

```
V1,25.11.09-14:11:10,[0002:01]CB,DLA,,,,,0102,11,14,1,,
```

8.2.2.3 Sending CDRs via e-mail

With an appropriate configuration, you can send corresponding CDRs of outgoing and incoming calls as e-mail. Bear in mind that the mail server must be configured in the [Mail] section of the `pabx.cfg`, as described in Chapter 5.2.2 SMTP-client configuration. The sender is given as cdr and the system’s name appears in the subject box. The text box contains the CDR information according to the format for the entry in Log=/data/cdr.log @<account> @<domain>. A space must appear between cdr.log and @<account>; @<domain> is optional. You can also send CDR entries via e-mail to an e-mail recipient.

Enter an @ sign to send each CDR entry as e-mail:

```
Log=/data/cdr.log @<e-mail account>@<domain>
```

If you enter a ! the entire cdr.log will be sent as an e-mail attachment:

```
Log=/data/cdr.log !<e-mail account>@<domain>
```

8.3 Ported number screening

Ported number screening is a very useful functionality to avoid high routing costs for numbers that have been ported to another network operator.

Number portabilty refers to the ability to transfer either an existing fixed-line or mobile telephone number to another network operator. This way telecommunications subscribers can change operators without having to change their telephone numbers. Routing ported numbers, however, can become very cost intensive due to differences in tariffs.

With ported number screening, an external database is queried to find out if a number has been ported. Either use the iMNP for querying the external database or query the database directly. The iMNP is a proxy that remembers the database information for a defined period of time and renews the query to the external database only after that period has passed. Since every query to the external database costs money, the iMNP helps to reduce querying costs.

The qery result is used in a routing to route the call via the right network operator.

To implement ported number screening, make sure to meet the system requirements which are:

- An active license for number portability.
- An iMNP server or another appropriate server.

Also ensure to adjust your configuration:

- To connect to the number portability database, you must set the entries described in Chapter 5.2.3 Number portability settings.
- Configure your `route.cfg` file to activate ported number screening:

  `DTMFWaitDial=<sec>`
  Set the time that the gateway waits for additional digits.

  `MapAll<num>=|$<prefix><num><<<count>`
  Enable digit collection (pipe symbol) and collect up to the number of digits that is specified under `count`. Add the dollar sign to search the routing table again. Prefix the number with a freely chosen prefix to prepare it for sending to the database.

  `MapAll<prefix>=|D@<num><<01`
  Send the prefixed number to the database. Make sure that the prefix used here matches the above prefix.

  `MapAllQN<query result>=<controller>`
  Map the query result to the respective controller. The query result consists of the LAIN and the number including the country code. Do not forget to also configure routings for numbers that haven't been found in the database or that do not exist and also provide a routing when the server does not respond in a defined period of time.

In the following example, 14 digits are collected (11 digits plus $ph) and a maximum time of 5 seconds is waited for each digit. Every incoming call with a leading digit of 0 results in an iMNP query. The SIM card's LAIN is used instead of controller numbers. All numbers that come back from the iMNP with the LAIN for Carrier_1 (26211) are then routed through

Carrier_1’s SIM card. The same applies for Carrier_2 (26212), Carrier_3 (26213) and Carrier_4 (26214). Numbers that the iMNP sends back as non-existing (00000) are rejected. Numbers that may exist but are not found in the database (99999) are routed as they come in (normal). If the iMNP does not respond within two seconds (D@0), the call is routed as it comes in, whether it is ported or not.

Example 8.23 Ported number screening

```
DTMFWaitDial=5
MapAll0=|$ph0<<14
MapAllph=|D@0<<01

MapAllQN26211=26211
MapAllQN26212=26212
MapAllQN26213=26213
MapAllQN26214=26214
MapAllQN00000=&81
MapAllQN99999=$normal
MapAllD@0=$normal1
; not in Database
;Carrier_1
MapAllnormal0151=262110151
MapAllnormal0160=262110160
MapAllnormal0170=262110170
MapAllnormal0171=262110171
MapAllnormal0175=262110175
;Carrier_2
MapAllnormal0152=262120152
MapAllnormal0162=262120162
MapAllnormal0172=262120172
MapAllnormal0173=262120173
MapAllnormal0174=262120174
;Carrier_3
MapAllnormal0155=262130155
MapAllnormal0163=262130163
MapAllnormal0177=262130177
MapAllnormal0178=262130178
;Carrier_4
MapAllnormal0159=262140159
MapAllnormal0176=262140176
MapAllnormal0179=262140179
```

8.4 Digit collection (enblock/overlap receiving)

This function makes it possible to collect up to 24 digits and transmit calls when a specific number of digits has been dialed. The entire call number is required for the call to be set up with a mobile phone or the mobile gateway. Since most numbers have a uniform number of digits, the mobile gateway can collect digits when calls enter the gateway in overlap mode. Digit collection occurs through the following mapping command:

`MapAll<direct>=|<num><<<digits>`

The **|** (pipe) signifies that the following digits will be collected before they are transmitted, and <digits> is the total number of the port digits and the digits of the called party number. This figure can range between 00 and 24 and must be entered in double digits. The parameter **`DTMFWaitDial`** defines the number of seconds the system waits between the individual digits (default 5). Please bear in mind that you can configure a maximum of 11 digits in the first part of the command and 19 (including a special character, for example #) in the second. The call will be forwarded as soon as the specified number of digits has been dialed or a time-out limit has been reached.

The following example shows a call with the prefix 01555. The **|** (pipe) signifies that the following digits will be collected before they are transmitted. The 14 at the end is the sum of the port digits and the digits of the called party number (for example |#20=3, 01555899666=11, 3+11=14).

Example 8.24 Digit collection

```
...
MapAll01555=|#2001555<<14
...
DTMFWaitDial=5
...
```

8.5 Rejecting data calls and specified numbers

This chapter describes the configuration options for exclusion of data calls, prefixes, or call numbers from the routing process.

8.5.1 Blacklist routing

The system will reject all calls directly if the MapAll entry contains the keyword & followed by the two-digit cause value (see ETS 300 102-1).

`MapAll<direct>=&<cause>`

A maximum of 5000 MapAll entries per time zone can be defined. For more than 5000 entries, please use the iMNP.

In the following example, all calls to the number 004915551234 and all service calls with the prefix 0180 are rejected with a busy signal. All other calls are sent to the VoIP profile DF.

Example 8.25 Blacklist routing

```
MapAll015551234=&91
MapAll004915551234=&91
MapAll0180=&91
MapAll0=40DF:0
...
MapAll9=40DF:9
```

8.5.2 Whitelist routing

The following entries enable exclusion of specific OADs or trunk groups:

`Restrict<ns>=<pl>`

`MapAll<pl>=&<cause>`

NS refers to the internal controller number and the call’s origination address.

A maximum of 1000 Restrict entries per time zone can be defined.

In the following example, the numbers 12345 and 12346 connected to the PBX at port 10 cannot make any international calls. All national calls are sent to the VoIP profile DF and all local calls are sent to the PSTN.

Example 8.26 Whitelist routing 1

```
Restrict1012346=int
MapAllint00=&91
MapAllint0=40DF:0
MapAllint1=91
...
MapAllint9=90
```

In the following example, all incoming calls from the mobile port trunk groups 26212 and 26213 are rejected with a busy signal.

Example 8.27 Whitelist routing 2

```
Restrict26212=not
Restrict26213=not
MapAllnot=&91
```

8.5.3 Rejecting calls with ISDN bearer capability data

ISDN data calls can be handled differently from voice calls depending on the configuration of the call types DATA or VOICE. This setting is especially interesting for VoIP or GSM calls.

`MapAll<direct>=&<cause> <mode>`

Analog modm connections are not included in this configuration, as they generally do not have a specified bearer capability.

In the following example, all ISDN data calls are rejected with the cause value AA (switching equipment congestion). All calls with the prefix 0170 are routed to the mobile trunk group 26211 and all other calls are routed through VoIP.

Example 8.28 Rejecting calls with ISDN bearer capability data

```
MapAll0=&aa DATA
...
MapAll9=&aa DATA
...
MapAll0170=262110170
MapAll0=40DF:0
...
MapAll9=40DF:9
```

8.5.4 Specific routing of data calls via VoIP

In the ISDN network, data calls have a special service type. When an ISDN PBX is connected to a VoIP network, it must continue to work without any problems (for example PBX remote maintenance calls or ISDN terminal adapter). In the case of VoIP, a specific RTP payload type is used: trp, ccd or gnx64.

In the following example, two VoIP profiles are configured, so that all calls are routed, regardless of whether they are data calls or voice over IP calls. The first one is for outgoing voice calls and all calls from VoIP to ISDN. The second profile is exclusively for outgoing data calls, so that signaling consists solely of clear mode in SDP.

Example 8.29

```
MapAll0=40DATA:0 DATA
...
MapAll9=40DATA:9 DATA
MapAll0=|40DF:0<<24
...
MapAll9=|40DF:9<<24
Restrict40=In
MapAllIn=10
[Voip:DF]
VoipDirection=IO
...
VoipCompression=g711a g729 trp t38
...
[Voip:DATA]
VoipDirection=Out
...
VoipCompression=trp
VoipECE=No
...
```

8.6 CLIP and CLIR

8.6.1 Routing CLIP and CLIR calls

This function allows you to route calls with Calling Line Identification Presentation (CLIP) differently from calls with Calling Line Identification Restriction (CLIR). For example, all CLIP calls can be rejected, so that only calls that do not present the calling number or calls without a calling party number (for example analog) are transmitted through the CELLX.

Use the following configuration to define the various routing methods.

Example 8.30 Routing CLIP and CLIR calls

```
...
InsertCLIR=On
...
Restrict9=OK 01
Restrict|9=OK 01
Restrict90=FAIL 01
...
MapInOK00491555=2200491555
MapInFAIL=&aa
...
```

InsertCLIR=On activates this mode. 01 is the service indicator for telephony (analog and ISDN) and is used to differentiate these calls from remote administration calls. Restrict9=OK 01 means that all telephony calls without a calling number are put through. Restrict|9=OK 01 means that all CLIR telephony calls are put through. Restrict90=FAIL 01 means that all CLIP telephony calls are rejected with No Channel Available as rejection cause when they are mapped to MapInFAIL=&aa.

8.6.2 Routing calls without CLIR

Use this function to bypass CLIR for calls through the defined mobile port. The following configuration in `pabx.cfg` activates this function.

Example 8.31 Routing calls without CLIR

```
Subscriber<xx>=...GSM[...,!CLIR]...
```

When this function is configured, the SIM's telephone number (and not originating telephone) is always transmitted to the B subscriber.

8.6.2.1 Setting CLIR

Setting a hash (#) in front of a call number makes it possible to suppress the presentation of the origination number of calls regardless of how the call comes into the system.

The following syntax is used: `MapAll<num>=#<port><num>.`

The following example shows an appropriate configuration. With this entry, all calls beginning with 00491555 are sent to the port with the address 22 and the presentation of the number is restricted.

Example 8.32 Setting CLIR

```
MapAll00491555=#2200491555
```

8.6.2.2 Setting CLIP

Setting an exclamation point (!) in front of a call number makes it possible to force the presentation of the origination number of calls regardless of how the call comes into the system.

The following syntax is used: `MapAll<num>=!<port><num>`

The following example shows an appropriate configuration. With this entry, all calls beginning with 004930 are sent to the port with the address 9 and the presentation of the origination number is allowed.

Example 8.33 Setting CLIP

```
MapAll004930=!9004930
```

8.7 Conversion of call numbers

The conversion of call numbers makes it possible, for example, to implement number portability or to redirect calls when the user can be reached at another number. In the following mapping command, the call number 015550123456 is changed to 015559876543 and sent to the mobile channel (MapAll...=20..)

Example 8.34 Conversion of call numbers 1

```
MapAll015550123456=20015559876543
```

The following example presents an alternative, in which the routing file is searched through again after conversion of the call number to determine the route for the prefix `01555`. Please bear in mind that you can configure a maximum of 1499 mapping entries with no more than 11 digits in the first part of the command and 19 in the second.

Example 8.35 Conversion of call numbers 2

```
MapAll015550123451=$Reception
MapAll015550123452=$Reception
MapAll015550123453=$Reception
MapAllReception=015559876543
```

8.8 Overwriting OAD

In some cases where the OAD is not transmitted or shall for other reasons be overwritten, the following parameter can be used to send another OAD:

```
Restrict<port>=<OAD to be sent> 15 or
Restrict<port><original OAD>=<OAD to be sent> 15
```

For PRI ports only: extend this parameter by a text in quotation marks to transmit a display info element in a PRI or SIP call. If supported by the receiving telephone, the content of the display field is displayed in addition or instead of the OAD.

```
Restrict<port>=<OAD to be sent> 15 “display field“ or
Restrict<port><original OAD>=<OAD to be sent> 15 “display field“
```

In the following example, the text “Teles Marketing” and / or the OAD 456 are displayed to the called person for calls that have been received via port 10.

Example 8.36 Overwriting OAD

```
Restrict10=456 15 “TELES Marketing“
```

8.9 Setting number type in OAD/DAD

In some cases it may be necessary to set a specific number type for the OAD or DAD. There are different methods for the various interfaces. The following number types can be set.

Table 8.6 Number types

Type	Description
u	Unknown
s	Subscriber number
n	National number
i	International number

OAD

Use the following entry to set a specific number type in the OAD:

`Restrict<port><num>=<type> 15`

For the national and international types, remove the 0(s) at the beginning of the number:

`Restrict<port>0=n 15`

`Restrict<port>00=i 15`

In the following example, the bit is set in the caller’s origination number for a call via BRI controller 01.

Example 8.37 Setting number type in OAD 1

```
Restrict90=n 15
Restrict900=i 15
```

You can set a u (unknown type of number) in the Restrict entry to change transmission of the national/international bit to 0 or 00 at the beginning of the OAD. As in a mapping entry, the national/international bit will always appear left of the equal sign as 0 or 00.

```
Restrict<port>0=u0 15
Restrict<port>00=u00 15
```

In the following example, the area code 030 with a 0 at the beginning of the OAD of the PBX's extension is set as a digit and transmitted along with the number.

Example 8.38 Setting number type in OAD 2

```
Restrict10555=u030555 15
```

ⓘ `Restrict` entries are handled from general to specific from top to bottom.

The keyword `RestrictR` allows you to set a second modification in the OAD in the case of a redirect.

In the first part of this example, OADs beginning with 030 are changed to 4930.

Example 8.39 OAD changed ...

```
Restrict9030=4930 15
```

In the second part of this example, if the call is not connected and a Redirect3 is configured for the call, the OAD will be changed from 4930 to 004930 as follows.

Example 8.40 ... and RestrictR applied

```
RestrictR4930=004930 15
```

DAD

Enter one of the four specific number types in the DAD as follows:

`MapAll<num>=<port><type><num>`

In the case of a VoIP controller, enter the following:

`MapAll<num>=<port><voip profile>:<type><num>`

The number type will then be defined at the port. For the national and international types, remove the 0(s) at the beginning of the number:

In the following example, the international bit is set for all calls to Italy (0039) and the number is transmitted with 39. For the area code 012, the national bit is set and the number is transmitted with 12.

Example 8.41 Setting number type in DAD 1

```
MapAll0039=40iG1:i39 VOICE
MapAll012=40iG1:n12 VOICE
```

In the following example, a 1:1 routing entry for the individual PRI controllers to VoIP appears in addition to the international flag from PRI to VoIP. A placeholder routing entry is used (in1 or in2), in which the PRI ports are directly assigned to a mapping. Traffic at PRI port 9 is sent directly to VoIP port 40 with the VoIP profile iG1. Traffic from PRI port 10 is sent to VoIP port 40 with the profile iG2.

Example 8.42 Setting number type in DAD 2

```
Restrict9=in1
Restrict900=i 15
Restrict10=in2
Restrict1000=i 15

MapAllin100=40iG1:i
MapAllin200=40iG2:i
```

The `restrict` entries for the individual ports must appear in the following order: placeholder, OAD international flag, DAD routing with international flag.

8.10 Setting the screening indicator

You can set the screening indicator to define whether the calling-party number sent is specified as user provided verified and passed or network provided:

User provided verified and passed: v

Network provided: p

In the following Restrict example, the calling party number sent is specified as user provided verified and passed.

Example 8.43 Setting the screening indicator 1

```
Restrict10=v 15
```

In the following Restrict example, the calling party number sent is specified as network provided.

Example 8.44 Setting the screening indicator 2

```
Restrict10=p 15
```

If you also want to define a number type (see Chapter 8.9 Setting number type in OAD/DAD), it must appear in front of the screening indicator:

In the following Restrict example, the screening indicator is specified as network provided, and the number type is international.

Example 8.45 Setting the screening indicator 3

```
Restrict10=ip 15
```

Please bear in mind that this entry will not work if you set a minus sign (-) behind `VoipOad=<num>`.

8.11 Setting a default OAD

Use the Restrict command to set a default origination number (*<oad> 15) when the OAD is restricted (<num>):

`Restrict<port><oad>=*<num> 15`

In the following example, 12345 replaces the original OAD. When the destination number begins with 030, the call is sent through controller 10.

Example 8.46 Setting a default OAD 1

```
Restrict9=*12345 15
MapAll030=10030
```

Use the entry Restrict<port><oad>=<num> 15 if digits at the beginning of the OAD are the only ones to be restricted.

In the following example, the digits 004930 are replaced with 030 followed by the remaining digits. The destination number begins with 030 and is sent through port 10.

Example 8.47 Setting a default OAD 2

```
Restrict9004930=030 15
MapAll030=10030
```

8.12 Setting or removing sending complete byte in setup

In some cases the ISDN or H323 peer system may require this byte for routing, or the byte may disrupt signaling.

Setting sending complete

The following entry ensures that the Setup includes a Sending Complete:

`MapAll<direct>=)<num>`

The) causes inclusion of Sending Complete in the ISDN Setup or in the H323 Setup.

In the following example, all calls beginning with 0 are sent with a Sending Complete to controller 9.

Example 8.48 Setting sending complete

```
MapAll0=)90
```

Removing sending complete

The following entry ensures that the Setup never includes a Sending Complete:

```
MapAll<direct>=(<num>
```

The (causes removal of Sending Complete in the ISDN Setup or in the H323 Setup.

In the following example, all calls beginning with 0 are sent without a Sending Complete to VoIP controller 40. The VoIP profile is `DF`.

Example 8.49 Removing sending complete

```
MapAll0=(40DF:0
```

8.12.1 Exclusion from SIM minutes counter

The keyword DDNC (daily duration not counted) will exclude the mapping entry from the minutes counter:

MapAll<num>=<port><num> DDNC

Example 8.50 Exclude mapping entry from SIM minutes counter

```
MapAll030=26212030 DDNC
```

8.13 Miscellaneous routing methods

In the following scenarios it may occur that some call numbers must be routed with differing lengths or that some call numbers may require additional number conversion:

- Calls without a destination number
- Connection to a PBX with an extension prefix
- Routing based on the length of the destination number

8.13.1 Routing calls without a destination number

Enter the following configuration in the `route.cfg` if the CELLX must route calls that come in without a destination number:

```
Restrict<port>=<pl>
MapAll<pl><num>=<port><num>
MapAll<pl>=<port>
```

Incoming calls from the configured port will be assigned a placeholder and then all calls beginning with the placeholder will be routed to the placeholder's placeholder's mapping.

In the following example, all calls from controller 9 are routed to controller 10, regardless of whether a destination number appears in the setup.

Example 8.51 Routing calls without a destination number

```
Restrict9=pl
MapAllpl=10
```

8.13.2 Routing calls based on extension prefix or the length of the DAD

To route calls with a DAD differently from those without a DAD, you must activate the block feature in the `pabx.cfg` and restart the system:

```
Block=1
```

Set all other parameters in the `route.cfg`. First define the port from which the incoming calls are to be routed. Incoming calls from the configured port will be assigned a placeholder and then digit collection will occur for all calls beginning with the placeholder. The $ in the mapping entry, followed by the defined placeholder (MMM), causes a second search of the routing file when the number is complete:

```
DTMFWaitDial=<sec>
Restrict<port>=<pl>
MapAll<pl>=|$MMM<<98
```

The second routing-file search is based on the routing entry with the leading placeholder (MMM):

```
MapAllMMM<digits>=<dest><digits>
```

In the following example, digit collection is activated for all calls that come into port 9. Calls with the destination number 2222 are sent to the VoIP controller with the profile DF and the destination number is replaced with the SIP account Betty. Calls with the number 3333 are sent to VoIP with the SIP account Al. All other calls with a destination number are sent to controller 10. Calls without a destination number are sent to the number 12345 at port 10.

Example 8.52 Routing calls based on an extension prefix or on the length of the destination number

```
DTMFWaitDial=5
Restrict9=pl
MapAllpl=|$MMM<<98
MapAllMMM2222=40DF:Betty
MapAllMMM3333=40DF:Al
MapAllMMM0=100
MapAllMMM1=101
MapAllMMM2=102
MapAllMMM3=103
MapAllMMM4=104
MapAllMMM5=105
MapAllMMM6=106
MapAllMMM7=107
MapAllMMM8=108
MapAllMMM9=109
MapAllMMM=1012345
```

8.14 Changing cause values

It is possible to group cause values together into a single defined cause value so that rejected calls can be handled in a specified manner by the switch sending the call to the CELLX. The following cause value groups can be defined in the `pabx.cfg`:

Group 0 cause values

All connections that are rejected with a group 0 cause value (0x80-0x8f) can be mapped to a single cause value by entering `TranslateG0Cause=<cau>`, whereby `<cau>` represents a cause value in hexadecimal form.

Group 1 cause values

All connections that are rejected with a group 1 cause value (0x90-0x9f) can be mapped to a single cause value by entering `TranslateG1Cause=<cau>`, whereby `<cau>` represents a cause value in hexadecimal form.

Group 2 cause values

All connections that are rejected with a group 2 cause value (0xa0-0xaf) can be mapped to a single cause value by entering `TranslateG2Cause=<cau>`, whereby `<cau>` represents a cause value in hexadecimal form.

Group 3 cause values

All connections that are rejected with a group 3 cause value (0xb0-0xbf) can be mapped to a single cause value by entering `TranslateG3Cause=<cau>`, whereby `<cau>` represents a cause value in hexadecimal form.

Translating individual cause values

The following parameter allows you to translate any of these cause values to any other one: `Translate<cause>=<cause>`. The values entered must be in hexadecimal notation between 00 and 7f.

Translating SIP causes to ISDN and vice versa

You can define a specific translation from SIP responses (4xx - 6xx) to ISDN cause values and vice versa. If nothing is set, the translation occurs as described in `draft-kotar-sipping-dss1-sip-iw-01.txt`.

Use the following parameter to translate a cause from ISDN to a specific SIP response:

```
SipCause<ISDN cause>=<SIP Response>
```

Repeat the entry to initiate an additional translation.

Use the following parameter to translate a cause from SIP to ISDN:

```
SipEvent<SIP Response>=<ISDN Cause>
```

The following range of values applies:

400<= <SIP Cause> <=699	(defined in RFC 3261)
0<= <ISDN Cause> <=127	(DSS1 decimal cause number)

8.15 Call forking

The call forking functionality is mostly used to send inbound calls to your PBX extension and mobile phone simultaneously. Calls can be answered from either the extension or the mobile phone.

To activate this feature, you need to add two parameters to the `route.cfg`: `GlobalIn` and `GlobalOut`.

Before adding call forking functionality to your `route.cfg,` check whether the necessary mapping of a destination number to a port plus new destination number exists:

`MapAll<destinationNumber>=<port><newDestinationNumber>`

For routing to a VoIP port, the following syntax is used:

`MapAll<destinationNumber>=<voipPort><voipProfile>:<newDestinationNumber>`

The parameters destination number and new destination number do not need to be identical.

To activate call forking, add the following entries behind the mappings:

Assign the port plus new destination number to the parameter `GlobalIn<int>`. You can define up to 32 different `GlobalIn` configurations.

`GlobalIn<int>=<port><newDestinationNumber>` where int >= 1

For VoIP, the following syntax is used:

`GlobalIn<int>=<voipPort><voipProfile>:<newDestinationNumber>` where int >= 1

Assign the numbers which you want to call simultaneously to the `GlobalOut<int>` parameter. The integer value is used to map `GlobalIn` to `GlobalOut.` You can define up to 32 different `GlobalOut` configurations.

`GlobalOut<int>=<destinationNumber1> <destinationNumber2>` where int >= 1

Be aware that for both destination numbers the `MapAll` parameter needs to be set. This must be done before the forking.

In this example two mappings exist: calls to the destination number 543211234 are mapped to the port 10 and the new destination number 543211234. Calls to the destination number 01721234554 are mapped to the port 20 and the new destination number 01721234554. Before the call is sent out, the system checks the `route.cfg` for a matching `GlobalIn` parameter. A match is found for calls to 10543211234 and calls to this number are sent to 543211234 and 01721234554 simultaneously.

Example 8.53 Call forking

```
MapAll543211234=10543211234
MapAll01721234554=2001721234554
...
GlobalIn1=10543211234
GlobalOut1=543211234 01721234554
```

9 Additional VoIP parameters

9 Additional VoIP parameters

You can enter the following additional parameters in the `route.cfg` to adjust the configuration for improved communication with the VoIP peer.

9.1 Signaling parameters

Table 9.1 Customized parameters: protocol-independent VoIP signaling

Protocol-Independent VoIP Signaling Parameters
`VoipComprMaster=<mode>` This parameter defines which side the first matching codec comes from: `Yes`: Default. Priority is determined by the order of the system's parameter list. `No`: Priority is determined by the peer.
`VoipConnectOnCallProc=<mode>` Enter Yes (default No) to change an H.323 Call Proceeding/Call Progress and Alert, or a SIP 180 or 183, into an H.323 Connect or SIP 200 message. This parameter can be used if an announcement that plays before the Connect requires a Connect on the other side to put through the call.
`VoipDad=<num>` The digits/numbers defined here will appear in front of the original DAD as described for the parameter VoipOad below. If the parameter is to be valid in only one direction, you must also set the parameter VoipDadIn or VoipDadOut .
`VoipDadIn=<string>` Specifies that the parameter VoipDad is for incoming calls only. If this parameter is not set (default), VoipDad will apply for both directions.
`VoipDadOut=<string>` Specifies that the parameter VoipDad is for outgoing calls only. If this parameter is not set (default), VoipDad will apply for both directions.
`VoipDataBypassPayloadType=<num>` Defines the payload type for the RTP packets when the call is sent as a data call. Default 96.
`VoIPEarlyT38=<mode>` Enter `yes` to enable T38 before the call has been connected. Enter `no` to enable T38 after the call has been connected. Default yes.
`VoIPEarlyVP=<mode>` Activates the vocoder chip caused by an incoming SIP 180 or 183 without sdp. Fax tones coming from ISDN are already recognized before the connect and can thus prepare or even allow a switch to T38. Default `no`.
`VoipG72616PayloadType=<num>` Changes the SIP payload type for G.726 16 b/s. Default is 35. A common alternative is one of the dynamic payload types from 96 to 127.
`VoipG72624PayloadType=<num>` Changes the SIP payload type for G.726 24 b/s. Default is 36. A common alternative is one of the dynamic payload types from 96 to 127.
`VoipG72632PayloadType=<num>` Changes the SIP payload type for G.726 32 b/s. Default is 2. A common alternative is one of the dynamic payload types from 96 to 127.
`VoipG72640PayloadType=<num>` Changes the SIP payload type for G.726 40 b/s. Default is 38. A common alternative is one of the dynamic payload types from 96 to 127.

