TELES.MANUALS

TELES.VoIPBOX GSM and TELES.VoIPBOX CDMA

Software version 13.0

TELES
Communication Systems

TABLE OF CONTENTS

1 ABOUT THIS MANUAL

This manual is set up to guide you through the step-by-step installation of your TELES.VoIPBOX GSM or TELES.VoIPBOX CDMA, so that you can follow it through from the front to the back. Unless otherwise specified, the TELES.VoIPBOX GSM and TELES.VoIPBOX CDMA will be referred to collectively as TELES.VoIPBOX GSM/CDMA throughout this manual. Quick-installation instructions appear in Chapter 4.8, "Startup with TELES.Quickstart". Make sure you familiarize yourself thoroughly with the safety and security precautions detailed in Chapter 2 ⇨ before you begin to install your TELES.VoIPBOX GSM/CDMA. 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 TELES.VoIPBOX GSM/CDMA 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 TELES.VoIPBOX GSM/CDMA and its implementation scenarios.
- **Chapter 4, "TELES.VoIPBOX GSM/CDMA Installation"** contains information on how to connect and configure the system so that it is ready for operation.
- **Chapter 5, "Configuration Files"** describes the TELES.VoIPBOX GSM/CDMA's individual configuration files and parameters.
- **Chapter 6, "Routing Examples"** contains useful examples and descriptions of scenario-based configurations in the `route.cfg`.
- **Chapter 7, "System Maintenance and Software Update"** describes system messages that are saved in the protocol file, as well as trace options.
- **Chapter 8, "Signaling and Routing Features"** describes configuration settings in the `route.cfg` used for adjusting signaling and customizing the configuration for specific scenarios.
- **Chapter 9, "Mobile Configuration Options"** describes mobile configuration entries.
- **Chapter 10, "Least Cost Routing"** describes configuration options for various routing processes.
- **Chapter 11, "Online Traffic Monitor"** contains the configuration for monitoring the system's statistics and CDRs.
- **Chapter 12, "DLA/Callback Services"** contains money-saving features that expand the functionality of your TELES.VoIPBOX to include callback capability and DTMF services.
- **Chapter 13, "Feature Packages"** contains a description of options that expand the TELES.VoIPBOX GSM/CDMA's functionality.
- **Chapter 14, "Additional VoIP Parameters"** contains additional configuration entries to fine-tune communication with the VoIP peer.
- **Chapter 15, "Optional Function Modules"** contains information on expansion modules.

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 2 ⇨ – 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

Please be sure and take time to read this section to ensure your personal safety and proper operation of your TELES.VoIPBOX GSM/CDMA.

To avoid personal injury or damage to the system, please follow all safety instructions before you begin working on your TELES.VoIPBOX GSM/CDMA.

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

2.1 SAFETY MEASURES

Danger of electric shock - the power supplies run on 230 V. Do not open the TELES.VoIPBOX GSM/CDMA or its power supply.

Make sure to install the TELES.VoIPBOX GSM/CDMA near the power source and that the power source is easily accessible.

Bear in mind that telephone and WAN lines are also energized and can cause electric shocks.

Be sure to respect country-specific regulations, standards or guidelines for accident prevention.

If you do not use the Ethernet cable included in the package contents, make sure you use a shielded Ethernet cable.

2.2 FCC / INDUSTRY CANADA NOTICE

The following information applies for the TELES.VoIPBOX GSM only.

In accordance with the manufacturer's specifications, the TELES.VoIPBOX GSM comes installed with modular transmitters Q24CL001 (FCC ID: O9EQ24CL001)and Q24PL001 (FCC ID: O9EQ24PL001).

The antenna gain, including cable loss, must not exceed 3 dBi at 1900 MHz / 1.4 dBi at 850 MHz for mobile operating configurations and 7 dBi at 1900 MHz / 1.4 dBi at 850 MHz for fixed mounted operations, as defined in 2.1091 and 1.1307 of the rules for satisfying RF exposure compliance.

The antenna(s) used for this transmitter must be installed to provide a separation distance of at least 20 cm from all persons and must not be collocated or operating in conjunction with any other antenna or transmitter.

The TELES.VoIPBOX GSM 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 POWER SUPPLY

The included power supply is to be used exclusively for operation of your TELES.VoIPBOX GSM/CDMA.

Make sure you read this chapter thoroughly and save the instructions for future reference. Use only the power supply GSP-1216TLS/1 included in the package contents of your TELES.VoIPBOX GSM/CDMA.

2.3.1 TECHNICAL DATA

The following list includes technical information on the power supply:

- Type: GSP-1216TLS/1 for TELES.VoIPBOX GSM/CDMA
- Input voltage: 230V~ +/-15% 50-60Hz; 0.40A
- Output voltage: 12V ⎓; 1.6A
- Weight: 96g
- Tested and certified as per EN60950-1

2.3.2 SYMBOLS

The symbols on the power supply have the following meanings:

Table 2.1 Power Supply Symbols

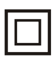
Symbol	Meaning
	Certified to conform with European norms.
	Protective insulation provided.
	For indoor use only.

Table 2.1 Power Supply Symbols

Symbol	Meaning
	Not for public disposal. Make sure you dispose of the power supply properly.
O - + + - O	Indicates the output polarity of the power supply.

2.3.3 INSTRUCTIONS FOR USE

Use only the power supply GSP-1216TLS/1 included in the package contents of your TELES.VoIPBOX GSM/CDMA.

Plug the power supply directly into the outlet. The power supply provides safety-low voltage with limited capacity for your TELES.VoIPBOX GSM/CDMA.

The devices are designed for constant use in dry, indoor locations. However, we recommend that you unplug them if you do not intend to use them for an extended amount of time. Make sure the power outlet is easily accessible at all time.

2.3.4 SAFETY PRECAUTIONS

Make sure you follow these safety precautions:

- Electrical devices may not be used by individuals who are not aware of the dangers of electricity and/or incorrect use thereof.
- Make sure you use only the correct input voltage.
- Make sure the installation site is sufficiently ventilated.
- Use the device only in dry, indoor locations, and protect it from humidity.
- Do not subject the device to direct sunlight.
- Unplug the device if you do not intend to use it for an extended amount of time.
- Hold the device by its housing when you unplug it. Wall outlets can become mechanically overloaded; do not pull on the cord.
- The room temperature may not exceed 35°C.
- Do not use the device if it is damaged or if there are signs of misfunction. In this case, send it to TELES Service or dispose of it properly (not with the public trash).

2.4 JACKS

The jacks on the TELES.VoIPBOX GSM/CDMA have fulfilled the requirements of the following safety standards.

- ETH jacks: SELV
- FXS jacks: TNV3

2.5 TIPS FOR EMC PROTECTION

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

2.6 SYSTEM SECURITY

This section describes all points crucial to the TELES.VoIPBOX GSM/CDMA's system security.

The TELES.VoIPBOX GSM/CDMA's location must support normal operation 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 in a clean, dry, dust-free location. If possible, the site should be air-conditioned. 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 0 and 35°C. 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: TELES.VoIPBOX GSM 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.

Servicing the TELES.VoIPBOX GSM/CDMA

Regular servicing ensures that your TELES.VoIPBOX GSM/CDMA 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.

2.6.1 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!

Make sure your hard disk or flash disk contains enough storage space. Downloading the log files and deleting them from the TELES.VoIPBOX GSM/CDMA on a regular basis will ensure your TELES.VoIPBOX GSM/CDMA's reliability.

Be careful when deleting files that you do not delete any files necessary for system operation.

TELES.vGATE Control Unit:
Do not use Ctrl/Alt/Del (Task Manager) to shut down vGATEDesktop or vGATECtrl. Do not perform queries on the database. This can result in damages to the database. Do not use any MySQL tools, such as MySQL-Front to make changes in or perform tests on the database.

2.7 CDR FILES

Call Detail Records are intended for analysis of the TELES.VoIPBOX GSM/CDMA'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.

Inaccuracies in the generation of CDRs may occur for active connections if traffic is flowing on the system while modifications in configuration or routing files are activated.

2.8 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.VoIPBOX GSM/CDMA if you connect it to the Internet:

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

The firewall must support the following features:

- Protection against IP spoofing
- Logging of all attempts to access the TELES.VoIPBOX GSM/CDMA

The firewall must be able to check the following information and only allow trusted users to access the TELES.VoIPBOX GSM/CDMA:

- 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.VoIPBOX GSM/CDMA, open only the following ports only when the indicated services are used:

Table 2.2 Ports Used for Specific Services

Service	Protocol	Port
For TELES.VoIPBOX GSM/CDMA		
FTP	TCP	21 (default, can be set)
Telnet (for TELES debug access only)	TCP	23
SMTP	TCP	25
DNS Forward	UDP	53
HTTP	TCP	80 (default, can be set)
SNTP	UDP	123
SNMP	UDP	161
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)
TELES.GATE Manager	TCP	4445 (default, can be set)
SIP Signaling	UDP / TCP	5060 (default, can be set)
RTP/RTCP	UDP	29000-29015 (default, can be set)
For TELES.NMS		

Table 2.2 Ports Used for Specific Services *(continued)*

Service	Protocol	Port
FTP	TCP	21
Telnet	TCP	23
The following ports are used for communication between the TELES.NMS Desktop and TELES.NMS:		
MySQL database	TCP	3306
TELES.NMS protocol	TCP	5000
TELES.NMS update	TCP	5001
TELES.NMS task	TCP	5002
TELES.NMS task	TCP	5003

3 OVERVIEW

TELES.VoIPBOX GSM/CDMA is a combined mobile and VoIP gateway solution for carrier networks and for corporate customers wanting to connect their PBX to mobile and VoIP services. This full-featured gateway can be added to analog and IP environments in a cost-effective and convenient manner. TELES.VoIPBOX GSM/CDMA converts fixed-to-mobile into mobile-to-mobile calls, terminating calls to mobile networks at lower rates than possible via fixed-net interconnection.

3.1 WHAT'S NEW IN VERSION 13.0

This manual includes descriptions of the following new features implemented since Version 12.0:

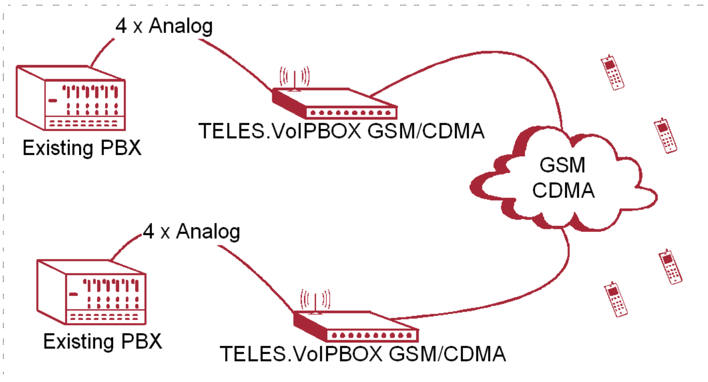
- New kernel and file system to improve system performance
- Expanded GUI (Graphical User Interface) functionality:
  - Unlimited configuration of VoIP, registrar and gatekeeper profiles
  - Full, flexible routing configuration now supported in addition to the quick routing settings
  - New statistic output
- Caller ID for FXS ports
- Optional automatic recognition of RTP peers' IP address
- Expanded error diagnostics especially for IP feature settings
- New VoIP functionality SIP:
  - Immediate switch to t.38 possible for interconnection with a fax server
  - Renegotiation of codecs in a fax if peer does not support t.38.
  - Refer method (RFC 3515)
  - VoipP-Preferred-Identity and P-Asserted-Identity (RFC 3325)
  - Additional possibilities for address type manipulations for OAD and DAD
  - DTMF relay with SIP INFO messages
  - Automatic and fixed fallback of DTMF tone transmission from RFC 2833 to SIP Info or inband
  - Possible to set SIP transaction/dialog matching to occur strictly as per RFC 3261
  - Radius support
- New VoIP functionality H323
  - Gatekeeper Registration: Terminal alias contains RasID plus prefixes
  - Radius support
  - Support of STUN for gatekeeper
  - DTMF relay with H.323 Info

3.2 IMPLEMENTATION SCENARIOS

These are the most commonly used implementation scenarios:

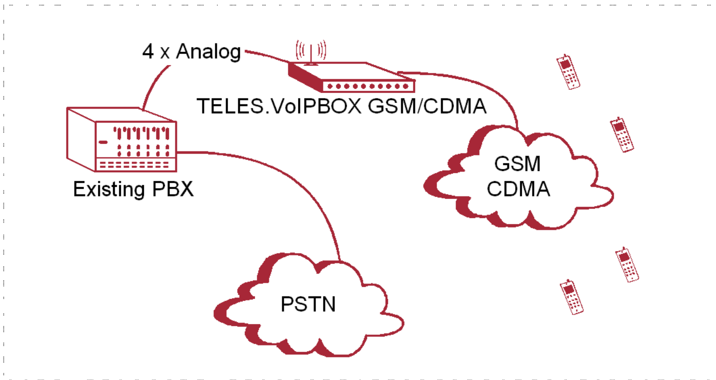
Mobile NT

The TELES.VoIPBOX GSM/CDMAs are connected to the customer's PBX with up to four analog lines, and to the mobile carrier's network via GSM or CDMA. The mobile gateway can multiplex the available mobile channels, as well as directly connect analog subscribers.



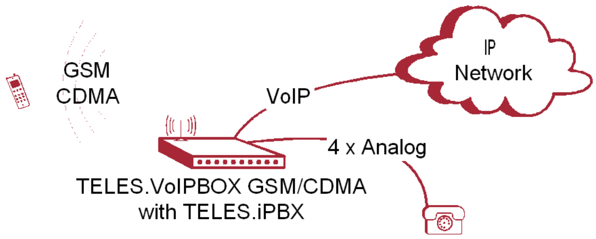
Corporate Scenario

The TELES.VoIPBOX GSM/CDMA only receives mobile calls from the customer's PBX. Calls are terminated directly in the mobile network. The sophisticated routing algorithms allow you to route calls to specific carriers if SIM cards from various carriers are used.



Mobile Gateway with VoIP

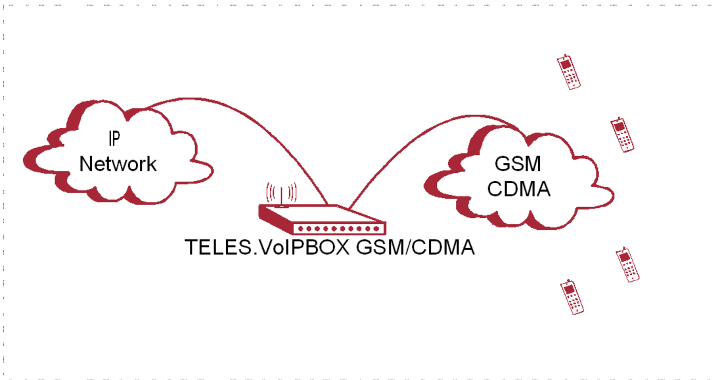
The TELES.VoIPBOX GSM/CDMAs are set up in small or medium-sized enterprises. The mobile gateway can multiplex the available mobile channels for an attached PBX and/or a Soft PBX. It can also connect analog subscribers directly. The sophisticated routing algorithms allow additional VoIP communication via SIP and/or H.323. Various voice codecs ensure universal connection to different VoIP destinations. Fax transmission occurs via T.38.



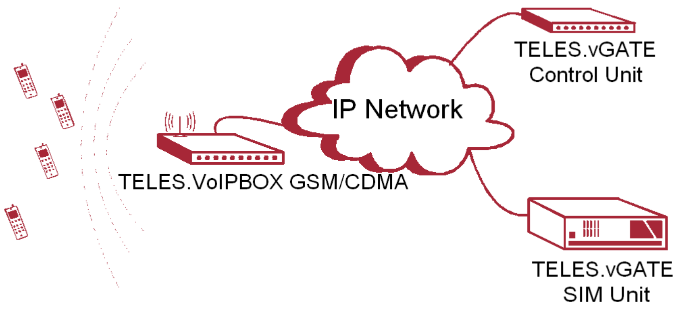
The TELES.VoIPBOX GSM/CDMA recognizes calls to the mobile network and sends them through the mobile gateway to the mobile network. All other calls are terminated via the VoIP carrier.

Carrier Scenario

One or more mobile gateways are connected to the carrier network via VoIP.The carrier network routes mobile connections to the individual mobile gateways, which then terminate the mobile calls.



Connection to a centralized SIM server (TELES.vGATE**):** The mobile gateways are integrated in the TELES.vGATE through the IP network. All SIM cards in the TELES.vGATE network are installed in and maintained from a central server, so that it is no longer necessary to install SIM cards into each TELES.VoIPBOX GSM/CDMA. The vGATEDesktop makes it possible to assign SIMs virtually to random ports and various times without physically removing the SIMs from the TELES.vGATE Sim Unit.



3.3 FEATURES

Mobile

- 4 GSM or CDMA channels
- 1 SIM card or R-UIM per channel
- 1 antenna
- Built-in SIM-card server support for unlimited SIMs per channel with TELES.vGATE Sim Unit
- Individual timers for each SIM /call

VoIP

- 8 media channels
- H.323 v.4 / SIP v.2 signaling (RFC 3261), operating in parallel
- Various audio codecs: G.711, G.723.1, G.726, G.728, G.729, GSM, iLBC, Fax T.38, Data: clear channel
- RTP multiplexing (reduces bandwidth required for RTP data by up to 60%)
- ENUM client
- Echo cancellation G.168–2000
- Silence suppression, comfort noise generation, voice activity detection
- Support for multiple gatekeepers and multiple registrars
- STUN client
- Traffic shaping

Analog

- 4 analog lines (FXS)
- Fax/modem detection (UDT)
- Charging impulse (12/16kHz)
- Integrated line echo cancellation

FXS

- Power feeding for FXO devices
- Generates dial tone and ring tone

LCR Engine

- Multiple VoIP-carrier logins
- Multilevel alternative routing
- Dynamic fallback to VoIP

General

- User-friendly HTTP user interface with easy and advanced mode configuration settings
- Ringtone generation
- Configurable ToS/DivServ
- Integrated DSL router (PPPoE)
- 2nd separate 10/100 Base-T Ethernet interface
- Status indication via LEDs
- Number portability

4 TELES.VOIPBOX GSM/CDMA INSTALLATION

This section contains information on basic installation and configuration of your TELES.VoIPBOX GSM/CDMA. Follow the easy instructions to set up your TELES.VoIPBOX GSM/CDMA in a matter of minutes.

Implementation of individual scenarios require 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 analog lines to the PBX
4. Check functionality (using the LEDs)
5. Using TELES.Quickstart, set the configuration (IP address and VoIP configuration)
6. Secure the LAN connection
7. Secure connection with the configuration program

4.2 PACKAGE CONTENTS

Your TELES.VoIPBOX GSM/CDMA 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 TELES.VoIPBOX GSM/CDMA
- 1 power supply
- 1 RJ-45 LAN cable with gray connectors

4.3 TELES.VOIPBOX GSM/CDMA HARDWARE DESCRIPTION

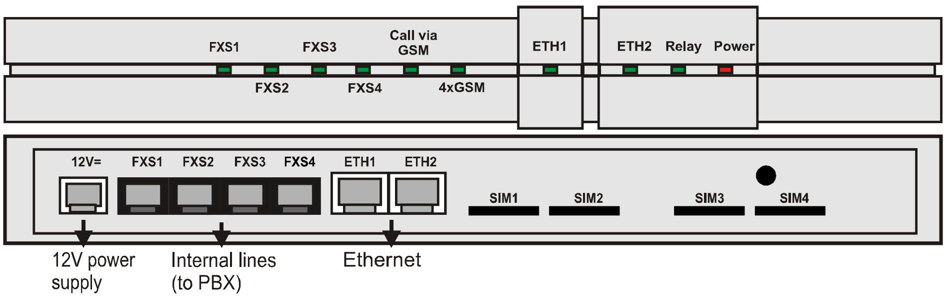


Figure 4.1 TELES.VoIPBOX GSM/CDMA: Front and Rear View

The TELES.VoIPBOX GSM/CDMA handles traffic on up to 8 media channels. The following pages describe installation of the TELES.VoIPBOX GSM/CDMA.

Figure 4.1 shows the front and rear view of a TELES.VoIPBOX GSM/CDMA.

4.4 INSTALLATION REQUIREMENTS

Before installing your TELES.VoIPBOX GSM/CDMA, make sure you have the following connections in place:

- Ethernet connection

> **if you do not use the included cable, make sure you use only a shielded Ethernet cable!**

- Analog connection to PBX
- Power

4.4.1 ANALOG WIRING

The FXS ports connect to the PBX. You can connect the TELES.VoIPBOX GSM/CDMA to a second analog outlet for the second analog interface.

Figure 4.2 shows the standard pin assignment for each FXS analog port.

Figure 4.2 Analog Wiring Scheme

4.4.2 ETHERNET WIRING

To connect the TELES.VoIPBOX GSM/CDMA's Ethernet port to your local network, connect the system to an Ethernet switch or hub in your network. Use the three meter cable with gray connectors.