Table 9.1 Customized parameters: protocol-independent VoIP signaling *(continued)*

Protocol-Independent VoIP Signaling Parameters
`VoipHideOadByRemove=<mode>` If Yes is configured and call setup is to VoIP, the OAD will be removed from signaling if presentation restricted or user-provided, not screened is set in the calling party's presentation or screening indicator. No (default) means no change will occur. NOTE: If the SIP protocol is used, Anonymous will always appear as the account in the From field. Transmission of the OAD can occur in the P-asserted header.
`VoipIgnoreDADType=<mode>` Enter yes to change the DAD type to unknown, for example from international. The type is lost, for example the leading 00 bit is removed. Default no.

Table 9.1 Customized parameters: protocol-independent VoIP signaling *(continued)*

Protocol-Independent VoIP Signaling Parameters

`VoipOad=<num>`

In the following examples, the OAD is 5175551212.

- The digits/numbers defined here will be transmitted in front of the original OAD. In the following example, 001 will appear in front of the number 5175551212.
  EXAMPLE: `VoipOad=001`
- To transmit only OADs consisting of more digits than those defined, enter a !, followed by the number of digits, at the end of the entry. If the OAD has more digits, it will be transmitted unchanged. If it has the same number or fewer digits, only the string in front of the ! will be added in front of the OAD. In the following example, the digits 0015175551212 will appear::
  EXAMPLE: `VoipOad=001!10`
- If a minus (-) is entered, the original OAD will not appear. In the following example, no number will be transmitted.
  EXAMPLE: `VoipOad=-`
- If integers are entered before the minus sign, only the digits entered in front of the minus sign will be displayed. In the following example, 789 will appear.
  EXAMPLE: `VoipOad=789-`
- If an integer is entered after the minus sign, this number of digits will be removed from the beginning of the OAD. In the following example, 517 will be removed from the OAD:
  EXAMPLE: `VoipOad=-3`
- If -\ is entered followed by the first part of an OAD, that part will be removed. If the string does not match, the OAD will be transmitted unchanged. In the following example, only 1212 will be transmitted.
  EXAMPLE: `VoipOad=-\517555`
- To modify the original OAD, enter random<x>, whereby x represents a number of random digits between 1 and 10 that will appear in the OAD. In the following example, 001 plus 2 random digits plus the original OAD will appear.
  EXAMPLE: `VoipOad=001random2`
- If --\ is entered followed by a part of an OAD (not necessarily the first part), the OAD will not be transmitted at all. If the string does not match, the OAD will be transmitted unchanged. In the following example, the OAD is not transmitted.
  EXAMPLE: `VoipOad=--\555`
- If -<num>-\ is entered followed by a part of an OAD (not necessarily the first part), the entered number of digits is cut from the OAD and the remaining OAD is searched for the entered part. If it is found, the OAD will not be transmitted. If the string does not match, the remaining OAD will be transmitted.
  In the following example, the OAD is not transmitted.
  EXAMPLE: `VoipOad=-3-\555`
  In the following example, the remaining OAD 5551212 is transmitted.
  EXAMPLE: `VoipOad=-3-\444`
- Enter -\x\y\ to replace the digits x by the digits y. x must be at the beginning of the OAD. x and y can differ in length.
  In the following example, the digits 517 are replaced by the digits 00517.
  EXAMPLE: `VoipOad=-517\00517\`
- Enter -\?\y\ to replace any first digits with y. ? is used as a placeholder for one digit. ? and y can have different lengths.
  In the following example, the first three digits are replaced by 00517.
  EXAMPLE: `VoipOad=-???\00517\`
- Use a semicolon to concatenate several `VoipOad` commands. The commands are processed from left to right.
  In the following example, the first three digits are removed from the beginning of the OAD. In the remaining OAD, the first three digits are replaced by 444. The OAD 4441212 is transmitted.
  EXAMPLE: `VoipOad=-3;-???\444\`

NOTE: If the parameter is to be valid in only one direction, you must also set the parameter VoipOadIn or VoipOadOut .

`VoipOadIn=<string>`

Specifies that the parameter VoipOad is for incoming calls only. If this parameter is not set (default), VoipOad will apply for both directions.

Table 9.1 Customized parameters: protocol-independent VoIP signaling *(continued)*

Protocol-Independent VoIP Signaling Parameters
`VoipOadOut=<string>` Specifies that the parameter VoipOad is for outgoing calls only. If this parameter is not set (default), VoipOad will apply for both directions.
`VoipProgress=<int>` For H.323: 0=progress indicator is not transmitted. 1 (default)=progress indicator is transmitted. 2=address complete message is transmitted. 3=call proceeding message type changed in alerting message type. For SIP: 0=183 response ignored and not sent. 1=183 response changed to a progress message with inband-info-available at the ISDN interface (default). 2=183 response changed to an address complete message at the ISDN interface. 3=183 response changed to an alerting at the ISDN interface.
`VoipSignalCLIR=<string>` When the configured string appears at the beginning of the OAD and the parameter VoipHideOadByRemove is set, the OAD is removed from signaling, regardless of the presentation bits in the calling party field. If the parameter VoipHideOadByRemove is not set (default), the presentation bits are set at presentation restricted (CLIR) if <string> is -. If the string matches the first digits of the OAD and it comes in with CLIP, the call will be sent to VoIP using CLIR. If the call comes in with CLIR, the string will be added to the beginning of the OAD and CLIR will be removed in the signaling.
`VoipSingleTcpSession=<mode>` Enter Yes to send all outgoing VoIP connections in a single TCP session. Enter No (default) for an extra TCP session for each VoIP connection.
`VoipSuppressInbandInfoAvailableIndicatorInCallProceeding=<mode>` Enter yes to send or receive the Progress Indicator in the Q.931 Call Proceeding message. Default no.
`VoipTrpPayloadType=<num>` Defines the payload type for data calls when trp (transparent/clear mode) is used as codec in VoipCompression=<list>. Default is 56. A common value is 102.

Table 9.2 Customized parameters: H.323 signaling

H.323 Signaling Parameters
`VoipCanOverlapSend=<mode>` Enter off to deactivate overlap sending during setup (default on).
`VoipH245Transport=<int>` This option determines the H.245 offer. 0 (default)=all signaling variants are offered; 1=FastStart only; 2=H.245 tunneling only; 3=extra session.
`VoipMapAddressType=<mode>` For calls from PSTN to VoIP only. Enter `yes` to change the 00 at the beginning of a number to international and 0 to national.
`VoipMCinRLC= <mode>` Enter Yes to cause the system to send the media channel capability in the reverse logical channel parameters as part of the H.245 negotiation (default No).
`VoipRejectIncomingNonMatchingFaststart=<mode>` Setup will be rejected if the RTP codecs offered for incoming H.323 call setup with Faststart elements do not match those configured in the parameter VoipCompression. Default No.
`VoipRestrictTCS=<mode>` If Yes is entered, the response in the H.323 tunneling terminal capability set contains only the codecs offered by the peer and not those configured in the system. Default No.

Table 9.2 Customized parameters: H.323 signaling *(continued)*

H.323 Signaling Parameters

`VoipService=0x<service indicator>`

This parameter sets the ISDN bearer capability. For example, it can be used for calls coming from VoIP with the bearer capability data. You can define the service indicator as it is in the 1TR6 code:
101 - ISDN 3,1kHz
102 - analog
103 - ISDN 7kHz
201 - Fax 2
202 - Fax 3
203 - Fax 4
700 - Data
Normally 101 is used. You can send another value to a switch that wants to handle VoIP calls differently from PSTN calls.
EXAMPLE:
`VoipService=0x101`

`VoipSetupAck=<int>`

1=setup acknowledge is transmitted; 0= setup acknowledge is not transmitted; 2 (default) =transmitted with H.323 information.

Table 9.3 Customized parameters: SIP signaling

SIP Signaling Parameters

`VoipAckWithSdp=<mode>`

Enter `yes` to send the SDP content in the SIP ack message. Default no.

`VoipAllow=<list>`

The allow header shows the supported methods and can be set here.
EXAMPLE: VoipAllow=INVITE,BYE
The default setting includes the following:
INVITE,ACK,CANCEL,BYE,UPDATE,REGISTER,PRACK,INFO,NOTIFY,REFER
It may be necessary to remove some of these entries for some peers.

`VoipAllowTlsAoc99=<yes/no>`

If the AOC99 information (charging information) is sent within a SIP-info-message and this parameter is set to yes, the AOC information is evaluated. The default value is yes.

`VoipAngleBracketIsReserved=<yes/no>`

Enter `yes` to to replace < or > in the SIP uri by their ASCII encoding %3c or %3e. Is needed when a peer does not accept < or >. Default no.

`VoipContact=<account@domain>`

Used for the `Contact` field in Sip-Invite and Sip-Response messages.

`VoipContactParam=<string>`

Sets additional header parameters in the contact field.

`VoipDadSource=<int>`

SIP only: defines the field from which field the called party number coming from SIP is to be taken:
0 = URL or URI in the Invite request that contains only digits, otherwise the To: field is used (default)
1 = To: field
2 = Remote-Party-ID with party = called
4 = URL or URI in the Invite request that can also contain letters
8 = Diversion header (in case a redirect number is to be used as destination number)

Table 9.3 Customized parameters: SIP signaling *(continued)*

SIP Signaling Parameters
`VoipDelayDisc=<mode>` Yes (default) delays confirmation transmission during call teardown. That means the release tone is audible when the peer tears down the call. NOTE: For versions 13.0c or lower: To improve ASR, we recommend that you set this parameter to `Yes` if you use the parameter `VoipMaxChan`.
`VoipInfoSamOnly=<mode>` This parameter determines the behavior in the case of overlap sending (VoipOverlap must also be set). Yes means that the contents of the SubsequentNumber field in info method will be attached to the URI's available digits or to the invite message's To field. No (default) means that the digit contents of the SubsequentNumber field will be used.
`VoipOadSource=<int>` SIP only: defines the field from which field the calling party number coming from SIP is to be taken: 0 = From: field (default) 1 = Remote-Party-ID 2 = P-Preferred-Identity 4 = P-Asserted-Identity 8= Display field NOTE: If 2 or 4 are entered, the number in the field must begin with tel: Going to SIP, the OAD is written in the following field: 0 = From: field (default) 1 = Remote-Party-ID (if VoipOwnAddress is not set) For the fields P-Preferred-Identity and P-Asserted-Identity, please check the corresponding parameters. If the number is sent with CLIR to SIP, the From: field contains anonymous@anonymous.invalid. If the number has to appear in the From: Field, the decimal value of the parameter must be increased by 8.
`VoipOverlap=<mode>` SIP only. Enter `yes` to activate signaling with overlap sending, as per draft-zhang-sipping-overlap-01.txt. That means digit collection is no longer necessary in the routing when the digets come from ISDN with overlap sending. When this parameter is active, VoipPrack is automatically set to yes. Default is no.
`VoipOwnAddress=<account@domain>` Used for the `From` field in Sip-Invite and Sip-Response messages. If only the domain is entered, the origination address (for example from ISDN) followed by an `@` sign will automatically be set at the beginning. If the keyword IMSI appears in the parameter and the call is from GSM to VoIP, the SIM cards IMSI is transmitted in the FROM field: EXAMPLE: VoipOwnAddress=IMSI@CELLX01
`VoipOwnDisplay=<string>` The entry is sent as Display Name in the `From` Field in SIP transmissions. The keyword `MSN` causes the calling telephone's MSN to be transmitted as Display Name. The keyword `DSP` causes the ISDN display information element to be transmitted as Display Name. Default: no setting. Example: From: "John" <sip:493011111@teles.de>
`VoipP-Asserted-Identity=<string>` Sets the P-Asserted-Identity field in the SIP invite message. The following settings are possible toward SIP: * The OAD coming from ISDN is transmitted. `<string>` The defined string is transmitted A combination of both is possible. Examples: 030* or tel:* or sip:user@carrier.de

Table 9.3 Customized parameters: SIP signaling *(continued)*

SIP Signaling Parameters
`VoipP-Preferred-Identity=<string>` Sets the P-Preferred-Identity field in the SIP invite message. The following settings are possible towards SIP: `*` The OAD coming from ISDN is transmitted. `<string>` The defined string is transmitted. A combination of both is possible. Examples: 030* or tel:* or sip:user@carrier.de
`VoipPrack=<mode>` SIP only: Enter yes to activate Provisional Response Messages in the signaling, as per RFC 3262 "Reliability of Provisional Responses in the Session Initiation Protocol (SIP)". Default is no.
`VoipSdpProxy=<mode>` SIP only. Enter `yes` to activate proxy mode for SDP signaling for SIP to SIP calls. The parameters for RTP signaling will be forwarded from one leg to the next and RTP is not handled by the system. Default is no.
`VoipSipStatusIncompatibleCodec=<reject cause>` An INVITE request is rejected if the codec that is sent with this INVITE does not match the codecs that have been defined in the VoIP profile. The INVITE is rejected with the reject cause given in this parameter, as per RFC 3261 (default 415).
`VoipUseMaxPTime=<mode>` SIP only. Enter yes to set the field mptime (max packet time) with the values set in VoipTxm (ptime). Default no. The parameter VoipUseMaxPTime is used when VoipUseMPTime is 0, 1 or 2.
`VoipUseMPTime=<int>` This parameter is used to configure packet time signaling in SDP: 0 = set attribute ptime with each individual codec description (default). 1 = set attribute ptime once as the first attribute after the m- line (media type). 2 = set attribute mptime (multiple ptime) once as the first attribute with the list of the codecs' corresponding ptimes. 3 = remove attribute ptime or mptime in SDP signaling. The parameter VoipUseMaxPTime is used when VoipUseMPTime is 0, 1 or 2.
`VoipUserAgent=<mode>/<string>` When Yes is set (default), the system type and software version is used in the SIP user agent header. Enter no to deactivate this setting, or enter a string to change it.
`VoipUtuFormat=<int>` Enter 1 to transfer ISDN user to user information to SIP and vice versa in the following special format: A separate contents block is created (using boundaries) or evaluated that contains the ISDN user to user info. Default is 0, which transports the ISDN user to user information in a SIP header field "User-to-User:", according to the recommendation in http://tools.ietf.org/html/draft-johnston-sipping-cc-uui-09.

9.2 Registrar parameters

The following parameters can be used in the VoIP profile when the SIP agent wants to register with the CELLX.

Table 9.4 Customized parameters: location server

Location Server Parameters
`VoipAuth=<mode>` Defines the authentication procedure `www` (default) or `proxy`.
`VoipExpires=<sec>` Defines the maximum number of seconds the agent's registration applies (default 3600).
`VoipOwnPwd=<string>` Defines the password the agent uses to register.
`VoipOwnUser=<string>` Defines the username the agent uses to register.

The following example creates an account for a user agent with the username 130 and password test130. Authentication occurs with the procedure www.

Example 9.1 Registrar parameters

```
MapAll130=40U1:130

[Voip:U1]
VoipDirection=IO
VoipIpMask=0x00000000
VoipOwnUser=130
VoipOwnPwd=test130
VoipExpires=300
VoipAuth=www
VoipCompression=g711a g711u g729 g729a g729b g729ab
VoipSilenceSuppression=no
VoipSignalling=1
VoipMaxChan=8
VoipTxM=2
VoipDtmfTransport=0
VoipRFC2833PayloadType=101
VoipMediaWaitForConnect=Tone
```

9.3 Routing parameters

Table 9.5 Customized parameters: VoIP routing

VoIP Basic Parameters
`VoipEnumDomain=<string>` Use this parameter to modify the domain name for the enum query (default is `e164.arpa`).

`VoipOadMask=<num>`
`VoipDadMask=<num>`
It is also possible to define the profile by destination or origination number (and not only by the IP address). That means you can use different parameters not only for different IP addresses, but also for different numbers (for example other codec, WaitForConnect, and so on). For example, you can define a number for the head of the company, so that her MSN always uses G.711.
It is possible to configure a list of numbers for a total of up to 80 characters per line. You must define the entry again if you need more numbers. You can also use a wildcard * at the end of the number to match all calls with OADs or DADs beginning with the digits entered. Use a coma to separate the numbers.
Example:
`VoipDadMask=123, 345*, 567, ....,`
`VoipDadMask=912, 913*, 914, ....,`
Bear in mind that you must enter numbers from specific to global (as for normal routing in the route.cfg). That means you must enter a profile with more specific numbers above a profile with more global numbers.

`VoIPOwnIpAddress=<ip addr>` If the system is behind a NAT firewall that does not translate H.323 or SIP, the NAT firewall’s public IP address is transmitted as own IP address in the H.323 or SIP protocol stack (not the private IP address). In this case, the public IP address must be defined. Bear in mind that the NAT firewall transmits the ports for signaling and voice data to the CELLX’s private IP address.
`VoipUseEnum=<mode>` Enter yes (default no) to activate an ENUM query to the called number before the call is set up via VoIP or PSTN. Using a standard DNS query, ENUM changes telephone numbers into Internet addresses. If a number is found, the call is set up via VoIP. If not, call setup occurs via PSTN or with another VoIP profile. NOTE: The query must include country and area codes.
`VoipUseIpStack=<mode>` Enter Yes to facilitate direct use of an xDSL or dial-up connection if the corresponding profile is defined. Default is No.
`VoipUseStun=<mode>` Enter yes (default yes) to use the STUN values for the VoIP profile.

9.4 Quality parameters

Table 9.6 Customized parameters: VoIP quality

VoIP Quality Parameters

SigTos=<num>

Enter a value between 0 and 255 (default is 0) to set the TOS (type of service) field in the SIP/SIPS packet IP header. Possible values are described in Table 9.7. If your IP network uses differentiated services, you can also define the DSCP (differentiated services codepoint) for the SIP/SIPS packets. The DSCP is the first six bits in the TOS octet.

NOTE: SigTos is set in the [System] section of the pabx.cfg file.

`VoipAGC=<x y z>`

This parameter allows automatic gain control of input signals from PSTN or IP. Enabling this feature compensates for near-far gain differences:
`x` - direction (`0` for signals from TDM, `1` for signals from IP)
`y` - gain slope (controls gain changing ratio in -dBm/sec, values 0 to 31, default 0)
`z` - target energy (determines attempted signal energy value in -dBm, values 0 to 63, default 19
Gain Slope:

0 - 00.25dB	1 - 00.50dB	2 - 00.75dB
3 - 01.00dB	4 - 01.25dB	5 - 01.50dB
6 - 01.75dB	7 - 02.00dB	8 - 02.50dB
9 - 03.00dB	10 - 03.50dB	11 - 04.00dB
12 - 04.50dB	13 - 05.00dB	14 - 05.50dB
15 - 06.00dB	16 - 07.00dB	17 - 08.00dB
18 - 09.00dB	19 - 10.00dB	20 - 11.00dB
21 - 12.00dB	22 - 13.00dB	23 - 14.00dB
24 - 15.00dB	25 - 20.00dB	26 - 25.00dB
27 - 30.00dB	28 - 35.00dB	29 - 40.00dB
30 - 50.00dB	31 - 70.00dB	

`VoipAutoRtpAddr=<mode>`

Some application scenarios require automatic RTP IP address and port detection for VoIP calls, for example if a firewall or NAT changes the IP address of incoming RTP data. Enter Yes to activate automatic detection. When No is set, RTP packets sources other than those processed are rejected. Default No.

`VoipBandwidthRestriction=<mode>`

Enter Yes to include the VoIP profile in traffic shaping. Default is No. For a description of the functionality, please refer to VoipMaximumBandwidth in Table 5.18.

`VoipBrokenDetectionTimeout=<ms>`

When this parameter is set, the system processes an interruption in the transmission of RTP/RTCP data in the VoIP connection following the set number of milliseconds (default 0). This parameter is necessary to set up an IntraSTAR call immediately when the IP connection is disrupted. Bear in mind that VoipSilenceSuppression=No must appear in the VoIP profile.

`VoipCallGroup=<name>`

All outgoing VoIP calls for VoIP profiles with the same VoipCallGroup name are distributed cyclically to these profiles.

`VoipConnBrokenTimeout=<sec>`

An entry is generated in the protocol.log file and the connection is terminated after a connection broken exists for the number of seconds entered (default 300). If `0` is entered, no entry will be generated and the connection will not be terminated.

`VoipDJBufMaxDelay=<count>`

Enter a value in milliseconds (0-320) to set a maximum jitter buffer limit (default 150). For fax transmission (t.38) it is fixed to 200ms.

NOTE: VoipDJBufMaxDelay must be greater than VoipDJBufMinDelay.

Table 9.6 Customized parameters: VoIP quality *(continued)*

VoIP Quality Parameters
`VoipDJBufMinDelay=<count>` Enter a value in milliseconds (0-320) to set a minimum jitter buffer limit (default 35). For fax transmission (t.38) it is fixed to 200ms. NOTE: VoipDJBufMaxDelay must be greater than VoipDJBufMinDelay.
`VoipDJBufOptFactor=<count>` Enter a value between 0 and 13 to set the balance between low frame erasure rates and low delay (default 7).
`VoipECE=<mode>` Enter yes (default) to set ITU G. 168 echo cancellation. Enter no to disable echo cancellation.
`VoipEcl=<ms>` This parameter defines the required tail length for echo cancelation. The following values in ms are possible: 32 64 (default) 128
`VoipInputGain=<num>` The volume of VoIP calls coming from ISDN or mobile. The range is 0-63. The default value of 32 is 0 dB.
`VoipIntrastar=<mode>` Enter Yes to activate the IntraSTAR feature. When the IP connection results in poor quality, an ISDN call is sent to the peer and the voice data is automatically transmitted via ISDN.
`VoipMediaWaitForConnect=<mode>` This parameter allows you to influence the system’s behavior in relation to voice channel negotiation (RTP stream). The following settings are possible: No (default): RTP data is transmitted immediately after negotiation for RTP. SIP: Early Media is activated; SDP is sent with 183 or 180. Yes: The negotiation of RTP data is sent only after the connection has been established. SIP: SDP is sent only with 200 and ack. Tone: The VoIP peer or the connected PBX requires generation of inband signaling tones (alert, busy, release). NOTE: If Tone is entered, the tones are not played in the direction of the PBX if RTP is already exchanged before connect (inband is switched through). Bear in mind that the parameter SWITCH in the VoIP controller’s Subscriber line must be removed if the tones are played for the PBX. If Tone is entered and the tones are played to VoIP, the VoIP media channel cannot be released following an ISDN call disconnect as long as the tones are being transmitted. This can result in CDR errors on the peer side.
`VoipOverflow=<name>` When the value entered in VoipMaxChan is reached, all overflow calls will be sent to the profile defined here. An alternative VoIP profile can also be used if the default profile can no longer be used as a result of poor quality.

Table 9.6 Customized parameters: VoIP quality *(continued)*

VoIP Quality Parameters
`VoipPCMPacketInterval=<int>` This parameter changes the default interval for PCM codecs (G.711, G.726). That means the VoipTxm factor is muliplied using this interval: For 16-channel chips: 0 = 20ms (default) 1 = 5 ms 2 = 10 ms 3 = 20 ms For 8-channel chips: 0 = 10ms (default)) 1 = 5 ms 2 = 10 ms 3 = 20 ms
`VoipQualityCheck=<type minsamples limit recovertime>` type: Enter one of the following: ASR1, ASR2, RoundTripDelay, Jitter or FractionLost **When type is ASR1 or ASR2**: minsamples: Minimum number of calls for which ASR shall be calculated with: limit: A value between 0 and 100 recovertime: Seconds to block the profile. **When type is RoundTripDelay**: minsamples: Minimum number of seconds RTD must be above: limit: The highest acceptable value for RTD (in milliseconds) recovertime: Seconds to block the profile. **When type is Jitter**: minsamples: Minimum number of seconds jitter must be above: limit: The highest acceptable value for jitter (in milliseconds) recovertime: Seconds to block the profile. **When type is FractionLost**: minsamples: Minimum number of seconds FL must be above: limit: The highest acceptable value for FL (percentage between o and 100) recovertime: Seconds to block the profile NOTE: If you base VoipQualityCheck on the ASR values: During setup, calls are calculated as not connected, which lowers the number of connected calls. Example: If minsamples is set at 20, with a limit of 80%, 4 calls in the setup phase will lower the ASR of the previous 20 calls to 80% and the profile will be blocked.
`VoipResetVocoderOnReinvite=<yes/no>` Enter yes to create an audible delay in a voice call. Default no.
`VoipRtcpTos=<num>` Enter a value between 0 and 255 to set the TOS (type of service) field in the RTCP packet IP header. Possible values are described in Table 9.7. If your IP network uses diferentiated services, you can also define the DSCP (differentiated services codepoint) for the RTCP packets. The DSCP is the first six bits in the TOS octet. NOTE: VoipUseIpStack must be 0 (default).
`VoipRtpTos=<num>` Enter a value between 0 and 255 (default is 0) to set the TOS (type of service) field in the RTP packet IP header. Possible values are described in Table 9.7. If your IP network uses differentiated services, you can also define the DSCP (differentiated services codepoint) for the RTP packets. The DSCP is the first six bits in the TOS octet.
`VoipSilenceSuppression=<mode>` Activates silence suppression (see Table 5.26).
`VoipT301=<sec>` An outgoing VoIP calls will be canceled in the state of Alerting (for H323) or Ringing (for SIP) if the number of seconds entered has passed and there is no response from the IP or VoIP carrier.

Table 9.6 Customized parameters: VoIP quality *(continued)*

VoIP Quality Parameters

`VoipT303=<sec>`
If this parameter is entered in a SIP profile, transmission of the INVITE is canceled after the number of seconds entered has passed (default 10). The call can then be redirected, for example to PSTN. This improves the reliability of the system when an IP or VoIP carrier's service fails.
EXAMPLE:
`Redirect340DF:=A`
`MapAllA=9`
`[Voip:DF]`
`.....`
`VoipT303=5`

`VoipT304=<sec>` An outgoing VoIP calls will be canceled in the state of Setup Acknowledge (for H323) or Trying (for SIP) if the number of seconds entered has passed and there is no response from the IP or VoIP carrier.
`VoipT310=<sec>` An outgoing VoIP calls will be canceled in the state of Call Proceeding (for H323) or Session Progress (for SIP) if the number of seconds entered has passed and there is no response from the IP or VoIP carrier.
`VoipTcpKeepAlive=<mode>` Enter yes (default) to send the RoundTripDelayRequest message every 10 seconds (necessary for long calls with firewalls using TCP aging).
`VoipVoiceVolume=<num>` The volume of VoIP calls coming from the Ethernet. The range is 0-63. The default value of 32 is 0 dB.

The following specifications for Quality of Service correspond with RFC791 and RFC1349.

Table 9.7 Quality of service values

Bit Distribution	0	1	2	3	4	5	6	7
	Precedence			TOS				MBZ
Bit	Description							
0-2	Precedence							
3	TOS: 0=normal delay, 1=low delay							
4	TOS: 0=normal throughput, 1=high throughput							
5	TOS: 0=normal reliability, 1=high reliability							
6	TOS: 0=normal service, 1=minimize monetary cost							
7	MBZ: must be 0 (currently not used)							
Precedence	Description							
111	Network control							
110	Internetwork control							
101	CRITIC/ECP							
100	Flash override							

Table 9.7 Quality of service values *(continued)*

011	Flash
010	Immediate
001	Priority
000	Routine

9.5 Compression parameters

The following parameters are for RTP multiplexing, which aggregates RTP packets (voice user data) for individual VoIP calls into a packet. The header (for Ethernet, IP, UDP and RTP) is sent only once for all calls instead of for each individual call. The relationship between header and payload benefits the payload when several calls occur simultaneously. This compression does not result in any loss in voice quality.

This feature is possible with a Teles peer and requires the following entries in the VoIP profile:

Table 9.8 Customized parameters: VoIP compression

VoIP Compression Parameters

`VoipAggOwnDataPort=<port>`
`VoipAggOwnDataPort=29500`
Enter the own port number used for aggregated packets. Default: 29500.

`VoipAggRemoteDataPort=<port>`
`VoipAggRemoteDataPort=29500`
Enter the port for the VoIP peer that is used for aggregated packets (compressed data). Default: 29500.

`VoipAggRemoteRtpPort=<port>` Enter the port for the VoIP peer that is the first RTP port. The next port is always the corresponding RTCP port. The port that is two numbers higher will be used for the next VoIP channel. Default 29000.
`VoipAggRemoteRtpPortSpacing=<count>` Defines the space between the ports used for the peer’s individual RTP streams (default 2).

9.6 Fax/modem parameters

Table 9.9 Customized parameters: VoIP fax

VoIP Fax/Modem Parameters

`VoipAPartyCanSwitchToT38=<mode>`
Enter Yes (default) to activate fax detection in both directions. When No is set, fax detection and translation to T.38 are carried out only when the call comes from VoIP.

`VoipFaxBypassPayloadType=<num>`
Defined the payload type for a fax's RTP packets when T.38 is not used (default 102).

`VoipFaxDisableAfterConnect=<sec>`
Deactivates fax detection after the number of seconds entered has passed. This is intended to avoid conversional disruption resulting from fax transmission noises when a fax machine is next to a phone that has an established call.

`VoipFaxECM=<mode>`
You can use this parameter to enable the error correction mode for fax transmission: `yes`=enabled, `no`=disabled (default).

`VoipFaxMaxRate=<num>`
If the peer does not support auto negotiation or has a fixed transmission rate, you can define the fixed rate:
0 - 2400 Bit/sec   1 - 4800
2 - 7200   3 - 9600 (default)
4 - 12000   5 - 14400
EXAMPLE:
`VoipFaxMaxRate=5`

`VoipFaxTransport=<int>`
0 = fax detection is switched off, no codec fallback at all (default).
1 = signaling will switch to T.38 (framesize 40ms). The codec will change when the system detects a fax or modem connection on the channel.
2 = fallback to G711a (framesize 40ms).

NOTE: Bear in mind that if T.38 is defined in the `VoipCompression=<list>` parameter of the VoIP profile, the system will switch only when it detects a modem connection. Fax calls will still be transmitted using T.38.

`VoipSuppressInitialT38Signalling=<mode> (SIP only)`
Enter Yes to suppress the SDP header m=image t38 in all SIP messages until reinvite of the actual fax detection occurs (default No).

`VoipT38Enforce=<mode>`
Enter Yes to switch to T38 although the B party does not signalize T38. (default Yes).

`VoipT38Version=<int>`
0 = T38 version 0 (default)
1 = T38 version 1
2 = T38 version 2002 ASN1 syntax TPKT enabled
3 = T38 version V34 V33 support 2002 ASN1 syntax TPKT enabled. Is needed to send v34 faxes via T38.

The following parameters are responsible for setting the modem transport method if a modem connection is detected.

`VoipV21Transport=<mode>`
`0`=disabled (must be set to 0).

`VoipV22Transport=<mode>`
`0`=disabled (default), `2`=bypass.

`VoipV23Transport=<mode>`
`0`=disabled (default), `2`=bypass.

Table 9.9 Customized parameters: VoIP fax *(continued)*

VoIP Fax/Modem Parameters
`VoipV32Transport=<mode>` `0`=disabled (default), `2`=bypass .
`VoipV34Transport=<mode>` `0`=disabled (default), `2`= bypass.

9.7 DTMF parameters

Table 9.10 Customized parameters: VoIP DTMF

VoIP DTMF Parameters
`VoipComprDtmfInband=<list>` This parameter always forced DTMF-tone inband transmission for the configured voice codes, regardless of what is configured for: VoipDtmfTransport=. The same codecs can be configured as for VoipCompression=<list>
`VoipDtmfFallback=<int>` If VoipDtmfTransport=3 is set and the peer does not support DTMF transmission according to RFC 2833, the following settings apply: 2 = automatic fallback to inband 0 = automatic fallback to signaling messages (default)
`VoipDtmfTransport=<int>` 0 (H323) = DTMF relayed with H.225 signaling information. 0 (SIP) = DTMF relayed with SIP INFO. 1 = DTMF taken from audio stream and relayed to remote, meanwhile no rtp data is sent. 2 (default) = DTMF and MF kept in audio stream and not relayed. 3 = DTMF taken from audio stream and relayed to remote as per RFC2833, meanwhile rtp data with silence is sent. 4 = For SIP, SIP INFO messages will be converted to DTMF. For H.323, the H.245 user input will be converted to DTMF. 6 (H.323 only) DTMF relayed with H245 user input. 8 DTMF taken from audiostream to be further processed in the gateway.
`VoipIBSDetectDir=<int>` Enter `1` and DTMF tones (and all other inband signaling) will be detected from the Ethernet side. Enter `0` for DTMF tones to be detected from the PCM side (default). DTMF tones from the Ethernet side are transmitted to the host as ISDN dialing information only if 1 is entered. In this case, set VoipDtmfTransport to 1 or 3. NOTE: If `1` is entered, fax detection is not supported.
`VoipMinDigitOnTime=<ms>` Defines the minimum length of DTMF tones, to ensure DTMF tone detection. Default 0.
`VoipMinInterDigitTime=<ms>` Sets a time interval for DTMF tone detection. Default 0.
`VoipRFC2833PayloadType=<num>` This parameter changes the DTMF payload type. The default value is 96, a common value is 101.

10 System maintenance and software update

10.1 Configuration errors

When typographical errors are made in the configuration files, an entry appears in the `protocol.log` when the configuration is activated. This entry includes the line number and its contents.

10.2 Status and error messages

The `protocol.log` file – assigned as the file for logging the protocol in the configuration file (`ActionLog=`*file*) – contains information on all activities within the system.

In the example below, you can see that all activities are recorded beginning with the date and time. If functions were activated by key combinations from terminal devices you can identify these along with the service ID.

Example 10.1 Status and error messages

```
16.05.06-11:51:31,[990]Start STATUS - TELES.iGATE V11.7a (007f)
16.05.06-12:10:57,[01A]ERR: Layer1
16.05.06-12:10:58,[000]ERR: OK
16.05.06-12:10:58,[010]ERR: OK
16.05.06-12:12:06,Remote Control from IP 192.168.1.2
16.05.06-12:12:06,Remote Control: OK
16.05.06-12:12:16,Activate Configuration System
16.05.06-12:16:26,Remote Control Terminated
16.05.06-14:00:00,Activate Configuration Night2
16.05.06-14:00:00,Time Switch Operation
16.05.06-18:00:00,Activate Configuration Night3
16.05.06-18:00:00,Time Switch Operation
```

Table 10.1 Event log messages

Message	NMS	Definition
Status Program		
[990] Start STATUS	X	TELES system software and status program have been started.
System Start		
[999] System-Boot	X	System restarted by timer.
[999] Remote Control: Reboot		System restarted by remote administration command.
Configuration Changes		
Activate configuration <num> OK		Configuration `<num>` successfully loaded. Initiator displayed in next line.
Activate configuration <num> failed [<err>]		Configuration `<num>` could not be loaded.
Remote Control: Date & Time changed		Date and/or time were changed via remote administration.

Table 10.1 Event log messages *(continued)*

Message	NMS	Definition
Time Switch Operation		The configuration change was made by the timer.
Remote Administration		
Remote Control from <peer>, <RemoteCode>, <service>, 0		Remote administration access from number or IP address.
Remote Control: OK		Successful remote administration access.
[993]Remote Control: wrong password	X	Remote administration access was denied because of a wrong password.
[994]Remote Control: wrong number	X	Remote administration access was denied because the call originated from an unauthorized number (RemoteOrigination).
Remote Control Terminated <start time>,<end time>, <num>, <RemoteCode>, <service>, 0		Remote administration session from <num> ended. Session length is indicated by start time and end time.
Errors Reported by the Status Program		
[<port><i>] ERR: Problem at Port <num>	X	A Layer 1 or Layer 2 error occurred on `<num>`. <i> indicates error type: A Layer 1 error ; Layer 2 error 0 Layer 1&2 operational. 4 RSSI (for mobile only) If the error persists, a differentiation is possible through 'status of the ports'. If this message appears, status inquiry connections via remote administration are accepted and NMS downloads the protocol.log file. NOTE: If the RSSI falls below the value configured in the pabx.cfg, the port will shut down automatically.
Attention: No Callback-Call <num> Arrived		Callback with DTMF: the Callback Provider <num> did not call back within approx. 20 sec. Direct Line Access with DTMF: the call was accepted but disconnected again within x sec. (as defined by MapCallBackWaitDisc).

Table 10.1 Event log messages *(continued)*

Message	NMS	Definition
Write error		Access to the disk drive on which the data is to be stored was not possible because it is set for read-only, full or because of faulty hardware or software.
[995] Msg-Memory > 75%	X	This message appears when message memory is over 75% full. If this message appears, status inquiry connections via remote administration are accepted and NMS downloads the protocol.log file.

The following options are available for monitoring the Mobile Boards' status or the status of each mobile channel. You can access status information through data recorded in the `protocol.log` file or in the **Layer 1** column in the GATE Manager's **Port Status** window.

The following status and error messages appear in the `protocol.log` file when `ALARM` appears in the VoIP port's subscriber line:

Table 10.3 Protocol log status and error messages

Message	Definition
System Configuration (a)	
config: <num> duplicate profile	Specified line in pabx.cfg or route.cfg contains duplicate profile.
config: <num> invalid	Specified line in pabx.cfg or route.cfg is invalid.
config: evaluation errcode <num>	Internal error.
Port-Specific Entries	
[<port>]Unblock Port	The <port> has been unblocked. This can occur via remote access for all controller types.
[<port>]Block Port	The <port> has been blocked. This can occur via remote access for all controller types.
[<port>]Restart Port	The <port> has been blocked. This can occur via remote access for all controller types.
Ethernet Interface	
[99d]ERR: emac<num><state>	The Ethernet controller's status is checked every minute and any change in status is noted. <num> Number of the EMAC interface (0 or 1). <state> up Ethernet link is active down Ethernet link is inactive
!resolve ip-address	ARP request for specified IP address failed.

Table 10.3 Protocol log status and error messages *(continued)*

Message	Definition
pingcheck failed	Ping to configured server failed for configured amount of time; host might reboot this port.
Voice Packetizer Task (b)	
[<port>]ERR: OK, <count> devices	The number (<count>) of DSPs were loaded during startup without errors. The first VoIP controller appears in [<port>].
[<port>]ERR: init failed	A DSP could not be loaded. This DSP or the first VoIP controller is defined in [<port>].
VP: <channel> <msg>	Voice-packetizer chips report fatal error on specified channel, with specified message.
VoIP (c)	
GK <name> URC	Successful UnRegister from specified gatekeeper.
GK <name> GRJ <num>	GatekeeperRequest was rejected
GK <name> RCF	Successful RegistrationRequest (Registration-Confirm).
GK <name> RRJ <num>	RegistrationRequest was rejected.
GK <name> ARJ <dad> <num>	AdmissionRequest was rejected.
GK <name> !ACF dad	AdmissionRequest was not answered.
GK <name> !GCF	GatekeeperRequest was not answered.
no profile for ipaddress	Incoming VoIP call from specified IP address was rejected due to no matching VoIP profile.
registrar <name>: registration done	Successful registration at SIP registrar.
registrar <name>: wrong auth-type <num>	Registrar does not perform MD5 for authentication.
registrar <name>: gives no nonce	Nonce missing in response from registrar (possible error in registrar configuration).
registrar <name>: registration forbidden	Registration with specified registrar is not allowed.
registrar <name> not answering	Specified registrar does not respond.
voipconn oad->dad broken	Voice codec chips report broken RTP connection.
voip FdInitAll failed <cause>	Internal failure.
voip ISDNListen failed	Internal failure.
voipIpSocketInit failed	Internal failure.
!DNS-lookup <hostname>	DNS lookup for specified host name failed (DNS not activated? Missing or invalid DNS server?).
message from <ip addr> not decodable	H323, ASN1 packet cannot be decoded.
vGATE	

Table 10.3 Protocol log status and error messages *(continued)*

Message	Definition
[99]ERR: SimUnit !connect	An outgoing connection to the vGATE SIM Unit could not be established.
[99]ERR: ControlUnit <ip addr> !connect	An outgoing connection to the vGATE Control Unit could not be established.
Number Portability	
[99i]ERR: np !connect	Connection to the iMNP could not be established.
[99i]ERR: np connect <ip addr>	Connection to the iMNP reestablished.
System Kernel (e)	
task <name> suspended	specified task was suspended due to internal error; host might reboot this port.
Mail (f)	
cdr !connect <ip addr>	sending CDR: TCP connect to specified IP address failed.
mail !connect <ip addr>	sending e-mail: TCP connect to specified IP address failed.
Radius (g)	
!DNS-lookup <hostname>	DNS lookup for specified host name failed (DNS not activated? Missing or invalid DNS server?).
timeout auth <ip addr>	Authentication request to specified Radius server failed due to timeout.
timeout acnt <ip addr>	Accounting request to specified Radius server failed due to timeout.
!rsp-auth <ip addr>	Response authenticator from specified Radius server was invalid (wrong secret/password?).
!auth <ip addr> <num>	Authentication denied by specified Radius server.
Configuration Errors in the `ip.cfg`	
Error in `ip.cfg` line <line>: section [<section_name>] unknown	
Error in `ip.cfg` line <line>: parameter `"`<parameter_name>`"` in [<section_name>] unknown	
Error in `ip.cfg` line <line>: parameter `"`<parameter_name>`"` does not belong to any Section	
There is an error in the NAT Configuration The NAT was not loaded, please check the Configuration for mistakes	
There is an error in the DHCPD Configuration The DHCP SERVER was not loaded, please check the Configuration for mistakes	
There is an error in the ALTQD Configuration The ALTQD SERVER was not loaded, please check the Configuration for mistakes	
There is an error in the FIREWALL Configuration The FIREWALL was not loaded, please check the Configuration for mistakes	
Error in <dsl_interface> Connection failed. Please, connect a cable in the <ethernet> port	

Table 10.3 Protocol log status and error messages *(continued)*

Message	Definition
Error in <dsl_interface>: Connection Failed. Please, revise your Username/Password configuration	
Error in <dsl_interface>: Connection Failed. Please, revise the DSL Modem	

10.3 Software update

You may find that you would like to implement features that are only possible with a more recent software version. To update the software on your system, follow these instructions:

Follow the below listed hints:

- Make sure no traffic is running on the system while updating the system. Do not turn the system off during the update.
- Never mix the different driver versions.
- Make sure there is enough available memory for the new version. We recommend that you delete unnecessary log files and backups.
- Do NOT delete or rename existing software files before updating.
- Upload the new files ONLY via GATE Manager. Do not use any other process (e.g. FTP) to update the software files. This can lead to irreversible damage to the operating system.
- If an error message appears during the update process, do NOT restart or turn off the system! Make a note of the error message and the update steps that have been taken and contact TELES service.

1. To get the necessary software files for your system, you need to contact your local sales representative.
2. To update your system software, download the following software files to your hard disk:
   `start`
   `netbsdz`
   `netbsdfs.gz`
   `ip4.vnd`
   `netbsdi`
   `xgate.vnd`
   and one of the following:
   CELLX GSM: `igate.tz1` or
   CELLX CDMA: `cgate.tz1` or
   CELLX UMTS: `igate.tz1`
3. To update your GUI, download the following software files to your hard disk:
   `httpd.izg`
   `httpd.tz2`
4. Start GATE Manager and connect to the system you want to update.
5. Check the software version running on your system to make sure the one you want to install is newer.
6. In the navigation bar, click **Directory**.
7. Right-click the **Directory** window and click **Send To System**.
8. Once the files have been completely transferred, check the file size and reboot the system.

9. As soon as you can reach the system via GATE Manager again, check the version number of the running software.

These files form a unit and belong to the same software version. To avoid compatibility conflicts, check with TELES service before you update the software.

An update of the following optional function modules occurs in the same way. For a description of how to update the software, please refer to Chapter 10.3 Software update.

Make sure the file extension has the same running number as that of the file on the system:

- SNMP agent: snmpd.tz0
- DNS forwarder: dnsmasg.tz2
- IP update - DynDNS client: ipupdate.tz2

Since these features are only required in individual cases, they are not part of the default software packet. They can be installed as stand-alone modules for the desired function. The description of the functionality of individual modules appears in their respective chapters.

Following completion of transmission, you must adjust the module’s configuration and reboot the CELLX. Once you have rebooted the system, you can use the required features.

10.4 SNMP agent

The following software package must be installed: snmpd.tz0.

This module allows you to connect the systems and their functions to an SNMP-based network monitoring system. With this module, SNMP requests are answered and alarm messages (for example Layer 1 errors on E1 lines) and error recovery messages are sent via SNMP trap. The trap consists of a generic trap `linkUp` or `linkDown`, and a specific trap.

Table 10.4 contains a list of the most easily verifiable specific traps:

Table 10.4 Specific trap

Error Code	Definition
0	Port registered/up
1	Wrong PIN
4	Bad RSSI
7	Unknown error, port restarted
9	Port restart
11	Layer 2 problem
12	SIM blocked
14	SIM barred

Table 10.4 Specific trap *(continued)*

Error Code	Definition
17	Layer 1 problem / port down
18	SIM missing
21	Registration failed
39	Call limit reached
40	SMS limit reached
41	Assign limit reached
42	SIM inserted
47	ACD limit reached
48	SIM removed
0	eth0 / eth1 up
52	eth0 / eth1 down

SNMP requests retrieve the value of a variable or list of variables (GET/GETBULK) referred to as OIDs. Each OID identifies a variable that can be read via SNMP, as described in Table 10.5.

Table 10.5 Description of OIDs

OID	Description	Type
1.3.6.1.4.1.2170.1.2.1.1.1	Port number	INTEGER
1.3.6.1.4.1.2170.1.2.1.1.2	Port address	DisplayString
1.3.6.1.4.1.2170.1.2.1.1.3	Port type	DisplayString
1.3.6.1.4.1.2170.1.2.1.1.4	State of port line 1	DisplayString
1.3.6.1.4.1.2170.1.2.1.1.5	State of port line 2	DisplayString
1.3.6.1.4.1.2170.1.2.1.1.6	Number of currently active connections on port	INTEGER
1.3.6.1.4.1.2170.1.2.1.1.7	Overall state of port	INTEGER
1.3.6.1.4.1.2170.1.2.1.1.8	Calls rejected due to no channel available	INTEGER
1.3.6.1.4.1.2170.1.2.1.1.9	Calls rejected due to busy	INTEGER
1.3.6.1.4.1.2170.1.2.1.1.10	Calls rejected due to no user	INTEGER
1.3.6.1.4.1.2170.1.2.1.1.11	Calls disconnected	INTEGER
1.3.6.1.4.1.2170.1.2.1.1.12	Outgoing calls	INTEGER
1.3.6.1.4.1.2170.1.2.1.1.13	Incoming calls	INTEGER
1.3.6.1.4.1.2170.1.2.1.1.14	Channels in use more than 90 percent	INTEGER
1.3.6.1.4.1.2170.1.2.1.1.15	Accumulated call duration	INTEGER

Table 10.5 Description of OIDs *(continued)*

OID	Description	Type
1.3.6.1.4.1.2170.1.2.2.1.1	Number of Ethernet controller	INTEGER
1.3.6.1.4.1.2170.1.2.2.1.2	Overall state of Ethernet controller	INTEGER
1.3.6.1.4.1.2170.1.1.1.1.1.1	Number of VoIP profile	INTEGER
1.3.6.1.4.1.2170.1.1.1.1.1.2.1	Name of profile 1	DisplayString
1.3.6.1.4.1.2170.1.1.1.1.1.2.2	Name of profile 2	DisplayString
1.3.6.1.4.1.2170.1.1.1.1.1.2.3	Name of profile 3	DisplayString
1.3.6.1.4.1.2170.1.1.1.1.1.3	Peer address of profile	INTEGER
1.3.6.1.4.1.2170.1.1.1.1.1.3.1	Peer address of profile 1	DisplayString
1.3.6.1.4.1.2170.1.1.1.1.1.3.2	Peer address of profile 2	DisplayString
1.3.6.1.4.1.2170.1.1.1.1.1.3.3	Peer address of profile 3	DisplayString
1.3.6.1.4.1.2170.1.1.1.1.1.4	Registration state of profile	INTEGER
1.3.6.1.4.1.2170.1.1.1.1.1.4.1	Registration state of profile 1	INTEGER
1.3.6.1.4.1.2170.1.1.1.1.1.4.2	Registration state of profile 2	INTEGER
1.3.6.1.4.1.2170.1.1.1.1.1.4.3	Registration state of profile 3	INTEGER
1.3.6.1.4.1.2170.1.1.1.1.1.5	Profile registration cause	INTEGER

Traps are generated for all line or mobile ports. The running number in the trap corresponds with the port. The module also monitors whether the voice codec chips are functioning correctly.

The traps for the IP interfaces are also generated in ascending order according to the following list:

Table 10.6 Traps for IP interfaces

Port Number	Interface
0	Ethernet 1
1	Ethernet 2
2	Loopback
3	xppp= (if used)
4	pppoe= (if used)

If more than one pppoe<x> profile is configured, the number will also increase.

Bear in mind that the keyword ALARM must be entered in the appropriate PRI, BRI or mobile port's Subscriber line in the pabx.cfg. The MIBs (Management Information Bases) are included on the product CD in the folder MIB. The module name snmpd.tz0 must have the ending tz0!

The following settings are possible in the [snmpd] section in the `pabx.cfg` file:

Table 10.7 Settings in the `[snmpd]`section

Parameter	Definition
Port=<port>	Defines the target port for the `get` service (default 161). The port for the trap server is fixed at 162.
TrapServer=<ip addr>	Enter the SNMP trap server's IP address. Example for listing more than one: `TrapServer=192.168.0.10 192.168.0.12`
Community=<password>	Enter a password for a community (group). The default password is `public`. The current implementation evaluates the configured password only when sending a trap. GET requests use the default value.

Example 10.2 Settings in the `[snmpd]`section

```
[snmpd]
TrapServer=192.168.0.1
Community=ABCDE
```

10.5 DNS forwarder

The following software package must be installed: dnsmasg.tz2.

With this module, the system can function as a DNS server for the clients in the local network. The system in the local network sent the DNS query to the CELLX, which forwards the queries to a known DNS server address if no valid entry for the query is known.

The advantage is that the clients always enter the CELLX's address as DNS server address, so that no public DNS server address is required. The CELLX functions in this scenario as a router.

Of course, the DNS server's address can also be transmitted to the clients using the integrated DHCP server. If the CELLX is used as a DSL router or if it sets up a dial-up connection, no entry is required in the pabx.cfg for the parameter NameServer. The DNS server's address that is negotiated through this connection will be used.

10.6 ipupdate - DynDNS client

The following software package must be installed: ipupdate.tz2.

This function allows you to assign a defined hostname to an IP address that changes dynamically. That means that you can always reach a device or service through the public IP network, even if, for example, it is a common DSL connection with dynamic IP address allocation. Several providers support this service.

Make the following entries in the system's `pabx.cfg`, in the [DynDNS] section:

Table 10.8 pabx.cfg: DynDNS

DynDNS Parameters

`service=<type>`
Specifies which provider is used. The following providers are supported:

dhs	http://www.dhs.org
dyndns	http://www.dyndns.org
dyndns-static	
dyns	http://www.dyns.cx
ezip	http://www.ez-ip.net
easydns	http:/www.easydns.com
easydns-partner	
gnudip	http://www.gnudip.cheapnet.net
heipv6tb	
hn	http://www.hn.org
pgpow	http:www.justlinux.com
ods	http://ods.org
tzo	http://www.tzo.com
zoneedit	http://zoneedit.com

`user=<username:password>`
Defines the username and password for the DNS service provider.

`host=<domain_name_of_dns_service>`
Enter the domain name that is used.

`interface=<If>`
Defines the interface to be used. Possible entries are `emac0`, `emac1`, `pppoe0`. The dynamic IP address for this interface is transmitted to the service provider.

`max-interval=<sec>`
Defines the value in seconds in which actualization of the name in the DNS database must occur. 2073600 seconds (24 days) is the default value. The shortest interval allowed is 60 seconds. Bear in mind that this setting may cause the provider to block the domain name, since multiple registrations in short intervals are often not allowed. You must clear this with your provider.

In the following example, the DynDNS service is used and the domain name is host.domain.de; the username is user and the password is pwd. The CELLX works as DSL router and the dynamically allocated IP address of the PPPoE interface is used.

Example 10.3 DynDNS

```
[DynDNS]
service=dyndns
user=user:pwd
host=host.domain.de
interface=pppoe0
max-interval=2073600
```

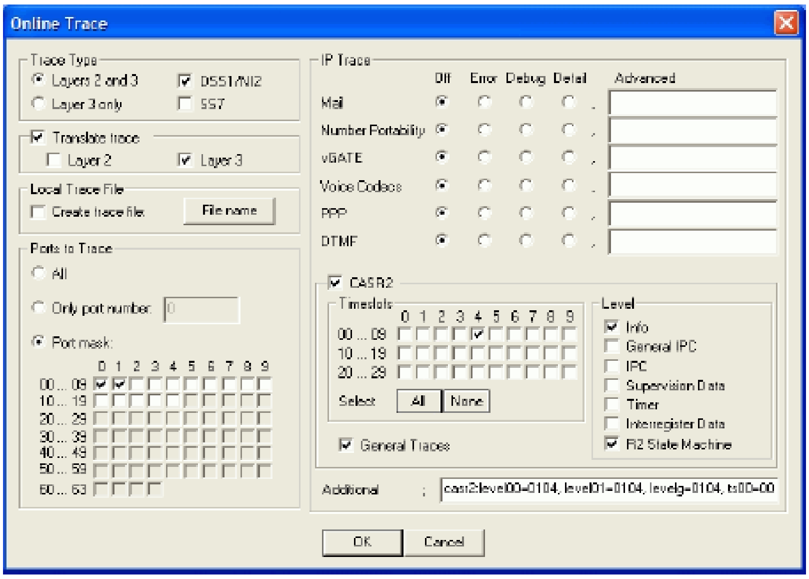
Included in the possible uses for this feature is remote access to the CELLX when the IP connection does not have a fixed IP address. In this case, you can access the system, for example with the GATE Manager, if the host name is used in the Remote Number dialog. Example entry in the Remote Number dialog: IP:host.domain.de.