If you want to connect the TELES.VoIPBOX GSM/CDMA directly to your computer and a connection cannot be established after you plug the cable in, use a cable with the following pin assignment:

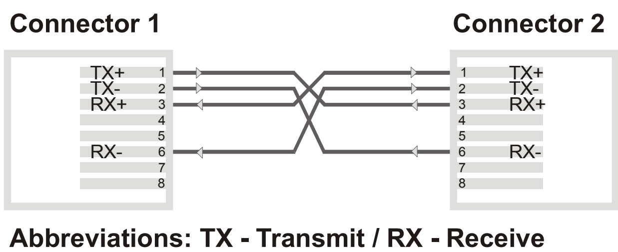


Figure 4.3 Ethernet Wiring Scheme

4.5 PREPARING FOR INSTALLATION

Each computer that is to communicate with the TELES.VoIPBOX GSM/CDMA requires a network connection. Please have the following information for connection to your network available:

- IP address in your local network for the TELES.VoIPBOX GSM/CDMA to be configured
- Netmask for the TELES.VoIPBOX GSM/CDMA to be configured
- Default gateway for TELES.VoIPBOX GSM/CDMA to be configured
- DNS server address
- NTP server address

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

4.6 HARDWARE CONNECTION

Connect your computer with the local network

- Connect the TELES.VoIPBOX GSM/CDMA with the local network
- Connect the TELES.VoIPBOX GSM/CDMA with your PBX according to the port configuration.
- Connect the TELES.VoIPBOX GSM/CDMA with the power supply.

4.7 LED FUNCTIONALITY

Each TELES.VoIPBOX GSM/CDMA has the following status LEDs:

Table 4.3 TELES.VoIPBOX GSM/CDMA LEDs

LED	Description
Green 1-4	Off: Telephone on the hook. On: Telephone off the hook. Blinking: Call active.

Table 4.3 TELES.VoIPBOX GSM/CDMA LEDs *(continued)*

LED	Description
Green 5	On: At least one SIM is active. Blinking: Call via GSM.
Green 6	On: All four SIMs are active.
Green 7-8	Off: Link is down. On: Link is active, no traffic. Blinking: Link is active with traffic.
Green 9	Off: Relay inactive. On: Relay active.
Red	Off: Power off. On: Power on.

4.8 STARTUP WITH TELES.QUICKSTART

TELES.Quickstart is an application that helps you to configure the basic settings of your TELES.VoIPBOX GSM/CDMA quickly and conveniently.

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

- Windows 98 SE
- Windows NT
- Windows ME
- Windows 2000
- Windows XP

If you are using any of these operating systems, please follow the instructions in this chapter. If you are using a non-Windows operating system (e.g. Linux) follow the instructions in Chapter 4.9 ⇨.

4.8.1 INSTALLING TELES.QUICKSTART

Make sure the TELES.GATE Manager is not running on your computer. To install TELES.Quickstart on your computer, insert the CD and select TELES.Quickstart from the menu. Follow the Windows instructions to begin installation of the TELES.Quickstart. Once installation begins, click **Next** to install TELES.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**.

The next dialog asks you where you want to install the program's icons. To install them in the folder that appears, click **Next**. If you want to install them in another location, select a folder from the list or enter a new folder name. Then click **Next**.

To start TELES.Quickstart immediately following installation, activate the checkbox **I would like to launch** TELES.Quickstart. Make sure the checkbox is inactive if you do not want to start the program now. Click **Finish**.

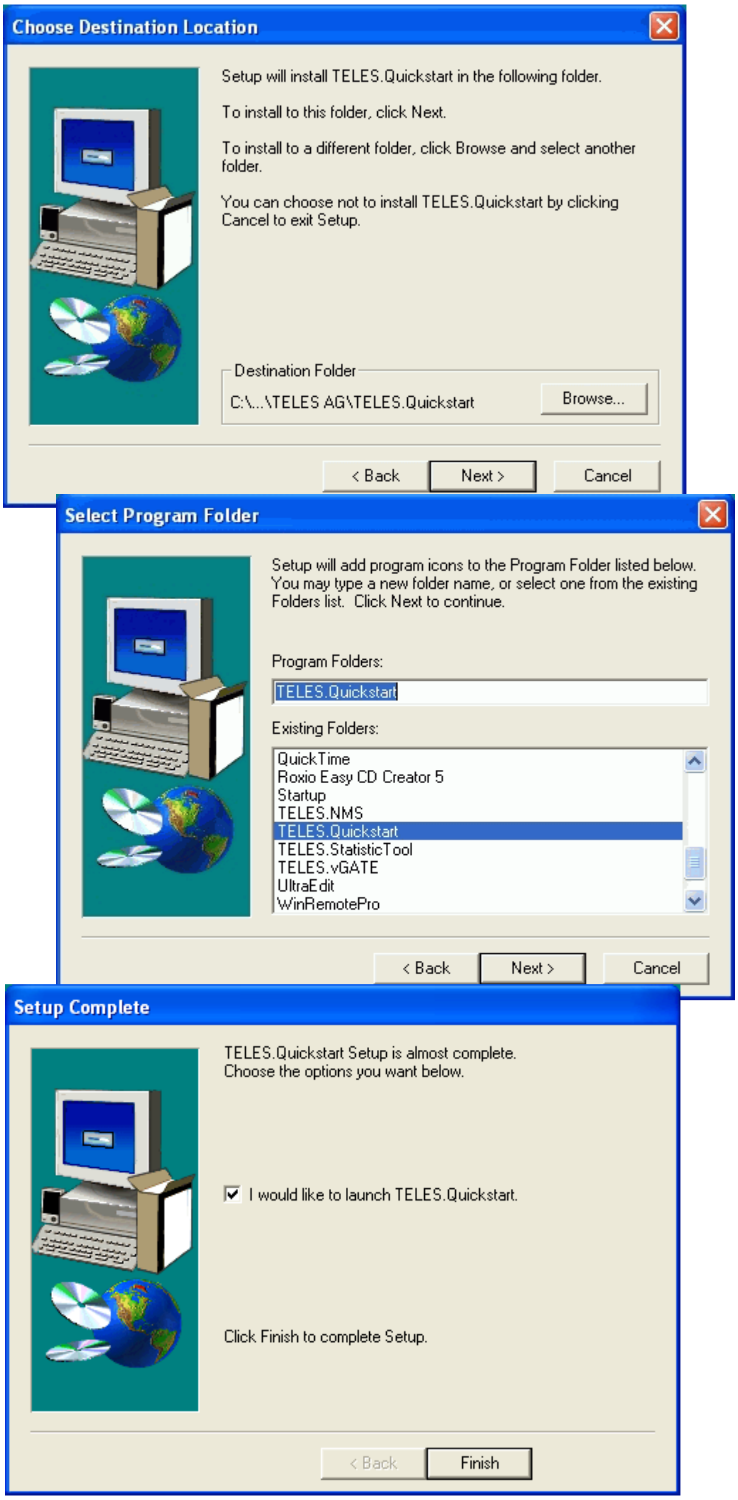


Figure 4.4 TELES.Quickstart Installation

4.8.2 CONFIGURATION WITH TELES.QUICKSTART

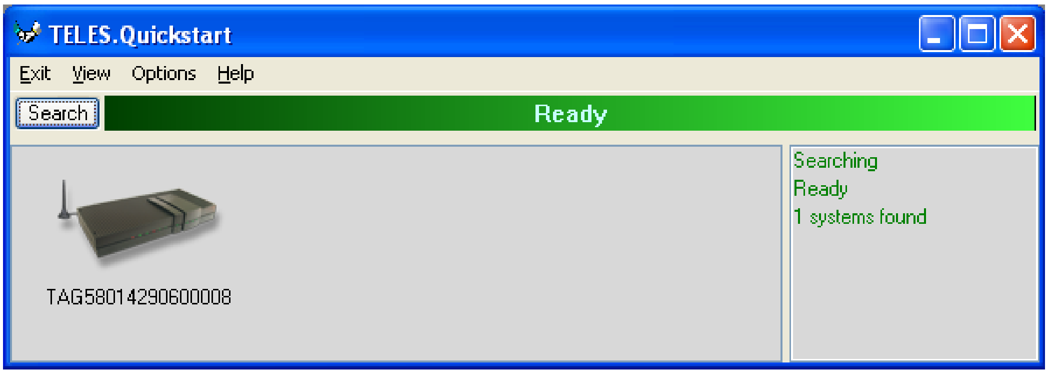


Figure 4.5 TELES.Quickstart

Now you can use TELES.Quickstart to set up your TELES.VoIPBOX GSM/CDMA. Open TELES.Quickstart`.exe`. The program will automatically search for your TELES.VoIPBOX GSM/CDMA in the local network. For TELES.Quickstart, the source UDP port is 57445. It might be necessary to change the firewall rules on your system.

Click the **Search** button to restart the search. When the program has found your TELES.VoIPBOX GSM/CDMA, it will appear in the main window. As soon as it appears, you can end the search by clicking **Stop**. The window on the right provides a running tally of the system's status.

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.

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

Table 4.4 TELES.Quickstart Details View Columns

Heading	Definition
Identifier	This column lists the system's serial number.
IP Address	This column lists the system's IP address.
Configured	An X means the system contains the configuration files.
# of VoIP Ctrls	This column lists the number of TELES.G729 Modules installed in the system. It will always be 1.
VoIP Channels	This column shows the number of VoIP channels per TELES.G729 Module.
Type	Lists the type of system.
Box	An X means the system is a TELES box-based system.
CF Mounted	This column is not relevant for TELES box-based systems.

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 are using a DHCP server, activate the checkbox **DHCP**. This will deactivate the next four 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** and the **Time Server** in the corresponding dialog boxes. Select the **Time Zone** for the location of the system. Click **Next**.

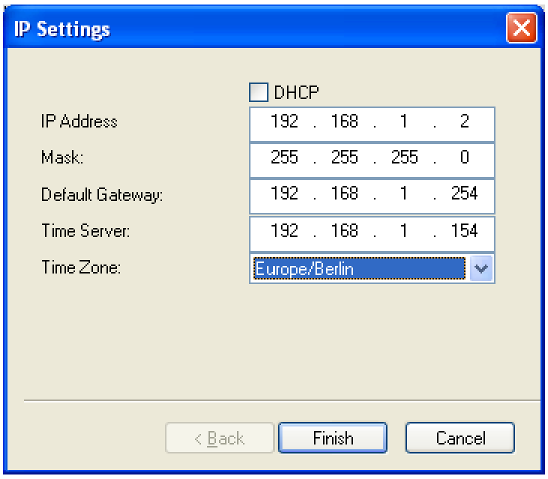


Figure 4.6 TELES.Quickstart Configuration: IP Settings

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.

Now the system is 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. When the system can be reached at the new IP address, all PSTN ports and routing entries will be set by sending the created configuration files to the system.

If you right-click the system's icon in the main window, you can also 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.

4.9 STARTUP VIA FTP

If you are using a computer that does not use a Windows operating system, you can preconfigure the system via FTP. The system's default IP address is 192.168.1.2. To configure the system using FTP, you must assign your computer an IP address from network range 192.168.1.0 Class C and then access the system via FTP.

The default user is `teles` and the default password is `tcs-ag`. To configure the system, use the default configuration file example on the CD in the `Configfiles` directory and the following subdirectories:

- `IPconfig`
  This subdirectory contains the file (`ip.cfg`) responsible for configuration of the Ethernet interface
- `portconfig`
  This subdirectory contains the file (`pabx.cfg`) responsible for configuration of the analog interface

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

4.10 SELF PROVISIONING WITH TELES.NMS

With a management connection to the TELES.NMS (Network Management System), the TELES.VoIPBOX GSM/CDMA can retrieve its configuration files from the configured TELES.NMS. That means that custom configuration of the device occurs automatically when the device is started. The following setting must be made in the `[System]` section of the `pabx.cfg`:

`AlarmCallback=<ip address` NMS server>

RemoteCallback=<ip address NMS server> <time> <days of week + holiday>

As soon as the device is started, it connects automatically with the TELES.NMS, which uses the device's TAG number to send a prepared configuration. For further information on configuration of the TELES.NMS, please refer to the TELES.NMS Systems Manual.

4.11 REMOTE ACCESS AND ACCESS SECURITY

After the system has been configured and all cables are connected, remote administration and maintenance can occur with the TELES.GATE Manager (Chapter 4.11.1 ⇨), the HTTP user interface (Chapter 4.11.2 ⇨), or via FTP (Chapter 4.11.3 ⇨).

4.11.1 TELES.GATE MANAGER

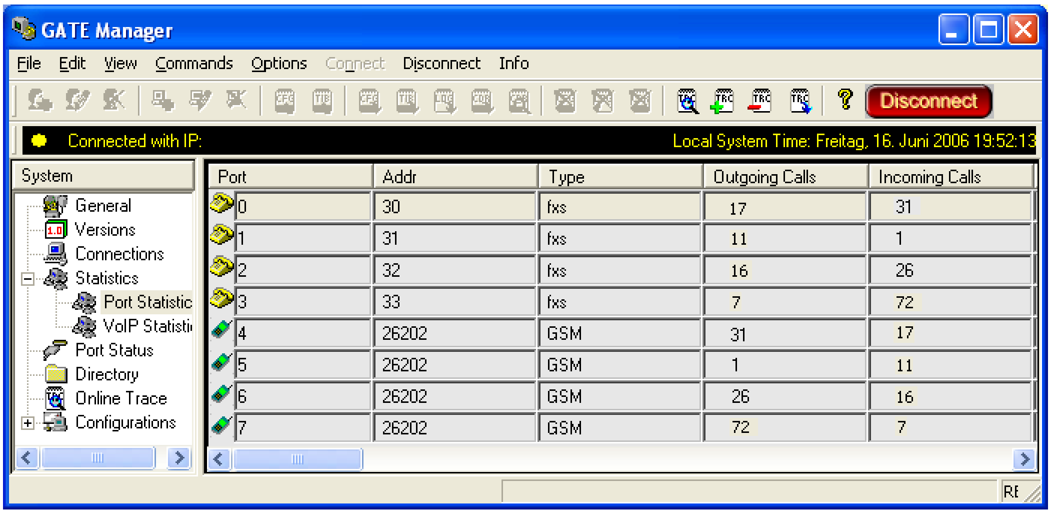


Figure 4.7 TELES.GATE Manager

The TELES.GATE Manager administration and maintenance software offers a broad range of functions. The TELES.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, e.g. to display and break down trace data.
- Update the system software (firmware) 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 TELES.GATE Manager. For a detailed description of installation and implementation of the TELES.GATE Manager, please refer to the TELES.GATE Manager and Utilities Programs Manual.

TELES.GATE Manager remote access can occur via IP. TELES.GATE Manager access via IP uses port 4444 as origination TCP port and port 4445 as destination port. The following default value (4445) is configured in the `pabx.cfg` file for the system's port:

```
MoipPort=4445
```

4.11.2 HTTP USER INTERFACE

Figure 4.8 HTTP User Interface

Remote access can occur via the HTTP user interface. 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.

The following administrative levels apply:

Carrier Mode (Full Access)

User: `teles-carrier`

Password: `tcs-carrier`

All configuration pages can be accessed in this mode.

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>
```

Example:

```
[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.

The user interface is divided into the following main sections:

Table 4.5 HTTP User Interface: Sections

Section	Description
User Data	Here you can change the user passwords and the language for the HTTP interface.
System Settings	IP Settings: Settings for the Ethernet interfaces and related services. Port Settings: Settings for the TELES.VoIPBOX GSM/CDMAs ports. VoIP Settings: VoIP settings for the SIP or H.323 carrier. Telephony Routing:Routings for telephone numbers. POTS Settings:Settings for the TELES.VoIPBOX GSM/CDMA's analog interfaces.
System Overview	Overview of system information and drivers.
Telephony Routing	VoIP settings for the SIP or H.323 carrier and routings for telephone numbers.
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.

4.11.3 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 ensure the security of your FTP access:

Table 4.6 FTP Security Entries

FTP Security
FtpdPort=<port> Defines the FTP access port (default 21).
RemotePassword=<password> Defines the password for FTP and TELES.GATE Manager access. Please refer to Chapter 4.11.4 ⇨ 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 TELES.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, also change to the directory `/boot`.

The following commands can be carried out via FTP access:

Table 4.7 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.

4.11.4 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:

```
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 TELES.GATE Manager occurs without a password, and FTP access occurs with the default password `tcs-ag`.

5 CONFIGURATION FILES

This chapter describes the basic setup and the most commonly used entries for the configuration files. Configuration of TELES.VoIPBOX GSM/CDMAs is managed in the following three files:

Table 5.8 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 routing entries.

Changing configuration data 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 to such changes. Please thoroughly check any changes you or a third party have made to your configuration.

The system comes without the files. 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 system using TELES.Quickstart, TELES.GATE Manager or via FTP (user `teles`, password `tcs-ag`). If you use the HTTP user interface 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.

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 5.9 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.

Table 5.9 Required Configuration File Sections *(continued)*

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

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, e.g. 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 5.10 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 TELES.VoIPBOX GSM/CDMA 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.
`[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:

```
[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:

```
[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 following settings are possible for the sections `[emac0]` and `[emac1]`:

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

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

Example:

```
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 is occurs in bridge mode.

`IpAddress=up`

5.1.3 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:

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

5.1.4 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 5.11 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 connectionsxppp<0>Protocol used for 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 redirects packets from one port and IP address to another.	
<interface>	Defines the translated interface or protocol: emac1 The system's second Ethernet interface pppoe0 Protocol used for DSL connectionsppp<0>Protocol used for 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.

Table 5.11 NAT Configuration *(continued)*

`<protocol>`	Defines the protocol. `tcp` and `udp` are possible.

Example: 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`.

```
[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
```

5.1.5 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 the carrier network and terminate VoIP traffic directly.

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:

`[pppoe<x>]`
`PppoeIf=emac1`
`User=<user>`
`Pwd=<pwd>`
`AuthProto=<pap|chap>`
`Route=<ip_gw>` (optional)

Table 5.12 Settings in the `[pppoe<x>]` Section of the `ip.cfg`

[pppoe<x>]
PppoeIf=<interface> Enter the Ethernet interface used for the DSL connection (usually `emac1`).
User=<username> Enter the username used for DSL access.

Table 5.12 Settings in the `[pppoe<x>]` Section of the `ip.cfg` *(continued)*

`[pppoe<x>]`
Pwd=<password> Enter the password used for DSL access.
AuthProto=<protocol> Enter `chap` or `pap` for the protocol used for authentication.
Route=<ip-addr> (optional) Enter the target IP address range, e.g. 0.0.0.0 (default route). All packets that are not defined for the local network will be sent through this interface. In this case, the parameter `DefaultGW` in the `System` section (Chapter 5.1.1 ⇨) must remain empty. Only network ranges can be routed. The syntax in this case is `Route=<target range> -netmask <ip mask>`. If several different network ranges are used, you must enter the `Route` parameter for each range.

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.

Example: 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:

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

5.1.6 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.

WARNING: 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.