10.7 Trace

During operation, the trace readouts of the CELLX can be saved in a file or transmitted with remote maintenance directly. The trace options must be turned on in the GATE Manager (offline or online trace) or via FTP raw commands (see Chapter 4.10.3 FTP). Trace results presented here are for PRI,VoIP, GSM/CDMA/UMTS interfaces and for the following services in various levels:

Table 10.9 Trace options

Option	Definition
Mail	Output for all SMTP packets.
NumberPortability	Output of all packets for communication with the iMNP.
VoiceCodecs	Output of RTCP information described under VP module.
PPP	Output of PPP connection information.
DTMF	Output for DTMF tone detection.
Remote	Output for GATE Manager and NMS communication. For offline traces only.



Online Trace

Trace Type
● Layers 2 and 3 ☑ DSS1/NI2
○ Layer 3 only □ SS7

☑ Translate trace
□ Layer 2 ☑ Layer 3

Local Trace File
□ Create trace file

Ports to Trace
○ All
○ Only port number: 0
● Port mask:

	0	1	2	3	4	5	6	7	8	9
00 ... 09	☑	☑	□	□	□	□	□	□	□	□
10 ... 19	□	□	□	□	□	□	□	□	□	□
20 ... 29	□	□	□	□	□	□	□	□	□	□
30 ... 39	□	□	□	□	□	□	□	□	□	□
40 ... 49	□	□	□	□	□	□	□	□	□	□
50 ... 59	□	□	□	□	□	□	□	□	□	□
60 ... 63	□	□	□	□						

IP Trace

	Off	Error	Debug	Detail		Advanced
Mail	●	○	○	○	,	
Number Portability	●	○	○	○	,	
vGATE	●	○	○	○	,	
Voice Codecs	●	○	○	○	,	
PPP	●	○	○	○	,	
DTMF	●	○	○	○	,	

☑ CAS R2

Timeslots

	0	1	2	3	4	5	6	7	8	9
00 ... 09	□	□	□	□	☑	□	□	□	□	□
10 ... 19	□	□	□	□	□	□	□	□	□	□
20 ... 29	□	□	□	□	□	□	□	□	□	□

Level
☑ Info
□ General IPC
□ IPC
□ Supervision Data
□ Timer
□ Interregister Data
☑ R2 State Machine

☑ General Traces

Additional ; casr2:level00=0104, level01=0104, levelg=0104, ts00=00

Figure 10.1 GATE Manager: online trace activation window

CELLXs offer two different types of trace:

- Online - trace information is immediately displayed in the GATE Manager’s trace window.
- Offline - trace information is written to a file on the CELLX.

CELLX systems create trace files when the TraceLog=file entry is present in the pabx.cfg. Traces can be activated via remote administration (GATE Manager or FTP).

The following table describes the fields that are on the online trace activation window. For offline traces, some fields are not provided.

Table 10.10 Online trace activation window

Field	Description
Trace Type	
Layer 2 and 3	Select this option if the trace is to contain layer 2 and 3 messages.
Layer 3 only	Select this option if the trace is to contain layer 3 messages only.
DSS1/NI2	Select this option if the trace is to contain DSS1/NI2 messages only.
SS7	For SS7 messages, select this option.
Translate Trace	
Layer 2	To translate layer 2 messages into plain text, select this option.
Layer 3	To translate layer 3 messages into plain text, select this option.
Local Trace File	
Create Trace File	Check this box if you want to store a trace file locally on your computer.
File Name	Click this button to change the default directory and file name of the local trace file.

Table 10.10 Online trace activation window *(continued)*

Field	Description
Ports to Trace	
All	Select this option to trace all ports.
Only Port Number	Enter here the port which you want to trace.
Port Mask	Select from this port mask which ports you want to trace.
IP Trace	
Mail, Number Portability, vGATE, ...	Contains a list of port-independent trace options. The different options can be limited to error messages, debug messages, or detailed trace output. The Advanced field is for TELES support usage only. For an explanation of the different trace option please refer to Chapter 10.9 Trace options on page 157.
CASR2	
Timeslots	Select which of the channels you want to trace for the CASR2 ports.
General Trace	Check this box if you want to run a general trace for the different timeslots.
Level	Select which special type of information you want to trace for the different timeslots.
Additional	The following additional entries are possible. The number behind the colon referes to the trace level. 0 = no traces, 1 = few traces, 2 = many traces, 3 = very many traces (only nstanl). autodial:0 for auto dial trace messages fctrans:trclevel=debug for ISDN facility to SIP converter messages nstanl:0 for call routing trace messages radius:0 for RADIUS trace messages rtman:0 for routing manager trace messages s0:0 for BRI trace messages s2m:0 for PRI and switching matrix trace messages voip:noreg to suppress traces of REGISTER and responses

Please bear in mind that the volume of trace readouts can grow quite large, so that faulty transmission of the trace data may result with remote maintenance. A trace at full capacity can cause the system to crash.

Trace output format

The following entries appear at the beginning and end of each trace:

- DD.MM.YY-hh:mm:ss.ss, Start
- DD.MM.YY-hh:mm:ss.ss, End
  - DD = day
  - hh = hour
  - MM = month
  - mm = minute
  - YY = year
  - ss.ss = hundredths of seconds

Traces appear in the following format:

- [<hh:mm:ss>] <module>[<port>]: <trace>
- <module>
  - s = send for PRI/BRI or mobile ports
  - r = receive for PRI/BRI or mobile ports
  - x = send to VoIP destinations
  - y = receive from VoIP destinations
  - i = information messages and internal trace outputs between VoIP and the other interfaces (ISDN, mobile)
  - a = VoIP controllers RTCP output
  - m = mail output
  - g = remote output
- <port>
  - port number (controller number in the pabx.cfg) or 255 if a service is used
- <trace>
  - output in the defined syntax for the module

10.7.1 ISDN trace output

Trace output for DSS1 are in hexadecimal notation. You can use the external tool TraceView.exe to translate offline trace output. You will find the tool in the **Software** folder on the enclosed CD. The GATE Manager's trace window can also display translated online traces.

The following example shows an untranslated DSS1 trace.

Example 10.4 ISDN trace output

```
17.05.06-09:54:40,Start 11.7a (L3)
[09:55:14.58] r[00]: 00 01 02 02 08 02 00 02 05 04 03 80 90 a3 18 03 a1 83 81 6c
02 81 31 70 06 81 31 32 33 34 35 7d 02 91 81
[09:55:14.58] s[00]: 02 01 02 04 08 02 80 02 0d 18 03 a9 83 81
[09:55:14.58] s[01]: 00 01 a8 9a 08 02 00 46 05 04 03 80 90 a3 18 03 a1 83 89 6c
02 81 31 70 06 81 31 32 33 34 35 7d 02 91 81
[09:55:14.58] r[01]: 02 01 9a aa 08 02 80 46 0d 18 03 a9 83 89
[09:55:14.86] r[01]: 02 01 9c aa 08 02 80 46 01
[09:55:14.86] s[00]: 02 01 04 04 08 02 80 02 01
[09:55:16.73] r[01]: 02 01 9e aa 08 02 80 46 07 29 05 05 07 01 09 33 4c 07 01 81
31 32 33 34 35
[09:55:16.73] s[01]: 00 01 aa a0 08 02 00 46 0f
[09:55:16.73] s[00]: 02 01 06 04 08 02 80 02 07 29 05 05 07 01 09 32 4c 07 01 81
31 32 33 34 35
[09:55:16.73] r[00]: 00 01 04 08 08 02 00 02 0f
[09:55:44.30] r[00]: 00 01 06 08 08 02 00 02 45 08 02 80 90
[09:55:44.35] s[01]: 00 01 ac a0 08 02 00 46 45 08 02 80 90
[09:55:46.71] r[01]: 02 01 a0 ae 08 02 80 46 4d
[09:55:46.71] s[01]: 00 01 ae a2 08 02 00 46 5a
[09:55:46.71] s[00]: 02 01 08 08 08 02 80 02 4d
[09:55:46.71] r[00]: 00 01 08 0a 08 02 00 02 5a
17.05.06-09:51:33,End
```

10.7.2 GSM/CDMA/UMTS trace output

The trace output for GSM appears in hexadecimal notation. Its format is the same as that for ISDN output. Table 10.11 and Table 10.12 describe the contents of GSM trace output.

Table 10.11 Request messages to the GSM module

Hex Value	Description
00	Setup
01	Connect
02	Disconnect
03	SMS
04	DTMF
05	Set Config
06	Get Config
07	LED
08	Restart
09	Switch SIM

Table 10.12 Incoming, indication message from the GSM module

Hex Value	Description
0B	Alert
0C	Voice Indication
0D	Connect
0E	DTMF
0F	Setup
10	Disconnect
11	SMS
12	SMS Confirmation
13	Error
16	Get Config Confirmation
18	Dial-End Call Proceeding
19	USSD
1A	Restart Indication

The following example shows a GSM call through the fourth GSM controller.

Example 10.5 GSM call through fourth GSM controller

```
Status Request
[14:57:51.80] s[04]: 06
Status Information:
[14:57:51.80] r[04]: 16
Setup Request:
[14:57:52.29] s[04]: 00 4c 93 04 00 00 00 35 36 36 37 00 35 38 2c 36 34 36 2c 33
30 2c 2c 2c 30 2c 2c 2c 30 2c 32 36 32 2c 30 37 2c 00 72 64 09 75 70 20 7b 64 35
7d 20 27 2e 2e 2b 43 43 45 44 3a 20 32 36 32 2c 30 37 2c 34
Dial End:
[14:57:55.47] r[04]: 18
Alert:
[14:57:55.63] r[04]: 0b
Connect Indication:
[14:57:56.63] r[04]: 0d
Disconnect Request:
[14:59:54.13] s[04]: 02 4c 93 00
Disconnect Indication:
[14:59:54.19] r[04]: 10
```

10.7.3 VoIP trace output

As described above in Chapter 10.7 Trace, there are four modules for VoIP traces. The groups x (send), y (receive) and i (information and internal output) appear when a Layer2 or Layer3 offline or online trace is started. Group a (RTCP output) only appears when the module Voice Codecs is active.

Particularly in the case of VoIP connections (protocols H.323 and SIP), the trace output is quite extensive and abbreviations make it difficult to keep track of the results. The following list contains a description of H.323 output.

Output for the signaling protocol SIP is transmitted in ASCII and translated for better legibility. Since they are displayed unabridged, no description is necessary. Information and internal output traces correspond with the H.323 output and are described in the following tables. For ENUM, please refer to Chapter 10.7.3.5 ENUM output.

In general, the following rules apply for this trace output:

Table 10.13 H.323 output

Packet	Description
h225	H.225-protocol messages.
h245	H.245-protocol messages.
pstn	Messages of the internal protocol interface that provides the interface to the other interfaces PRI, BRI and GSM.
rcv	Coming from the IP network or the internal protocol interface; appears with <dir> in the trace lines.
snd	Sending to the IP network or the internal protocol interface; appears with <dir> in the trace lines.

The information is thoroughly analyzed where it is received (all rcv messages).

10.7.3.1 Interface IP network

Establish H.323 session

Usually there is trace output that displays a new H.323 session. The direction is crucial (whether the call is going into or coming out of the IP network).

The outut syntax is `h225connect to <ip address> cr <cr> s <si>` for calls going into the IP network and `h225accept from <ip address> s <si>` for calls coming out of the IP network.

Table 10.14 H.323 session

Trace Output	Description
connect to	Outgoing VoIP call
accept from	Incoming VoIP call
<ip address>	Peer's IP address
cr <cr>	Call reference (corresponds with the internal protocol interface's PSTN call reference)
s <si>	Session ID

H.225 signaling output

The following trace results are for a call coming from the IP network. rcv will appear at <dir> and signifies the direction: `h225<dir> tpkt msg 0x<mt> h225cr <cr> addr <ip address>`.

Table 10.15 H.225 signaling

Trace Output	Description
<mt>	The ETS message type in hexadecimal; can consist of values listed in Table 10.16.
<hcr>	H.225 call reference in hexadecimal (does not have to be unique when calls come from multiple peers).
<ip address>	The peer's IP address.

Table 10.16 ETS message types

Hex Value	Message Type
1	Alerting
2	Call Proceeding
3	Progress
5	Setup
7	Connect
D	Setup Acknowledge

Table 10.16 ETS message types *(continued)*

Hex Value	Message Type
5A	Release Complete
62	Facility
6E	Notify
7B	Information
7D	Status

The following lines show the packet contents in detail:

```
h225 decode rc 0, q931 msg 0x<mt> = 0, len <length>
```

```
h225<type> <mt> voipcfg addr <ip address> rc 0 compr <codec>
```

```
h225<type>  <mt>  h225cr  <hcr>  FS:<bool>  (<codec>,<ip  address>,<port>)
TUNN:<bool> H245:<bool>(<ip address>,<port>)
```

```
h225<type> <mt> h225cr <hcr> cr <cr>
```

Table 10.17 Incoming VoIP calls

Trace Output	Description
<mt>	Message type in hexadecimal as per ETS standard (see Table 10.16) or written out as a name.
len <length>	Packet length in bytes.
h225<type>	H.225 `rcv` or `send`; received or sent from the IP network.
addr <ip address>	Peer's IP address.
compr <codec>	Peer's compression list (see Table 10.18).
FS<bool>	FastStart offered in the signaling packet or not.
(<codec>,	Lists codecs offered (seeTable 10.38).
<ip address>,	Peer's IP address for RTP data.
<port>)	Peer's port for RTP Data.
Tunn<bool>	Shows whether or not tunneling is offered as a signaling variant.
H245<bool>	Shows an extra H.245 session.
(ip address,	Peer's IP address.
port)	Peer's port.
h225cr <hcr>	H.225 message's call reference (does not have to be unique when calls come from multiple VoIP peers).
cr <cr>	Internal call reference (always unique for the call).

Table 10.18 Compression codecs used

Synonym	Codec
A	G.711Alaw64k
B	G.711Ulaw64k
C	G.7231
D	G.728
E	G.729
F	gsmFullRate
G	T.38fax
O	G.729A
P	G.72616
Q	G.72624
R	G.72632
S	G.729B
T	G.729AB
U	G.729E
V	G.723L
W	Transparent
X	G.721
Y	iLBC20
Z	iLBC30

When the call is sent in the direction of the IP network, the trace will include only the most important information: `h225<type> <mt1> dad <num> cr <cr>.`

Table 10.19 Calls to the IP network 1

Trace Output	Description
<mt>	Message type written out; if a decimal number appears here, it will be translated as per Table 10.16.
<num>	Called party number.
<cr>	Call reference.

Or: `h225<type> callproc typ <mt> cr <cr>.`

Table 10.20 Calls to the IP network 2

Trace Output	Description
<mt>	The ETS message type in hexadecimal.
<cr>	Call reference.

RTP/RTCP output

The RTP/RTCP output displays whether the signaling information corresponds with the contents of the compression chips. The output occurs when a media channel is set up or torn down:

```
rtp start cr <cr> ch <ch> li <li> ri <ri> st <st> fx <fx> cp <comp> txm
<factor>.
```

Table 10.21 RTP/RTCP output

Trace Output	Description
<cr>	Call reference.
<ch>	The internal media channel used.
<li>	1 appears when the local RTP address (and port) has been defined.
<ri>	1 appears when the remote RTP address (and port) have been established.
<st>	0 appears if the channel's voice packetizer has not yet been started. 1 appears if the voice packetizer can receive, but not send. 2 appears when the voice packetizer can receive and send.
<fx>	1 appears when T.38 (fax) is used, otherwise 0.
<comp>	The codec used, as per Table 10.18.
<factor>	Multiplication factor for default frame size (20ms, 30 ms for G.723).

The RTP stop message has the following syntax: `rtp stop cr <cr>1 ch <ch>.`

Table 10.22 RTP stop message

Trace Output	Description
<cr>	Call reference.
<ch>	The internal media channel used.

The following output shows the RTCP packet statistics:

```
rtcp <ch>: SR <dir> pc <pc> oc <oc> ji <ji> rt <rt> fl <fl> cl <cl>.
```

Table 10.23 RTCP packet statistics

Trace Output	Description
<ch>	The internal media channel used.
SR<dir>	Rx sender report (received) is more interesting, since it comes from the peer. Tx sender report (transmitted).
<pc>	Packet count (number of packets transmitted/received).
<oc>	Octet count (number of octets transmitted/received).
<ji>	Delay jitter [msec].
<rt>	Round-trip local<->remote, round-trip delay [msec].
<fl>	Fraction lost: Fraction of packets lost [8lsb].
<cl>	Cumulative lost: number of lost packets [24lsb].

The following output shows the jitter buffer status:

```
a[<controller>]: <VoIPcodecChipType> ch <ch> jitter buffer n1 n2 n3n4 n5
n6 n7 n8.
```

Table 10.24 Jitter buffer status

Trace Output	Description
n1	SteadyStateDelay in milliseconds
n2	NumberOfVoiceUnderrun
n3	NumberOfVoiceOverrun
n4	NumberOfVoiceDecoderBfi (bfi = bad frame interpolation)
n5	NumberOfVoicePacketsDropped
n6	NumberOfVoiceNetPacketsLost
n7	NumberOfIbsOverrun (ibs = in band signaling)
n8	NumberOfCasOverrun

An RTP connection has ended when the following trace output appears:

```
a[<controller>]: <VoIPcodecChipType> stop ch=<ch>.
```

Table 10.25 RTP stop message (VP module)

Trace Output	Description
`<ch>`	The internal media channel used.

The following output results when the codec changes for a fax connection:

```
a[<controller>]: ac49x ch <ch> fax/data n1 n2 n3.
```

Table 10.26 Codec change for fax

Trace Output	Description
`n1`	Fax bypass flag: `0` Voice, data bypass or fax relay `1` Fax bypass
`n2`	Signal detected on decoder output (see Table 10.27)
`n3`	Signal detected on encoder input (see Table 10.27)

Table 10.27 fax or data signal event

Value	Definition	Description
`0`	SILENCE_OR_UNKNOWN	Undefined (unknown signal or silence)
`1`	FAX_CNG	CNG-FAX (calling fax tone, 1100 Hz)
`2`	ANS_TONE_2100_FAX_CED_OR_MODEM	FAX-CED or modem-ANS (answer tone, 2100 Hz)
`3`	ANS_TONE_WITH_REVERSALS	ANS (answer tone with reversals)
`4`	ANS_TONE_AM	ANSam (AM answer tone)
`5`	ANS_TONE_AM_REVERSALS	ANSam (AM answer tone with reversals)
`6`	FAX_V21_PREAMBLE_FLAGS	FAX-V.21 preamble flags
`7`	FAX_V8_JM_V34	FAX-V.8 JM (fax call function, V.34 fax)
`8`	VXX_V8_JM_VXX_DATA	V.XX-V.8 JM (data call function, V-series modem)
`9`	V32_AA	V.32 AA (calling modem tone, 1800 Hz)
`10`	V22_USB1	V.22 USB1 (V.22(bis) unscrambled binary ones)

Table 10.27 fax or data signal event *(continued)*

Value	Definition	Description
11	V8_BIS_INITIATING_DUAL_TONE	V.8bis initiating dual tone (1375 Hz and 2002 Hz)
12	V8_BIS_RESPONDING_DUAL_TONE	V.8bis responding dual tone (1529 Hz and 2225 Hz)
13	VXX_DATA_SESSION	V.XX data session
14	V21_CHANNEL_2	V.21 channel 2 (mark tone, 1650 Hz)
15	V23_FORWARD_CHANNEL	V.23 forward channel (mark tone, 1300 Hz)
16	V21_CHANNEL_1=18	V.21 channel 1 (mark tone, 980 Hz)
17	BELL_103_ANSWER_TONE	Bell 103 answer tone, 2225 Hz
18	TTY	TTY
19	FAX_DCN	FAX-DCN (G.3 fax disconnect signal)

Fax relay is activated for the corresponding channel:

```
a[<controller>]: Ac49xActivateFaxRelayCommand(1) ch <ch> rc <cr>.
```

The following output shows various values for fax transmission (see Table 10.28 for a description of the values):

```
a[<controller>]: ac49x ch <ch> faxrelay: n1 n2 n3 n4 n5 n6 n7 n8 n9 n10 n11
n12 n13 n14 n15 n16 n17.
```

Table 10.28 Fax status

Value	Field Name	Description
n1	UnableToRecoverFlag (0 no, 1 yes)	Unable to recover lost packet
n2	IllegalHdlcFrameDetectedFlag (...)	Illegal HDLC frame detected
n3	FaxExitWithNoMcfFrameFlag	Fax exit with no MCF Frame
n4	HostTransmitOverRunFlag	Channel received a buffer overrun
n5	HostTransmitUnderRunFlag	Channel received a buffer underrun
n6	InternalErrorFlag	Internal error
n7	ReceivedBadCommandFlag	A bad command was received. Words 8 and 9 describe the bad command
n8	TimeOutErrorFlag	A timeout error occurred

Table 10.28 Fax status *(continued)*

Value	Field Name	Description
`n9`	T30RxState	T.30 receive state 0 Init (initialization state) 1 CNG (calling tone) 2 CED (called terminal identification) 3 V21 (switch to V21) 4 NSF (non-standard facilities) 5 NSC (non-standard (facilities) command) 6 CSI (called subscriber identification) 7 CIG (calling subscriber identification) 8 DIS (digital identification signal) 9 DTC (digital transmit command) 10 NSS (non-standard services) 11 TSI (transmitting subscriber ID) 12 DCS (digital command signal) 13 CTC (continue to correct) 14 CRP (command repeat) 15 DCN (disconnect) 16 Pre-message response 17 Post-message response 18 Post-message command 19 VXX (receive illegal high-speed signal) 20 TCF (training check function 21 Fax image 22 V.8 CI (calling indicator (reserved)) 23 V.8 CM (calling menu) 24 V.8 (joint menu)
`n10`	T30TxState	T.30 transmit state. For values, see the entry for T30RxState above.
`n11`	NumberOfTransferredPages	Number of transmitted or received pages positively confirmed by answering fax.
`n12`	BadInputPacketId	Sequence number of Bad Input UDPTL packet, or the first two bytes of the input TCP packet. The field is relevant if the “Bad CMD” or the “Unable Recover” bit is set.
`n13`	BadInputPacketTotalSize	The total number of bytes in a bad fax packet.
`n14`	FaxBitRate	0 Undefined or 300 bpx (T.30 control signals) 1 2400 bps 2 4800 bps 3 7200 bps 4 9600 bps 5 12000 bps 6 14400 bps 7-15 Reserved

Table 10.28 Fax status *(continued)*

Value	Field Name	Description	
n15	DemodulationStatus	0 1	EQM and Timing Offset fields are invalid. EQM and Timing Offset fields are valid.
n16	EyeQualityMonitor (EQM)	Eye Quality Monitor measure for TCF or fax image. The value of 0 corresponds to the best quality of demodulation. Values greater than 100 have a negative influence on the quality of the fax image.	
n17	TimingOffsetPpm	Timing Offset (ppm) recovered during demodulation of TCF or fax image. The offset is a short integer equal to the sum of the sampling rate error and the sending fax baud rate offset. The value is in integers and and may not exceed 50 ppm in the following ranges during fax transmission (T30TxState): 65535 – 65486 (-50ppm) 0 – 50 (+50ppm)	

The following output appears when the compression chip detects DTMF tones:

```
a[<controller]: ac49x ch <ch> ibs <dtmf> <dir> <mode> <lev> <dur>.
```

Table 10.29 DTMF tone detection

Trace Output	Description
<ch>	Media channel
<dtmf>	Detected DTMF tone in the stream or as per RFC2833
<dir>	Direction 0 Coming from BRI/analog 1 Coming from VoIP
<mode>	0 Tone has ended 1 Tone has been detected
<lev>	Signal level in -dBm
<dur>	Tone duration

10.7.3.2 Internal protocol interface (to ISDN, mobile)

These trace outputs always begin with the keyword `pstn`, followed by the direction and the message type. The message is then either concluded or other information follows:

```
pstn<type> <mtl> dad <num> oad <num> cc <value> cc <cc> id <id> c/c <ctrl>/
<ch> cr <cr>.
```

Table 10.30 Internal protocol interface

Trace Output	Description
`<type>`	Direction from (`rcv`) or to (`snd`) the internal protocol interface.
`<mtl>`	Message type written out; if a decimal number appears, it will be translated as per Table 10.16.
`<num>`	`DAD<num>` = called party number, `OAD<num>` = calling party number.
`<cc>`	Value for OAD field element: ▪ 20 OAD network type provided ▪ 40 OAD with sending complete ▪ 80 OAD presentation restricted
`<id>`	Internal ident for the call leg.
`<ctrl>`	Controller used (counting starts with 0).
`<ch>`	Media channel used (counting starts with 1).
`<cr>`	Internal call reference /, followed by 0 or call reference of the VoIP leg.

Output also appears when a call comes from the internal protocol interface and is assigned to a VoIP profile. The characters appear in front of the colon in the routing entry:

```
pstnrcv get_voipcfg <voip profile> compr <list>
```

Table 10.31 Received from PSTN 1

Trace Output	Description
`<voip profile>`	Defines the VoIP profile to be used.
`<list>`	Defines the compression algorithms offered.

Assignment of media channel used for the internal interface and the ISDN call reference for the VoIP call's appears as follows:

```
pstnrcv bchanind cr <cr> ch <chan> isdncr <icr>
```

Table 10.32 Received from PSTN 2

Trace Output	Description
`<cr>`	Call reference.
`<chan>`	Media channel used for the internal protocol interface (DSS1).
`<icr>`	Call reference for the internal protocol interface (DSS1).

10.7.3.3 H.245 messages

The following trace output is possible: `h245<dir>(<tt>) cr <cr>.`

Table 10.33 H.245 messages

Trace Output	Description
`<dir>`	The message's direction; `rcv` (incoming from the peer) or `snd` (sent message).
`<tt>`	H.245 transport type.
`<cr>`	Internal call reference.