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

```
[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 5.13 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` permits access `block` denies 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.	
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 (e.g. 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 (e.g.: `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 (e.g.: proto tcp, proto udp, proto icmp).	
port eq <num> <num> defines the port as number (e.g.: port eq 4445).	

Table 5.13 Settings in the `[firewall]` Section of the `ip.cfg` *(continued)*

[firewall] fw=<mode> <direction> <list>
keep state Ensures that the firewall checks packets from the beginning to the end of a session. This is necessary, as the firewall does not know 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:

```
[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
```

5.1.7 BANDWIDTH CONTROL

In many implementation scenarios, the TELES.VoIPBOX GSM/CDMA in router mode (e.g. 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 5.24, in the following entries:

`H225Port`

`SipPort`

`VoipRtp Port`

`VoipRtpPortSpacing`

Different ports can be 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 5.14 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 apples (e.e. `pppoe0`).
`<bw>`	Sets the bandwidth that is available on the interface in Kbit/s (e.g. 256K).
priq	Priority qeueing. A higher priority class is always served first.
class priq <interface> <class> root priority <prio>	
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 (e.g. 15), the higher the priority.
filter <interface> <class> <values>	
Defines the individual rules.	
`<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

Example: In the following example, prioritization is set for an eight-channel VoIP connection. The SIP signaling port 5060 and the RTP/RTCP ports 29000 to 29015 are prioritized at level 7. All other services are set at level 0:

```
[altqd]
interface pppoe0 bandwidth 256K 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 29014 0 0 17
  filter pppoe0 realtime_class 0 0 0 29014 17
  filter pppoe0 realtime_class 0 29015 0 0 17
  filter pppoe0 realtime_class 0 0 0 29015 17
class priq pppoe0 regular_class root priority 0 default
```

5.1.8 DHCP SERVER SETTINGS

The DHCP (Dynamic Host Configuration Protocol) server provides a mechanism for allocation of IP addresses to client hosts. The section `[dhcpd]` 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 5.15 Settings in the `[dhcpd]` Section of the `ip.cfg`

; Global dhcp 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 option 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.

Table 5.15 Settings in the `[dhcpd]` Section of the `ip.cfg` *(continued)*

; Global dhcp parameters
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:

```
[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;
}
```

5.1.9 PPP CONFIGURATION FOR CDMA DIAL-UP

The point-to-point protocol is used for dial-up connection via a mobile CDMA connection. That means the system can set up an Internet connection, which can be used for all local users or to transmit VoIP calls via CDMA dial-up. Bear in mind that you must configure the firewall and NAT options accordingly.

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

The settings are entered as follows:

Table 5.16 Settings in the `[xppp]` Section of the `ip.cfg`

[xppp<num>]
Dad=<num> Enter the dial-up number.
User=<username> Enter a username.
Pwd=<password> Enter a password.
Route=<ip-addr> Enter the target IP address range, e.g. 0.0.0.0 (default route).
AuthProto=<protocol> Enter `chap` (default) or `pap` for the protocol used for authentication.
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: 120 for CDMA dial-up.
Rfc1662=<val> Framing to be use: 1 for CDMA.

Example:

5.1.10 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>
```

Example: 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:

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

5.1.11 EXAMPLES

5.1.11.1 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:

```
[System]
DefaultGw=192.168.1.254

[emac0]
IpAddress=192.168.1.1/24
```

5.1.11.2 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:

```
[System]
DefaultGw=192.168.1.254

[emac0]
IpAddress=192.168.1.1/24

[emac1]
IpAddress=up

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

5.1.11.3 INTEGRATED DSL-ROUTER SCENARIO FOR VOIP TRAFFIC WITH AN ACTIVE DHCP SERVER AND FIREWALL

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 carrier network without an additional router when the connection is used only for VoIP data. A DHCP server is used for dynamic IP-address allocation:

```
[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;
```

5.1.11.4 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:

```
[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` is divided into the `[System]` section and the optional `[Night<num>]`, `[Mail]` and `[Snmpd]` sections.

5.2.1 SYSTEM SETTINGS

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

- Hardware
- Log files
- Night configuration
- Controllers
- Subscribers
- Global Settings

5.2.1.1 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 5.17 pabx.cfg: Log File Entries

Entry	Description
`ActionLog=/boot/protocol.log`	System events

Table 5.17 pabx.cfg: Log File Entries *(continued)*

Entry	Description
`Log=/boot/cdr.log`	CDR entries
`RRufLog=/boot/failed.log`	Unconnected calls
`TraceLog=/boot/trace.log`	System trace

The available internal memory is approximately 8 MB. Make sure you monitor the available memory.

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

- 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 7 files to be generated.

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

Table 5.18 pabx.cfg: Log Parameters

Log=/boot/<file> <day> <size> <count>	
`<file>`	The name of the log file is generated as follows: `[file]yymmdd[0-9|A-Z].log`.
`<day>`	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 `<day>`, the file will be saved when the `<size>` has been reached. **NOTE: We recommend a file size of a multiple of 60kB.**
`<count>`	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, e.g. 5 files of 1MB each. If the fifth file is full, the first one will automatically be overwritten.

Bear in mind that file size will be unlimited if no parameters are defined.

Example 1 In the following entry, the file `cdr.log` is renamed every day. Up to 35 CDR files will be saved on the system.

```
Log=/boot/cdr.log daily - 35
```

Example 2 In the following entry, the file `failed.log` is renamed once a week. Up to 10 failed files will be saved on the system.

```
RrufLog=/boot/failed.log weekly - 10
```

Example 3 In the following entry, the file `protocol.log` is renamed when the file has reached 1MB. Up to five log files will be saved on the system.

```
ActionLog=/boot/protocol.log - 1000 5
```

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

5.2.1.2 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 5.19 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 daily at that time on the defined day.
`<day>`	Use a bitmask to set the weekdays on which the configuration applies here. The day-mask appears in the following order: `HoSaFrThWeTuMoSu`.

Example: The configuration section is activated Fridays, Wednesdays and Mondays at noon unless the day in question is a holiday:
`Night2=12:00 00101010`

The configuration section switches back to the default configuration (`System` section) every day at 8:00 p.m:
`NightResetTime=20:00 11111111`

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 5-55). The active route(s) (`MapAll`, `Restrict` and `Redirect` entries) and VoIP, registrar and gatekeeper profiles must appear in the `Night` section of the `route.cfg` (see Chapter 5.3 on page 5-55).

5.2.1.3 CONTROLLERS

This category defines the parameters that apply to the ports.

The individual ports are defined with the following parameters:

Table 5.20 pabx.cfg: Controller Parameters

Controller<port>=<bus> <type>	
`<port>`	Defines the running (physical) port number.
`<bus>`	Defines the configured (virtual) port number. In the default configuration, mobile ports are 20 and FXS ports are 10. VoIP ports are 40.
`<type>`	Defines the connection type: `ANA` internal (Foreign eXchange Subscriber) `CDMA` CDMA port `VOIP` VoIP module `DTMF` virtual controller for activating DTMF tone recognition

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

Example:

```
Controller00 = 30 ANA
Controller01 = 31 ANA
Controller02 = 32 ANA
Controller03 = 33 ANA
Controller04 = 20 GSM
Controller05 = 20 GSM
Controller06 = 20 GSM
Controller07 = 20 GSM
Controller08 = 40 VOIP
```

5.2.1.4 SUBSCRIBERS

Features for each port can be defined using this entry. Changes become active following a restart:

Table 5.21 pabx.cfg: Subscriber Parameters

Subscriber<port>=<list>	
`<port>`	Defines the running (physical) port number.
The `<list>` variable may contain one or more of the following keywords:	
`DEFAULT`	The standard configuration will be used.
`TRANSPARENT ROUTER`	Only the number is sent as caller ID (without the virtual port address).
`ALARM`	Activates the monitoring mode for the respective port. If a relevant error occurs at the port, a remote call is placed to the number defined in `RemoteCallBack`.
`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.
`CHMAX[x]`	Defines the number of VoIP channels (8) or DTMF channels. A maximum of two concurrent channels are possible for DTMF recognition if the callback platform is used.

Table 5.21 pabx.cfg: Subscriber Parameters *(continued)*

Subscriber<port>=<list>	
ANA[<country-code>,<charge-freq>,<cid-T5>,<cid-max-short>,<cid-min-long>,<cid-seizure-bits>,<cid-level>,<cid-mode>,<audio-law>]	Analog FXS ports only. <country-code> Impedance for individual analog controllers. 0 = TBR21 standard (default) 0001 = USA 0049 = Germany 0090 = Turkey <charge-freq> frequency in kHz of charge impulse: 16 (default) or 12 <cid-T5> After first ring: pause (in ms) before generation of caller ID Default: 800 <cid-max-short> Max duration of short pause (in ms) between first and second ring Default: 250 <cid-min-long> Min duration of long pause [ms] between first and second ring Default: 1500 <cid-seizure-bits> Length of caller ID signal: 80 (default) or 180 <cid-level> Caller ID signal level between 0 and 65535 0 = caller ID off (default) We recommend a value of 12800 <cid-mode> Caller ID standard 1 = V.23 (default) 2 = BELL 202 (usuall used in North America) <audio-law> 1 = a-law (default) 2 = μ-law default

Additional parameters for mobile controllers are described in Table 5.22 and Table 5.23. The parameters listed in

```
Subscriber00 = TRANSPARENT ROUTER ALARM
Subscriber01 = TRANSPARENT ROUTER ALARM
Subscriber02 = TRANSPARENT ROUTER ALARM
Subscriber03 = TRANSPARENT ROUTER ALARM
Subscriber04 = TRANSPARENT ROUTER ANA[0049,16,800,250,1500,80,12800,2,1]
Subscriber05 = TRANSPARENT ROUTER ANA[0049,16,800,250,1500,80,12800,2,1]
Subscriber06 = TRANSPARENT ROUTER ANA[0049,16,800,250,1500,80,12800,2,1]
Subscriber07 = TRANSPARENT ROUTER ANA[0049,16,800,250,1500,80,12800,2,1]
Subscriber08 = TRANSPARENT ROUTER SWITCH CHMAX[8] ALARM
```

Table 5.22 are required for mobile controllers and those listed in Table 5.23 are optional, depending on the implementation scenario.

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.

The following parameters are required:

Table 5.22 Required Parameters in `pabx.cfg`

Subscriber<port>=<type> [<pin>,<lain>,<SMSC>,<sim>,<loudGSM>,<loudPCM>,SIM<x>,...]	
`<port>`	Defines the running (physical) port number.
`<type>`	Defines whether a `GSM`, `CDMA` or `UMTS` module is used.
`<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. This prevents roaming into another mobile network. The LAIN is always five digits. 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. Only the default value 1 is possible. Activate configuration suffices to activate changes. **NOTE: Please see the example following Table 5.23 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.

Table 5.22 Required 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`	The number entered (4) refers to the number of slots read out to activate the mobile ports. **NOTE: This parameter cannot be used in combination with `SIMS`.**

Optional Mobile Parameters

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

Table 5.23 Optional Parameters in `pabx.cfg`

Optional Mobile Parameters
<IMSI> This keyword causes the IMSIs to be recorded in each CDR. This parameter appears after `SIM<x>`. Activate configuration suffices to activate changes.
SIMS Define this keyword to connect the system to a TELES.vGATE. `SIM1` must be defined in the appropriate mobile controller `Subscriber` line. **NOTE: This parameter cannot be used with the parameters `SIM4`. Bear in mind that no SIM-card carrier is to be inserted in the TELES.VoIPBOX GSM/CDMA.** EXAMPLE: `Subscriber00=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIMS]`

Example: The following example uses the German impedance value and 16kHz for the FXS ports. The mobile controllers have two SIM groups. Different SMS center numbers are set for mobile controllers 04-05 and 06-07. The SIM cards in the TELES.VoIPBOX GSM/CDMA are used. The routing process to mobile is based on the LAIN (**CHADDR**). VoIP is active:

```
Subscriber00 = TRANSPARENT ROUTER ANA[0049,16]
Subscriber01 = TRANSPARENT ROUTER ANA[0049,16]
Subscriber02 = TRANSPARENT ROUTER ANA[0049,16]
Subscriber03 = TRANSPARENT ROUTER ANA[0049,16]
Subscriber04 = TRANSPARENT ROUTER GSM[0000,00000,+495555555,1,1,1,SIM4] CHADDR ALARM NEXT
Subscriber05 = TRANSPARENT ROUTER GSM[0000,00000,+495555555,1,1,1,SIM4] CHADDR ALARM
Subscriber06 = TRANSPARENT ROUTER GSM[0000,00000,+496666666,1,1,1,SIM4] CHADDR ALARM NEXT
Subscriber07 = TRANSPARENT ROUTER GSM[0000,00000,+496666666,1,1,1,SIM4] CHADDR ALARM
Subscriber08 = TRANSPARENT ROUTER SWITCH CHMAX[8] ALARM
```

5.2.1.5 GLOBAL SETTINGS

This category contains the following system parameters:

Table 5.24 pabx.cfg: IP Configuration System Parameters

System Parameters
VoipGlobalMaxChan=<count> Max. number of channels for the entire system.
VoipRtpPort=<port> Defines the starting UDP port used to transmit RTP packets (default 29000).
VoipRtpPortSpacing=<count> Defines the space between the ports used for individual RTP streams (default 2).
H225Port=<port> Endpoint-to-endpoint port (default 1720).
SipPort=<port> SIP signaling port (default 5060).
VoipMaximumBandwidth=<int> Defines an upper limit for available bandwidth for the VoIP profiles to be configured (see `VoipBandwidthRestriction` in Table 14.106) if traffic shaping is active for the corresponding VoIP profile. Individual codecs are assigned the following values: 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 `no`.
StunServerAddress=<ip addr> When this parameter is active, the TELES.VoIPBOX GSM/CDMA 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.**

Table 5.24 pabx.cfg: IP Configuration System Parameters *(continued)*

System Parameters
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 TELES.VoIPBOX GSM/CDMA's time zone and time zone 0 (Greenwich Mean Time). Enter the continent and a large city (usually the capital) in the time zone.
NtpServer=<ip addr> Sets the IP address at which the `TELES.VoIPBOX GSM/CDMA`'s SNTP server queries the standard time. The query occurs every four hours.
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 TELES.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`.

Table 5.24 pabx.cfg: IP Configuration System Parameters *(continued)*

System Parameters
Tftpd=<mode> Activates (`on`) or deactivates (`off`) FTP access. Default `off`.
RemotePassword=<password> Defines the password for FTP and TELES.GATE Manager access. Please refer to Chapter 4.11.4 ⇨ 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 TELES.GATE Manager occurs without a password, and FTP access occurs with the default password `tcs-ag`.
SimCtrlUnitAddress=<ip addr> Enter the TELES.vGATE Control Unit's IP address. For a detailed description of TELES.VoIPBOX GSM/CDMA configuration for connection to a TELES.vGATE, see Chapter 9.1 ⇨.
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:

```
VoipGlobalMaxChan=8
H225Port=1720
SipPort=5060
VoipRtpPort=29000
VoipRtpPortSpacing=2
StunServerAddress=172.16.0.1
StunServerPollInterval=600
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.

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 TELES.VoIPBOX GSM/CDMA. The connection to the SMTP server can be used to send CDR files or alarm messages.

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

The following features are possible:

- Displaying incoming calls via e-mail
- Setting up connections using e-mail
- Sending announcements via e-mail
- Sending CDRs via e-mail
- Sending alarm messages via e-mail

SmtpServer=<ip addr> In `<ip addr>`, enter the IP address of the destination SMTP server that is to receive the e-mail messages.
MailUser=<username> Enter a username for e-mail authentication.
MailPwd=<password> Enter a username for e-mail authentication.
MailAuthEncr=<type> Enter an encryption method for e-mail authentication (default base64).
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=<domain> In `<domain>`, enter the source domain, the source address and an @ sign. If the source address is already complete (with an @ sign), `<domain>` is not added.
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.

MailToHostDelay=<count> Milliseconds until en e-mail message is sent to the LCR (this timer runs separately for each MaxMailsToHost message).
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.
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 5.
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.

Example:

```
[Mail]
SmtpServer=172.16.0.10
MailRcpt=teles.de
MailFrom=172.16.0.100
MailRcvMax=500
MailRcptMax=100
MaxMailsToHost=2
MailToHostDelay=3000
MailToHostRetries=10
MailSendRetries=10
MailAllowedPeers=172.16.0.10
```

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:

```
...
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 13.2 ➪.

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

MNPQAddress=<ip addr> Enter the IP address to which the number portability query is to be sent. The service comes from an external provider. It is also used as the TELES.iMNP address if the parameter MNPQSum=Yes is set.
MNPQPort=<port> Enter the port to which the number portability query is to be sent.
MNPQAddress2=<ip addr> 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 on is not online.
MNPQPort2=<port> Enter the port to which the second number portability query is to be sent.
MNPQSum=<mode> This parameter must be activated (Yes) if a TELES.iMNP is used.
E2EMRSAddress=<ip addr> Enter the IP address to which the number portability query is to be sent. The service comes from an external provider.
E2EMRSPort=<port> Enter the port to which the number portability query is to be sent.

```
[NumberPortability]
MNPQAddress=172.16.0.100
MNPQPort=9003
MNPQSum=Yes
```

5.2.4 SNMP SETTINGS

The Simple Network Management Protocol facilitates network management and monitoring of TELES.VoIPBOX GSM/CDMA network devices and their functions. For a detailed description of SNMP configuration, please refer to Chapter 15.4 ⇨.

i	**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. Simply copy all `Subscriber` lines into the `Night` Section without making any changes.

5.3 CONFIGURATION FILE ROUTE.CFG

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

- `[System]`
- `[Night<num>]`
- `[VoIP=<name>]`
- `[GateKeeper=<name>]`
- `[Registrar=<name>]`
- `[Radius=<name>]`

5.3.1 ENTRIES IN THE SECTIONS [SYSTEM] AND [NIGHT<NUM>]

The sections `[System]` and `[Night<num>]` contain the following entries.

5.3.1.1 MAPPING

Mapping entries begin with the keyword `MapAll`.

Table 5.25 route.cfg: Map Parameters

MapAll<direct>=<num> <mode>	
`<direct>`	Defines the prefix or telephone number for which the entry applies.

Table 5.25 route.cfg: Map Parameters *(continued)*

MapAll<direct>=<num> <mode>	
`<num>`	Defines the following 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 digit.
`<mode>`	`VOICE` Applies for calls with the service indicator **voice** (default). `DATA` Applies for calls with the service indicator **data**.

Example: In the following example, all international calls are sent to the VoIP carrier (`40`) with the profile name **DF**. All national calls are sent to the analog controller with the number `9`:

```
MapAll00=40DF:00
MapAll0=90
```

5.3.1.2 RESTRICT

This entry is for controller-specific routing entries. These entries apply only for a single controller and can be set for an OAD base number or an MSN:

Table 5.26 route.cfg: Restrict Parameters

Restrict<ns>=<num> <sin>	
`<ns>`	Defines the virtual controller number plus an optional base number or a specific calling number. The special symbol `?` may be used as a wildcard to represent any digit.
`<pl>`	Stands for a virtual placeholder used for the mapping entry that routes calls for the the `Restrict` command.

Table 5.26 route.cfg: Restrict Parameters *(continued)*

Restrict<ns>=<num> <sin>	
`<sin>`	The service indicator variable `sin` restricts the command to one service. Without a `sin`, the `Restrict` 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

Example: In the following example, all calls coming from the GSM network (controller 20) are sent to FXS controller `10` (PBX) without regard to the routing file:

```
Restrict20=pl
MapAllpl=10
```

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 5.27 route.cfg: Redirect Parameters

Redirect<type><num>=<redirect> <sin> <time>	
`<type>`	Enter `1`, `2`, `3` or `5` to set the following types: `1` call forwarding immediately `2` call forwarding no answer `3` call forwarding when busy `5` call forwarding when busy and no answer
`<num>`	Defines the number for which calls will be redirected. The special symbol `?` may be used as a wildcard to represent any digit.

Table 5.27 route.cfg: Redirect Parameters *(continued)*

Redirect<type><num>=<redirect> <sin> <time>	
`<redirect>`	Defines the placeholder used in the two-target routing entry and the number to which calls to <x> will be redirected.
`<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: 00 All services 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 Teletex 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.**
<time>	Optional. For type 2 redirect entries, a timer (in seconds) can be defined after the service indicator entry. **NOTE: In the entry is to apply for all service indicators, the value 00 must be defined for `<sin>`.**

Example: In the following example all international calls (beginning with `00`) are sent to VoIP controller `40` with the carrier profile `DF`. 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 GSM controller `20`.

```
MapAll00=40DF:00
Redirect340DF:=A
MapAllA=20
```

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 TELES.VoIPBOX GSM/CDMA 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 TELES.VoIPBOX GSM/CDMA 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>]

Example: 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.

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

5.3.2 VOIP PROFILES

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

Basic Parameters

Table 5.28 route.cfg: VoIP Basic Parameters

VoIP Basic Parameters
[Voip=<name>] Name of the routing profile. The name must begin with a letter and should be short and meaningful.
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.
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`.
VoipSignalling=<int> Determines the profile's signaling protocol for outgoing VoIP calls. In the case of incoming calls, autorecognition ensures that each call from the peer is accepted, regardless of the protocol: 0=H.323 (default), 1=SIP udp, 2=SIP tcp.

Table 5.28 route.cfg: VoIP Basic Parameters *(continued)*

VoIP Basic Parameters

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`
These ADPCM codecs have various bit rates: g72616 = 16kBit/s (compression ratio 1:4), g72624 = 24kBit/s and g72632 = 32kBit/s (compression ratio 1:2).
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.
`iLBC20`, `iLBC30`
The Internet Low Bitrate Codec works with a frame size of 20 or 30ms. iLBC20 has a bit rate of 15,2 kbit/s and iLBC30 has a bit rate of 13,3 kbit/s.
`gsm`
GSM-FR (full rate) has a bit rate of 13 kbit/s.
The following codecs are also possible: g721
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: ccd
Clear mode (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`).

Table 5.28 route.cfg: VoIP Basic Parameters *(continued)*

VoIP Basic Parameters
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.**
VoipSilenceSuppression=<mode> `Yes` (default) activates silence suppression, CNG (comfort noise generation) and VAD (voice activity detection). `No` 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. 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 8 ⇨ for information on other possible entries.

Management Parameters

Table 5.29 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.
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

5.3.3 GATEKEEPER PROFILES

Gatekeeper profiles are used to connect the TELES.VoIPBOX GSM/CDMA 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 5.30 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> TELES.VoIPBOX GSM/CDMA's defined prefix(es). Use a space to separate entries.
RasId=<name> The alias used for gatekeeper registration.
GkId=<name> The gatekeeper's alias.
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`).

Table 5.30 route.cfg: Gatekeeper Parameters *(continued)*

Gatekeeper Parameters
GkTerminalTypeWithPrefix=<mode> Enter `no` to deactivate sending the Dialed Prefix Information in the Registration Request (default `yes`).

5.3.4 REGISTRAR PROFILES

Registrar profiles are used to register the TELES.VoIPBOX GSM/CDMA 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 5.31 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`.
RegContact=<name or ip addr> Used in the `Contact:` field.
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 TELES.VoIPBOX sends an empty UDP packet to the registrar's IP address. The packet is essentially an alive packet to avoid possible firewall problems.

5.3.5 RADIUS PROFILES

Radius profiles are used to connect the TELES.VoIPBOX GSM/CDMA 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 5.32 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.
User=<name> Enter a username for authorization.
Pwd=<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.

Table 5.32 route.cfg: Radius Parameters *(continued)*

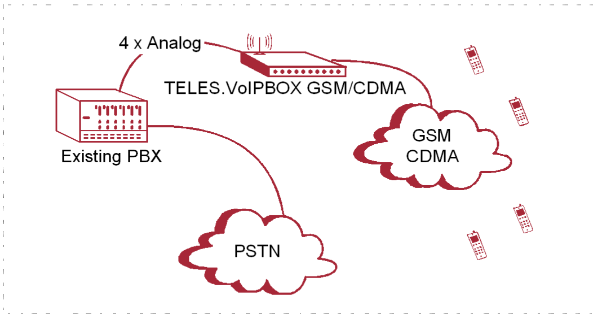
Radius Parameters
CallType=<int> Enter one of the following to define the call type: 3 = VoIP and telephony 2 = VoIP only 1 = Telephony only
FramedProtocol=<int> Enter one of the following to define the framed protocol (see RFC 2865, Chapter 5.7): 1 = PPP 2 = SLIP 3 = AppleTalk Remote Access Protocol (ARAP) 4 = Gandalf proprietary SingleLink/MultiLink protocol 5 = Xylogics proprietary IPX/SLIP 6 = X.75 Synchronous

6 ROUTING EXAMPLES

The following examples describe possible implementation scenarios for the TELES.VoIPBOX GSM/CDMA.

6.1 MOBILE GATEWAY IN A CORPORATE SCENARIO

In the following example, the TELES.VoIPBOX GSM/CDMA is used in a corporate scenario for termination of mobile calls. SIM cards from two carriers are inserted in the TELES.VoIPBOX GSM/CDMA. The PBX sends only calls that must be terminated directly through the mobile networks through the four analog lines. Calls with the mobile carrier prefixes 01555 and 01556 are terminated through the mobile ports with the LAIN 26212. Calls with the prefixes 01665 and 01666 are terminated through the mobile ports with the LAIN 26213.



Configuration in the pabx.cfg

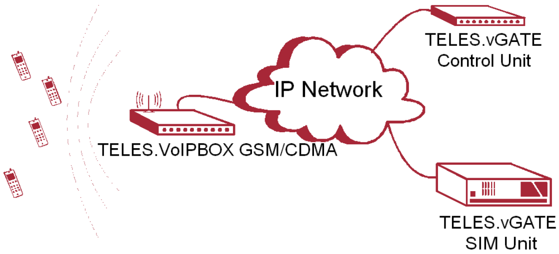
```
...
Subscriber04 = TRANSPARENT ROUTER GSM[0000,00000,+491555555,1,1,1,SIM4,IMSI] CHADDR ALARM NEXT
Subscriber05 = TRANSPARENT ROUTER GSM[0000,00000,+491555555,1,1,1,SIM4,IMSI] CHADDR ALARM
Subscriber06 = TRANSPARENT ROUTER GSM[0000,00000,+491666666,1,1,1,SIM4,IMSI] CHADDR ALARM
Subscriber07 = TRANSPARENT ROUTER GSM[0000,00000,+491666666,1,1,1,SIM4,IMSI] CHADDR ALARM
...
```

Configuration in the route.cfg

```
[System]
DTMFWaitDial=5

MapAll01555=|2621201555<<17
MapAll01556=|2621201556<<17
MapAll01666=|2621301666<<17
MapAll01665=|2621301665<<17
```

6.2 TELES.VOIPBOX GSM/CDMA INTEGRATION IN A CARRIER NETWORK



In the following example, a TELES.VoIPBOX GSM/CDMA is integrated in a carrier network via VoIP. The VoIP traffic is accepted only from the IP address 192.168.100.1. Auto-negotiation is active for VoIP signaling, so that SIP and H.323 are both possible. The voice codecs G.729a and G.711a are used. The TELES.VoIPBOX GSM/CDMA is connected to a TELES.vGATE and receives SIM-card information from a centralized SIM-card server. The IP address for the TELES.vGATE Control Unit is 172.16.0.100. The parameter `SIMS` is used in `SIM<x>` to connect the mobile controller with the TELES.vGATE. All calls coming from VoIP with the prefixes 01555 and 01556 are sent to the carrier's mobile networks. 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.

Configuration in the pabx.cfg

```
....
Subscriber04 = TRANSPARENT ROUTER GSM[0000,00000,+491555555,1,1,1,SIMS,IMSI] CHADDR ALARM NEXT
Subscriber05 = TRANSPARENT ROUTER GSM[0000,00000,+491555555,1,1,1,SIMS,IMSI] CHADDR ALARM
Subscriber06 = TRANSPARENT ROUTER GSM[0000,00000,+491555555,1,1,1,SIMS,IMSI] CHADDR ALARM
Subscriber07 = TRANSPARENT ROUTER GSM[0000,00000,+491555555,1,1,1,SIMS,IMSI] CHADDR ALARM
Subscriber08 = TRANSPARENT ROUTER SWITCH CHMAX[8] ALARM
```

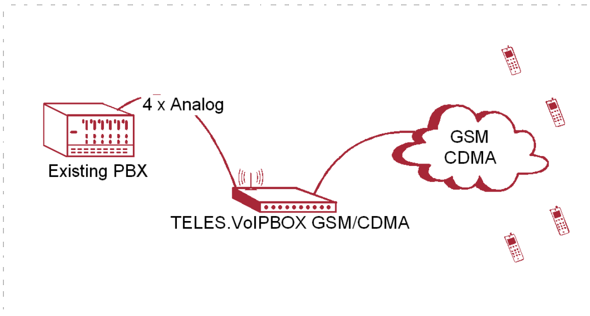
Configuration in the route.cfg

```
[System]
DTMFWaitDial=5
MapAll01555=|2621201555<<17
MapAll01556=|2621201556<<17

[Voip=DF]
VoipDirection=In
VoipPeerAddress=192.168.100.1
VoipIpMask=0xffffffff
VoipSignalling=1
VoipCompression=g729a G711a
VoipSilenceSuppression=Yes
VoipMaxChan=4
VoipTxM=2
```

6.3 MOBILE NT

In the following example, an analog PBX is connected directly with the GSM carrier in mobile NT mode. All FXS ports are bundled together in a trunk group with the name 30. The GSM controllers are bundled together in trunk group 20. The SIM cards are inserted in the TELES.VoIPBOX GSM/CDMA (`SIM4`) and the carrier uses them in a group. That means that the call is sent to another group when one SIM card is used. The routing process for the TELES.VoIPBOX GSM/CDMA is configured without changes from GSM toward analog and vice versa.



Configuration in the pabx.cfg

```
...
Controller00 = 30 ANA
Controller01 = 30 ANA
Controller02 = 30 ANA
Controller03 = 30 ANA
Controller04 = 20 GSM
Controller05 = 20 GSM
Controller06 = 20 GSM
Controller07 = 20 GSM
...
Subscriber00 = TRANSPARENT ROUTER ANA[0049,16]
Subscriber01 = TRANSPARENT ROUTER ANA[0049,16]
Subscriber02 = TRANSPARENT ROUTER ANA[0049,16]
Subscriber03 = TRANSPARENT ROUTER ANA[0049,16]
Subscriber04 = TRANSPARENT ROUTER GSM[0000,00000,+491555555,1,1,1,SIM4,IMSI] ALARM NEXT
Subscriber05 = TRANSPARENT ROUTER GSM[0000,00000,+491555555,1,1,1,SIM4,IMSI] ALARM
Subscriber06 = TRANSPARENT ROUTER GSM[0000,00000,+491555555,1,1,1,SIM4,IMSI] ALARM
Subscriber07 = TRANSPARENT ROUTER GSM[0000,00000,+491555555,1,1,1,SIM4,IMSI] ALARM
...
```

Configuration in the route.cfg

```
[System]
DTMFWaitDial=5

Restrict30=pl
MapAllpl=|20<<24

Restrict20=pbx
MapAllpbx=30
```

.

7 SYSTEM MAINTENANCE AND SOFTWARE UPDATE

7.1 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.

7.2 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.

```
16.05.06-11:51:31,[990]Start STATUS - TELES.VoIPBOX 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 7.33 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.

Table 7.33 Event Log Messages *(continued)*

Message	NMS	Definition
`Remote Control: Date & Time changed`		Date and/or time were changed via remote administration.
`Time Switch Operation`		The configuration change was made by the timer.
Remote Administration		
`Remote Control from <peer>, <Remote-Code>, <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) Should the error persist, 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 Call-back-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 `MapCallBack-WaitDisc`).

Table 7.33 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 status and error messages appear in the `protocol.log` file when `ALARM` appears in the VoIP port's subscriber line:

Table 7.34 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 or automatically via TELES.vGATE for mobile channels.
`[<port>]Block Port`	The `<port>` has been blocked. This can occur via remote access for all controller types or automatically via TELES.vGATE for mobile channels.
`[<port>]Restart Port`	The `<port>` has been blocked. This can occur via remote access for all controller types or automatically via TELES.vGATE for mobile channels.
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 7.34 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 (RegistrationConfirm).
`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.

Table 7.34 Protocol Log Status and Error Messages *(continued)*

Message	Definition
`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	
`[99]ERR: SimUnit !connect`	An outgoing connection to the TELES.vGATE Sim Unit could not be established.
`[99]ERR: ControlUnit <ip addr> !connect`	An outgoing connection to the TELES.vGATE Control Unit could not be established.
Number Portability	
`[99i]ERR: np !connect`	Connection to the TELES.iMNP could not be established.
`[99i]ERR: np connect <ip addr>`	Connection to the TELES.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`	

Table 7.34 Protocol Log Status and Error Messages *(continued)*

Message	Definition
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	
Error in <dsl_interface>: Connection Failed. Please, revise your Username/ Password configuration	
Error in <dsl_interface>: Connection Failed. Please, revise the DSL Modem	

7.3 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:

Check the software version running on your system to make sure the one you want to install is newer.

The basic software consists of the following files:

```
start
netbsdz
netbsd.gz
gbox.tz1
```

Once you have the same version number of all of these files, you can upload them via TELES.GATE Manager. Make sure there is enough available memory for the new version. This is better than uploading them vie FTP (binary mode), especially in the case of low bandwidth, because the old files will not be overwritten until the new file has been completely uploaded onto the system. It may take some time to upload and verify all of the files.

Once the files have been completely transferred, check the file size and reboot the system. As soon as you can reach the system via TELES.GATE Manager again, check the version number of the running software.

An update of the following optional function modules (see Chapter 15 ➪) occurs in the same way. Make sure the file extension has the same running number as that of the file on the system:

- HTTP User Interface:
  `httpd.tz2`
  `httpd.izg`
- TELES.iPBX:
  `ipbx.tz2`
  `ipbx.izg`
- DNS forwarder:
  `dnsmasg.tz2`
- SNMP agent:
  `snmpd.tz0`
- IP update:
  `ipupdate.tz2`

The only exception is that you must shut down the modules that have `*.izg` files before updating. To shut down these modules, change the name of or delete the corresponding `*.tz*` file and restart the system.

Following transfer of the `*.izg` file, you must rename the `*.tz.*` file again and restart the system.

7.4 TRACE

During operation, the trace readouts of the TELES.VoIPBOX GSM/CDMA can be saved in a file or transmitted with remote maintenance directly. The trace options must be turned on in the TELES.GATE Manager (offline or online trace) or via FTP raw commands (see Chapter 4.11.3 ➪). Trace results presented here are for analog, mobile and VoIP interfaces, and for the following services in various levels:

Table 7.35 Trace Options

Option	Definition
Mail	Output for all SMTP packets.
NumberPortability	Opuput of all packets for communication with the TELES.iMNP.
vGATE	Ouput of all packets for communication with the TELES.vGATE.
VoiceCodecs	Output of RTCP information described under VP module.
PPP	Output of PPP connection information.
DTMF	Output for DTMF tone recognition.
Remote	Output for TELES.GATE Manager and TELES.NMS communication.

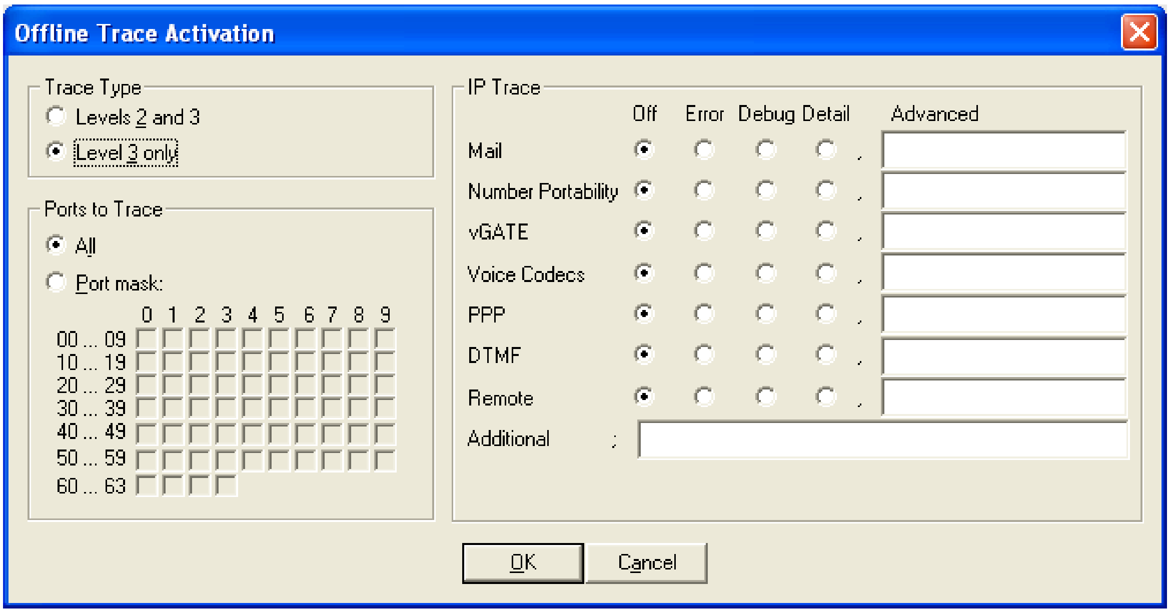


Figure 7.9 TELES.GATE Manager: Offline Trace Activation Window

TELES.VoIPBOX GSM/CDMAs offer two different types of trace:

- Online - trace information is immediately displayed in the TELES.GATE Manager's trace window.
- Offline - trace information is written to a file on the TELES.VoIPBOX GSM/CDMA.

TELES.VoIPBOX GSM/CDMA systems create trace files when the `TraceLog=file` entry is present in the `pabx.cfg`. Traces can be activated via remote administration (TELES.GATE Manager or FTP).

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 mobile ports
  - `r` = receive for mobile ports
  - `stt`, `app`, `mid` and `ton` are described in Table 7.36
  - `x` = send to VoIP destinations
  - `y` = receive from VoIP destinations
  - `i` = information messages and internal trace outputs between VoIP and the other interfaces (POTS, 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

7.4.1 POTS TRACE OUTPUT

Trace output for FXS and FXO controllers is in ASCII (text notation. the `<trace>` part (see Chapter 7.4 ⇨) is consists of `<trace type>`, followed directly by `<trace output>`.

7.4.1.1 TRACE TYPES

Possible notations for <trace type> are described in Table 7.36.

Table 7.36 POTS Trace Type

Trace Type	Description
`stt`	Output related to a status change in the POTS controller.
`app`	Output related to the driver.
`mid`	Output from the interface to the physical layer.
`ton`	Information about tone recognition in the B-channel.

7.4.1.2 SST TRACE OUTPUT

The most important information appears in the controller status output `stt`, which appears only if the controller's state changes. Table 7.37 contains a description of possible trace output.

Table 7.37 sst Trace Output

Trace Output	Description
IDLE	Idle state.
UP	A-subscriber has picked up the phone.
DIAL	A-subscriber is dialing.
ALERT	Alert state.
CONNECT	Connecting.
AWAIT_HOOKDOWN	Waiting for A-subscriber to hang up the phone.

Example: In the following output example, FXS controller 05 (calling party) has been in the `ALERT` state. This state changes to `CONNECT` as soon as the called party picks up the phone:

```
[20:33.54] i[05]: sttALERT              -> CONNECT
```

7.4.1.3 APP TRACE OUTPUT

The application output `app` provides important information about the connection process.

Table 7.38 app Trace Output

Trace Output	Description
CONN_IND	Connect indication
CONN_REQ	Connect request
CONN_ACK	Connect acknowledgment
ALRT_IND	Alert indication
CONN_CNF	Connect confirmation
CONN_RSP	Connect response
DSC_IND	Disconnect indication
DSC_REQ	disconnect request
DSC_CNF	Disconnect confirmation

Table 7.38 app Trace Output *(continued)*

Trace Output	Description
`info:`	Dialed digit

7.4.1.4 MID TRACE OUTPUT

`mid` traces are intended for monitoring the state of the physical layer and are described in Table 7.39.

Table 7.39 mid Trace Output

Trace Output	Description
`+`	A-subscriber has picked up the phone.
`bchan`	B-channel is connected.
up '<DTMF>'	A-subscriber has entered a digit via DTMF.
`dn cif <count>`	Number of impulses for each charge unit.
`-`	A-subscriber has hung up the phone.

Example: In the following example, the digit 0 has been dialed:

```
[30:23.21] i[05]: midup '0'
```

7.4.1.5 TON TRACE OUTPUT

The trace output `ton` provides information about tone recognition in the B-channel. Since they appear unabridged and are not of importance, no description is necessary.

7.4.2 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 7.40 and Table 7.41 describe the contents of GSM trace output.

Table 7.40 Request Messages to the GSM Module

Hex Value	Description
`00`	Setup
`01`	Connect
`02`	Disconnect
`03`	SMS

Table 7.40 Request Messages to the GSM Module *(contin-*

Hex Value	Description
04	DTMF
05	Set Config
06	Get Config
07	LED
08	Restart
09	Switch SIM

Table 7.41 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-In Call Proceeding
19	USSD
1A	Restart Indication

Example: The following example shows a GSM call through the 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
```

7.4.3 VOIP TRACE OUTPUT

As described above in Chapter 7.4 ⇨, 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 7.4.3.6 ⇨.

In general, the following rules apply for this trace output:

Table 7.42 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).

7.4.3.1 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).

```
h225connect to <ip address> cr <cr> s <si>
h225accept from <ip address> s <si>
```

Table 7.43 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 7.44 H.225 Signaling

Trace Output	Description
`<mt>`	The ETS message type in hexadecimal; can consist of values listed in Table 7.45.
`<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 7.45 ETS Message Types

Hex Value	Message Type
1	Alerting
2	Call Proceeding
3	Progress
5	Setup
7	Connect
D	Setup Acknowledge
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 7.46 Incoming VoIP Calls

Trace Output	Description
`<mt>`	Message type in hexadecimal as per ETS standard (see Table 7.45) 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 7.47).
`FS<bool>`	FastStart offered in the signaling packet or not.
`(<codec>,`	Lists codecs offered (seeTable 7.69).

Table 7.46 Incoming VoIP Calls *(continued)*

Trace Output	Description
`<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).
`ALT:<ip ad-dress>:<port>,<DAD>`	Optional alternative values for IP address port or a new destination number for a facility message with the cause `call forwarded`.

Table 7.47 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

Table 7.47 Compression Codecs Used *(continued)*

Synonym	Codec
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 7.48 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 7.45.
<num>	Called party number.
<cr>	Call reference.

Or:

```
h225<type> callproc typ <mt> cr <cr>
```

Table 7.49 Calls to the IP Network 2

Trace Output	Description
<mt>	The ETS message type in hexadecimal.
<cr>	Call reference.

7.4.3.2 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 7.50 RTP/RTCP Output

Trace Output	Description
<cr>	Call reference.
<ch>	The internal media channel used.
	`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 7.47.
<factor>	Multiplication factor for default frame size (20ms, 30 ms for G.723).

```
rtp stop cr <cr>1 ch <ch>
```

Table 7.51 RTP Stop Message

Trace Output	Description
<cr>	Call reference.
<ch>	The internal media channel used.

VP Module

This module's output shows the controller packets for the voice connections. That means that the RTCP packets and relevant information also appear.