Following this trace output, either a detailed description of the message and its corresponding message type, including negotiating information, or trace output elements that are explained later appear. The most important message types that contain further information elements are as follows:

```
TerminalCapabilitySet peer=<comp> cfg=<comp>
```

```
TerminalCapabilitySet <comp>
```

Table 10.34 Codec used

Trace Output	Description
`<comp>`	List of compression codecs offered (see Table 10.18), the list of the peer's codecs appears behind peer, and `cfg` shows which codecs are defined in the VoIP profile

The output syntax for the logical channel parameters is as follows:

```
OpenLogicalChannel cn=<cn> cpr=<comp> sessid=<sid> ctrl=<ip address>:<rtcp
port>
```

```
OpenLogicalChannelAck cn=<cn> sessid=<sid> media=<ip address>:<rtp port>
```

Table 10.35 Logical channel parameters

Trace Output	Description
`<cn>`	H.245 channel number per H.225 connection.
`<sid>`	Session ID.
`<comp>`	Codec used (see Table 10.18).
`<ip address>`	Protocol peer IP address.
`<rtcp port>`	Port used for the protocol RTCP.
`<rtp port>`	Port used for the protocol RTP.

The trace output is as follows when the message type is not translated or is ignored:

```
h245<dir>(<tt>) cr <cr> unknown msg <hmt> <hmi>
```

Table 10.36 H.245 parameters

Trace Output	Description
`hmt`	The H.245 message type (multimedia system control message type), (Table 10.37).
`hmi`	The H.245 message ID (see Table 10.38, Table 10.39, Table 10.40, Table 10.41).

Table 10.37 Multimedia system control message types

ID	Message
0 (Table 10.38)	Request
1 (Table 10.39)	Response
2 (Table 10.40)	Command
3 (Table 10.41)	Indication

Depending on the system control message type, one of the following message IDs appear:

Table 10.38 Message IDs for request message

ID	Message
0	NonStandard
1	MasterSlaveDetermination
2	TerminalCapabilitySet
3	OpenLogicalChannel
4	CloseLogicalChannel
5	RequestChannelClose
6	MultiplexEntrySend
7	RequestMultiplexEntry
8	RequestMode
9	RoundTripDelayRequest
10	MaintenanceLoopRequest
11	CommunicationModeRequest
12	ConferenceRequest
13	MultilinkRequest
14	LogicalChannelRateRequest

Table 10.39 Message IDs for response message

ID	Message
0	NonStandard
1	MasterSlaveDeterminationAck
2	MasterSlaveDeterminationReject
3	TerminalCapabilitySetAck
4	TerminalCapabilitySetReject
5	OpenLogicalChannelAck
6	OpenLogicalChannelReject
7	CloseLogicalChannelAck
8	RequestChannelCloseAck
9	RequestChannelCloseReject
10	MultiplexEntrySendAck
11	MultiplexEntrySendReject
12	RequestMultiplexEntryAck

Table 10.39 Message IDs for response message *(continued)*

ID	Message
13	RequestMultiplexEntryReject
14	RequestModeAck
15	RequestModeReject
16	RoundTripDelayResponse
17	MaintenanceLoopAck
18	MaintenanceLoopReject
19	CommunicationModeResponse
20	ConferenceResponse
21	MultilinkResponse
22	LogicalChannelRateAcknowledge
23	LogicalChannelRateReject

Table 10.40 Message IDs for command message

ID	Message
0	NonStandard
1	MaintenanceLoopOffCommand
2	SendTerminalCapabilitySet
3	EncryptionCommand
4	FlowControlCommand
5	EndSessionCommand
6	MiscellaneousCommand
7	CommunicationModeCommand
8	ConferenceCommand
9	h223MultiplexReconfiguration
10	NewATMVCCommand
11	MobileMultilinkReconfigurationCommand

Table 10.41 Message IDs For indication message

ID	Message
0	NonStandard
1	FunctionNotUnderstood
2	MasterSlaveDeterminationRelease

Table 10.41 Message IDs For indication message *(continued)*

ID	Message
3	TerminalCapabilitySetRelease
4	OpenLogicalChannelConfirm
5	RequestChannelCloseRelease
6	MultiplexEntrySendRelease
7	RequestMultiplexEntryRelease
8	RequestModeRelease
9	MiscellaneousIndication
10	JitterIndication
11	h223SkewIndication
12	NewATMVCIndication
13	UserInput
14	h2250MaximumSkewIndication
15	McLocationIndication
16	ConferenceIndication
17	VendorIdentification
18	FunctionNotSupported
19	MultilinkIndication
20	LogicalChannelRateRelease
21	FlowControlIndication
22	MobileMultilinkReconfigurationIndication

10.7.3.4 RAS (registration, admission, status)

As a general rule, the most important terminal and gatekeeper messages appear written out with the gatekeeper's IP address (<ip addr>).

Here is the output syntax for gatekeeper discovery requests:

```
H225 GatekeeperRequest to <ip addr> (s 131)
H225 GatekeeperConfirm <ip addr>
H225 GatekeeperReject <ip addr> reason <reason>
```

Table 10.42 RAS

Trace Output	Description
`<reason>`	Gatekeeper reject reason, see Table 10.46.

Here is the output syntax for endpoint registration requests:

```
H225 GkRegistration to <ip addr>
H225 RegistrationConfirm <ip addr>
H225 RegistrationReject <ip addr> reason <reason>
```

Table 10.43 Gatekeeper 1

Trace Output	Description
`<reason>`	Registration reject reason, see Table 10.47.

Below you see the output syntax for bandwith availability indication:

```
H225 GkResourcesAvailableIndicate to <ip addr> (<act chan> <max chan>)
H225 ResourcesAvailableConfirm <ip addr>
```

Here is the output syntax for call admission requests:

```
H225 GkAdmission cr <cr> to <ip addr>
H225 AdmissionConfirm <ip addr> cr <cr>
H225 AdmissionReject <ip addr> reason <reason>
```

Table 10.44 Gatekeeper 2

Trace Output	Description
`<reason>`	Admission reject reason, see Table 10.48.

Here is the output syntax for call teardown requests:

```
H225 GkDisengage cr <cr> to <ip addr>
H225 DisengageConfirm <ip addr>
```

Here is the output syntax for registration release requests:

```
H225 UnregistrationRequest <ip addr>
H225 GkUnregistrationConf to <ip addr>
```

All other messages appear as follows:

```
H225 unknown msg from Gk <ip addr>: <code>
```

Table 10.45 Gatekeeper 3

Trace Output	Description
`<code>`	Unknown gatekeeper message, see Table 10.49.

Table 10.46 Gatekeeper reject reason

ID	Reject Reason
0	resourceUnavailable
1	terminalExcluded
2	invalidRevision
3	undefinedReason
4	securityDenial
5	genericDataReason
6	neededFeatureNotSupported

Table 10.47 Registration reject reason

ID	Reject Reason
0	DiscoveryRequired
1	InvalidRevision
2	InvalidCallSignalAddress
3	InvalidRASAddress
4	DuplicateAlias
5	InvalidTerminalType
6	UndefinedReason
7	TransportNotSupported
8	TransportQOSNotSupported
9	ResourceUnavailable
10	InvalidAlias
11	SecurityDenial
12	RullRegistrationRequired
13	AdditiveRegistrationNotSupported
14	InvalidTerminalAliases
15	GenericDataReason
16	NeededFeatureNotSupported

Table 10.48 Admission reject reason

ID	Reject Reason
0	CalledPartyNotRegistered
1	InvalidPermission

Table 10.48 Admission reject reason *(continued)*

ID	Reject Reason
2	RequestDenied
3	UndefinedReason
4	CallerNotRegistered
5	RouteCallToGatekeeper
6	InvalidEndpointIdentifier
7	ResourceUnavailable
8	SecurityDenial
9	QosControlNotSupported
10	IncompleteAddress
11	AliasesInconsistent
12	RouteCallToSCN
13	ExceedsCallCapacity
14	CollectDestination
15	CollectPIN
16	GenericDataReason
17	NeededFeatureNotSupported

Table 10.49 Unknown gatekeeper messages

ID	Message
0	GatekeeperRequest
1	GatekeeperConfirm
2	GatekeeperReject
3	RegistrationRequest
4	RegistrationConfirm
5	RegistrationReject
6	UnregistrationRequest
7	UnregistrationConfirm
8	UnregistrationReject
9	AdmissionRequest
10	AdmissionConfirm
11	AdmissionReject
12	BandwidthRequest

Table 10.49 Unknown gatekeeper messages *(continued)*

ID	Message
13	BandwidthConfirm
14	BandwidthReject
15	DisengageRequest
16	DisengageConfirm
17	DisengageReject
18	LocationRequest
19	LocationConfirm
20	LocationReject
21	InfoRequest
22	InfoRequestResponse
23	NonStandardMessage
24	UnknownMessageResponse
25	RequestInProgress
26	ResourcesAvailableIndicate
27	ResourcesAvailableConfirm
28	InfoRequestAck
29	InfoRequestNak
30	ServiceControlIndication
31	ServiceControlResponse

10.7.3.5 ENUM output

This output is assigned to group `i` and occurs with Layer2 and Layer3 traces:

```
i[<controller>]: enum_query cr <CR> ch <CH>: <num> -> <length> <<answer pattern>>.
```

Table 10.50 ENUM output

Trace Output	Description
`<cr>`	Call reference.
`<ch>`	Media channel.
`<num>`	Phone number converted into ENUM domain format.
`<length>`	Length of the answer field in the DNS response in bytes. `0` appears if the number was not found.
`<answer pattern>`	Displays the DNS response. `0` appears if the number was not found.

10.7.3.6 Examples

The following examples are offline traces. You can generate them using the GATE Manager or FTP commands. The filename is trace.log. The following cases appear in the examples:

- Incoming H.323 call with FastStart (please see "Incoming H.323 call with FastStart" on page 182)
- Outgoing H.323 call with FastStart (please see "Outgoing H.323 call with FastStart" on page 183)
- Fax call (please see "Fax call" on page 184)

Incoming H.323 call with FastStart

Example 10.6 Incoming H.323 call with FastStart

```
[15:25:13.65] i[02]: h225accept from 172.16.0.200 s 4
[15:25:13.75] y[02]: h225rcv tpkt msg 5 h225cr 8006 addr 172.16.0.200 pt 0
[15:25:13.75] y[02]: h225 decode rc 0, q931 msg 5 (0), len 364
[15:25:13.75] y[02]: h225rcv setup voipcfg addr 172.16.0.200 rc 0 <DF> compr EABG
[15:25:13.75] y[02]: h225rcv faststart <A1B1E1G0>
[15:25:13.75] y[02]: h225rcv setup oad 01 00 <111> <> dad 01 <123456> rad <> bc 038090a3 0101
[15:25:13.75] y[02]: h225rcv setup h225cr 8006 FS:1(E,172.16.0.200,29000) TUNN:1 H245:0(0,0)
[15:25:13.75] y[02]: h225rcv setup h225cr 8006 cr 7
[15:25:13.75] i[02]: pstnsnd setup dad 123456 oad 1 cr 7 s 4
[15:25:13.75] s[00]: 00 01 52 4c 08 02 00 08 05 04 03 80 90 a3 18 03 a1 83 87 6c 04 81 31 31 31 70 07 81
31 32 33 34 35 36 7d 02 91 81
[15:25:13.75] i[02]: pstnrcv connresp cr 7 acc 5 ch 1
[15:25:13.75] x[02]: h225snd callproc typ d cr 7 pri 0
[15:25:13.75] r[00]: 00 01 01 54
[15:25:13.75] r[00]: 02 01 4c 54 08 02 80 08 0d 18 03 a9 83 87
[15:25:13.75] s[00]: 02 01 01 4e
[15:25:14.33] r[00]: 02 01 4e 54 08 02 80 08 01
[15:25:14.33] s[00]: 02 01 01 50
[15:25:14.33] i[02]: pstnrcv alert cr 7 cls ff
[15:25:14.33] i[02]: rtp start cr 7 ch 1 li 1 ri 1 st 2 fx 0 cp E txm 1
[15:25:14.33] x[02]: h225snd callproc typ 1 cr 7 pri 8
[15:25:14.34] a[02]: vp start(201) ch=0 local=29000 remote=ac1000c8:29000 agg=0 pcm=0
[15:25:14.38] a[02]: vp rtcp 0: RR Tx pc 0 oc 0 ji -1 rt 0 fl -1 cl -1
[15:25:14.38] a[02]: vp ch 0: in 0 out 74
[15:25:15.57] r[00]: 02 01 50 54 08 02 80 08 07 29 05 06 03 18 0f 17 4c 06 01 81 31 37 33 31
[15:25:15.57] s[00]: 00 01 54 52 08 02 00 08 0f
[15:25:15.57] i[02]: pstnrcv connresp cr 7 acc 10 ch 255
[15:25:15.57] x[02]: h225snd callproc typ 7 cr 7 pri 0
[15:25:15.58] r[00]: 00 01 01 56
[15:25:17.01] a[02]: vp rtcp 0: SR Rx pc 110 oc 1816 ji 158 rt -1 fl 2 cl 1
[15:25:20.09] a[02]: vp rtcp 0: SR Tx pc 277 oc 5496 ji 164 rt 0 fl 0 cl 0
[15:25:20.09] a[02]: vp ch 0: in 18166 out 20646
[15:25:20.09] a[02]: vp rtcp 0: SR Rx pc 258 oc 4634 ji 208 rt -1 fl 0 cl 1
[15:25:23.32] a[02]: vp rtcp 0: SR Tx pc 441 oc 8776 ji 176 rt 0 fl 0 cl 0
[15:25:23.32] a[02]: vp ch 0: in 28966 out 32900
[15:25:24.68] y[02]: h225rcv tpkt msg 5a h225cr 8006 addr 172.16.0.200 pt 800e7800
[15:25:24.68] y[02]: h225 decode rc 0, q931 msg 5a (5), len 33
[15:25:24.68] y[02]: h225rcv relack h225cr 8006 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[15:25:24.68] y[02]: h225rcv relack h225cr 8006 cau 0x10
[15:25:24.68] i[02]: rtp hold cr 7 ch 1
[15:25:24.68] s[00]: 00 01 56 52 08 02 00 08 45 08 02 80 90
[15:25:24.68] i[02]: h225 connection 4 terminated
[15:25:24.69] r[00]: 00 01 01 58
[15:25:25.89] r[00]: 02 01 52 58 08 02 80 08 4d
[15:25:25.89] s[00]: 00 01 58 54 08 02 00 08 5a
[15:25:25.94] i[02]: pstnrcv terminate connection (3201) cr 7 cau 1 err 16 state 17 ch 1 rsid 1
[15:25:25.94] i[02]: rtp stop cr 7 ch 1
[15:25:25.94] r[00]: 00 01 01 5a
[15:25:25.94] a[02]: vp ch 0: in 34096 out 38154
[15:25:25.94] a[02]: vp stop ch=0
```

Outgoing H.323 call with FastStart

Example 10.7 Outgoing H.323 call with FastStart

```
[15:04:09.12] r[00]: 02 01 46 48 08 02 22 54 05 04 03 80 90 a3 18 03 a9 83 94 6c 06 01 81 31 31 31 31 70
04 81 33 32 31 7d 02 91 81
[15:04:09.12] s[00]: 02 01 01 48
[15:04:09.12] s[00]: 00 01 48 48 08 02 a2 54 0d 18 03 a9 83 94
[15:04:09.12] i[02]: pstnrcv setup dad DF:321 oad 1111 cc 0 id 15d006
[15:04:09.12] i[02]: pstnrcv get_voipcfg <DF>
[15:04:09.12] i[02]: h225connect to 172.16.0.200 cr 6
[15:04:09.12] x[02]: h225snd setup dad 1 cr 6
[15:04:09.12] r[00]: 00 01 01 4a
[15:04:09.15] y[02]: h225rcv tpkt msg d h225cr 6 addr 172.16.0.200 pt 80412800
[15:04:09.15] y[02]: h225 decode rc 0, q931 msg d (11), len 32
[15:04:09.15] y[02]: h225rcv msg d (11) h225cr 6 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[15:04:09.50] y[02]: h225rcv tpkt msg 1 h225cr 6 addr 172.16.0.200 pt 80412800
[15:04:09.50] y[02]: h225 decode rc 0, q931 msg 1 (3), len 121
[15:04:09.50] y[02]: h225rcv faststart <E1>
[15:04:09.50] y[02]: h225rcv alert h225cr 6 FS:1(E,172.16.0.200,29000) TUNN:1 H245:0(0,0)
[15:04:09.50] i[02]: rtp start cr 6 ch 1 li 1 ri 1 st 2 fx 0 cp E txm 1
[15:04:09.50] s[00]: 00 01 4a 48 08 02 a2 54 01 1e 02 80 88
[15:04:09.50] a[02]: vp start(201) ch=0 local=29000 remote=ac1000c8:29000 agg=0 pcm=0
[15:04:09.50] r[00]: 00 01 01 4c
[15:04:09.53] a[02]: vp rtcp 0: RR Tx pc 0 oc 0 ji -1 rt 0 fl -1 cl -1
[15:04:09.53] a[02]: vp ch 0: in 0 out 74
[15:04:11.79] y[02]: h225rcv tpkt msg 7 h225cr 6 addr 172.16.0.200 pt 80412800
[15:04:11.79] y[02]: h225 decode rc 0, q931 msg 7 (2), len 79
[15:04:11.79] y[02]: h225rcv connect h225cr 6 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[15:04:11.79] i[02]: pstnsnd connect cr 6
[15:04:11.79] s[00]: 00 01 4c 48 08 02 a2 54 07
[15:04:11.80] r[00]: 02 01 48 4e 08 02 22 54 0f
[15:04:11.80] s[00]: 02 01 01 4a
[15:04:12.50] a[02]: vp rtcp 0: SR Rx pc 21 oc 394 ji 201 rt -1 fl 0 cl 0
[15:04:16.13] a[02]: vp rtcp 0: SR Tx pc 192 oc 3236 ji 196 rt 0 fl 0 cl 0
[15:04:16.13] a[02]: vp ch 0: in 14612 out 13796
[15:04:17.98] y[02]: h225rcv tpkt msg 5a h225cr 6 addr 172.16.0.200 pt 80412800
[15:04:17.98] y[02]: h225 decode rc 0, q931 msg 5a (5), len 33
[15:04:17.98] y[02]: h225rcv relack h225cr 6 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[15:04:17.98] y[02]: h225rcv relack h225cr 6 cau 0x10
[15:04:17.98] i[02]: rtp hold cr 6 ch 1
[15:04:17.98] s[00]: 00 01 4e 4a 08 02 a2 54 45 08 02 80 90
[15:04:17.98] i[02]: h225 connection 4 terminated
[15:04:17.99] r[00]: 00 01 01 50
[15:04:18.04] r[00]: 02 01 4a 50 08 02 22 54 4d 08 02 84 90
[15:04:18.04] s[00]: 00 01 50 4c 08 02 a2 54 5a
[15:04:18.06] i[02]: pstnrcv terminate connection (3201) cr 6 cau 90 err 16 state 17 ch 1 rsid 1
[15:04:18.06] i[02]: rtp stop cr 6 ch 1
[15:04:18.06] r[00]: 00 01 01 52
[15:04:18.06] a[02]: vp ch 0: in 21288 out 20708
[15:04:18.06] a[02]: vp stop ch=0
```

Fax call

Example 10.8 Fax call

```
[16:00:40.44] i[02]: h225accept from 172.20.0.200 s 4
[16:00:40.49] y[02]: h225rcv tpkt msg 5 h225cr 8007 addr 172.20.0.200 pt 0
[16:00:40.49] y[02]: h225 decode rc 0, q931 msg 5 (0), len 251
[16:00:40.49] y[02]: h225rcv setup voipcfg addr 172.20.0.200 rc 0 <DF> compr EABG
[16:00:40.49] y[02]: h225rcv faststart <E0G0>
[16:00:40.49] y[02]: h225rcv setup oad 00 00 <> <> dad 01 <123456> rad <> bc 038090a3 0101
[16:00:40.49] y[02]: h225rcv setup h225cr 8007 FS:1(E,172.20.0.200,29000) TUNN:1 H245:0(0,0)
[16:00:40.49] y[02]: h225rcv setup h225cr 8007 cr 14
[16:00:40.49] i[02]: pstnsnd setup dad 123456 oad  cr 14 s 4
[16:00:40.49] s[00]: 00 01 5a 54 08 02 00 09 05 04 03 80 90 a3 18 03 a1 83 88 70 07 81 31 32 33 34 35 36
7d 02 91 81
[16:00:40.49] i[02]: pstnrcv connresp cr 14 acc 5 ch 1
[16:00:40.49] x[02]: h225snd callproc typ d cr 14 pri 0
[16:00:40.50] r[00]: 02 01 54 5c 08 02 80 09 0d 18 03 a9 83 88
[16:00:40.67] r[00]: 02 01 56 5c 08 02 80 09 01
[16:00:40.67] i[02]: pstnrcv alert cr 14 cls ff
[16:00:40.67] i[02]: rtp start cr 14 ch 1 li 1 ri 1 st 2 fx 0 cp E txm 2
[16:00:40.67] x[02]: h225snd callproc typ 1 cr 14 pri 8
[16:00:40.70] a[02]: vp start(201) ch=0 local=29000 remote=ac1000c8:29000 agg=0 pcm=0
[16:00:40.74] a[02]: vp rtcp 0: RR Tx pc 0 oc 0 ji -1 rt 0 fl -1 cl -1
[16:00:40.74] a[02]: vp ch 0: in 0 out 74
[16:00:40.90] r[00]: 02 01 58 5c 08 02 80 09 07 29 05 06 03 18 0f 3b 4c 08 01 81 31 32 33 34 35 36
[16:00:40.90] s[00]: 00 01 5c 5a 08 02 00 09 0f
[16:00:40.90] i[02]: pstnrcv connresp cr 14 acc 10 ch 255
[16:00:40.90] x[02]: h225snd callproc typ 7 cr 14 pri 0
[16:00:41.98] a[02]: vp rtcp 0: SR Rx pc 134 oc 1340 ji 195 rt -1 fl 0 cl 0
[16:00:43.29] y[02]: h225rcv tpkt msg 62 h225cr 8007 addr 172.20.0.200 pt 80410800
[16:00:43.29] y[02]: h225 decode rc 0, q931 msg 62 (6), len 123
[16:00:43.29] y[02]: h225rcv facility h225cr 8007 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:43.29] i[02]: h245rcv(1) cr 14 TerminalCapabilitySet peer=<EG> cfg=<EABG>
[16:00:43.29] i[02]: h245snd(1) cr 14 TerminalCapabilitySetAck
[16:00:43.29] i[02]: h245snd(1) cr 14 TerminalCapabilitySet <EABG>
[16:00:43.51] y[02]: h225rcv tpkt msg 62 h225cr 8007 addr 172.20.0.200 pt 80410800
[16:00:43.51] y[02]: h225 decode rc 0, q931 msg 62 (6), len 63
[16:00:43.51] y[02]: h225rcv facility h225cr 8007 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:43.51] i[02]: h245rcv(1) cr 14 TerminalCapabilitySetAck
[16:00:43.72] y[02]: h225rcv tpkt msg 62 h225cr 8007 addr 172.20.0.200 pt 80410800
[16:00:43.72] y[02]: h225 decode rc 0, q931 msg 62 (6), len 74
[16:00:43.72] y[02]: h225rcv facility h225cr 8007 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:43.72] i[02]: h245rcv(1) cr 14 RequestMode t38=1
[16:00:43.72] i[02]: h245snd(1) cr 14 RequestModeAck
[16:00:43.73] i[02]: h245snd(1) cr 14 CloseLogicalChannel cn=1
[16:00:43.73] i[02]: h245snd(1) cr 14 OpenLogicalChannel cn=1 cpr=G sessid=1 ctrl=172.20.0.100:29001
[16:00:43.73] y[02]: h225rcv tpkt msg 62 h225cr 8007 addr 172.20.0.200 pt 80410800
[16:00:43.73] y[02]: h225 decode rc 0, q931 msg 62 (6), len 68
[16:00:43.73] y[02]: h225rcv facility h225cr 8007 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:43.73] i[02]: h245rcv(1) cr 14 CloseLogicalChannel cn=1 (1)
[16:00:43.73] i[02]: h245snd(1) cr 14 CloseLogicalChannelAck cn=1
[16:00:43.73] y[02]: h225rcv tpkt msg 62 h225cr 8007 addr 172.20.0.200 pt 80410800
[16:00:43.73] y[02]: h225 decode rc 0, q931 msg 62 (6), len 92
[16:00:43.73] y[02]: h225rcv facility h225cr 8007 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:43.73] i[02]: h245rcv(1) cr 14 OpenLogicalChannel cn=1 cpr=G sessid=1 ctrl=172.20.0.200:29001
[16:00:43.73] i[02]: h245snd(1) cr 14 OpenLogicalChannelAck cn=1 sessid=1 media=172.20.0.100:29000
[16:00:43.73] y[02]: h225rcv tpkt msg 62 h225cr 8007 addr 172.20.0.200 pt 80410800
[16:00:43.73] y[02]: h225 decode rc 0, q931 msg 62 (6), len 64
[16:00:43.73] y[02]: h225rcv facility h225cr 8007 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:43.73] i[02]: h245rcv(1) cr 14 CloseLogicalChannelAck cn=1
[16:00:43.73] y[02]: h225rcv tpkt msg 62 h225cr 8007 addr 172.20.0.200 pt 80410800
[16:00:43.73] y[02]: h225 decode rc 0, q931 msg 62 (6), len 83
[16:00:43.73] y[02]: h225rcv facility h225cr 8007 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:43.73] i[02]: h245rcv(1) cr 14 OpenLogicalChannelAck cn=1 sessid=1 media=172.20.0.200:29000
```

Example 10.8 Fax call *(continued)*

```
[16:00:43.73] i[02]: rtp start cr 14 ch 1 li 1 ri 1 st 3 fx 0 cp G txm 2
[16:00:43.73] i[02]: rtp start cr 14 ch 1 li 1 ri 1 st 3 fx 1 cp G txm 2
[16:00:43.74] a[02]: vp start2 ch=0 remote=ac1000c8:29000
[16:00:43.74] a[02]: vp start(401) ch=0 local=29000 remote=ac1000c8:29000 agg=0 pcm=0
[16:00:47.70] a[02]: vp rtcp 0: SR Tx pc 13 oc 352 ji 132 rt 0 fl 0 cl 0
[16:00:53.63] a[02]: vp rtcp 0: RR Tx pc 13 oc 352 ji -1 rt 0 fl -1 cl -1
[16:00:59.14] a[02]: vp rtcp 0: RR Tx pc 13 oc 352 ji -1 rt 0 fl -1 cl -1
[16:01:02.12] a[02]: vp rtcp 0: RR Tx pc 13 oc 352 ji -1 rt 0 fl -1 cl -1
[16:01:07.16] a[02]: vp rtcp 0: RR Tx pc 13 oc 352 ji -1 rt 0 fl -1 cl -1
[16:01:11.82] a[02]: vp rtcp 0: RR Tx pc 13 oc 352 ji -1 rt 0 fl -1 cl -1
[16:01:18.06] a[02]: vp rtcp 0: RR Tx pc 13 oc 352 ji -1 rt 0 fl -1 cl -1
[16:01:21.15] a[02]: vp rtcp 0: RR Tx pc 13 oc 352 ji -1 rt 0 fl -1 cl -1
[16:01:26.10] a[02]: vp rtcp 0: RR Tx pc 13 oc 352 ji -1 rt 0 fl -1 cl -1
[16:01:28.89] a[02]: vp rtcp 0: RR Tx pc 13 oc 352 ji -1 rt 0 fl -1 cl -1
[16:01:33.14] y[02]: h225rcv tpkt msg 5a h225cr 8007 addr 172.20.0.200 pt 80410800
[16:01:33.14] y[02]: h225 decode rc 0, q931 msg 5a (5), len 33
[16:01:33.14] y[02]: h225rcv relack h225cr 8007 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:01:33.14] y[02]: h225rcv relack h225cr 8007 cau 0x10
[16:01:33.14] i[02]: rtp hold cr 14 ch 1
[16:01:33.15] s[00]: 00 01 5e 5a 08 02 00 09 45 08 02 80 90
[16:01:33.15] i[02]: h225 connection 4 terminated
[16:01:33.19] r[00]: 02 01 5a 60 08 02 80 09 4d
[16:01:33.19] s[00]: 00 01 60 5c 08 02 00 09 5a
[16:01:33.19] i[02]: pstnrcv terminate connection (3201) cr 14 cau 1 err 16 state 17 ch 1 rsid 1
[16:01:33.19] i[02]: rtp stop cr 14 ch 1
[16:01:33.23] a[02]: vp ch 0: in 85542 out 4346
[16:01:33.23] a[02]: vp stop ch=0
```

10.7.4 Remote output

This trace option provides output for communication with the GATE Manager or NMS. To activate this option, activate the section **Remote** in the GATE Manager. You can choose the depth of the trace output: **Error** is limited to error messages; **Debug** provides information; **Detail** provides the entire packet.

Output is defined with a `g`, and the port number is 99.

The following output shows an established GATE Manager connection:

`g[99]:moip: accept rc=2 ipad=<ip address> port=<port>`.

Table 10.51 Remote output

Trace Output	Description
`<ip address>`	Remote system's IP address with GATE Manager.
`<port>`	Origination port for the GATE Manager connection.

The following output shows the direction (In/Out) for the packet and the size.

```
g[99]:moip: <direction> <length>.
```

Table 10.52 Remote output

Trace Output	Description	
`<direction>`	recv send write read	Packets received from the remote system Packets sent to the remote system Output for communication with the internal remote interface Output for communication from the internal remote interface
`<length>`	Data length in bytes.	

All other trace output appears in detail mode in ASCII and are also translated.

10.7.5 SMTP trace output

This trace option provides output for communication with the mail server that occurs when status information or files are sent, or in the other direction, which e-mails are received and converted to SMS or USSD.

To activate this option, activate the section **Mail** in the GATE Manager. You can choose the depth of the trace output: **Error** is limited to error messages; **Debug** provides information; **Detail** provides the entire packet.

Output is defined with a `m`, and the port number is 99.

Sending files or status information

Here is the global message output syntax:

```
m[99]:mail: sendmail (<length>)
```

Table 10.53 SMTP output: sending files or status info

Trace Output	Description
`<length>`	Data length in bytes.

The detailed message output looks like this:

```
m[99]:mail: sendmail: <Faccount> <ip address> <Taccount> <domain> <subject> <content>.
```

Table 10.54 SMTP output: sending fles or status info

Trace Output	Description
`<Faccount>`	Sender’s e-mail account (cdr, alarm, file, and so on).
`<ip address>`	SMTP server’s IP address.

Table 10.54 SMTP output: sending fles or status info *(continued)*

Trace Output	Description
`<Taccount>`	Recipient's e-mail account.
`<domain>`	Recipient's domain.
`<subject>`	Content of the subject field; serial number of the sender system.
`<content>`	Content of the message's body.

All other trace output appears in detail mode in ASCII and are also translated.

Receiving e-mail messages and sending them as SMS or USSD

The following output displays communication of an incoming SMTP connection:

```
m[99]:mail: accept: ipad=<ip address> port=<port>
```

Table 10.55 SMTP output: receiving e-mail and sending as SMS or USSD

Trace Output	Description
`<ip address>`	The SMTP peer system's IP address.
`<port>`	The SMTP peer system's origination port.

The following output displays which packets are sent to the SMTP peer:

```
m[99]:mail: mysend <<content>>
```

Table 10.56 SMTP output: receiving e-mail and sending as SMS or USSD

Trace Output	Description
`<content>`	Content of the transmitted packet.

All other trace output appears in detail mode in ASCII and are also translated.

The following output displays which packets are received from the SMTP peer:

```
m[99]:mail: recv (<length>)
```

Table 10.57 SMTP Output: receiving e-mail and sending as SMS or USSD

Trace Output	Description
`<length>`	Data length in bytes.

All other trace output appears in detail mode in ASCII and are also translated.

The following output shows that the SMTP connection is being closed:

```
m[99]:mail: terminate_session
```

The mail module now converts the e-mail message to the internal format and then sent as SMS or USSD. Bulk mail (several recipient entries for the same e-mail) appear as individual messages:

```
m[99]:mail: newMail2Host r=<Taccount> f=<Faccount> s=<subject> d=<content>
```

Table 10.58 SMTP output: receiving e-mail and sending as SMS or USSD

Trace Output	Description
`<Faccount>`	One entry from the sender’s To field.
`<Taccount>`	Content of the From field.
`<subject>`	Content of the subject field; usually not used.
`<content>`	Content of the message’s body; is sent as SMS or USSD.

The following output appears when the message has been successfully sent:

```
m[99]:mail: rcvmail <Faccount> -> <Taccount>, done
```

This is converted in the confirmation message, with the subject `sent`. The output in the subsequent communication with the mail server are identical to those described above in Chapter Sending files or status information.

The following output appears when errors occur during transmission of the SMS or USSD message:

- Message transmission was faulty and will be repeated: `m[99]:mail: rcvmail <Faccount> -> <Taccount>, failed, will retry (<num>)`
- Retried message transmission was also faulty, and an e-mail will be generated: `m[99]:mail: rcvmail <Faccount> -> <Taccount>, failed <num> times`

Table 10.59 SMTP output: transmission error

Trace Output	Description
`<num>`	Current number of retries.

The output in the subsequent communication with the mail server are identical to those described above in Chapter Sending files or status information.

Receiving SMS or USSD and sending as e-mail

The following output shows the internal format when an SMS or USSD message is sent to the mail module. This output is generated when transmission of the SMS or USSD message was not possible:

```
m[99]:mail: DATA_IND (<length>).
```

All other trace output appears in detail mode in ASCII and are also translated. The output in the subsequent communication with the mail server are identical to those described above in Chapter Sending files or status information.

10.7.6 Number portability trace output

This trace option provides output for the communication with the iMNP database. To activate this option, activate the section **Number Portability** in the GATE Manager. Output is defined with an `n`, and the port number is 99.

The following output appears when the system sets up a TCP session with the iMNP is being set up:

```
n[99]:np: connecting to <ip addr>.
```

Table 10.60 Number portability output: connection with iMNP

Trace Output	Description
`<ip address>`	The iMNP system's IP address.

The following output shows that the connection has been established:

```
n[99]:np: connect to <ip addr> ok.
```

The following output shows that the connection attempt failed:

```
n[99]:np: connect to <ip addr> failed.
```

The following output shows a keep alive packet from the iMNP to keep the TCP session open:

```
n[99]:np: recv <>.
```

Response to a number portability request that results in the call's routing:

```
n[99]:np: recv <N<num>>.
```

Table 10.61 Number portability output: response

Trace Output	Description
`<num>`	Ported or unported number provided by the database.

10.7.7 DTMF tone trace output

Output about the setup of connections with the DTMF module and DTMF tone detection are debugged. The output differentiates between the groups `err` and `inf`. Output is defined with a `d`, and the port number is that of the virtual DTMF controller:

The following output shows incoming call setup to the DTMF module:

```
d[<ctrl>]: dtmf: msg <call state>, unknown id <id>, from 14.
```

Table 10.62 DTMF output: incoming call setup

Trace Output	Description
`<ctrl>`	The virual controller’s running number.
`<call state>`	3101 Incoming setup 3201 Disconnect request
`<id>`	Call identification number.

The following output shows transmitted signaling messages depending on the call state:

```
d[<ctrl>]: dtmf <message type> <id> <call state> 0.
```

Table 10.63 DTMF output: signaling messages

Trace Output	Description
`<message type>`	Send_d_connect For setup acknowledge and connect. send_alert_ind For alert. send_disconnect For disconnect
`<id>`	Call identification number.
`<call state>`	3110 Incoming setup 3102 Disconnect request 3804 Alert 3202 Disconnect confirmation

The following output shows that the media channel has been designated for DTMF tone detection:

```
d[<ctrl>]: dtmf send_alloc <b_chan id_unset> <ctrl>/<b chan>.
```

Table 10.64 DTMF output: media channel designation

Trace Output	Description
`<b chan>`	Internal media channel used.
`<b_chan id_unset>`	Media channel identification (in unset state).

The following output shows the status for the media channel will is/was used for DTMF tone recognission:

```
d[<ctrl>]: dtmf: msg <msg>, id <b_chan id>, from 1, id <id>/<b_chan
id_unset>.
```

Table 10.65 DTMF output: media channel designation

Trace Output	Description	

`<msg>`	`502`
	`102`
	`602`
		Media channel confirmation
		Connect confirmation
		Media channel free confirmation

The following output shows the output for negotiated DTMF tones:

```
d[<ctrl>]: dtmf send_info_ind <id> <<dtmf tone>>.
```

11 Feature packages

11 Feature packages

The feature packages are modular expansion applications that provide services in addition to those offered with the standard software. Feature packages can be activated separately or in combination with one another, so that you can design your system according to your own needs.

The following feature packages are available:

- Two Stage Dialing/Callback Services (please see Chapter 11.2 on page 194)
- Least Cost Routing (please see Chapter 11.3 on page 200)
- Online Traffic Monitor (please see Chapter 11.4 on page 206)
- Ported Number Screening (please see Chapter 11.5 on page 213)
- Call recording (please see Chapter 11.6 on page 215)

11.1 Activating the license

Each feature package requires a license. Once you have ordered a feature package, you can activate the license:

The `/boot/` directory of each system contains a file called license.key, which contains information on the system's ID, the included components, which feature packages are active and the license number:

Example 11.1 Activating the license

```
[IDENTIFICATION]
SYSTEM: TELES.iGATE
SERNO:  VT810011
AUTOR:  create   Wed Sep 09 15:01:09 2006

[COMPONENTS]
...
CARD99:11 d1 S0  PB900034
...

[FEATURES]
PRI:Max
GSM:Max
IP:Max
VoIP:Max
SIM manager: On
DDI and call back: Off
least cost routing: On
statistics and CDR: On
SMS gateway: On
ported number screening: Off
roaming: Off

[SIGNATURE]
00000000000license0number00000000000
```

You will receive a new license.key file any time you order a new license package. Simply save the new file, overwriting the old file, and restart the system.

Deleting or making changes in the `license.key` file will delete any feature package licenses, causing the system to revert to the standard configuration!

11.2 Two stage dialing/callback server functionality

This package contains money-saving features that expand the functionality of your to include DTMF services (two stage dialing) and callback capability. It is particularly useful for companies with employees who travel often, because it eliminates expensive roaming fees.

In a two stage dialing scenario, the caller dials a number which the connects with the integrated DTMF platform. He then enters the destination number via DTMF. The establishes the connection to the destination number.

Callback takes place, when the initial call is released and the calling party is called back in a second call as a response.

Depending on your , various intelligent solutions as a call server are possible. The most important scenarios and properties are described here. The scenarios can also be combined to suit your needs.

- Announcements
- Two stage dialing with DTMF
- Callback with DTMF and OAD as callback number
- Callback with DTMF and preconfigured callback number
- Callback to OAD with predefined destination number
- Callback with PIN and preconfigured callback number

To indicate that the number transmitted using DTMF tones is complete, the caller can press the # key. Otherwise, there is a default timer set to 5 seconds, after which DTMF transmission will automatically end.

To correct a wrong destination number, the caller can press the * key. The DTMF announcement is replayed and the destination number can be entered again.

CDR entries for calls routed as callback with DTMF include the connection times for the A and B parties. The times are separated by a slash (`/`). If no connection is established to the B party, an entry recording the A party's connection time is generated in the `failed.log` file.

Activating DTMF tone recognition

The can recognize DTMF tones and use them to initiate calls. In the `pabx.cfg`, enter a virtual DTMF controller, as described in Table 5.14. The corresponding Subscriber entry contains the options:

`TRANSPARENT ROUTER CHMAX[5]`

The `5` refers to the maximum number of simultaneous channels used for DTMF recognition.

Example 11.2 Activating DTMF tone recognition

```
...
Controller20 = 41 DTMF
...
Subscriber20 = TRANSPARENT ROUTER CHMAX[5]
...
```

The must be restarted to activate this configuration.

11.2.1 Announcements

An announcement can be played immediately after the connection has been established. The announcement's file format must be G.711. A converter tool to convert wave files to the G.711 format is available from TELES free of charge. The announcement can be defined in the virtual DTMF controller's `Subscriber` line using the following entry in the `pabx.cfg` file:

`DTMF[<sec>,/<dir>/<file>]`

`<sec>` refers to the maximum number of seconds after which the connection to the DTMF controller is teared down. Please enter a time that is longer than the announcement time. The entry `<dir>` refers to the directory in which the announcement file is saved. The directory name can be `boot` or `data`. The file extension must be 711.

The `Subscriber` line can be extended to contain up to 26 additional announcement files:

```
DTMF[<sec>,/<dir>/<defaultfile>,/<dir>/<file_a>,/<dir>/<file_b>,...,/
<dir>/<file_z>]
```

A mapping needs to be added to the `route.cfg` file for every announcement to map to the port that is necessary for recognizing DTMF tones:

`MapAll<number>=<DTMFport>DTMF` (for the default announcement file)
`MapAll<number>=<DTMFport>DTMFa` (for file_a)
`...`
`MapAll<number>=<DTMFport>DTMFz` (for file_z)

To stop the announcement after the entry of the first digit, a [needs to be added behind DTMF to the respective MapAll parameter:

`MapAll<number>=<DTMFport>DTMF[`

In this example, a maximum of 5 channels can recognize DTMF tones and change them into dialing data. The default announcement is named `DTMF1.711` and must be available in the `boot` directory. There is one additional announcement file named `DTMF2.711` which must also be available in the `boot` directory. Users calling in from GSM hear the default announcement. After entry of the first digit, the annoucement stops. Calls coming in from analog sources trigger the announcement `a` stored in `file_a`.

Example 11.3 Announcements configuration in the `pabx.cfg`

```
Controller20 = 41 DTMF
Subscriber20 = TRANSPARENT ROUTER DTMF[30,/boot/DTMF1.711,/boot/DTMF2.711]
CHMAX[5]
```

Example 11.4 Announcements configuration in the `route.cfg`

```
Restrict20 = IN1
Restrict10 = IN2
MapAllIN1 = 41DTMF[
MapAllIN2 = 41DTMFa[
```

11.2.2 Two stage dialing with DTMF

The user dials a number in the system that is connected with the virtual DTMF controller. He then enters the number he wants to be connected to.

Make the following entries in the `route.cfg` to connect a call directly:

`MapAll<number>=<DTMFport>DTMF`

Maps the calls to the DTMF port.

`MapAllDLA=<port>`

Sends all digits that have been entered as DTMF to the specified port. To prevent abuse, the following entry can be made to configure a PIN before the actual call number:

`MapAllDLA<pin>=<port>`

In addition, the mapping to the DTMF port can be extended by the capital letters C to Z to send the digits to different ports:

```
MapAll<number>=<DTMFport>DTMFC
MapAll<number>=<DTMFport>DTMFD
...
MapAll<number>=<DTMFport>DTMFZ
```

must be combined with:

```
MapAllDLC=<port>
MapAllDLD=<port>
…
MapAllDLZ=<port>
```

In the following example, all calls coming from GSM are connected to the virtual DTMF controller and the call that comes in as DTMF tones is directed to port 9. Calls to the number 12345 are connected to the virtual DTMF controller and are directed to port 10.

Example 11.5 Two stage dialing with DTMF 1

```
Restrict20=IN
MapAllIN=41DTMF
MapAllDLA=9
MapAll12345=41DTMFC
MapAllDLC=10
```

In the folllowing example, two announcement files are listed in the `pabx.cfg`. All calls coming from GSM are connected to the virtual DTMF controller and the default announcement (DTMF1.711) is played. The call that comes in as DTMF tones is directed to port 9. Calls to the number 12345 are connected to the virtual DTMF controller and the DTMF2.711 announcement is played. The call that comes in as DTMF tones is directed to port 10.

Example 11.6 Two stage dialing with DTMF 1 (`pabx.cfg`)

```
Controller20 = 41 DTMF
Subscriber20 = TRANSPARENT ROUTER DTMF[30,/boot/DTMF1.711,/boot/DTMF2.711]
CHMAX[5]
```

Example 11.7 Two stage dialing with DTMF 1 (`route.cfg`)

```
Restrict20=IN
MapAllIN=41DTMF
MapAllDLA=9
MapAll12345=41DTMFaC
MapAllDLC=10
```

11.2.3 Callback with DTMF and OAD as callback number

The user calls a number to trigger callback to his calling number (OAD). After the alert tone, the system or the user hangs up and the user is called back. When he takes the call, he is requested by announcement to enter the destination number using DTMF tones. The connection to the destination number is then established.

The following entries in `route.cfg` will initiate callback to the calling party's number:

`MapAll<number>=CALLB`

This entry contains the number that needs to be called to trigger callback.

`MapCallbackAlert=<sec>`

If this optional parameter is configured, the will release the trigger call after the given number of seconds. The default value is 180. If you enter the value 1, no alert tone is sent. For all other entries an alert tone is sent. The range of possible values is between 1 and 255.

`MapCallbackDelay=<sec>`

If `MapCallbackAlert=<sec>` is configured, `MapCallbackDelay=<sec>` is needed to define the time between the release of the trigger call and the call back. The parameter is used to set up the call back. A callback delay of 2 to 5 seconds is recommended.

`MapCallbackRejCause=<cause value>`

If this optional parameter is configured, the trigger call to the gateway is rejected with the here entered cause value. The default value is 16 (Normal call clearing). The values 1 - 127 are possible. This parameter is not available for mobile ports.

`MapAllCB=<port>`

Calls back the calling party's number via the defined port.

`MapAllDTMF=<DTMFport>DTMF`

Maps to the port that is needed for recognizing DTMF tones.

`MapAllDLA=<port>`

Sends all digits that have been entered as DTMF to the specified port.

The letter "B" in `CALLB` can be replaced by any capital letter from C to Z to allow for different routings. Correspondingly, "B" in `MapAllCB` and "A" in `MapAllDLA` need to be replaced by a matching capital letter C to Z:

```
MapAll<number>=CALLC
MapAll<number>=CALLD
...
MapAll<number>=CALLZ
```

corresponds with:

```
MapAllCC=<port>
MapAllCD=<port>
...
MapAllCZ=<port>
```

corresponds with:

```
MapAllDLC=<port>
MapAllDLD=<port>
...
MapAllDLZ=<port>
```

`MapCallbackMaxDuration=<sec>`

This optional parameter defines the length of time that the system tries to call back the user. The entered value can be between 1 and 180. If the time limit has been exceeded, the system terminates the call.

`CallbackOAD=<OAD>`

Configure this optional parameter in the `[System]` section of the `pabx.cfg` file to display the here defined OAD when the gateway calls back the user. Not possible for mobile ports.

In this scenario, there is no need to adjust the `MapAllDTMF=<DTMFport>DTMF` command.

You need to activate digit collection if the call goes to the mobile network or VoIP (using SIP)

In this example, the number 123 is called to trigger a callback. After 3 seconds the releases the call. The calling party's number (OAD) is called back via port 9. The user is requested to enter the destination number as DTMF. The digits that come in as DTMF are sent to port 9.

Example 11.8 Callback with DTMF and OAD as callback number 1

```
MapAll123=CALLB
MapCallbackAlert=3
MapCallbackDelay=5
MapAllCB=9
MapAllDTMF=41DTMF
MapAllDLA=9
MapCallbackMaxDuration=180
```

This example is similar to the above one except that the port for the destination call is now port 10. The capital letter "C" is used in the mappings to allow for this additional routing.

Example 11.9 Callback with DTMF and OAD as callback number 2

```
MapAll456=CALLC
MapCallbackAlert=3
MapCallbackDelay=5
MapAllCC=9
MapAllDTMF=41DTMF
MapAllDLC=10
MapCallbackMaxDuration=60
```

11.2.4 Callback with DTMF and preconfigured callback number

This feature is especially useful when the calling party's number is not transmitted. The user calls a predefined number that is mapped to a defined callback number. An alerting takes place. The user hangs up and is called back at the defined callback number. After the user has accepted the call, he must enter the destination number via DTMF. The connection is set up when he finishes dialing.

Make the following entries in `route.cfg` to initiate callback to a fixed number:

```
MapAll<number>=CALL<callback number>
MapAllDTMF=<DTMFport>DTMF
MapAllDLA=<port>
```

In the following example, calls to the number 123 are connected with the number 03012345. The number that comes in as DTMF is directed to port 9.

Example 11.10 Callback with DTMF and preconfigured callback number

```
MapAll123=CALL903012345
MapAllDTMF=41DTMF
MAPAllDLA=9
```

11.2.5 Callback to OAD with predefined destination number

The user calls a predefined number in the system. An alerting occurs. The user hangs up and is called back on his calling party number (OAD). After the user accepts the call, he is connected to a fixed, preconfigured number (e.g. operator or corporate central office).

Make the following entries in `route.cfg`:

```
MapAll<number>=CALLB
MapAllCB=<port>
MapAllDTMF=<port><destination number>
```

Different routings are possible by using the capital letters "C" to "Z".

In the following example, the caller dials 123456 and his OAD is called back through port 9. He is then connected with the operator's number 0 through port 10.

Example 11.11 Callback to OAD with predefined destination number 1

```
MAPAll123456=CALLB
MapAllCB=9
MapAllDTMF=100
```

In the following example, the caller dials 987654 and his OAD is called back through port 9. He is then connected to the destination number 03011111 through port 9.

Example 11.12 Callback to OAD with predefined destination number 2

```
MAPAll987654=CALLC
MapAllCC=9
MapAllDTMFC=903011111
```

11.2.6 Callback with PIN and preconfigured callback number

The user dials a number in the system that is connected to the DTMF platform. He then enters a predefined PIN that maps him to a predefined fixed number that is to be called back. He then hangs up. After he takes the callback, he can enter the destination number using DTMF tones.

Make the following entries in route.cfg:

```
MapAll<number>=<DTMFport>DTMF
MapAllDLA<pin>=CALL<port><callback number>
MapAllDTMF=<DTMFport>DTMF
MapAllDLA=<port>
```

The number 123456 is dialed and the PIN 123# is entered. The call is then connected to the pre-configured callback number 004930123456 through port 9. After connect, the destination number is entered using DTMF and transmitted through port 9.

Example 11.13 Callback with PIN and preconfigured callback number 1

```
MAPAll123456=41DTMF
MapAllDLA123=CALL9004930123456
MapAllDTMF=41DTMF
MapAllDLA=9
```

The number 987654 is dialed and the PIN 456# is entered. The call is then connected to the pre-configured callback number 004930987654 through port 9. The capital letter “C” is used in the mappings to allow for this additional callback number. After connect, the destination number is entered using DTMF and transmitted through port 9.

Example 11.14 Callback with PIN and preconfigured callback number 2

```
MAPAll987654=41DTMFC
MapAllDLC456=CALL9004930987654
MapAllDTMF=41DTMF
MapAllDLA=9
```

The user must enter a # following the PIN. Otherwise the callback to the predefined number will not take place.

11.3 Least cost routing

s are connected between the customer’s private branch exchange (PBX) and the public telephone network (ISDN) and/or VoIP. The customer saves connection charges and can effortlessly and automatically connect to the corporate network as needed using one of six routing methods:

- Carrier selection
- Dedicated lines
- Direct line access with subaddressing
- Direct line access with DTMF
- Callback with subaddressing
- Callback with DTMF

This manual contains information only on carrier selection. If you would like to configure any other variation, please contact TELES.

Calls are routed transparently for the PBX and its users. s can generate charges and route calls using alternate settings in case of network failures. The provider can access the system via ISDN for routine maintenance and monitoring.

The following additional services are supported by this feature package:

- Generation of charges
- Time-controlled configuration
- Alternative routing

11.3.1 Carrier selection

Carrier selection is currently one of the most commonly used routing methods supported by the . In the , this routing process also includes direct calls into the mobile network or through a VoIP network. That means the system is a full-fledged second generation LCR.

11.3.1.1 Routing entries

Use the MapAll command to route calls using Carrier Selection.

Use the following syntax for connections routed via the provider:

`MapAll<AreaCode>=9<CarrierSelection><AreaCode>`

where <AreaCode> is the number or number range to be routed and <CarrierSelection> is the access number required to reach the provider’s network.

For unrouted connections (placed via the public telephone network), use:

`MapAll<AreaCode>=9<AreaCode>`

To block undesired carrier selection prefixes use:

`MapAll<CarrierSelection>=&91;(Busy signal)`

In the following example, calls to international destinations are terminated through the VoIP interface. The profile names iG1 and iG2 in the routing entries refer to different VoIP carriers. All other national long distance and local calls are routed through an alternative carrier (01019). All calls from the PSTN to the PBX are put through transparently.

Example 11.15 Least cost routing 1

```
MapAll001=40iG1:001
MapAll0044=40iG2:0044
...
MapAll01=90101901
MapAll02=90101902
...
MapAll09=90101909

MapAll1=9010191
MapAll2=9010192
...
MapAll9=9010199

Restrict9=10
```

Be sure to enter phone numbers in the routing file in ascending order.

11.3.2 Alternative routing settings

Alternative routing refers to the ability to establish connections using a different (alternative) network in case of provider failure (e.g. all mobile controllers are in use). Alternative routing ensures uninterrupted operation of the attached PBX. In such cases, connections are often made via the public network using the Redirect command:

```
MapAll<num>=<port><num>
Redirect3<port><num>=<placeholder>
```

`MapAll<placeholder>=<alt port><num>`

Example 11.16 Least cost routing 2

```
MapAll01555=2621201555
Redirect32621201555=A
MapAllA=901555
```

11.3.3 Charge models

s can either generate charge information or transmit received charges from the public or corporate networks to the attached PBX. Charge simulation is achieved using variables, which ensure a great degree of flexibility for the implementation of many different charge models including:

- Charge units per time unit
- Flat rate (initial charge without time interval)
- Initial charge plus time interval
- Initial charge plus time interval after delay
- Time interval and/or flat rate plus received charges
- Received charges only or no charge information
- Initial toll-free period with retroactive charge generation afterwards
- Price-per-minute (with whole second accuracy)

In this chapter, **unit** means that charge information is transmitted as a whole number value, and **currency** means that the charge information is sent as a currency amount (e.g. EUR 3.45). The charge impulse generation options can be set for each mapping by adding charge-specific arguments to the MapAll commands, as shown below. The use of each variable is explained in Table 11.1.