The following results occur for a new RTP connection:

```
a[<controller>]: <VoIPcodecChipType> start(val) ch=<ch> local=<port> remote=<ip address:port> agg=<bool>
```

Table 7.52 RTP/RTCP Output (VP Module)

Trace Output	Description
<controller>	Running number for the VoIP controller.
<VoIPcodecChipType>	Stands for the type designation for the compression chips used (e.g. Ac49x).
`<val>`	Shows which connection is set up.
`<ch>`	The internal media channel used.
`<port>`	RTP port.
`<ip address>`	Peer's IP address in hexadecimal.
`agg=<bool>`	`1` means an RTP-multiplex connection is used (default `0`).

The following output shows the channel’s state in the compression chip during a startup or change of codec:

```
a[<controller>]: <VoIPcodecChipType>OpenChannelConfiguration ch=<ch> rc=0
a[<controller>]: <VoIPcodecChipType>T38ChannelConfiguration ch=<ch> rc=0
a[<controller>]: <VoIPcodecChipType>ActivateRegularRtpChannelConfiguration ch=<ch> rc=0
```

The following output shows whether the compression chip starts sending and receiving packets:

```
a[<controller>]: <VoIPcodecChipType> ch <ch> establish
```

Sent and received bytes appear with the following output results:

```
a[<controller>]: <VoIPcodecChipType> ch <ch>: in <byte> out <byte>
```

Table 7.53 RTP Packet Statistics

Trace Output	Description
`<ch>`	The internal media channel used.
`<byte>`	The call's received or sent bytes.

```
a[<controller>]: <VoIPcodecChipType> ch <ch> rtcp<dir> <num> ji <ji> rt <rt> fl <fl> in <byte> out <byte>
```

Table 7.54 RTCP Packet Statistics

Trace Output	Description
`<ch>`	The internal media channel used.
`rtcp<dir>`	R sender report (received) is more interesting, since it comes from the peer. T sender report (transmitted).
`<num>`	`0` ReceiverReport packet `1` SenderReport packet `2` Packet requested by the driver
`<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 7.55 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 7.56 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 7.57 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 7.58)
n3	Signal detected on encoder input (see Table 7.58)

Table 7.58 faxordatasignalevent

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 mo-dem)
9	V32_AA	V.32 AA (calling modem tone, 1800 Hz)

Table 7.58 faxordatasignalevent *(continued)*

Value	Definition	Description
10	V22_USB1	V.22 USB1 (V.22(bis) unscrambled binary ones)
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 7.59 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
```

Table 7.59 Fax Status

Value	Description
n1	UnableToRecoverFlag (0 no, 1 yes)
n2	IllegalHdlcFrameDetectedFlag (...)
n3	FaxExitWithNoMcfFrameFlag
n4	HostTransmitOverRunFlag
n5	HostTransmitUnderRunFlag
n6	InternalErrorFlag
n7	ReceivedBadCommandFlag

Table 7.59 Fax Status *(continued)*

Value	Description
n8	TimeOutErrorFlag
n9	TxRxFlag (0 receive, 1 transmit)
n10	T30State 0 FAX_RELAY_T30_STATE__INITIALIZATION 1 FAX_RELAY_T30_STATE__CNG 2 FAX_RELAY_T30_STATE__CED 3 FAX_RELAY_T30_STATE__V21 4 FAX_RELAY_T30_STATE__NSF 5 FAX_RELAY_T30_STATE__NSC 6 FAX_RELAY_T30_STATE__CSI 7 FAX_RELAY_T30_STATE__CIG 8 FAX_RELAY_T30_STATE__DIS 9 FAX_RELAY_T30_STATE__DTC 10 FAX_RELAY_T30_STATE__NSS 11 FAX_RELAY_T30_STATE__TSI 12 FAX_RELAY_T30_STATE__DCS 13 FAX_RELAY_T30_STATE__CTC 14 FAX_RELAY_T30_STATE__CRP 15 FAX_RELAY_T30_STATE__DCN 16 FAX_RELAY_T30_STATE__PRE_MESSAGE_RESPONSE 17 FAX_RELAY_T30_STATE__POST_MESSAGE_RESPONSE 18 FAX_RELAY_T30_STATE__POST_MESSAGE_COMMAND 19 FAX_RELAY_T30_STATE__VXX 20 FAX_RELAY_T30_STATE__TCF 21 FAX_RELAY_T30_STATE__IMAGE
n11	NumberOfTransferredPages
n12	BadInputPacketId
n13	BadInputPacketTotalSize

Table 7.59 Fax Status *(continued)*

Value	Description
n14	FaxBitRate 1 FAX_BIT_RATE__300_BPS 2 FAX_BIT_RATE__2400_BPS 3 FAX_BIT_RATE__4800_BPS 4 FAX_BIT_RATE__7200_BPS 5 FAX_BIT_RATE__9600_BPS 6 FAX_BIT_RATE__12000_BPS 7 FAX_BIT_RATE__14400_BPS

The following output appears when the compression chip recognizes DTMF tones:

```
a[<controller]: ac49x ch <ch> ibs <dtmf> <dir> <mode> <lev> <dur>
```

Table 7.60 DTMF Tone Recognition

Trace Output	Description
`<ch>`	Media channel
`<dtmf>`	Recognized 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 recognized
`<lev>`	Signal level in -dBm
`<dur>`	Tone duration

7.4.3.3 INTERNAL PROTOCOL INTERFACE (TO ISDN, POTS, 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> <mt1> dad <num> oad <num> cr <cr> s <si> ch <chan> isdncr<icr>
```

Table 7.61 Internal Protocol Interface

Trace Output	Description
`<type>`	Direction from (`rcv`) or to (`snd`) the internal protocol interface.
`<mt1>`	Message type written out; if a decimal number appears, it will be translated as per Table 7.45.
`<num>`	`DAD<num>` = called party number, `OAD<num>` = calling party number.
`<cr>`	Call reference.
`<si>`	Session ID.
`<chan>`	Media channel used.
`<icr>`	Call reference for the internal protocol interface (DSS1).

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>
```

Table 7.62 Received from PSTN 1

Trace Output	Description
`<voip profile>`	Defines the VoIP profile to be used.

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 7.63 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).

7.4.3.4 H.245 MESSAGES

The following trace output is possible:

```
h245<dir>(<tt>) cr <cr>
```

Table 7.64 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 7.65 Codec Used

Trace Output	Description
<comp>	List of compression codecs offered (see Table 7.47), the list of the peer's codecs appears behind peer, and `cfg` shows which codecs are defined in the VoIP profile

```
... OpenLogicalChannel cn=<cn> cpr=<comp> sessid=<sid> ctrl=<ip address>:<rtcp port>
... OpenLogicalChannelAck cn=<cn> sessid=<sid> media=<ip address>:<rtp port>
```

Table 7.66 Logical Channel Parameters

Trace Output	Description
<cn>	H.245 channel number per H.225 connection.
<sid>	Session ID.
<comp>	Codec used (see Table 7.47).

Table 7.66 Logical Channel Parameters *(continued)*

Trace Output	Description
`<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 7.67 H.245 Parameters

Trace Output	Description
`hmt`	The H.245 message type (multimedia system control message type), (Table 7.68).
`hmi`	The H.245 message ID (see Table 7.69, Table 7.70, Table 7.71, Table 7.72).

Table 7.68 Multimedia System Control Message Types

ID	Message
`0` (Table 7.69)	Request
`1` (Table 7.70)	Response
`2` (Table 7.71)	Command
`3` (Table 7.72)	Indication

Depending on the system control message type, one of the following message IDs appear:

Table 7.69 Message IDs for Request Message

ID	Message
`0`	NonStandard
`1`	MasterSlaveDetermination
`2`	TerminalCapabilitySet
`3`	OpenLogicalChannel
`4`	CloseLogicalChannel

Table 7.69 Message IDs for Request Message *(continued)*

ID	Message
5	RequestChannelClose
6	MultiplexEntrySend
7	RequestMultiplexEntry
8	RequestMode
9	RoundTripDelayRequest
10	MaintenanceLoopRequest
11	CommunicationModeRequest
12	ConferenceRequest
13	MultilinkRequest
14	LogicalChannelRateRequest

Table 7.70 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
13	RequestMultiplexEntryReject

Table 7.70 Message IDs for Response Message *(continued)*

ID	Message
14	RequestModeAck
15	RequestModeReject
16	RoundTripDelayResponse
17	MaintenanceLoopAck
18	MaintenanceLoopReject
19	CommunicationModeResponse
20	ConferenceResponse
21	MultilinkResponse
22	LogicalChannelRateAcknowledge
23	LogicalChannelRateReject

Table 7.71 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 7.72 Message IDs For Indication Message

ID	Message
0	NonStandard
1	FunctionNotUnderstood
2	MasterSlaveDeterminationRelease
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

7.4.3.5 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>`):

```
H225 GatekeeperRequest to <ip addr> (s 131)
H225 GatekeeperConfirm <ip addr>
H225 GatekeeperReject <ip addr> reason <reason>
```

Table 7.73 RAS

Trace Output	Description
`<reason>`	Gatekeeper reject reason, see Table 7.77.

```
H225 GkRegistration to <ip addr>
H225 RegistrationConfirm <ip addr>
H225 RegistrationReject <ip addr> reason <reason>
```

Table 7.74 Gatekeeper 1

Trace Output	Description
`<reason>`	Registration reject reason, see Table 7.78.

```
H225 GkResourcesAvailableIndicate to <ip addr> (<act chan> <max chan>)
H225 ResourcesAvailableConfirm <ip addr>
```

```
H225 GkAdmission cr <cr> to <ip addr>
H225 AdmissionConfirm <ip addr> cr <cr>
H225 AdmissionReject <ip addr> reason <reason>
```

Table 7.75 Gatekeeper 2

Trace Output	Description
`<reason>`	Admission reject reason, see Table 7.79.

```
H225 GkDisengage cr <cr> to <ip addr>
H225 DisengageConfirm <ip addr>
```

```
H225 UnregistrationRequest <ip addr>
H225 GkUnregistrationConf to <ip addr>
```

All other messages appear as follows:

```
H225 unknown msg from Gk <ip addr>: <code>
```

Table 7.76 Gatekeeper 3

Trace Output	Description
`<code>`	Unknown gatekeeper message, see Table 7.80.

Table 7.77 Gatekeeper Reject Reason

ID	Reject Reason
0	resourceUnavailable
1	terminalExcluded
2	invalidRevision
3	undefinedReason
4	securityDenial
5	genericDataReason
6	neededFeatureNotSupported

Table 7.78 Registration Reject Reason

ID	Reject Reason
0	DiscoveryRequired
1	InvalidRevision

Table 7.78 Registration Reject Reason *(continued)*

ID	Reject Reason
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 7.79 Admission Reject Reason

ID	Reject Reason
0	CalledPartyNotRegistered
1	InvalidPermission
2	RequestDenied
3	UndefinedReason
4	CallerNotRegistered
5	RouteCallToGatekeeper
6	InvalidEndpointIdentifier
7	ResourceUnavailable
8	SecurityDenial

Table 7.79 Admission Reject Reason *(continued)*

ID	Reject Reason
9	QosControlNotSupported
10	IncompleteAddress
11	AliasesInconsistent
12	RouteCallToSCN
13	ExceedsCallCapacity
14	CollectDestination
15	CollectPIN
16	GenericDataReason
17	NeededFeatureNotSupported

Table 7.80 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
13	BandwidthConfirm
14	BandwidthReject

Table 7.80 Unknown Gatekeeper Messages *(continued)*

ID	Message
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

7.4.3.6 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 7.81 ENUM Output

Trace Output	Description
`<cr>`	Call reference.
`<ch>`	Media channel.
`<num>`	Phone number converted into ENUM domain format.

Table 7.81 ENUM Output *(continued)*

Trace Output	Description
`<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.

7.4.3.7 EXAMPLES

The following examples are offline traces. You can generate them using the TELES.GATE Manager or FTP commands. The filename is `trace.log`. The following cases appear in the examples:

- Incoming H323 Call with FastStart (Chapter ⇨)
- Outgoing H323 Call with FastStart (Chapter ⇨)
- Fax Call (Chapter ⇨)

Incoming H323 Call with FastStart

```
[15:04:09.12] i[04]: h225accept from 172.16.0.100 s 4
[15:04:09.15] y[04]: h225rcv tpkt msg 5 h225cr 8006 addr 172.16.0.100 pt 0
[15:04:09.16] y[04]: h225 decode rc 0, q931 msg 5 (0), len 361
[15:04:09.16] y[04]: h225rcv setup voipcfg addr 172.16.0.100 rc 0 <DF> compr EABG
[15:04:09.16] y[04]: h225rcv faststart <A4B4E4G0>
[15:04:09.16] y[04]: h225rcv setup oad 01 00 <1111> <> dad 01 <321> rad <> bc 038090a3 0101
[15:04:09.16] y[04]: h225rcv setup h225cr 8006 FS:1(E,172.16.0.100,29000) TUNN:1 H245:0(0,0)
[15:04:09.16] y[04]: h225rcv setup h225cr 8006 cr 5
[15:04:09.16] i[04]: pstnsnd setup dad 1 oad 1111 cr 5 s 4
[15:04:09.16] s[02]: 02 ff 03 08 01 02 05 04 03 80 90 a3 18 01 89 6c 06 01 81 31 31 31 31 70 04 81 33 32
31 7d 02 91 81
[15:04:09.16] i[04]: pstnrcv connresp cr 5 acc 5 ch 1
[15:04:09.16] x[04]: h225snd callproc typ d cr 5 pri 0
[15:04:09.50] r[02]: 00 81 20 1a 08 01 82 01 18 01 89
[15:04:09.50] s[02]: 00 81 01 22
[15:04:09.50] i[04]: pstnrcv alert cr 5 cls ff
[15:04:09.50] i[04]: rtp start cr 5 ch 1 li 1 ri 1 st 2 fx 0 cp E txm 2
[15:04:09.50] x[04]: h225snd callproc typ 1 cr 5 pri 8
[15:04:09.52] a[04]: ac49x start(201) ch=0 local=29000 remote=ac100064:29000 agg=0
[15:04:09.52] a[04]: Ac49xOpenChannelConfiguration ch=0 rc=0
[15:04:09.52] a[04]: Ac49xT38ChannelConfiguration ch=0 rc=0
[15:04:09.52] a[04]: Ac49xActivateRegularRtpChannelConfiguration ch=0 rc=0
[15:04:09.63] a[04]: ac49x ch 0 rtcpR 0 ji -1 rt -1 fl 65535 in 0 out -1
[15:04:09.63] a[04]: ac49x ch 0 establish
[15:04:09.98] a[04]: ac49x ch 0 jitter buffer 75 0 0 0 1 0 0 0
[15:04:10.94] a[04]: ac49x ch 0 jitter buffer 115 5 0 5 1 0 0 0
[15:04:11.79] r[02]: 00 81 22 1a 08 01 82 07 4c 03 00 80 31
[15:04:11.79] s[02]: 02 81 1a 24 08 01 02 0f
[15:04:11.79] i[04]: pstnrcv connresp cr 5 acc 10 ch 255
[15:04:11.79] x[04]: h225snd callproc typ 7 cr 5 pri 0
[15:04:11.89] r[02]: 02 81 01 1c
[15:04:12.49] a[04]: ac49x ch 0 rtcpT 1 ji 201 rt 0 fl 0 in 290 out 394
[15:04:12.49] a[04]: ac49x ch 0: in 1552 out 1646
[15:04:13.50] a[04]: ac49x ch 0 jitter buffer 125 1 0 1 0 0 0 0
[15:04:14.50] a[04]: ac49x ch 0 jitter buffer 145 3 0 3 1 0 0 0
[15:04:15.56] a[04]: ac49x ch 0 jitter buffer 145 0 0 0 1 0 0 0
[15:04:16.23] a[04]: ac49x ch 0 rtcpR 1 ji 196 rt 84 fl 0 in 3236 out 3236
[15:04:17.98] r[02]: 00 81 24 1c 08 01 82 45 08 02 80 90
[15:04:17.98] s[02]: 00 81 01 26
[15:04:17.98] i[04]: pstnrcv terminate connection (3201) cr 5 cau 90 err 0 state 16 ch 1 rsid 1
[15:04:17.98] i[04]: rtp stop cr 5 ch 1
[15:04:17.98] x[04]: h225snd relack cr 5 cau 0x90
[15:04:17.98] i[04]: h225connection s 4 close
[15:04:17.98] i[04]: CloseSysFd 4 (st 22)
[15:04:18.03] s[02]: 02 81 1c 26 08 01 02 4d
[15:04:18.03] a[04]: ac49x ch 0: in 20486 out 21288
[15:04:18.03] a[04]: ac49x stop ch=0
[15:04:18.06] a[04]: ac49x ch 0 rtcpR 2 ji 221 rt 84 fl 0 in 5012 out 5510
[15:04:18.24] r[02]: 02 81 01 1e
[15:04:18.28] r[02]: 00 81 26 1e 08 01 82 5a
```

Outgoing H323 Call with FastStart

```
[15:25:13.61] r[02]: 00 81 2a 1e 08 01 48 05 04 03 80 90 a3 18 01 83 6c 05 00 80 31 31 31 70 07 81 31 32
33 34 35 36 7d 02 91 81
[15:25:13.61] s[02]: 00 81 01 2c
[15:25:13.61] s[02]: 02 81 1e 2c 08 01 c8 0d 18 01 8a
[15:25:13.61] i[04]: pstnrcv setup dad DF:123456 oad 111 cc 0 id dd006
[15:25:13.61] i[04]: pstnrcv get_voipcfg <DF>
[15:25:13.61] i[04]: h225connect to 172.16.0.100 cr 6
[15:25:13.61] x[04]: h225snd setup dad 123456 cr 6
[15:25:13.69] r[02]: 02 81 01 20
[15:25:13.69] y[04]: h225rcv tpkt msg d h225cr 6 addr 172.16.0.100 pt 8018c000
[15:25:13.69] y[04]: h225 decode rc 0, q931 msg d (11), len 32
[15:25:13.69] y[04]: h225rcv msg d (11) h225cr 6 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[15:25:14.36] y[04]: h225rcv tpkt msg 1 h225cr 6 addr 172.16.0.100 pt 8018c000
[15:25:14.36] y[04]: h225 decode rc 0, q931 msg 1 (3), len 119
[15:25:14.36] y[04]: h225rcv faststart <E4>
[15:25:14.36] y[04]: h225rcv alert h225cr 6 FS:1(E,172.16.0.100,29000) TUNN:1 H245:0(0,0)
[15:25:14.36] i[04]: rtp start cr 6 ch 1 li 1 ri 1 st 2 fx 0 cp E txm 2
[15:25:14.36] s[02]: 02 81 20 2c 08 01 c8 01 1e 02 82 88
[15:25:14.39] a[04]: ac49x start(201) ch=0 local=29000 remote=ac100064:29000 agg=0
[15:25:14.39] a[04]: Ac49xOpenChannelConfiguration ch=0 rc=0
[15:25:14.39] a[04]: Ac49xT38ChannelConfiguration ch=0 rc=0
[15:25:14.39] a[04]: Ac49xActivateRegularRtpChannelConfiguration ch=0 rc=0
[15:25:14.41] r[02]: 02 81 01 22
[15:25:14.50] a[04]: ac49x ch 0 rtcpR 0 ji -1 rt -1 fl 65535 in 0 out -1
[15:25:14.50] a[04]: ac49x ch 0 establish
[15:25:14.71] a[04]: ac49x ch 0 jitter buffer 35 1 0 1 0 0 0 0
[15:25:15.59] y[04]: h225rcv tpkt msg 7 h225cr 6 addr 172.16.0.100 pt 8018c000
[15:25:15.59] y[04]: h225 decode rc 0, q931 msg 7 (2), len 77
[15:25:15.59] y[04]: h225rcv connect h225cr 6 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[15:25:15.59] i[04]: pstnsnd connect cr 6
[15:25:15.59] s[02]: 02 81 22 2c 08 01 c8 07 29 05 06 03 18 0f 19
[15:25:15.62] a[04]: ac49x ch 0 jitter buffer 145 15 0 17 5 2 0 0
[15:25:15.65] r[02]: 02 81 01 24
[15:25:15.93] r[02]: 00 81 2c 24 08 01 48 0f
[15:25:15.93] s[02]: 00 81 01 2e
[15:25:16.98] a[04]: ac49x ch 0 rtcpT 1 ji 158 rt 0 fl 2 in 2316 out 1816
[15:25:16.98] a[04]: ac49x ch 0: in 8836 out 7874
[15:25:17.57] a[04]: ac49x ch 0 jitter buffer 145 0 0 0 1 0 0 0
[15:25:18.60] a[04]: ac49x ch 0 jitter buffer 145 0 0 0 2 0 0 0
[15:25:20.10] a[04]: ac49x ch 0 rtcpT 1 ji 208 rt 0 fl 0 in 5376 out 4634
[15:25:20.10] a[04]: ac49x ch 0: in 20084 out 18802
[15:25:20.21] a[04]: ac49x ch 0 jitter buffer 145 1 0 1 0 0 0 0
[15:25:20.25] a[04]: ac49x ch 0 rtcpR 1 ji 164 rt 147 fl 0 in 5476 out 5496
[15:25:21.21] a[04]: ac49x ch 0 jitter buffer 155 1 0 1 1 0 0 0
[15:25:23.40] a[04]: ac49x ch 0 rtcpR 1 ji 176 rt 36 fl 0 in 8756 out 8776
[15:25:24.71] r[02]: 00 81 2e 24 08 01 48 45 08 02 80 90
[15:25:24.71] s[02]: 00 81 01 30
[15:25:24.71] i[04]: pstnrcv terminate connection (3201) cr 6 cau 90 err 0 state 16 ch 1 rsid 1
[15:25:24.71] i[04]: rtp stop cr 6 ch 1
[15:25:24.71] x[04]: h225snd relack cr 6 cau 0x90
[15:25:24.71] i[04]: h225connection s 4 close
[15:25:24.71] i[04]: CloseSysFd 4 (st 22)
[15:25:24.71] s[02]: 02 81 24 30 08 01 c8 4d
[15:25:24.79] a[04]: ac49x ch 0: in 37858 out 34096
[15:25:24.79] a[04]: ac49x stop ch=0
[15:25:24.83] a[04]: ac49x ch 0 rtcpR 2 ji 194 rt 36 fl 0 in 10116 out 8426
[15:25:24.92] r[02]: 02 81 01 26
[15:25:24.92] r[02]: 00 81 30 26 08 01 48 5a
```

Fax Call