`MapAllsrc=dst time start/wait` and

`MapCallBackOutprovsrc=dst mode time start/wait`.

Table 11.1 Charge variables

Variable	Purpose
time	Determines the length of each time interval (how long each unit lasts). The value is entered in seconds and hundredths or thousandths of a second (the maximum value accepted is 655.35 seconds, 65.535 if thousandths are entered). If time is set to zero or not present no charges are generated. External charge information is passed through if received.
start	Sets the initial unit level. Enter a value between 0 and 127 whole units. If you want to use a flat rate, set the desired number of units here and set the `wait` to 255 to turn off the time interval.
wait	Determines the delay after which charge generation begins. Once this time has elapsed, charge impulses are sent at the interval determined by `time`. Enter a value between 0 and 254 seconds. 255 deactivates the charge pulse. In this case, the time variable is ignored.

To generate charge impulses from the first second of the call, the following parameter needs to be configured in the [System] section of the `pabx.cfg` file:

`InitialCharge=On`

External charges can be added to the generated charges by adding 128 to the *start* value. (The value range for the initial unit level is still set from 0 to 127). The maximum supported number of units per connection is 32767 units.

Additional adjustments may be made to allow for the implementation of new charge models.

- When charge information is sent as Currency hervorheben, values can be expressed in thousandths for greater precision in charge calculation.
  For the internal Layer 3 protocols, charges can be specified to the third decimal place (thousandth) using the /Value option (Example: /Value:1.056). In this manner, charges can be generated for units of currency requiring accuracy to the third decimal place or for fractions such as tenths of a cent. This allows for greater flexibility in the transmission of charges to terminal devices. In order to make use of this option, connected devices must support "AOC-D Currency". In the current version, this option is only available for the DSS1 protocol.
- A multiplication factor can be specified for received or generated charges.
  During the charge generation process, each charge unit is multiplied by a preset factor. This factor appears in the mapping entry after the time and start/wait variables (`MapAllsrc=dst time start/wait*factor`).
  Each unit, for example, can be converted to 12 cents. The following example illustrates the use of this feature:

In the following example, all received charge units are multiplied by 12 and passed on. If `AOC-Currency` is set on the internal port, each unit appears as 12 cents.
The multiplication factor is also used to implement two new charge models:

- If the factor value exceeds 128, this indicates the use of an initial toll-free phase followed by retroactive charge generation.
- If the multiplication factor is set to 255, a "minute price" is used in place of the time variable.

Example 11.17 Charging pulse example

```
...
MapAll1=91 1 128/255*12
...
```

These charge models are explained under Retroactive charge generation after initial toll-free period.

11.3.4 Generating charges with the

To generate charges for the attached PBX, add the charge variables described in Table 11.1 to the MapAll commands according to the requirements of the corporate network environment.

In the following mapping example, `time`=1.65, `start`=131, `wait`=0. Three initial tariff units (131-128) are transmitted upon connection and a new unit is generated every 1.65 seconds and transmitted the next full second. Charges received from the public network for the connection to the corporate network dial-in node are added and transmitted (because 128 has been added to the start variable's value).

Example 11.18 Generating charges with the 1

```
...
MapAll0172=9123450172 1.65 131/0
...
```

Upon connection establishment, 3 initial tariff units (131-128) are transmitted. Then a 10-second delay (wait=10) elapses before charge impulses are generated according to the time variable (a new unit is generated every 1.65 seconds and transmitted the next full second). Charges received from the public network for the connection to the corporate network dial-in node are added and transmitted (because 128 has been added to the start variable's value).

Example 11.19 Generating charges with the 2

```
...
MapAll0172=9123450172 1.65 131/10
...
```

New charge models can be implemented by taking advantage of the multiplication factor in conjunction with the *time* and *start/wait* variables.

Retroactive charge generation after initial toll-free period

The charge generation process has been expanded to allow for the implementation of this new charge model. In this scenario, an initial period is free of charge, but after that charges are calculated for the entire call. For example: the first minute is free, but as soon as the second minute begins, charges are incurred for the first minute as well.
The multiplication factor is set to a base value of 128. If the value exceeds this base, the remaining value represents the number of units charged with each *time* interval. The following configuration generates one unit (129-128) per minute (*time*=60 seconds) retroactively after the first minute (*wait*=60 sec.).

Example 11.20 Retroactive charge generation after initial toll-free period

```
...
MapAll030=901019030 60 0/60*129
...
```

Price per minute

A price per minute charge model can be implemented as of version 5.01 in one of two ways:

- The attached PBX supports Advice of Charges as Currency
- If not, the PBX can be configured to assign one thousandth (1⁄1000) of a currency unit (€0.001 or 1⁄10 of a cent) to each charge unit.

If thousandths are defined, a maximum value of 65.535 is possible. If tenths are defined, a maximum value of 6553.5 is possible. Sind diese Zahlen im englischen Format, also 65 euros und 53 und ein halb cents bzw. 6553 euros und 50 cents?

This model does not always guarantee whole second accuracy (depending on the rates), but it is significantly more precise than the standard charge generation method.

If the attached PBX supports Advice of Charges as Currency, include the following line in the ’s pabx.cfg.

Example 11.21 Price per minute 1

```
...
Controller01=10 NTS2M DSS1 CRC4 UNIT:€ VALUE:0.001
...
```

If the PBX does not support this AOC model, but allows for the assignment of one thousandth (1⁄1000) of a currency unit (€0.001 or 1⁄10 of a cent) for each charge unit, the above entry need not be included. The configuration entries must make use of the multiplication factor for a single unit as shown below.

Example 11.22 Price per minute 2

```
...
MapAll902=90103002 1.00 0/0*4 ; each second costs €0.004 (€0.24 / minute)
MapAll909=90108809 1.00 0/0*5 ; each second costs €0.005 (€0.30 / minute)
...
```

If the minute price does not allow generated charges to “fit” evenly into a second (such as 20 cents per minute or 0.33 cents per second), the system can be configured to generate 10 “points” every 3 seconds (€0.01 or 1 cent).

Example 11.23 Price per minute 3

```
...
MapAll902=90101302 3.00 0/0*10 ; 3 seconds cost €0.01 (€0.20 / minute)
MapAll909=90105009 2.00 0/0*3  ; 2 seconds cost €0.003 (€0.09 / minute)
...
```

The “points” method allows for a more precise calculation of smaller intervals.

The price per minute can also be specified in each routing entry by setting the multiplication factor to 255, to signalize to the system that a minute price is being used instead of the interval usually specified with the time variable. The attached PBX must support Advice of Charges as Currency, and the appropriate settings must be made in the ’s pabx.cfg as described under Price per minute.

The examples below show sample entries with rates of 18 and 9 cents per minute.

Example 11.24 Price per minute 4

```
...
MapAll902=90101302 0.18 0/0*255 ; €0.18 / minute
MapAll909=90105009 0.09 0/0*255 ; €0.09 / minute
...
```

Example 11.25 Price per minute 5

```
...
Controller01=10 NTS2M DSS1 CRC4 UNIT:€ VALUE:0.010
...
```

If greater precision is desired (1⁄1000 of a currency unit – $0.001 or 1⁄10 of a cent), use settings such as the following.

Example 11.26 Price per minute 6

```
...
MapAll902=90101302 1.80 0/0*255 ; €0.18 / minute
MapAll909=90105009 0.90 0/0*255 ; €0.09 / minute
...
```

Example 11.27 Price per minute 7

```
...
Controller01=10 NTS2M DSS1 CRC4 UNIT:€ VALUE:0.001
...
```

11.4 Online traffic monitor

The Online Traffic Monitor allows you to collect and monitor statistics and call detail records (CDRs). The following functions are possible with this feature package:

- ASR calculation
- Generation of CDRs
- Generation of online CDRs using e-mail

11.4.1 Generating and retrieving CDRs

Do not configure both StatisticTimeReset and StatisticTime or StatisticTimeReset and StatisticCounter together.

With the `Log` and `failedlog` commands, you save CDRs and unconnected calls in the .

For these parameters (`Log` and `failedlog`), a folder and file name must always be specified after the equal sign. The function is not active (no data is recorded) until a file name is specified.

Example 11.28

```
Log=/data/cdr.log
failedlog=/data/failed.log
```

With recording of files, system maintenance increases. You have to be sure to download or delete files and ensure that there is enough disk space left on the hard drive.

The service indicator listed in the call log and missed calls list describes the type of connection as a four digit hexadecimal number. The coding is conducted according to the 1TR6 standard. A few frequently used values are listed below:

Table 11.2 1TR6 service indicators

Service Indicator	Definition
0101	ISDN-telephony 3.1 kHz
0102	analog telephony
0103	ISDN-telephony 7 kHz
0200	Fax group 2
0202	Fax group 3
0203	Data via modem
0400	Telefax group 4
0500	SMS or BTX (64 kbps)
0700	Data transfer 64 kbps
07…	Bit rate adaptation
1001	Video telephone – audio 3.1 kHz
1002	Video telephone – audio 7 kHz
1003	Video telephone – video

For detailed information on how to automatically divide the files (e.g. on a daily basis), please refer to the Chapter 5.2.1.2.

11.4.1.1 Call log

The following entry in the pabx.cfg configuration file activates the capability to generate CDRs in the :

```
Log=/boot/cdr.log
```

The cdr.log file is stored in the data directory. New entries are always added to the end of the file. The file is open only during editing.

Each line represents an outgoing call with the following information separated by commas:

Table 11.3 Call Log Entries

Column	Description
0	Version
1	Start time (format DD.MM.YY-hh.mm.ss)
2	End time (format DD.MM.YY-hh.mm.ss)
3	Source. The following format applies: [node number:automatically set internal channel number]
4	Destination. The following format applies: [node number:automatically set internal channel number]
5	IMSI (optional)
6	IP logging signaling: RTP (optional)
7	Audio codec used (optional)
8	Frame size (optional)
9	Service indicator (cf. Chapter 11.4.1 on page 206)
10	Call duration
11	Cause values
12	Charge from the public line (in units)
13	Charge generated from the system (in units) (if configured)
14	Cell ID (if mobile call)
15	RSSI (if mobile call)

Sample log file

The example below shows a sample log file.

Example 11.29 Sample log file

```
C1,25.11.09-10:16:20,25.11.09-10:16:27,9,111,,,,,0102,7,1f,0,,3663,10,,,
C1,25.11.09-10:35:16,25.11.09-10:35:26,9,111,,,,,0102,10,1f,0,3,38922,14,,,
C1,25.11.09-10:38:30,25.11.09-10:38:41,9,111,,,,,0102,11,90,0,3,38922,14,,,
```

Differentiating between ports in the same trunk group

To differentiate between ports with the same number in the CDRs, a specific node number must be defined. You can expand the subscriber configuration line with the keyword NODE[<no.>] for this purpose. <no.> can be a string of between 1 and 15 characters:

```
Subscriber<xx>=... NODE[<num>]
```

In the below formula, <num> consists of a four-digit number that is included in the CDR.

Example 11.30 Differentiating between ports in the same trunk group 1

```
C1,25.11.09-10:16:20,25.11.09-
10:16:27,[0000:01]9,[0006:01]111,,,,,0102,7,1f,0,,3663,10,,,
```

The following example shows the pabx.cfg configuration file changed according to the formula.

Example 11.31 Differentiating between ports in the same trunk group 2

```
...
Subscriber00=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0001]
Subscriber01=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0002]
Subscriber02=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0003]
Subscriber03=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0004]
Subscriber04=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0005]
Subscriber05=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0006]
Subscriber06=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0007]
Subscriber07=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0008]
Subscriber08=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0009]
Subscriber09=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0010]
Subscriber10=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0011]
Subscriber11=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0012]
Subscriber12=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0013]
Subscriber13=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0014]
Subscriber14=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0015]
Subscriber15=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] CHADDR ALARM
NEXT NODE[0016]
...
```

Differentiating between SIM cards

The CDR can contain the IMSI (International Mobile Subscriber Identity), which identifies each SIM card used.

Example 11.32 Differentiating between SIM cards 1

```
C1,25.11.09-10:35:16,25.11.09-
10:35:26,9,111,123456789123451,,,,0102,10,1f,0,3,38922,14
```

The following example shows the `pabx.cfg` configuration file changed according to the formula.

Example 11.33 Differentiating between SIM cards 2

```
...
Subscriber04=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM4,IMSI] CHADDR
ALARM NEXT
Subscriber05=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM4,IMSI] CHADDR
ALARM NEXT
Subscriber06=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM4,IMSI] CHADDR
ALARM NEXT
Subscriber07=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM4,IMSI] CHADDR
ALARM NEXT
...
```

Activating peer data for VoIP calls

To generate a VoIP-call CDR entry that includes IP addresses for the remote device’s signaling and voice data, audio codec and frame size, the entry `VoipIpLogging=Yes` must be included in the VoIP profile.

The following entry shows the `route.cfg` configuration file changed according to the formula.

Example 11.34 Activating peer data for VoIP calls 1

```
[Voip:DF]
VoipDirection=IO
VoipPeerAddress=192.168.0.2
VoipIpMask=0xffffffff
VoipCompression=g729 g711a t38
VoipMaxChan=30
VoipSilenceSuppression=Yes
VoipSignalling=0
VoipTxM=4
VoipIPLogging=Yes
```

The following CDR entry includes IP addresses for signaling and voice data, audio codec and frame size.

Example 11.35 Activating peer data for VoIP calls 2

```
C1,24.11.09-16:52:20,24.11.09-
16:52:22,401419,9777,,172.20.25.103:172.20.25.103,G711a,20,0101,2,10,0,,,11
```

CDRs for callback and two stage calls

In the case of CDR entries for Two stage dialing/Callback calls, the beginning and ending times for the first call leg is always used as the call time. The call time in seconds appears first for the first leg, followed by a slash and the connection time for the second leg.

Example 11.36 CDRs for callback and two stage calls

```
C1,24.11.09-17:15:29,24.11.09-
17:15:57,[0002:01]CB,[0008:01]DLA,,172.20.25.103:172.20.25.103,G711a,20,0102,28/
3,90,0,,,
```

11.4.1.2 Missed calls list

To avoid sending these values as the reason for call teardown, translate the cause values to standard values Chapter 8.14

All incoming calls that are not connected can be recorded in a list to facilitate return calls. Recording is activated using the `failedlog=<name>` entry in the `pabx.cfg`. Specify a file name, e.g. failedlog=failed.log. Once this setting is made, recording begins at once.

Each line represents an unaccepted incoming call with the following information separated by commas:

Table 11.4 Failed log entries

Column	Description
0	Version
1	Start time (format DD.MM.YY-hh.mm.ss)
2	Source. The following format applies: [node number:automatically set internal channel number]
3	Destination. The following format applies: [node number:automatically set internal channel number]
4	IMSI
5	IP logging signaling: RTP
6	Audio codec used
7	Frame size
8	Service indicator (cf. Chapter 11.4.1 on page 206)
9	Cause values
10	Call duration (if the call does not result in an Alerting, the entry will be -1)
11	Number of call attempts

Table 11.4 Failed log entries *(continued)*

Column	Description
12	Cell ID
13	RSSI

The example below shows a sample failed log file.

Example 11.37 Sample failed log 1

```
V1,24.11.09-16:13:08,[0006:01]IN,[0008:01]GSM,123456789123456,,,,0101,92,-
1,1,34193,9
V1,24.11.09-16:33:34,[0006:01]IN,[0008:01]GSM,123456789123456,,,,0101,92,-
1,1,34193,12
V1,24.11.09-16:35:19,[0006:01]IN,[0008:01]GSM,123456789123456,,,,0101,92,-
1,1,34193,11
V1,24.11.09-16:35:59,[0006:01]IN,[0008:01]GSM,123456789123456,,,,0101,92,-
1,1,34193,11
V1,24.11.09-16:37:29,[0006:01]IN,[0008:01]GSM,123456789123456,,,,0101,92,-
1,1,34193,11
V1,24.11.09-
16:39:17,[0006:01]IN,[0008:01]GSM,123456789123456,,,,0101,ff,7,1,34193,11
```

The reason the connection could not be established is specified using DSS1 codes:

91 – (user busy)

ff – call not answered (disconnected by calling party)

When callback with DTMF is configured and no connection is established to the B subscriber, an entry recording the A subscriber's connection time is generated in the failed.log file.

Example 11.38 Sample failed log 2

```
V1,24.11.09-
16:39:17,[0006:01]IN,[0008:01]GSM,123456789123456,,,,0101,ff,7,1,34193,11
```

The CDR contains the IP addresses for signaling and voice data. The first IP address is the signaling address and the second one is the RTP address.The IMSI is written behind the IP addresses.

Example 11.39 Sample failed log 3

```
V1,24.11.09-16:52:20,24.11.09-
16:52:22,[0008:01]401419,[0006:01]IN777,262032441017556,172.20.25.103:172.20.25.
103,G711a,20,0101,2,10,0,,34193,11
```

In the case of missed-call entries for Two stage dialing/Callback calls, dur is the connection time for the first leg.

Example 11.40 Sample failed log 4

```
V1,25.11.09-14:11:10,[0002:01]CB,DLA,,,,,0102,11,14,1,,
```

11.4.1.3 Sending CDRs via e-mail

With an appropriate configuration, you can send corresponding CDRs of outgoing and incoming calls as e-mail. Bear in mind that the mail server must be configured in the [Mail] section of the pabx.cfg, as described in Chapter 5.2.2. The sender is given as cdr and the system's name appears in the subject box. The text box contains the CDR information according to the format for the entry in Log=/data/cdr.log @<account> @<domain>. A space must appear between cdr.log and @<account>; @<domain> is optional. You can also send CDR entries via e-mail to an e-mail recipient.

Enter an @ sign to send each CDR entry as e-mail.

Example 11.41 Sending CDRs via e-mail 1

```
Log=/data/cdr.log @bob@example.com
```

If you replace the first @ sign with an !, the present cdr.log will be sent whenecver a new one is generated.

Example 11.42 Sending CDRs via e-mail 2

```
Log=/data/cdr.log daily 60 5 !bob@example.com
```

11.5 Ported number screening

Ported number screening is a very useful functionality to avoid high routing costs for numbers that have been ported to another network operator.

Number portabilty refers to the ability to transfer either an existing landline or mobile telephone number to another network operator. This way telecommunications subscribers can change operators without having to change their telephone numbers. Routing ported numbers, however, can become very cost intensive due to differences in tariffs.

With ported number screening, an external database is queried to find out if a number has been ported. Either use the iMNP for querying the external database or query the database directly. The iMNP is a proxy that remembers the database information for a defined period of time and renews the query to the external database only after that period has passed. Since every query to the external database costs money, the iMNP helps to reduce querying costs.

The qery result is used in a routing to route the call via the right network operator.

To implement ported number screening, make sure to meet the system requirements which are:

- An active license for number portability.
- An iMNP server or another appropriate server.

Also ensure to adjust your configuration:

- To connect to the number portability database, you must set the entries described in Chapter 5.2.3 Number portability settings.
- Configure your `route.cfg` file to activate ported number screening:

  `DTMFWaitDial=<sec>`
  Sets the time the gateway waits for additional digits.

  `MapAll<num>=|$<prefix><num><<<count>`
  Enable digit collection (pipe symbol) and collect up to the number of digits that is specified under `count`. Add the dollar sign to search the routing table again. Prefix the number with a freely chosen prefix to prepare it for sending to the database.

  `MapAll<prefix>=|D@<num><<01`
  Send the prefixed number to the database. Make sure that the prefix used here matches the above prefix.

  `MapAllQN<query result>=<controller>`
  Map the query result to the respective controller. The query result consists of the LAIN and the number including the country code. Do not forget to also configure routings for numbers that haven't been found in the database or that do not exist and also provide a routing when the server does not respond in a defined period of time.

In the following example, 14 digits are collected (11 digits plus $ph) and a maximum time of 5 seconds is waited for each digit. Every incoming call with a leading digit of 0 results in an iMNP query. The SIM card's LAIN is used instead of controller numbers. All numbers that come back from the iMNP with the LAIN for Carrier_1 (26211) are then routed through Carrier_1's SIM card. The same applies for Carrier_2 (26212), Carrier_3 (26213) and Carrier_4 (26214). Numbers that the iMNP sends back as non-existing (00000) are rejected. Numbers that may exist but are not found in the database (99999) are routed as they come in (normal). If the iMNP does not respond within two seconds (D@0), the call is routed as it comes in, whether it is ported or not.

Example 11.43 Ported number screening

```
DTMFWaitDial=5
MapAll0=|$ph0<<14
MapAllph=|D@0<<01

MapAllQN26211=26211
MapAllQN26212=26212
MapAllQN26213=26213
MapAllQN26214=26214
MapAllQN00000=&81
MapAllQN99999=$normal
MapAllD@0=$normal1
; not in Database
;Carrier_1
MapAllnormal0151=262110151
MapAllnormal0160=262110160
MapAllnormal0170=262110170
MapAllnormal0171=262110171
MapAllnormal0175=262110175
;Carrier_2
MapAllnormal0152=262120152
MapAllnormal0162=262120162
MapAllnormal0172=262120172
MapAllnormal0173=262120173
MapAllnormal0174=262120174
;Carrier_3
MapAllnormal0155=262130155
MapAllnormal0163=262130163
MapAllnormal0177=262130177
MapAllnormal0178=262130178
;Carrier_4
MapAllnormal0159=262140159
MapAllnormal0176=262140176
MapAllnormal0179=262140179
```

11.6 Call recording

Businesses may want to record calls to provide evidence of a business transactions, to ensure that a business complies with regulatory procedures, or to see that quality standards are being met.

Starting from version 16.2, TELES iGATE and VoIPBox PRI systems support call recording systems, such as Nice, to record signalling and voice data.

The is connected via ISDN PRI or VoIP to the different networks or directly to the subscribers. The call recording system is connected with the via VoIP (SIP).

For call recording for PBX systems, the and the call recording system can be put between a PBX and the PSTN to record calls between the PBX extensions and the PSTN.

As many calls as handled by the gateway are recorded. Should the connection to the call recorder fail, the call is rejected altogether. To make the calling parties aware of the recording, an announcement can be played.

The call recording option is available for all interfaces, such as ISDN PRI, GSM, 3G/UMTS, CDMA, VoIP SIP, VoIP H.323. To support recording, the gateway needs to have VoIP resources. Recording is done either through the outgoing leg (for calls going to VoIP), or by using additionally allocated vocoder resources (as a so-called local VoIP loop).

To enable the call recording feature, a licence is required.

11.6.1 Call recording procedure

The following steps describe how call recording works.

- A user sends a call.
- The call is routed through the .
- The call arrives at the called party.
- Signalling messages are sent from the gateway to the call recording system.
- The voice data is sent from the gateway to the call recording systems in two different RTP streams, one for each call direction.

The signalling messages that are sent for all calls within one SIP session are set up in the following manner:

- The gateway sends an INVITE to the call recording system.
- The call recording system sends back a SUBSCRIBE as reply.
- The gateway sends NOTIFYs for all calls
  - when the call starts
  - when an alert is sent
  - when the call fails
  - when the call is connected / disconnected.

11.6.2 Call recording configuration

The following parts must be configured:

- The VoIP profile for communication with the call recorder (in the `route.cfg` file). This profile is needed for sending signalling messages.
- The `VoipNiceMRA` parameter (in the `pabx.cfg` file) to point to this VoIP profile.
- A second VoIP profile for communication with the call recorder (`route.cfg`). This profile contains all data necessary for sending the voice data. If this profile is not different from the profile needed for sending signalling messages, it is not needed.
- A second `VoipNiceMRA` parameter (in the `route.cfg` file) to point to the RTP specific VoIP profile.
- This parameter needs to be included in a VoIP profile that is either used for routing the calls (if the calls go to VoIP) or as the local VoIP loop (for all calls not going to VoIP).
- Finally, if a local VoIP loop is used, the routing must be adjusted to include this loop.

The VoIP profile that is needed for the signalling messages contains the following parameters:

Table 11.5 VoIP profile for signalling messages (`route.cfg`)

Parameter	Description
`[VOIP:name of VoIP profile for signalling mesages]`	Enter here the name of the VoIP profile. The profile has to have the same name as the `VoipNiceMRA` parameter in the `pabx.cfg` file.
`VoipDirection=Out`	For communication with the call recorder, the direction needs to be "out".
`VoipPeerAddress=<IP address of call recorder>`	Enter here the IP address of the call recorder (or, alternatively, the domain name).
`VoipProxy=<IP address of proxy>`	Optional. Enter the IP address to which you want to send the INVITE, if the INVITE is not to be sent to the IP configured in `VoipPeerAddress`.
`VoipSignalling=<num>`	Enter 1 for SIP signalling via udp, 2 for SIP via tcp , or 3 for SIPS.
`VoipOwnAddress=<IP address of gateway>`	Optional. Enter here the IP address or domain name as requested by the peer.

To attach the gateway to the call recorder, an additional entry must be configured in the `System` section of the `pabx.cfg` file:

Table 11.6 VoipNiceMRA parameter for signalling messages (`pabx.cfg`)

Parameter	Description
`VoipNiceMRA=<name of VoIP profile for signalling mesages>`	The name entered here must be the name of the VoIP profile for signalling messages.

An RTP specific VoIP profile, if needed, is configured in the `route.cfg` file. Its name must comply with the entry in the second `VoipNiceMRA` parameter. This VoIP profile must contain a `VoIPCompression` parameter which needs to match the codec used for actual calls. All other parameters are configured as described in the VoIP profile above:

Table 11.7 VoIP profile for RTP data (`route.cfg`)

Parameter	Description
`[VOIP:name of VoIP profile for RTP data]`	Enter here the name of the VoIP profile. The profile has to have the same name as the `VoipNiceMRA` parameter in the route.cfg file.
`VoipDirection=Out`	For communication with the call recorder, the direction needs to be "out".
`VoipPeerAddress=<IP address of call recorder>`	Enter here the IP address of the call recorder (or, alternatively, the domain name).

Table 11.7 VoIP profile for RTP data (`route.cfg`) *(continued)*

Parameter	Description
`VoipProxy=<IP address of proxy>`	Optional. Enter the IP address to which you want to send the INVITE, if the INVITE is not to be sent to the IP configured in `VoipPeerAddress`.
`VoipSignalling=<num>`	Enter 1 for SIP signaling via udp, 2 for SIP via tcp , or 3 for SIPS.
`VoipOwnAddress=<IP address of gateway>`	Optional. Enter here the IP address or domain name as requested by the peer.
`VoipCompression=<codec>`	Needs to match the codec used for the actual call. Usually g711a, for the USA g711u.

Add the `VoipNiceMRA` parameter to the VoIP profile via which you route your calls to VoIP. That parameter must points to the RTP specific VoIP profile. If no RTP specific VoIP profile exists, `VoipNiceMRA` must point to the VoIP profile needed for signalling messages.

If you want to record calls that do not go to VoIP, create a VoIP profile that you use as local loop. Add `VoipNiceMRA` to this profile. The local loop needs the following entries:

Table 11.8 Local VoIP loop (`route.cfg`)

Parameters	Description
`VoIPDirection=IO`	The direction needs to be in/out.
`VoipPeerAddress=127.0.0.1`	Enter 127.0.0.1. Everything that is sent to this IP address is sent back to the gateway in a loop.
`VoipCompression=<codec>`	Needs to match the codec used for the actual call. Usually g711a, for the USA g711u.
`VoipSignalling=1`	Enter 1 for SIP signalling via udp.
`VoipNiceMRA=<name of VoIP profile for RTP data>`	The parameter points to the RTP specific VoIP profile.
`VoipDadIn=<digit/number>`	The digits/numbers defined here appear in front of the original DAD. This parameter is needed to adjust the mapping.
`VoipAnnounce=<path and name of announcement file>`	Add this parameter to play an announcement to the caller. The file must have the 711 format and must be in the boot or data folder.

For all calls not going to VoIP, the local VoIP loop profile must be included in the routing by adjusting the mapping. If, for instance, calls to the number 123 are routed via PRI (port 9), the original mapping would be:

```
MapAll123=9123
```

This mapping needs to be changed to include a local VoIP loop (named LL):

```
MapAll123=40LL:123
MapAllA=9
...
```

```
[Voip:LL]
VoipDadIn=A
```

For calls to the DAD 123 the `MapAll` parameters are searched, `MapAll123=40LL:123` matches the search criteria. As configured in the MapAll parameter, the call is then sent on a local VoIP loop through port 40 and the LL profile. In this profile, the original DAD is prefixed with the capital letter A as configured in `VoipDadIn=A`. The local loop sends back the call to the gateway. The `MapAll` parameters are searched again, a match is found here: `MapAllA=9`. This mapping sends the call to the PRI port 9.

If you want to play an announcement to the caller before connecting the call to the destination, add `VoipAnnounce=<path and name of announcement file>` to the local VoIP loop, or, for calls going to VoIP, to the VoIP profile through which you send the call.

The complete example can be found below. The local VoIP loop is named LL. There is only one VoIP profile used for the communication with the call recorder, named call_recorder. Both `VoipNiceMRA` parameters point to this profile. Calls to the PRI port 9 are sent through a local VoIP loop to enable call recording. An announcement is played to the caller.

Example 11.44 Call recording (`pabx.cfg`)

```
VoipNiceMRA=call_recorder
```

Example 11.45 Call recording (`route.cfg`)

```
MapAll123=40LL:123
MapAllA=9

...

[Voip:LL]
VoipDirection=IO
VoipPeerAddress=127.0.0.1
VoipCompression=g711a
VoipSignalling=1
VoipNiceMRA=call_recorder
VoipDadIn=A
VoipAnnounce=/boot/announcement.711

[Voip:call_recorder]
VoipDirection=OUT
VoipPeerAddress=172.20.27.161
VoipProxy=172.20.25.7
VoipCompression=g711a
VoipSignalling=1
VoipOwnAddress=ext2000@somelocation.com
```

12 Troubleshooting

12 Troubleshooting

This chapter discusses problems that you may encounter when operating your and offers solutions. The reasons are listed that may cause the different problems. Every reason is described in detail, the symptoms are identified and suggestions are made for solving the issue. If you cannot solve a problem using these instructions, please contact your service partner.

Troubleshooting suggestions exist for the following problem areas:

- No connection to the system possible (please refer to Chapter 12.1)
- Software update problems occurred (please refer to Chapter 12.3)

12.1 No connection to the system

The system cannot be accessed via GateManager, the Web interface, or FTP. There can be different reasons for this:

General

- System does not start correctly (please refer to Chapter 12.1.1)
- Web interface is not accessible (please refer to Chapter 12.1.2)

IP access

- IP address settings (please refer to Chapter 12.1.3)
- Firewall issues (please refer to Chapter 12.1.4)
- NAT / PAT settings (please refer to Chapter 12.1.4)

ISDN access

- Dial-in number missing or wrong (please refer to Chapter 12.1.5)
- ISDN port not loaded correctly / not active (please refer to Chapter 12.1.6)

12.1.1 System does not start correctly

The system is not accessible at all or it restarts approximately every two minutes.

Reasons

- A software update failed. For more information on software update problems, refer to Chapter 12.3 on page 231.
- Modifications in the `pabx.cfg` cause the problem.
- A hardware issue causes the problem.

Symptoms

- The system is not accessible via GateManager or FTP.
- The system is accessible, but restarts approximately every 2 minutes.

Solution

- Make sure that FTP is allowed in your network.
- Check whether you have access to the system using FTP.
- If you cannot access your , the operating system is no longer accessible. In that case you need to contact your service partner for help.
- If the system can be accessed but restarts approximately every two minutes, try to solve the problem by taking the following actions:
  - Connect to your via FTP.
  - Change to the boot directory.
  - Delete or rename the `*.tz1` file. This stops the recurring reboot of the system and gives you time to search for the problem.
  - Check the `pabx.cfg` for modifications that may have caused the problem and change or replace the file. Make sure that the `[System]` section is not missing and check the `Controller` entry for the correct numberings, matching `Controller` and `Subscriber` entries, and the correct hardware settings.
  - Upload the system-specific `*.tz1` file again or change it back to its real name.
  - Restart the system:
    - by resetting the power or
    - by going to your FTP client's command line interpreter and selecting restart. This option depends on your FTP client's functionality. For a description of the FTP commands, please refer to Table 4.8, "FTP commands" on page 34.
- If none of the above solutions solve the problem, there might be a hardware issue that causes the system behavior. Please contact your service partner for help in handling a possible hardware problem. Please do not open your on your own as you would thus lose warranty.

12.1.2 Web interface is not accessible

Your can be accessed through a web interface which requires installation in addition to the mandatory system files. Mismatches between system version and the version of the web interface, missing web interface files or no permission for port 80 are most likely the reason why access to a system through the web interface fails.

Reasons

- The system is not accessible through the web interface because the required files were not installed.
- The system cannot be accessed through the web interface because the required version does not match the system version.
- The system cannot be accessed through the web interface because you don't have permission to access port 80 (the firewall is blocking).

Symptoms

- There is no connection to the system through the web interface.

Solution

- In the GateManager, go to the boot directory and check if the `httpd.izg` and `httpd.tz2` files exist. If not, update your software. Follow the steps described in chapter 10.3, "Software update".
- If these files exist but if the system still cannot be accessed through the web interface, check if there is a mismatch between the system version and the version of the files for the web interface and replace the web interface files by matching ones. To do that, follow the steps described in chapter 10.3, "Software update".
- If the problem persists, check if port 80 is blocked by your firewall. For detecting firewall problems, please see also12.1.4, "Firewall issues and NAT / PAT settings" on page 223.

12.1.3 IP address settings

Faulty IP address settings or a disabled ICMP can impede connections to your system.

Reasons

- A syntax error in the IP address settings occurred in the `ip.cfg` file.
- The netmask was not set correctly in the `ip.cfg` file.
- The Internet Control Message Protocol (ICMP) is disabled in your network.
- The IP address is used more than once in your network.

Symptoms

- The system is not accessible via GateManager or FTP.
- The ping command does not work.

Solution

- If ICMP is disabled in your network, please enable it and check whether this solves your problem.
- If the problem still exists, change the IP address using Quickstart.

12.1.4 Firewall issues and NAT / PAT settings

Very often, firewall settings in your network prohibit access to your . Network address translation (NAT) or port address translation (PAT) issues can also cause connection problems.

Reasons

- The configuration of your firewall blocks access to your system.
- NAT / PAT settings block access to your system.

Symptoms

- The gateway cannot be accessed from remote.
- There is ethernet activity on the gateway.
- Ping works if you connect directly to the gateway.
- The gateway can be accessed via GateManager if you connect directly to the gateway.

Solution

- Make sure there is ethernet activity by checking the ethernet port LEDs (please see Chapter 4.9.1 on page 29. If the LED that indicates data traffic is not blinking, the ethernet cable is probably defect. Please replace the cable and check whether this solves your problem.
- If the problem persists, connect your computer directly to the gateway using a crossover cable. Make sure your computer uses the same IP range as your gateway.
- Send a Ping from your computer to the gateway and wait for the corresponding ICMP echo to make sure that the system can be accessed using ICMP.
- From the connected computer, use GateManager to access the system and make sure that this access is working as well.
- If steps 1 to 4 are working, most likely a firewall or NAT / PAT issue is causing the connection problem. Check your firewall or NAT / PAT settings and adjust them to let traffic pass through to your system.

12.1.5 ISDN access: dial-in number missing or wrong

In systems with a BRI / PRI interface, ISDN access can be impaired due to faulty or missing configurations of dial-in numbers.

Reasons

- The configuration of the `RemoteCode` parameter in the `pabx.cfg` is missing or does not match the corresponding dial-in number mapping in the `route.cfg`.
- There is no route configured for the ISDN dial-in number in the `route.cfg` .
- The ISDN dial-in number was configured in the `route.cfg` using the wrong number format (i.e. the trunk number or extension are missing).
- The ISDN dial-in number that has been configured in the `route.cfg` is used for other data routings as well.

Symptoms

- The system cannot be accessed from remote using ISDN.

Solution

- In the `pabx.cfg`, check the configuration of the `RemoteCode` parameter. If, for instance, the configuration is `RemoteCode=BBB`, `BBB` is defined as the dial-in prefix for ISDN remote access.
- Check the `route.cfg` file for a matching mapping of an ISDN dial-in number. The mapping that matches the above `RemoteCode` parameter is `MapAll<num>=BBB DATA` whereby `<num>` is the ISDN dial-in number.
- Ensure that you use the correct ISDN dial-in number format (i.e. trunk number plus extension or extension only) by changing the above mapping to the following catch-all parameter: `MapAll?=BBB DATA` and by tracing the routing attempt. The trace file tells you the correct number format. Adjust the mapping accordingly.
- Make sure that your ISDN dial-in number is not used for any other data routings by taking the same steps as above: change the mapping to `MapAll?=BBB DATA`, trace the routing attempt and adjust the mapping according to your findings.

12.1.6 ISDN port not loaded correctly / not active

Other issues that can inhibit ISDN remote access relate to the ISDN interface.

Reasons

- PRI only: The CRC4 mode does not match the peer’s CRC4 mode.
- The ISDN cable pin assignment is wrong.
- The TE or NT port configurations in the `pabx.cfg` file are missing or wrong.

Symptoms

- The system cannot be accessed from remote using ISDN.
- No calls are possible.

Solution

- Access your system with GateManager and go to port status.
- Verify that layer 1 of your ISDN port is active. If not, open `pabx.cfg` and examine the ISDN controller configuration for the correct CRC4 mode (PRI only). The CRC4 mode needs to comply with the mode of the PRI port's peer. If the peer has CRC4 switched on, the ISDN port also needs to have it switched on (controller entry CRC4). If the peer has CRC4 switched off, the ISDN port also needs to have it switched off (controller entry DF). Try to change CRC4 from on to off or from off to on, reboot the system and check if this addresses your problem.

Example 12.1 PRI controller with CRC4 and DF

```
Controller00=9 TES2M DSS1 CRC4 ;PRI port to DTAG with CRC4 on
Controller01=10 NTS2M DSS1 DF ;PRI port to PBX with CRC4 off
```

- If layer 1 is still inactive, make sure that there is no problem with the wiring. Check the ISDN cable pin assignment. You can do so by looking at the cable colors or by using a cable tester. Please refer to chapter 4.4.2, "PRI wiring" on page 22 for the correct pin assignment.
- Check the entry in the Layer 2 column. If the entry says "MFE" (Multiframe established), the ISDN layer 2 signaling is working correctly. If not, open `pabx.cfg` and check the port entries in the `Controller` /`Subscriber` line for the correct connection type (TE/NT). Make sure that a TE port is always connected to an NT port at the peer's side and vice versa. Please refer to Table 5.14 on page 57 and Table 5.15 on page 59 for an explanation of the `Controller` /`Subscriber` parameters.

12.2 No calls are possible

The system can be accessed but no calls are connected. There can be a number of reasons for such behavior, mostly relating to errors in the configuration.

To identify where your call problem originates, you need to trace the call attempt. You can distinguish the following call behaviors:

- The call does not arrive on the gateway (please refer to Chapter 12.2.1).
- The call arrives on the gateway, but is either rejected or not routed to the right destination address (please refer to Chapter 12.2.2).
- The call arrives on the gateway, is routed to the right destination address, but is rejected elsewhere (please refer to Chapter 12.2.3).

12.2.1 Call does not arrive on the gateway

If the trace of a failed call attempt shows that the call did not arrive on the gateway, follow the troubleshooting suggestions below.

Reasons

- **VoIP**: The firewall configuration in the `ip.cfg` blocks VoIP traffic.
- **VoIP**: The global VoIP settings cause problems.
- **VoIP**: Authentication failed.
- **ISDN**: The ISDN port is not active.
- **Mobile**: The called SIM card is not registered or not available.

Symptoms

- The incoming call does not arrive on the gateway.

Solution

VoIP: If a VoIP call does not arrive on the gateway, check your firewall settings in the `ip.cfg` file. Consider the following:

- Blocking all udp ports in your firewall causes SIP traffic to be blocked as well. If you block all udp ports, make sure to explicitly enable the SIP port, otherwise no SIP calls will be connected. Make sure as well to keep the right order: first enable the SIP port, then block all udp ports.
- The same applies to RTP ports: if all udp ports are blocked, the RTP ports need to be enabled first.
- If the VoIP profile's SIP proxy in the `route.cfg` has been configured using the domain name, make sure that the DNS port has been explicitly enabled in the firewall settings if all udp ports are blocked. First enable the DNS port, then block all udp ports.
- Check that the ports have the correct numbers.
- Keep your firewall entries in the correct order if you are using the quick command: place the more specific entries above the more general ones because the search is done from top to bottom and stops at the first match.

Example 12.2 udp 5060 (SIP) for IP 195.4.12.0/24 range enabled, all other udp ports blocked

```
fw=pass in quick on emac0 proto udp from 195.4.12.0/24 to any port eq 5060 keep
state
fw=block in quick on emac0 proto udp all keep state
```

VoIP: Problems with incoming calls often originate in the global VoIP settings in the `pabx.cfg` file. Global VoIP settings apply to the whole system, not just to any particular VoIP profile.

- Check that the SIP port has been set correctly and matches the sender's destination port. If not, an incoming VoIP call will not reach the gateway. If you are using the default SIP and H.225 ports (SIP: 5060, H.225: 1720), you don't need to set these two parameters in the global VoIP settings. For an explanation of the global VoIP settings please go to Chapter 5.2.1.6 Global settings.

Example 12.3 Global VoIP settings in the `pabx.cfg`

```
; IP Configuration
VoipGlobalMaxChan=16
H225Port=1720 ;default port
SipPort=5060 ;default port
...
```

VoIP: Authentication problems can also be the reason why a VoIP call does not arrive on the gateway.

- Often, gateways need to register at the VoIP carrier before being able to receive calls. Nowadays authentication is mostly done using registrar (SIP) or gatekeeper (H.323) profiles, which you configure in the `route.cfg` file. If a VoIP call does not arrive on the gateway, this might be due to incorrect or missing registrar or gatekeeper profiles. To look up the registration state of your gateway, access it with GateManager and go to Statistics - VoIP Statistics. Check the VoIP profile where the registrar or gatekeeper profile is used. If there is a problem, the "Registration State" column will contain one of these entries, "not registered", "denied", or "timeout". If so, check the username, password and registrar IP or gatekeeper IP in the associated profile. You can likewise look up the registration state in the `protocol.log` file. For an explanation of how to configure gatekeeper profiles please see Chapter 5.3.3 Gatekeeper profiles. To look up information on the registrar profile please read Chapter 5.3.4 Registrar profiles.

Example 12.4 Gatekeeper profile

```
[Gatekeeper:GK1]
RasPort=1719
OwnRasPort=1719
RasId=iGATE01
RasPrefix=01555 01556 01444 01445
GkId=GK
GkAdd=192.168.0.10
GkPwd=
GkTtl=300
GkMaxChan=30
GkDynMaxChan=Yes
```

Example 12.5 Registrar profile

```
[Registrar=reg]
RegId=office.teles.de
RegUser=4930399280
RegPwd=123456789
RegProxy=<ip adr.>
RegPing=20
RegExpires=3600
```

ISDN: If an incoming ISDN call does not arrive on the gateway, check the port status (using GateManager) and the configuration of the ISDN ports in the `pabx.cfg`.

- Access your system with GateManager and go to port status.
- Verify that layer 1 of your ISDN port is active. If not, open `pabx.cfg` and examine the ISDN controller configuration for the correct CRC4 mode (PRI only). For an explanation, please see Chapter 12.1.6 ISDN port not loaded correctly / not active.
- If layer 1 is still inactive, check the wiring of the ISDN cable. For an explanation, go to Chapter 12.1.6 ISDN port not loaded correctly / not active.
- If you see in the GateManager that layer 1 is active but layer 2 does not show the entry "MFE" (Multiframe established), check the port configuration in the `pabx.cfg` file. Refer to Chapter 12.1.6 ISDN port not loaded correctly / not active for details.

Mobile: If the called SIM card is not registered or not available, follow these instructions for troubleshooting.

- Open GateManager, access your system and check the port status of the SIM card.
- If the SIM card is not registered (check the entry in the "Layer 1" column) and no IMSI is displayed (check the entry in the "IMSI" column), this could be because the gateway is not able to read the SIM card. Check the `Subscriber` entries in your system's `pabx.cfg` file for the correct SIM card carrier. Your ECOTEL must have the entries SIM4, if SIM cards are inserted in your system or SIMS, if you are administer your SIM cards with a vGATE. Your iGATE can in addition have the entries SIM24 if you are using a SIM24 carrier.
- If the IMSI is displayed but the "Layer 1" column says "search" or "not registered", there is most likely a problem with the reception. Check if other SIM cards by the same carrier have the same problem. If yes, relocate the antenna. If not, check the SIM card in your mobile phone.
- If the SIM card's PIN has been misconfigured in the `pabx.cfg's Subscriber` line, GateManager notifies you about two failed attempts to access the SIM card. The Layer 1 column displays "2 x wrong PIN". In this case, remove the SIM card from the system and reset it's PIN using a mobile phone. Do not forget to adjust the PIN entry in the port's `Subscriber` line in the `pabx.cfg` file before reinserting the SIM card into the gateway.
- iGATE with SIM 24 carrier only: If GateManager informs you that no SIM card has been found, check the SIM card's position in the SIM24 carrier and insert the SIM card into the position which matches the configuration.

12.2.2 Call is rejected or not routed to the right destination address

The gateway receives the call, but refuses to route it, or routes it to the wrong destination address.

Reasons

- **Calls from and to mobile / VoIP / ISDN / analog**: Routes have been misconfigured in the `route.cfg`.
- **Calls from and to mobile / VoIP / ISDN / analog:** A license is missing, has expired, or contains LAIN restrictions.
- **Calls to mobile**: The configuration of `CHADDR` in `pabx.cfg` is causing problems.
- **Calls from and to VoIP**: The VoIP profile in `route.cfg` is causing problems with incoming or outgoing VoIP calls.
- **Calls from and to VoIP**: A technical prefix is causing the problem in the `route.cfg's MapAll` parameter.

Symptoms

- The call arrives on the gateway.
- The gateway refuses to route the call or it routes the call, but not to the right destination address.

Solution

- **Mobile / VoIP / ISDN / analog:** If you are using `Restricts` in your routes, make sure that every `Restrict` command has a corresponding `MapAll` command. Also ensure that the more specific `Restrict` parameters are placed below the more general ones, because they are searched bottom up. Please refer to Chapter 5.3.1.1 Restrict to look up an explanation of the `Restrict` command.
  When configuring routes, make sure to place the more specific `MapAll` commands above the more general ones since mappings are searched from top to bottom and the first match is taken. Also ensure that no routes are missing. If you encounter problems routing VoIP calls, ascertain that the VoIP profiles are included in the mappings, since VoIP calls are distinguished by different profiles, not by different trunk groups. Please

go to Chapter 5.3.1.2 MapAll to learn about the correct configuration of the `MapAll` command. For information on VoIP profiles, please read Chapter 5.3.2 VoIP profiles.

- **Mobile / VoIP / ISDN / analog**: A license issue can be the reason why a call is rejected by the gateway. Access your system using GateManager and check which licenses are available under General, and - in the case of trial licenses - what day they expire. If licenses are missing, they are not listed in the Licenses field. The date in License Term indicates if trial licenses have already expired. If the Licenses entry is empty, this means that you are either using the wrong software or the wrong license key. In addition, the `license.key` file tells you about possible LAIN restrictions. Licenses can be restricted by LAIN, so that gateways can only be used in certain countries and / or with certain mobile operators. Contact your service partner for help.
- **Mobile**: In the configuration for mobile ports, missing `CHADDR` parameters often cause problems with outgoing calls. If you are routing by LAIN, check that `CHADDR` exists in your `Subscriber` settings. Also check that the corresponding mappings include the correct LAIN instead of the address port (trunk group). Access your system using GateManager and check which trunk group or LAIN is displayed in the Address column under port status. Modify the `pabx.cfg` or `route.cfg` files, if needed. For a description of the `Subscriber` configuration please read Chapter 5.2.1.5 Subscribers.

Example 12.6 LAIN configuration in the `pabx.cfg`

```
...
Controller06=20 GSM
...
Subscriber06=TRANSPARENT ROUTER GSM[0000,00000,+000000,1,1,1,SIM4,IMSI,BAND(6)]
CHADDR ALARM NODE[0006]
...
```

Example 12.7 Corresponding mapping by LAIN in the `route.cfg`

```
MapAll0172=262070172
```

VoIP: Missing or incorrect VoIP profiles cause problems with incoming or outgoing VoIP calls. The gateway receives the call but refuses to route it. For an explanation of the configuration of VoIP profiles please read Chapter 5.3.2 VoIP profiles.

- In the case of an incoming call, the system searches the `route.cfg file` for a VoIP profile which matches the incoming VoIP call. That VoIP profile must be defined for incoming, or incoming and outgoing calls (`VoipDirection=In` or `VoipDirection=IO`). Check that the peer address and the VoIP mask match the IP range from which you expect VoIP calls. Verify that the codecs which have been entered in the `VoipCompression` parameter match the peer's codec. To look up the codec offered by the peer, you need to run a layer 2 / layer 3 trace.
- In the case of an outgoing call, set `VoipDirection=Out` or `VoipDirection=IO`. The `VoipPeerAddress` parameter must contain the IP address to which you send VoIP traffic. The `VoipIpMask` parameter is not needed. Again, the `VoipCompression` parameter needs to match the peer's codecs.
- In all cases, ensure to put the profile name in square brackets.
- In the following example, the gateway receives a call from the IP 195.4.13.14. There is only one VoIP profile for incoming VoIP calls in the gateway's `route.cfg` file. The VoIP profile's `VoipPeerAddress` parameter is 195.4.13.0. The `VoipIPMask` parameter contains the value `0xfffffff8` which translates into the subnet mask 255.255.255.248. This means that the IP addresses 195.4.13.0 to 195.4.13.7 can send calls. The IP address 195.4.13.14 which actually sent the call is not inside this range. The call is thus rejected by the gateway.

Example 12.8 Calls from 195.4.13.14 rejected

```
[Voip:GW1]
VoipDirection=In
VoipPeerAddress=195.4.13.0
VoipIpMask=0xfffffff8
VoipCompression=g711u g711a
VoipSilenceSuppression=Yes
VoipSignalling=1
VoipFaxTransport=1
VoipMaxChan=120
VoipTxM=2
VoipMediaWaitForConnect=Tone
VoipProgress=2
```

- To allow for calls coming from 195.4.13.14 the `VoipIpMask` parameter needs to be changed, as shown in the following example. The IP addresses 195.4.13.0 to 195.4.13.15 can now send VoIP traffic.

Example 12.9 Calls from 195.4.13.14 accepted

```
[Voip:GW2]
VoipDirection=In
VoipPeerAddress=195.4.13.0
VoipIpMask=0xfffffff0
VoipCompression=g711u g711a
VoipSilenceSuppression=Yes
VoipSignalling=1
VoipFaxTransport=1
VoipMaxChan=120
VoipTxM=2
VoipMediaWaitForConnect=Tone
VoipProgress=2
```

- **VoIP**: Some VoIP carriers require that called numbers include a technical prefix for authentication. If a call comes from a VoIP carrier who requires a prefix and is sent elsewhere, the called number (DAD) arrives on the gateway including the technical prefix. That prefix needs to be removed in the `MapAll` parameter. If a call comes in without a prefix and is routed to VoIP where it requires a prefix, the prefix needs to be added to the `MapAll` parameter. In the following example, all international calls are sent to VoIP and prefixed with 0815#.

Example 12.10 Outgoing VoIP call with technical prefix

```
MapAll00=40VoIP:0815#00
```

12.2.3 Call is rejected elsewhere

When a call is received and routed by the gateway but rejected elsewhere, this behavior can be caused by your routing settings.

Reasons

- **Mobile**: Calling line identity restriction (CLIR) has been set in the call routing but is either not allowed at the call destination or has not been activated at the mobile operator.

Symptoms

- The gateway receives a call.
- The gateway routes the call to the right destination address.
- The call is rejected.
- The system's `failed.log` file most likely contains the cause values b2 (requested facillity not subscribed) or 95 (call rejected).

Solution

Mobile: The CLIR configuration needs to be removed from the routing settings if CLIR is not allowed, or it needs to be activated by the mobile operator where it is allowed.

- To disable CLIR in your routing settings and thus transmit the calling number, you need to adjust your routing entries in the `route.cfg` file. Remove the `#` sign from all `MapAll` entries where it is used.
- In cases where CLIR is allowed, you need to request activation from the mobile operator.

12.3 Software update problems

In the following scenario, a software update failed.

Reasons

- The file transmission was interrupted.
- The system memory is full.
- The wrong file(s) has / have been transferred.
- Operating system and system files are not from the same version package, such as 16.1.
- The files have been loaded into the wrong directory (only when using FTP for transfer).

Symptoms

- The system cannot be accessed via GateManager.
- The system can be accessed via GateManager, but the system type that is displayed in the GateManager's General Page is wrong (i.e. **)**.
- The file size of the updated files does not match the size of the original files.
- The system restarts approximately every 2 minutes.

Solution

- If you cannot reach the system at all via FTP or GateManager, this is because the operating system is no longer accessible. You need to contact your service partner for help.
- If you have access to the system, the operating system starts but the `*.tz1` file is defective. Try to solve the problem by taking the following actions:
  - Connect with your via FTP.
  - Change to the boot directory.
  - Delete the defective `*.tz1` file..
  - Upload the `*.tz1` file again.
  - Restart the system:
    - by resetting the power or
    - by going to your FTP client's command line interpreter and selecting restart. This option depends on your FTP client's functionality. For a description of the FTP commands, please refer to Table 4.8, "FTP commands" on page 34.
  - Check whether the software upload was successful.

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