```
[16:00:33.87] r[02]: 00 81 54 2c 08 01 01 05 04 03 80 90 a3 18 01 83 7d 02 91 81
[16:00:33.87] s[02]: 02 81 2c 56 08 01 81 0d 18 01 89 1e 02 82 88
[16:00:36.99] r[02]: 00 81 56 2e 08 01 01 7b 70 02 81 31
[16:00:37.17] r[02]: 00 81 58 2e 08 01 01 7b 70 02 81 32
[16:00:37.33] r[02]: 00 81 5a 2e 08 01 01 7b 70 02 81 33
[16:00:37.54] r[02]: 00 81 5c 2e 08 01 01 7b 70 02 81 34
[16:00:40.46] s[02]: 02 81 2e 5e 08 01 81 02 1e 02 82 88
[16:00:40.46] i[04]: pstnrcv setup dad DF:1234 oad  cc 0 id 11d007
[16:00:40.46] i[04]: pstnrcv get_voipcfg <DF>
[16:00:40.46] i[04]: rtp start cr 7 ch 1 li 1 ri 0 st 1 fx 0 cp E txm 1
[16:00:40.46] i[04]: h225connect to 172.20.0.100 cr 7
[16:00:40.46] x[04]: h225snd setup dad 1234 cr 7
[16:00:40.46] y[04]: h225rcv tpkt msg d h225cr 7 addr 172.20.0.100 pt 800e7000
[16:00:40.46] y[04]: h225 decode rc 0, q931 msg d (11), len 32
[16:00:40.46] y[04]: h225rcv msg d (11) h225cr 7 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:40.54] a[04]: ac49x start(201) ch=0 local=29000 remote=0:0 agg=0
[16:00:40.54] a[04]: Ac49xOpenChannelConfiguration ch=0 rc=0
[16:00:40.54] a[04]: Ac49xT38ChannelConfiguration ch=0 rc=0
[16:00:40.54] a[04]: Ac49xActivateRegularRtpChannelConfiguration ch=0 rc=0
[16:00:40.69] y[04]: h225rcv tpkt msg 1 h225cr 7 addr 172.20.0.100 pt 800e7000
[16:00:40.69] y[04]: h225 decode rc 0, q931 msg 1 (3), len 119
[16:00:40.69] y[04]: h225rcv faststart <E4>
[16:00:40.69] y[04]: h225rcv alert h225cr 7 FS:1(E,172.20.0.100,29000) TUNN:1 H245:0(0,0)
[16:00:40.69] i[04]: rtp start cr 7 ch 1 li 1 ri 1 st 2 fx 0 cp E txm 1
[16:00:40.69] s[02]: 02 81 30 5e 08 01 81 01 1e 02 82 88
[16:00:40.70] a[04]: ac49x start2 ch=0 remote=ac100064:29000 rc=0
[16:00:40.77] a[04]: ac49x ch 0 rtcpR 0 ji -1 rt -1 fl 65535 in 0 out -1
[16:00:40.77] a[04]: ac49x ch 0 establish
[16:00:40.88] a[04]: ac49x ch 0 jitter buffer 35 1 0 1 0 0 0 0
[16:00:40.91] y[04]: h225rcv tpkt msg 7 h225cr 7 addr 172.20.0.100 pt 800e7000
[16:00:40.91] y[04]: h225 decode rc 0, q931 msg 7 (2), len 77
[16:00:40.91] y[04]: h225rcv connect h225cr 7 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:40.92] i[04]: pstnsnd connect cr 7
[16:00:40.92] s[02]: 02 81 32 5e 08 01 81 07 29 05 06 03 18 10 00
[16:00:41.91] a[04]: ac49x ch 0 jitter buffer 85 4 0 4 2 0 0 0
[16:00:41.95] a[04]: ac49x ch 0 rtcpT 1 ji 195 rt 0 fl 0 in 272 out 1340
[16:00:41.95] a[04]: ac49x ch 0: in 940 out 7926
[16:00:43.15] a[04]: ac49x ch 0 fax/data 0 0 1
[16:00:43.15] a[04]: ac49x ch 0 fax/data 0 0 0
[16:00:43.30] a[04]: Ac49xActivateFaxRelayCommand(1) ch 0 rc 0
[16:00:43.30] a[04]: ac49x ch 0 fax detected(1)
[16:00:43.30] i[04]: vpinfo fax detected cr 7 ch 1
[16:00:43.30] i[04]: h245snd(1) cr 7 TerminalCapabilitySet <EG>
[16:00:43.30] y[04]: h225rcv tpkt msg 62 h225cr 7 addr 172.20.0.100 pt 800e7000
[16:00:43.30] y[04]: h225 decode rc 0, q931 msg 62 (6), len 63
[16:00:43.30] y[04]: h225rcv facility h225cr 7 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:43.30] i[04]: h245rcv(1) cr 7 TerminalCapabilitySetAck
[16:00:43.33] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 1 2 0 0 0 0
[16:00:43.50] y[04]: h225rcv tpkt msg 62 h225cr 7 addr 172.20.0.100 pt 800e7000
[16:00:43.50] y[04]: h225 decode rc 0, q931 msg 62 (6), len 147
[16:00:43.50] y[04]: h225rcv facility h225cr 7 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:43.50] i[04]: h245rcv(1) cr 7 TerminalCapabilitySet peer=<EG> cfg=<EG>
[16:00:43.50] i[04]: h245snd(1) cr 7 TerminalCapabilitySetAck
[16:00:43.50] i[04]: h245snd(1) cr 7 RequestModeT38
[16:00:43.68] y[04]: h225rcv tpkt msg 62 h225cr 7 addr 172.20.0.100 pt 800e7000
[16:00:43.68] y[04]: h225 decode rc 0, q931 msg 62 (6), len 64
[16:00:43.68] y[04]: h225rcv facility h225cr 7 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:43.68] i[04]: h245rcv(1) cr 7 RequestModeAck
[16:00:43.68] i[04]: h245snd(1) cr 7 CloseLogicalChannel cn=1
[16:00:43.68] i[04]: h245snd(1) cr 7 OpenLogicalChannel cn=1 cpr=G sessid=1 ctrl=172.20.0.200:29001
[16:00:43.69] y[04]: h225rcv tpkt msg 62 h225cr 7 addr 172.20.0.100 pt 800e7000
[16:00:43.69] y[04]: h225 decode rc 0, q931 msg 62 (6), len 68
[16:00:43.69] y[04]: h225rcv facility h225cr 7 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:43.69] i[04]: h245rcv(1) cr 7 CloseLogicalChannel cn=1 (1)
[16:00:43.69] i[04]: h245snd(1) cr 7 CloseLogicalChannelAck cn=1
[16:00:43.69] y[04]: h225rcv tpkt msg 62 h225cr 7 addr 172.20.0.100 pt 800e7000
[16:00:43.72] y[04]: h225 decode rc 0, q931 msg 62 (6), len 92
[16:00:43.72] y[04]: h225rcv facility h225cr 7 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:43.72] i[04]: h245rcv(1) cr 7 OpenLogicalChannel cn=1 cpr=G sessid=1 ctrl=172.20.0.100:29001
[16:00:43.72] i[04]: h245snd(1) cr 7 OpenLogicalChannelAck cn=1 sessid=1 media=172.20.0.200:29000
```

```
[16:00:43.72] y[04]: h225rcv tpkt msg 62 h225cr 7 addr 172.20.0.100 pt 800e7000
[16:00:43.72] y[04]: h225 decode rc 0, q931 msg 62 (6), len 64
[16:00:43.72] y[04]: h225rcv facility h225cr 7 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:43.72] i[04]: h245rcv(1) cr 7 CloseLogicalChannelAck cn=1
[16:00:43.72] y[04]: h225rcv tpkt msg 62 h225cr 7 addr 172.20.0.100 pt 800e7000
[16:00:43.72] y[04]: h225 decode rc 0, q931 msg 62 (6), len 83
[16:00:43.72] y[04]: h225rcv facility h225cr 7 FS:0(-,0,0) TUNN:1 H245:0(0,0)
[16:00:43.72] i[04]: h245rcv(1) cr 7 OpenLogicalChannelAck cn=1 sessid=1 media=172.20.0.100:29000
[16:00:43.72] i[04]: rtp start cr 7 ch 1 li 1 ri 1 st 3 fx 0 cp G txm 1
[16:00:43.72] i[04]: rtp start cr 7 ch 1 li 1 ri 1 st 3 fx 1 cp G txm 1
[16:00:43.79] a[04]: ac49x start2 ch=0 remote=ac100064:29000 rc=0
[16:00:43.79] a[04]: ac49x start fax ch=0 doing fax already
[16:00:46.70] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 1 3 0 0 0 0
[16:00:48.95] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 1 8 0 0 0 0
[16:00:49.60] a[04]: ac49x ch 0 fax/data 0 0 6
[16:00:49.60] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 0 3 0 0 0 0
[16:00:51.53] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 0 12 0 0 0 0
[16:00:51.65] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 0 20 0 0 0 4
[16:00:52.94] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 0 20 0 0 0 4
[16:00:54.25] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 1 3 0 0 0 0
[16:00:55.73] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 1 16 0 0 0 0
[16:00:56.44] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 0 21 0 0 0 4
...
[16:01:25.93] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 0 21 0 0 0 4
[16:01:27.13] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 0 3 0 0 0 0
[16:01:28.26] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 0 18 0 0 0 0
[16:01:29.05] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 1 3 0 0 0 0
[16:01:30.56] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 1 17 1 0 0 0
[16:01:31.62] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 0 3 1 0 0 0
[16:01:32.72] a[04]: ac49x ch 0 faxrelay 0 0 0 0 0 0 0 0 0 15 1 0 0 0
[16:01:33.13] r[02]: 00 81 5e 34 08 01 01 45 08 02 80 90
[16:01:33.13] i[04]: pstnrcv terminate connection (3201) cr 7 cau 90 err 0 state 16 ch 1 rsid 1
[16:01:33.13] i[04]: rtp stop cr 7 ch 1
[16:01:33.13] x[04]: h225snd relack cr 7 cau 0x90
[16:01:33.13] i[04]: h225connection s 4 close
[16:01:33.13] i[04]: CloseSysFd 4 (st 22)
[16:01:33.16] s[02]: 02 81 34 60 08 01 81 4d
[16:01:33.16] a[04]: ac49x ch 0: in 5714 out 99508
[16:01:33.16] a[04]: ac49x stop ch=0
[16:01:33.21] a[04]: ac49x ch 0 rtcpR 2 ji 234 rt 15139031 fl 0 in 15139047 out 2228
[16:01:33.22] r[02]: 00 81 60 36 08 01 01 5a
```

7.4.4 REMOTE OUTPUT

This trace option provides output for communication with the TELES.GATE Manager or TELES.NMS. To activate this option, activate the section **Remote** in the TELES.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 TELES.GATE Manager connection:

```
g[99]:moip: accept rc=2 ipad=<ip address> port=<port>
```

Table 7.82 Remote Output

Trace Output	Description
`<ip address>`	Remote system's IP address with TELES.GATE Manager.
`<port>`	Origination port for the TELES.GATE Manager connection.

```
g[99]:moip: <direction> <length>
```

Table 7.83 Remote Output

Trace Output	Description
<direction>	recv — Packets received from the remote system send — Packets sent to the remote system write — Output for communication with the internal remote interface read — 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.

7.4.5 SMTP TRACE OUTPUT

This trace option provides output for communication with the mail server that occurs when status information or files are sent.

To activate this option, activate the section **Mail** in the TELES.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

Global message output:

```
m[99]:mail: sendmail (<length>)
```

Table 7.84 SMTP Output: Sending Files or Status Info

Trace Output	Description
<length>	Data length in bytes.

Detailed message output:

```
m[99]:mail: sendmail: <Faccount> <ip address> <Taccount> <domain> <subject> <content>
```

Table 7.85 SMTP Output: Sending Files or Status Info

Trace Output	Description
<Faccount>	Sender's e-mail account (cdr, alarm, file, etc.).
`<ip address>`	SMTP server's IP address.
`<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 7.86 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 7.87 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 7.88 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 7.89 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 "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>)
```

Table 7.90 SMTP Output: Transmission Error

Trace Output	Description
<num>	Current number of retries.

Retried message transmission was also faulty, and an e-mail will be generated:

```
m[99]:mail: rcvmail <Faccount> -> <Taccount>, failed <num> times
```

The output in the subsequent communication with the mail server are identical to those described above in "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 "Sending Files or Status Information".

7.4.6 NUMBER PORTABILITY TRACE OUTPUT

This trace option provides output for the communication with the TELES.iMNP database. To activate this option, activate the section **Number Portability** in the TELES.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 TELES.iMNP is being set up:

```
n[99]:np: connecting to <ip addr>
```

Table 7.91 Number Portability Output: Connection with TELES.iMNP

Trace Output	Description
`<ip address>`	The TELES.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 TELES.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 7.92 Number Portability Output: Response

Trace Output	Description
`<num>`	Ported or unported number provided by the database.

7.4.7 DTMF TONE TRACE OUTPUT

Output about the setup of connections with the DTMF module and DTMF tone recognition 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 7.93 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 7.94 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 recognition:

```
d[<ctrl>]: dtmf send_alloc <b_chan id_unset> <ctrl>/<b chan>
```

Table 7.95 DTMF Output: Media Channel Designation

Trace Output	Description
<b chan>	Internal media channel used.
`<b_chan id_unset>`	Media channel identification (in unset state).

```
d[<ctrl>]: dtmf: msg <msg>, id <b_chan id>, from 1, id <id>/<b_chan id_unset>
```

Table 7.96 DTMF Output: Media Channel Designation

Trace Output	Description	
<msg>	502	Media channel confirmation
	102	Connect confirmation
	602	Media channel free confirmation

The following output shows the output for negotiated DTMF tones:

```
d[<ctrl>]: dtmf send_info_ind <id> <<dtmf tone>>
```

8 SIGNALING AND ROUTING FEATURES

8.1 DIGIT COLLECTION (ENBLOCK/OVERLAP RECEIVING)

This function makes it possible to collect 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 SIP gateway. Digit collection occurs through the following mapping command:

```
...
MapAll01555=|40iG:01555<<16
...
DTMFWaitDial=5
...
```

The example above shows a call with the prefix 01555. The **|** (pipe) signifies that the following digits will be collected before they are transmitted. The 16 at the end is the sum of the port digits, the digits of the called party number, and the optional VoIP profile, as (e.g. |#40iG:=5, 01555899666=11, 5+11=16). This figure must be entered in double digits (e.g. `<<08` for 8 digits). The parameter `DTMFWaitDial` defines the amount of time the system waits between the individual digits. The system forwards the call to the VoIP profile as soon as the correct number of digits has been dialed. Please bear in mind that you can configure a maximum of 11 digits on the left of the equal sign and 19 (including a #, which activates calling number suppression) on the right. The call will be forwarded as soon as the specified number of digits has been dialed or a time-out limit has been reached.

8.2 REJECTING DATA CALLS AND SPECIFIED NUMBERS

The following configuration enables you to reject calls with bearer capability data and calls that should not be directed to specified numbers. In both cases the cause value **Switching Equipment Congestion** (aa) is used:

```
...
MapAll00491555=&aa DATA
...
MapAll004915551234=&aa VOICE
```

8.3 CLIP AND CLIR

8.3.1 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 (e.g. analog) are transmitted through the TELES.VoIPBOX GSM/CDMA.

Use the following configuration to define the various routing methods:

```
...
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.

8.3.2 ROUTING CALLS WITHOUT CLIR

This function enables you to bypass CLIR for calls through the defined mobile port. The following configuration in pabx.cfg activates this function:

```
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.**

8.3.3 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 sytax is used: MapAll<num>=#<port><num>

Example: 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:

```
MapAll00491555=#2200491555
```

8.3.4 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 sytax is used: `MapAll<num>=!<port><num>`

Example: 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.:

```
MapAll004930=!9004930
```

8.4 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` (`MapAll...=9..`):

Example 1

```
...
MapAll015550123456=9015559876543
```

Example 2 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 5000 mapping entries with no more than 11 digits in the first part of the command and 19 in the second.

Example 2

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

8.5 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 8.97 Number Types

Type	Definition
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`

Example: In the following example, the bit is set in the caller's origination number for a call via analog controller 01:

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

Example:

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:

```
Restrict10555=u030555 15
```

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

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:

Example: 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`:

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

8.6 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`

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

```
Restrict10=v 15
```

Network provided: `p`

Example: In the following `Restrict` example, the calling party number sent is specified as `network provided`:

```
Restrict10=p 15
```

If you also want to define a number type (see Chapter 8.5 ⇨), it must appear in front of the screening indicator:

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

```
Restrict10=ip 15
```

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

8.7 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`

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

```
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.

Example: 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.

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

8.8 SETTING SENDING COMPLETE BYTE IN SETUP

In some cases the 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 H323 Setup.

Example: In the following example, all calls beginning with 0 are sent with a Setup Complete to controller 9:

```
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.

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

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

8.9 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

8.9.1 ROUTING CALLS WITHOUT A DESTINATION NUMBER

Enter the following configuration in the `route.cfg` if the TELES.VoIPBOX GSM/CDMA 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.

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

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

8.9.2 ROUTING CALLS BASED ON EXISTENCE OF DESTINATION NUMBER

To route calls with a DAD differently from those with 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>`

Example: 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 num-ber 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:

```
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
```

8.9.3 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 PBX sending the call to the TELES.VoIPBOX GSM/CDMA. 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 paramter 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)

8.10 CALLER ID

Caller ID (caller identification) is a feature that transmits the calling party number to the called party's telephone when it rings unless the calling party has restricted number transmission. Caller ID is generated on FXS ports according to the V.23 standard. Bear in mind that the called party must have a telephone or device that displays the calling number in order for this feature to work.

The current firmware version supports the caller ID feature on FXS ports in the following cases:

- Calls from VoIP to FXS using the SIP protocol
- Calls from VoIP to FXS using the H.323 protocol
- Calls from GSM to FXS

9 MOBILE CONFIGURATION OPTIONS

Be sure to save a backup copy of the configuration files before making changes. Changing configuration data and/or SIM-card positions may lead to malfunctions and/or misrouting, as well as possible consequential damages. Make changes at your own risk. TELES is not liable for any damages resulting from or related to such changes. Therefore, please thoroughly check any configuration changes you or a third party have made.

9.1 CONNECTION TO A TELES.VGATE

The TELES.vGATE is a system that enables more convenient management of a network of TELES.VoIPBOX GSM/CDMA systems. All SIM cards in the network are installed in and maintained at a central server, so that it is no longer necessary to install SIM cards into each gateway. The TELES.VoIPBOX GSM/CDMAs connected to the TELES.vGATE do not require SIM-card carriers, as the TELES.vGATE contains SIM-card carriers for the entire network.

Bear in mind that no SIM-card carriers are to be inserted in TELES.VoIPBOX GSM/CDMAs connected to a TELES.vGATE.

The following parameters must be configured in the `pabx.cfg` of each TELES.VoIPBOX GSM/CDMA connected to the TELES.vGATE. After the parameters have been entered, you must restart the TELES.VoIPBOX GSM/CDMA to activate the changes:

SIMS

Enter this keyword in the `Subscriber` lines of the mobile controllers to connect the system to a TELES.vGATE.

EXAMPLE:

```
Subscriber04=TRANSPARENT ROUTER GSM[0000,00000,+00000,1,1,1,SIMS] CHADDR ALARM
```

SimCtrlUnitAddress=<ip addr>

Enter the TELES.vGATE Control Unit's IP address. Set this parameter in the IP configuration section.

EXAMPLE:

```
SimCtrlUnitAddress=192.168.0.1
```

9.2 MODULE DISTRIBUTION OF VARIOUS MOBILE NETWORKS

You can assign each mobile port in the TELES.VoIPBOX GSM/CDMA system either one mobile network or different access groups to different mobile networks. The port numbers in the TELES.VoIPBOX GSM/CDMA must be the same for the individual groups.

The keyword `NEXT` ensures equal distribution of calls.

The following configuration samples (from the `pabx.cfg` configuration file) show the changes:

Example 1 All ports in the following example must have the same number for all mobile channels to route calls to the same mobile network. The `subscriber` line of the first port must also contain the keyword NEXT to ensure the equal distribution of calls.

```
...
Controller04=20 GSM
Controller05=20 GSM
Controller06=20 GSM
Controller07=20 GSM
...
Subscriber04=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM4] ALARM NEXT
Subscriber05=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM4] ALARM
Subscriber06=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM4] ALARM
Subscriber07=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM4] ALARM
...
```

Example 2 In the following example, a group of 4 mobile channels is assigned to two different mobile networks. The `subscriber` line of the first port in each group must contain the keyword NEXT to ensure the equal distribution of calls.

```
...
Controller04=20 GSM
Controller05=20 GSM
Controller06=22 GSM
Controller07=22 GSM
...
Subscriber04=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM4] ALARM NEXT
Subscriber05=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM4] ALARM
Subscriber06=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM4] ALARM NEXT
Subscriber07=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM4] ALARM
...
```

9.3 NETWORK-SPECIFIC MOBILE ROUTING

9.3.1 USING A FIXED MOBILE PORT ADDRESS

The customer's network routes calls to the mobile network with a defined prefix. Because this code is not always uniform, the TELES.VoIPBOX GSM/CDMA might have to convert it. Conversion requires the following information:

- The mobile network's access number
- Destination number format (with or without prefixes, national or international)

conversion occurs according to the following formula:

```
<national/international><prefix[incoming]>=<destination number>
```

converted as:

```
<port><mobile network access number><destination number>
```

Be sure to make the following entry in the `pabx.cfg` configuration file to configure this conversion:

```
MapAll<nat./int.>prefix[incoming]>=<port><mobile network access number>
```

The TELES.VoIPBOX GSM/CDMA system converts *national* into one zero and *international* into two zeros.

The following configuration exemplifies this conversion as it might occur in Germany:

```
...
MapAll00491555=2001555
MapAll00491556=2101556
...
```

This example shows how the customer's network provides the prefix `international+49+1555+destination number` for one mobile network, and `international+49+1556` for the other. The configuration entries see to it that `00491555+destination number` is converted to `2001555+destination number` and `00491556+destination number` is converted to `2101556+destination number`. The calls to the carrier with prefix 01555 are routed to ports with the number 20 and calls to the carrier with prefix 01556 are routed to the ports with the number 21.

9.3.2 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`:

pabx.cfg

```
...
Controller04=20 GSM
Controller05=20 GSM
Controller06=20 GSM
Controller07=20 GSM
...
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.

route.cfg

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

9.3.3 FIXED LAIN FOR A MOBILE PORT

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

Example:

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

MapAllSMS=444 05
```

9.4 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 an FXS port:

```
Restrict20=30 01
Restrict21=31 01
```

The mobile controllers can also have the same identifier, so that all voice calls (service indicator 01) from controller 20 are sent to port 30. This number could, for example, serve a call center.

```
Restrict20=30 01
```

9.5 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 block a port (i.e. a mobile channel), enter the keyword `CHINC[...]` in the `Subscriber` line.

In Example 1, port 04 is blocked.

Example 1

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

To activate the port, remove the entry and enter **Activate configuration**.

It is also possible to block an entire port with the remote administration program TELES.GATE Manager. The CHINC[...] function is not necessary with this application. The port's status is displayed by remote administration. You can remove the block with **Activate a configuration** or with the **Unblock** option.

9.6 SETTING LIMITS

This function enables you to monitor time limits. A limit can be set, either to the defined time interval or in 10-second intervals (default value). When this value is reached, the current connection is torn down either immediately or when the call has been terminated; no more connections will be set up. An alarm also goes off and an entry is generated in the log file.

Setting SIM Time Limits

In the following example, the mobile channel shuts down when 6,000 intervals of 10 seconds each have passed:

```
...
Subscriber04=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM4] CHADDR ALARM NEXT LIMIT[6000]
...
```

Setting the Default Time Window

In the following example, the mobile channel shuts down when a calculated number of seconds has been reached. If ChargeUnitGenerate=<sec> is configured, the mobile channel will shut down when the value entered in LIMIT multiplied by the value entered here is reached. The default value for this parameter is 10 seconds. Bear in mind that the value entered in LIMIT may not exceed 65535. In the example, the mobile channel will shut down at 60,000 seconds.

```
...
ChargeUnitGenerate=1
...
Subscriber04=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM4] CHADDR ALARM NEXT LIMIT[60000]...
```

Setting Start Units

The following example shows how you can use the keyword ChargeUnitFirst=<seconds> to configure a starting unit for LIMIT. The following entries are used for SIM card tariffs where the first defined number of seconds are always charged (e.g. the first minute), followed by charges calculated in defined intervals (e.g. every 10 seconds):

```
...
ChargeUnitFirst=60
ChargeUnitGenerate=10
...
Subscriber04=TRANSPARENT ROUTER GSM[0000,00000,<SMSC>,1,1,1,SIM24] LIMIT[6000] ALARM NEXT...
...
```

Information about the active SIM cards can be found in TELES.GATE Manager and **Port Status**.

To remove limits from the configuration, follow these steps:

- Enter LIMIT[-].
- Activate the configuration.
- Delete the entry LIMIT[-].
- Activate the configuration again.

9.6.1 COUNT STATUS INFORMATION

To find information about the current counts and limits for each mobile channel, click **Statistics** in the TELES.GATE Manager.

For mobile ports:

- **Count A-F** For each mobile channel. Bear in mind that only one SIM card per channel is possible (**Count A**). Counts are the time slices (from 0 to the limit value) for the individual SIM cards. When a SIM's set time limit is reached, the channel will disconnect and the SIM will be blocked.
- Click **Reset/Set** Counters in the **Statistics** context menu in TELES.GATE Manager to reset the counters (default value is 0). You can also configure the counters to reset automatically (see Chapter 11.1 ⇨).

If LIMIT is configured, bear in mind that SIM 1 must be configured in the pabx.cfg for the corresponding mobile port.

9.7 DEFINING TIME LIMITS FOR CALLS

By entering the parameter CALL in the Subscriber line, you can terminate calls that reach a defined time limit. For each call the limit is reset at 0. You can define a value anywhere within <limit>-<random>, and you can define a maximum value for <random>.

If this parameter is configured once, it will be set in each configuration file. If you prefer not to use it for all SIMs, set the limit higher than any call is likely to last (e.g. 360,000 seconds).

Example:

```
ChargeUnitGenerate=1;defines the factor for the limit entry (600*1=600sec)
ChargeUnitDivisor=5 ;random value that defines the maximum call duration between 595 and 600secs
...
Subscriber05=TRANSPARENT ROUTER GSM[0000,00000,+000000,1,1,1,SIM24] CALL[600]
```

The following entry is required if the TELES.VoIPBOX GSM/CDMA is used in conjunction with a TELES.vGATE:

```
Subscriber05=TRANSPARENT ROUTER GSM[0000,00000,+000000,1,1,1,SIMS] CALL[600]
```

9.8 PAUSE BETWEEN TWO CALLS

If the parameter `WAIT` appears in the `Subscriber` line, the mobile controller will not be used after a successful connection for a random amount of time between 1 and 30 seconds

Example:

```
Subscriber04=TRANSPARENT ROUTER GSM[0000,00000,+00000,1,1,1,SIM24,IMSI] CHADDR WAIT ALARM
```

9.9 TIME-CONTROLLED SIM LOGOFF

You can define a time at which a mobile channel will log off. Before proceeding, please refer to Chapter 5.2.1.2 ⇨ for basic information.

You must define all time windows you would like to use in the `[System]` section of the `pabx.cfg`, in the sub-section `night configuration`.

The following entry in the configuration file `pabx.cfg` is necessary:

`Night<num>=<time> <day>`

Example: In the following example, two time windows are defined. The standard configuration is active every day from 6:00 a.m. to 8:00 p.m.. The time window `Night1` is active from 8:00 p.m. to 6:00 a.m.

```
;Night configuration
; --------------------
Night1=20:00 11111111
NightResetTime=06:00 11111111
```

To generate a `NightConfiguration` section for SIM logoff, copy the complete `Subscriber` subsection from the `[System]` section after making the appropriate entries in the `[Nightx]` section.

If a dash (`-`) is entered in the SIM-card position, the SIM card will log off automatically. Time-controlled activation of configuration files makes it possible to shut off unneeded SIMs, for example at night.

Example: In the following example, SIM cards log off in the section [`Night1]`. All `Subscriber` lines must be defined.

```
[System]
...
Subscriber04=TRANSPARENT ROUTER GSM[0000,00000,+49555,1,1,1,SIM4] ALARM
Subscriber05=TRANSPARENT ROUTER GSM[0000,00000,+49555,1,1,1,SIM4] ALARM
Subscriber06=TRANSPARENT ROUTER GSM[0000,00000,+49555,1,1,1,SIM4] ALARM
Subscriber07=TRANSPARENT ROUTER GSM[0000,00000,+49555,1,1,1,SIM4] ALARM
...
[Night1]
...
Subscriber04=TRANSPARENT ROUTER GSM[0000,00000,+49555,-,1,1,SIM4] ALARM
Subscriber05=TRANSPARENT ROUTER GSM[0000,00000,+49555,-,1,1,SIM4] ALARM
Subscriber06=TRANSPARENT ROUTER GSM[0000,00000,+49555,-,1,1,SIM4] ALARM
Subscriber07=TRANSPARENT ROUTER GSM[0000,00000,+49555,-,1,1,SIM4] ALARM
...
```

The only difference is in the active SIM card. Activate the configuration after the required files have been copied onto the system.

Bear in mind, that you must also make the appropriate entries in the corresponding `route.cfg` sections. For more information, please refer to Chapter 5.3 ⇨.

9.10 MOBILE-USER PBX CALLBACK

When the TELES.VoIPBOX GSM/CDMA 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:

> **Callback is not possible for VoIP calls.**

`DialBack=<hours>`

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

Example: 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:

```
DialBack=5

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

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

9.11 OPTIONAL MOBILE QUALITY PARAMETERS

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

Table 9.98 Optional Mobile Parameters

GSM=...	
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]`	
`ALERT[<sec>]`	Set this parameter to generate alert signal in the D channel immediately after 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.

Table 9.98 Optional Mobile Parameters *(continued)*

GSM=...	
ANNOUNCE	Set this parameter to define what happens when a recorded announcement is recognized: ▪ 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. ▪ ANNOUNCE[00,<sec>] A timeout for voice recognition is defined in seconds (default value: 120 seconds). After the interval entered has passed, the connection is torn down.
ASR[<limit>, <calls>]	Allows you to change the default value (40 calls at 30% ASR). For a definition of ASR, see Chapter 11.1 ⇨.
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]
STOP[<val1>, <val2>]	This entry allows you to define a maximum number of connection setups that always result in a recorded message (`<val1>`) without a call-connected signal or successful connection setup, or that are always accepted (`<val2>`). 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 recognized and blocked so that they are removed from the routing process. The default status of this function is off.
VOICE[1]	The B-channel is immediately connected (Siemens dial tone is heard).
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]

9.12 DEACTIVATING CLASS 2 RE-ROUTING

Configuring `Class2Next=Off` in the `pabx.cfg` file ensures that calls rejected with a class 2 cause value are not re-routed to the next available port.

9.13 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>]`

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

9.14 RECHARGING PREPAID SIMS

Prepaid SIM cards are an alternative to mobile telephone SIMs with a contract. Instead of being billed retroactively, prepaid SIMs are paid for in advance and then recharged when they run out.

The advantages of prepaid SIMs are:

- No monthly basic fee
- Cost control
- No surprises resulting from unexpectedly high mobile telephone bills

When the account is empty, it can be recharged. The recharging methods for prepaid SIM cards of different carriers vary:

- Recharging via SMS/USSD
- Recharging via call to a defined number
- Recharging via DTMF
- Automatic recharging via direct debit

RECHARGING PREPAID SIMS

When prepaid SIMs that do not recharge automatically (e.g. through a credit card) are used in a TELES.VoIPBOX GSM/CDMA, it is possible to recharge them directly from the system. The following requirements apply:

- The SIM is registered and no connection is active.
- Exact knowledge of the mobile carrier-specific recharging procedure exists.
- One valid prepaid voucher exists for one recharge.

!	**Transmission errors, truncated connections, incorrect or altered recharging procedures can prevent successful recharging. Please bear in mind that 3 incorrect recharge attempts (per SIM) can result in blocked SIMs. Recharge SIMs at your own risk. TELES is not liable for any possible loss.**

TELES.VoIPBOX GSM/CDMAs support of the following procedures:

- USSD message to the mobile carrier's account manager
  The TELES.GATE Manager sends the configured USSD message through the TELES.VoIPBOX GSM/CDMA to the account manager. USSD recharging is the recommended and most reliable procedure, as it consists of a digital message. Unfortunately, only a few mobile carriers currently support USSD recharging. Please ask your mobile carrier if he supports USSD recharging.
- SMS message to the mobile carrier's account manager
  The TELES.GATE Manager sends the configured SMS containing the voucher number through the TELES.VoIPBOX GSM/CDMA to the account manager. Unfortunately, only a few mobile carriers currently support SMS recharging. Please ask your mobile carrier if he supports SMS recharging.
- Connection setup to the account manager with subsequent menu selection and DTMF-tone transmission of the voucher number
  - Direct recharging: Connection setup from a telephone through the TELES.VoIPBOX GSM/CDMA to the account manager and manual DTMF-tone transmission.
  - Indirect recharging using the TELES.GATE Manager: The TELES.GATE Manager sets up a connections through the TELES.VoIPBOX GSM/CDMA to the account manager and sends the configured DTMF tones automatically.

i	**Direct recharging is the simplest procedure. Since a direct connection exists, it is possible to react to commands and error messages immediately. Indirect recharging by means of USSD is the most reliable and quickest way to recharge SIMs if the configuration in the TELES.GATE Manager and TELES.VoIPBOX GSM/CDMA is correct.**

9.14.1 RECHARGE PREPARATION

9.14.1.1 CHECKING THE ACTIVE SIM

To avoid recharging the wrong SIM card, be sure to check the mobile controller's active SIM using the TELES.GATE Manager:

TELES.GATE Manager

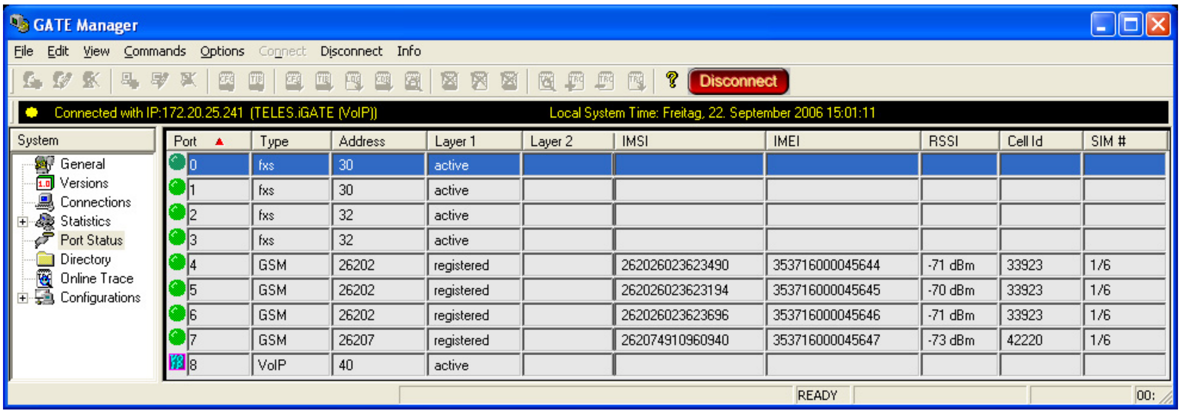


Figure 9.10 TELES.GATE Manager Port Status

Connect to the system and go to the **Port Status** window. The active position in the SIM-card carrier is displayed in the **SIM #** column. The mobile controller's active SIM is displayed in the **IMSI** column.

9.14.1.2 ADDRESSING SIMS USING PORT- AND CONTROLLER-SPECIFIC ROUTING

SIM recharging for a specific mobile controller requires configuration and activation of port- and controller-specific routing entries in the route.cfg or pabx.cfg configuration file. Usually SIM cards are assigned to a carrier's trunk group and all calls through the carrier's network are evenly divided between the mobile controllers in the group. This would also apply to recharge calls.

The routing entry defined here sets up a connection to the network:

`MapAll<in>=<port>*<ctrl>01:<num>`

When the number `<in>` is dialed, a connection to `<num>` is set up through `<port>*<ctrl>01:` You can now manually enter DTMF tones using a telephone.

To recharge all SIMs in the TELES.VoIPBOX GSM/CDMA, configure the following mapping, whereby 4400 is an example for a number that matches the first controller and 12345 is the number for the account manager:

```
MapAll4400=20*0001:12345
MapAll4401=20*0101:12345
MapAll4402=20*0201:12345
MapAll4403=20*0301:12345
MapAll4404=20*0401:12345
```

9.14.1.3 BLOCKING THE PORT CONTAINING THE RECHARGING SIM

If a call is active on the mobile port containing the SIM to be recharged, the recharging process will not occur. For this reason it is better to block the port before recharging the SIM. In the Connections window, you must check the status No Connection on the mobile port. Block Port does not tear down a connection, it only prevents a new connection from being set up.

Port- and controller-specific routing has a higher priority than the Block Port command. This ensures that normal calls are blocked, but recharge calls can be sent through the defined mobile port.

9.14.2 RECHARGING PROCEDURE

9.14.2.1 DIRECT RECHARGING VIA CALL

This is the easiest method when it is possible to set up a telephone connection to the TELES.VoIPBOX GSM/CDMA system via PSTN or VoIP. This call can be connected with the carrier's account manager over a defined mobile controller. This means the call is set up over the controller's active SIM. Then you simply follow the account manager's recharge instructions. After the SIM has been successfully recharged, it can be used again for a certain amount of time.

The call can be set up using a number of methods. This is also possible if there are not enough available telephone numbers to handle all of the system's available mobile controllers.

- DLA via DTMF
  The user calls a defined number in the system. The called number is connected with the DTMF platform. The digits that are transmitted via DTMF match those in the routing entries. When the connection to the account manager has been established, both legs will be connected (See Chapter 9.14.1.2 ➪).
- TELES.GATE Manager (described in Chapter 9.14.2.2 ➪ below)

That means no BRI connection is necessary for a telephone that is connected directly to the system!

9.14.2.2 INDIRECT RECHARGING VIA TELES.GATE MANAGER

This chapter describes automatic recharging of prepaid SIMs using the TELES.GATE Manager. It is not necessary to set up a telephone connection to the TELES.VoIPBOX GSM/CDMA. The TELES.GATE Manager can set up its own connection to the carrier's account manager and send the pattern of DTMF tones or a USSD message.

Recharging via Call and Transmission of Preconfigured DTMF Tones

This procedure requires exact knowledge of the when and what information the mobile carrier requests. The corresponding pauses following the connect, for menu selection, between the DTMF tones, for correct repetition of the DTMF tones must be correctly configured in the TELES.GATE Manager before the call is set up.

This DTMF-tone pattern can be established by testing the recharging process on a mobile phone or by following the directions in Chapter 2.1 and noting the pauses and transmitted digits.

After a connection has been set up between the TELES.VoIPBOX GSM/CDMA and the TELES.GATE Manager, select **Commands | Send Call.** The window must contain either **General**, **Version** or **Directory**.

Send Call opens a dialog to initiate calls or recharge SIMs.

To recharge SIMs, the **1st Number** (e.g. 12345) is the mapping to the prepaid platform through a defined controller (e.g. `MapAll12345=20*0101:12345`).

Optional: You can set up a second connection to hear the announcement from the prepaid platform if you enter your own number into the box **2nd Number**. Both connections will be torn down when the second number disconnects. If no number is entered in this box, the call will disconnect when the last DTMF tone has been transmitted or when the last pause interval has passed.

Activate the checkbox **Advanced** to open the **DTMF** box

Enter a series of DTMF tones in the **DTMF** box. Enter a **p** for a pause of 1 second and a **P** for a pause of 10 seconds.

Example: The number for the prepaid platform's account manager is 12345. The telephone number to listen along to the accounting procedure is 5554321, set up through controller 9 (no special routing configuration is defined in the configuration files). Leave this dialog box empty if the accounting process is not to be monitored.
The voucher key is: 55555555555 (a short pause can also be defined between individual digits: for example, 5p5p5p5p5p5p5p5p5p5p5).
To get to the voucher key query, the following pattern must be transmitted: P2ppppp1ppp. Following transmission of the voucher key and a 10-second pause, the call will be torn down.

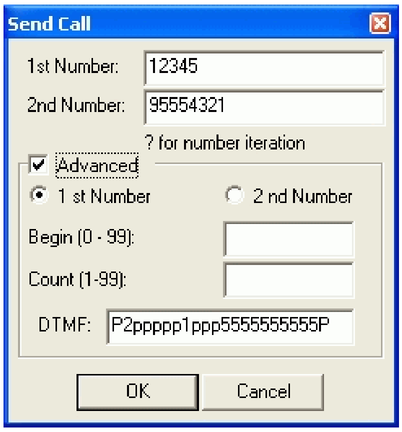


Figure 9.11 Recharging with DTMF Tones

Bear in mind that DTMF tones are only generated with connections into the mobile network. Test calls over the PRI, BRI or VoIP interfaces do not transmit DTMF tones and no tones can be heard!

Recharging via USSD Code (Unstructured Supplementary Services Data)

Recharging prepaid SIMs using the TELES.GATE Manager and USSD is the most convenient solution if the prepaid carrier offers this service. The USSD messages contains the prepaid voucher number.

Configure an additional controller in the last position for DTMF functionality as follows:

Example:

```
Controller36=41DTMF
```

The corresponding `Subscriber` line will look like this:

```
Subscriber36=TRANSPARENT ROUTER CHMAX[5]
```

The configuration file pabx`.cfg` must contain the following entry in the `[System]` section:

`MapAllDTMF=<dtmf port>DTMF`

`MapAll<place>??=<port>*??01:`

or

`MapAll<place>??=<LAIN>*??01:`

First a placeholder is defined, followed by `??` so that one mapping entry applies for the entire group of the carrier's mobile controllers. The right side of the mapping entry begins with the mobile port number or the port's LAIN.

Example: In the following example, prepaid SIMs from 2 different carriers are used in the system. The letters Y and Z are used as placeholders, and the carrier's LAINs are 26212 and 26213 (based on the German country code):

```
MapAllDTMF=41DTMF
MapAllY??=26212*??01:
MapAllZ??=26213*??01:
```

Set up a connection to the system through the TELES.GATE Manager and select **Commands | Send Call**. Enter DTMF in the **1st Number** dialog box. In the **2nd Number** dialog box, enter the carrier's placeholder (Y or Z) and the number of the controller in which the SIM card is active (15 or 05). Enter a 0 in front of single-digit controller numbers. Then enter the carrier's USSD code (*101*) and the voucher number (44444444444 or 55555555555). The USSD command ends with #.

Incorrect USSD commands can result in blocked SIMs or failure in the mobile module!

Configuration entries for recharging confirmation are described in Chapter 3.

Recharging via SMS (Short Message Service)

First of all, please check whether you have the license to send SMS messages on your system.

You will find it in the **General** view under **Licenses** when you connect to the system via TELES.GATE Manager.

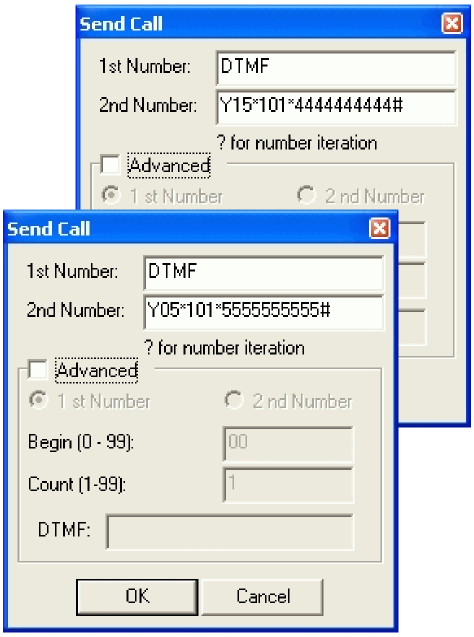


Figure 9.12 Recharging with USSD Codes

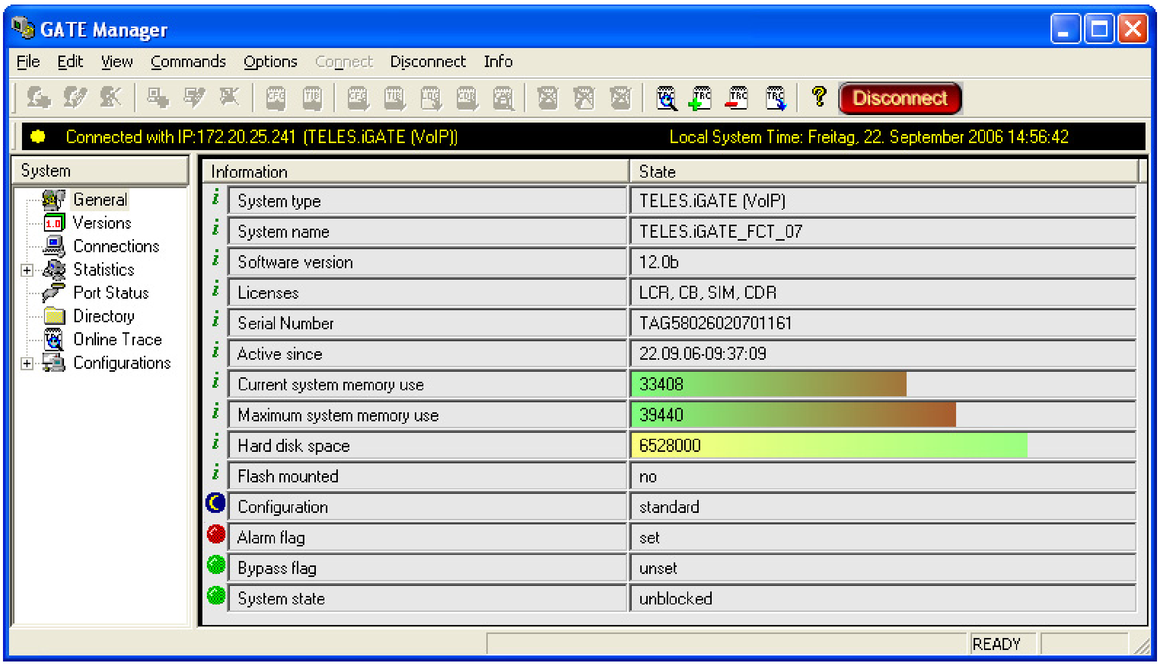


Figure 9.13 TELES.GATE Manager General View

The name for the license is SMS. This entry is required to send SMS.

You must configure the mail service in the `[Mail]` section of the file `pabx.cfg` if you want to send the SMS messages with an e-mail client through a mail server or directly to the TELES.VoIPBOX GSM/CDMA:

`[Mail]`

`SmtpServer=<server addr>`

MailRcpt=<domain>

MailFrom=<own address or name>

This entry is not necessary when using only the GateManager's Send SMS command.

The third entry in the mobile controller's `Subscriber` line is the SMS center number:

```
Subscriber00=TRANSPARENT ROUTER GSM[0000,00000,+00000,1,1,1,SIMS,IMSI] CHADDR ALARM
```

German example:

```
Subscriber00=TRANSPARENT ROUTER GSM[0000,00000,+491721111,1,1,1,SIMS,IMSI] CHADDR ALARM
```

You must restart the system to activate the changes.

Then enter the port-specific SMS settings:

```
MapAllSMS<shortnumber>=LAIN*0001:<number>
MapAllSMS<shortnumber>=port*0001:<number>
```

Example: In the following German example with LAIN numbers, the short number 00 is routed to the first controller. The number 12345 is sent to the SMS center.

```
MapAllSMS00=26212*0001:12345
MapAllSMS01=26212*0101:12345
MapAllSMS02=26212*0201:12345
MapAllSMS03=26212*0301:12345
MapAllSMS04=26212*0401:12345
MapAllSMS05=26212*0501:12345
MapAllSMS06=26212*0601:12345
……
```

If the SMS is sent with a normal e-mail client, the keyword **SMS** will appear in the **To** dialog box, followed by the short number, which indicates the mobile controller. An **@** sign and the TELES.VoIPBOX GSM/CDMA's IP address or name if the system is attached to a DNS server will follow. The message box contains the recharge code in the carrier's syntax.

```
SMS00@<ipaddr>
SMS00@<domain>
```

If the SMS is used with the Send SMS command in the TELES.GATE Manager's **Commands** menu, use only the short number and not the keyword SMS.

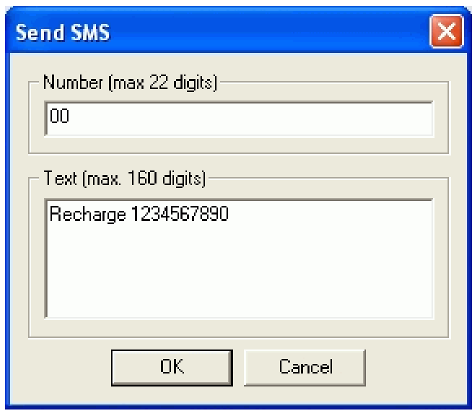


Figure 9.14 Send SMS

To save the recharge platform's confirmation e-mail, the e-mail can be sent to an account using the following entry in the `route.cfg`:

```
Restrict<port>=@<addressee> 05
```

It can also be saved into a file using the following entry in the `pabx.cfg`:

```
MsgLog=/data/msg.log
```

To save it into a file, the following entry in the `route.cfg` is also required:

```
Restrict<port>=@FILE 05
```

Example: In the following example for saving the SMS into a file, all incoming SMS to LAIN 26212 is saved into the file `msg.log`:

```
Restrict26212=@FILE 05
```

If the e-mail is sent to an account, the routing entry will look like this:

```
Restrict26212=@nase 05
```

Use the following entries in the `pabx.cfg` to connect the TELES.VoIPBOX GSM/CDMA to an e-mail server:

```
[Mail]
SmtpServer=<server address>
MailRcpt=<domain>
MailFrom=<own address or name>
```

9.14.3 PREPAID ACCOUNT STATUS QUERY

After a SIM card has successfully been recharged, you can query its current account status. The following variations are possible:

- Direct query via call
- Indirect query with the TELES.GATE Manager
- Listening in on the TELES.GATE Manager connection
- USSD account-status query

9.14.3.1 DIRECT ACCOUNT-STATUS QUERY

The same basic settings apply here as have already been described for SIM recharging.

That means routing configurations must be entered that set up a connection between the caller and the carrier's prepaid platform. Use the same routing configuration described in Chapter 9.14.2 ⇨. The only difference is that you will select the account-status query instead of account recharging from the menu.

9.14.3.2 INDIRECT ACCOUNT-STATUS QUERY

The indirect method with the TELES.GATE Manager and listening in on the connection requires a change in the pattern of DTMF tones. You must, of course, know what the pattern is beforehand, and then you must configure it in the **Send Call** dialog.

The indirect USSD account-status query corresponds with the USSD recharging procedure with altered USSD account-status-query command instead of the recharging command with cash code (voucher number).

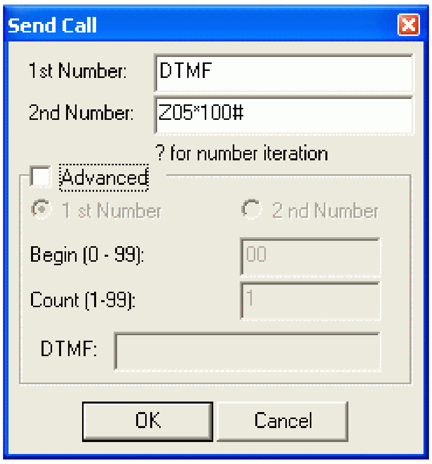


Figure 9.15 Status Query

i	**The USSD recharging procedure results in an immediate USSD response message, so that the TELES.VoIPBOX GSM/CDMA does not require an explicit query following USSD recharging.**

9.14.3.3 SAVING /FORWARDING THE ACCOUNT STATUS

Account-status information can be saved to a file in the TELES.VoIPBOX GSM/CDMA. The following entry in the `pabx.cfg` is required:

`MsgLog=/data/msg.log`

The corresponding routing entries in the `route.cfg` configuration file will look like this:

`Restrict<port>=@FILE 06`

`Restrict<LAIN>=@FILE 06`

Example: The following example shows incoming USSD messages for 2 carriers:

```
Restrict26212=@FILE 06
Restrict26213=@FILE 06
```

The USSD entry in the file will appear in 2 lines as follows:

`Date Time [Port] IMSI`

`USSD Message Text`

Example:

```
14.02.05-17:40:06 [04] 262125555555555
Current Cash Account: 143,83 Euros
```

9.15 DEFINING SPECIAL CHARACTERS FOR VOICE CALLS

In cases in which the called number includes special characters (e.g. * 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:

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

10 LEAST COST ROUTING

TELES.VoIPBOX GSM/CDMAs are connected between the customer's private branch exchange (PBX) and the public telephone network (PSTN) and/or VoIP. The customer saves connection charges and can effortlessly and automatically connect to the carrier as needed using the following routing method:

- Carrier selection

Calls are routed transparently for the PBX and its users. TELES.VoIPBOX GSM/CDMAs can generate charges and route calls using alternate settings in case of network failures.

The following additional services are supported by this feature package:

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

10.1 CARRIER SELECTION

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

10.1.1 ROUTING ENTRIES

Use the `MapAll` command to route calls using Carrier Selection.

a) 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.
b) For unrouted connections (placed via the public telephone network), use:
`MapAll<AreaCode>=9<AreaCode>`
c) 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 or VoIP to the PBX are put through transparently.

Example:

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

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

Restrict9=10
Restrict40=10
```

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

10.2 ALTERNATIVE ROUTING SETTINGS

Alternative routing refers to the ability to establish connections using a different (alternative) network in case of provider failure (e.g. the VoIP connection has been disrupted). 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:

```
MapAll001=40iG1:001
Redirect40iG1:=A
MapAllA=9
```

In addition, the TELES.VoIPBOX GSM/CDMA can identify if the maximum number of VoIP connections has been reached. In that case, any further requests for connections are routed over the alternative (public) network using the `EscapeCtrl8=9` command.

10.3 CHARGE MODELS

TELES.VoIPBOX GSM/CDMAs can generate charge information but cannot transmit received charges from the public or corporate networks to the attached PBX. Charge impulses from the PSTN can cause interferences on the established connection and must be deactivated. Charge simulation on the TELES.VoIPBOX GSM/CDMA 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
- Initial toll-free period with retroactive charge generation afterwards

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 (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 10.99.

`MapAll`*src=dst mode time start/wait* and
`MapCallBackOut`*provsrc=dst mode time start/wait*.

Table 10.99 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.
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 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.

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".

- 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 (`MapAll`*src=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:

```
...
MapAll1=91 60 0/60*12
...
```

  In the above example, all charge units are multiplied by 12. 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.

  These charge models are explained on page 10-149.

10.4 GENERATING CHARGES WITH THE TELES.VOIPBOX GSM/CDMA

To generate charges for the attached PBX, add the charge variables described in Table 10.99 to the `MapAll` commands according to the necessities of the corporate network environment.

Example 1 `MapAll0172=9123450172 1.65 131/0`

(*time*=1.65, *start*=131, *wait*=0)

In the mapping example above, 3 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 2 `MapAll0172=9123450172 1.65 131/10`

(*time*=1.65, *start*=131, *wait*=10)

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).

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 this period 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.):

```
...
MapAll030=901019030 60 0/60*129
...
```

11 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 TCP/IP
- Generation of online CDRs using e-mail

11.1 ASR CALCULATION AND RESETTING STATISTIC VALUES

When this function is configured in the `pabx.cfg` file, statistical values, such as the number of minutes, number of calls, ASR, etc., are calculated for the entire system at a defined time. These statistics are then copied into a specified file and reset at 0.

This information can also be sent to an e-mail recipient. The following syntax must be used:

`StatisticTime=<file> <hh:mm> <day> @<account>`

i	**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 ⇨.**

Example: In the following example, the system's statistic values are saved daily into the file `stati.log` and sent to an e-mail account.

```
StatisticTime=stati.log 00:00 11111111 @<account>
```

Example: If `??` appears instead of a specified hour, the ASR is written into the `stati.log` file once every hour. The values are reset to zero in the twenty-third hour:

```
StatisticTimeReset=stati.log ??:00
```

Example: The next example shows how the statistics appear in the file into which they are copied. The following information is listed in the following order: day and time of the entry, followed by the system name. Calls: connected calls followed by the total number of calls in parentheses. The total number of minutes terminated by the system, followed by the ASR1 value, the external ASR for the traffic source (ext) and the internal ASR for the TELES.VoIPBOX GSM/CDMA (int). These values can differ if a significant number of calls cannot be routed through the TELES.VoIPBOX GSM/CDMA or an insuffficient number of channels is available for a prefix. Finally, the average call duration (ACD) appears in the entry:

```
26.10.05-00:00:00,BoIPBOX: Calls: 19351 (29716) - Minutes: 46647 - ASR1: 65.12% -  ASR(ext): 65.12% -
ASR(int): 65.30% - ACD: 144.63s
```

`StatisticTimeReset=<file> <hh:mm> <day>` performs the same function as the `StatisticTime` parameter, but also resets the counters (A-F).

Example: In the following example, the system's statistic values are saved on the 15th of every month into the file `reset.log`.

```
StatisticTimeReset=reset.log 00:00 15.
```

It is not possible to configure both `StatisticTimeReset` and `StatisticTime`. ASR values reset to 0 when the SIM card is changed using the TELES.GATE Manager.

11.2 GENERATING AND RETRIEVING CDRS

With the `Log` and `RrufLog` commands, you save CDRs and unconnected calls in the TELES.VoIPBOX GSM/CDMA.

For these parameters (`Log` and `RrufLog`), 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:

```
Log=/boot/cdr.log
RRufLog=/boot/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 11.100 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

Table 11.100 1TR6 Service Indicators *(continued)*

Service Indicator	Definition
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 5.2.1.1 ➩.

11.2.1 CALL LOG

The following entry in the `pabx.cfg` configuration file activates the capability to generate CDRs in the TELES.VoIPBOX GSM/CDMA:

`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:

DD.MM.YY-hh:mm:ss[Start],DD.MM.YY-hh:mm:ss[End],src,dst,service,dur,cause,charge_publine,[charge_sys]

DD – Day	hh – Hour	*src* – source/extension	*dur* – duration
MM – Month	mm – Minute	*dst* – destination	*cause – reason for teardown*
YY – Year	ss – Seconds	*service* – service indicator	*charge_publine* – from the public line
			charge_sys – generated by the system

The charge is specified in units. The service indicator listed will be one of the values shown on Table 11.100. The example below shows a sample log file.

```
28.01.05-19:38:51,28.01.05-19:44:51,10611,9010193333333,0101,360,90,10
28.01.05-19:43:55,28.01.05-19:44:55,10610,26212015551111111,0101,60,90,3
28.01.05-19:32:54,28.01.05-19:44:55,10612,40iG2:004498989898,0101,721,90,15
28.01.05-19:41:34,28.01.05-19:45:34,10616,9010190123456,0101,240,90,4
28.01.05-19:44:19,28.01.05-19:45:49,10615,26212015553333333,0101,90,90,5
28.01.05-19:44:58,28.01.05-19:45:58,10610,26213015562222222,0101,60,90,3
28.01.05-19:46:01,28.01.05-19:47:12,10610,9010194444444,0101,71,90,5
28.01.05-19:46:18,28.01.05-19:47:48,10615,40iG1:001232323232323,0101,90,90,4
28.01.05-19:47:03,28.01.05-19:48:07,10610,9010195555555,0101,64,90,4
28.01.05-19:48:07,28.01.05-19:49:07,10610,9010190306666666,0101,60,90,3
```

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>]`

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

```
...
Subscriber00=TRANSPARENT ROUTER ALARM NODE[0000]
...
```

Example: In the following CDR entry, `<num>` consists of a four-digit number (`0000`) that is included in the CDR.

```
29.08.05-09:45:24,29.08.05-09:46:33,923456789,[0000:01]01771111111,0101,69,0
```

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:

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

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

```
21.08.07-11:54:09,21.08.07-11:54:14,40501,[0005:01]toSIM,262032441413482,172.20.25.210:172.20.25.210,G729,20,0101,5,90,0
```

In the case of CDR entries for DLA/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:`

```
20.10.05-15:27:36,20.10.05-15:30:36,2621201555555555,DLA1234567890,0101,180/168,10,0
```

11.2.2 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 `RRufLog=<name>` entry in the `pabx.cfg`. Specify a file name, e.g. `RRufLog=failed.log`. Once this setting is made, recording begins at once.

A new line of the following format is created for each incoming call that is not accepted:

DD.MM.YY-hh:mm:ss,src,dst,cause,dur,att

DD – Day	hh – Hour	*src* – source/extension	*cause* – reason for tear down
MM – Month	mm – Minute	*dst* – destination	*dur* – duration of call attempt
YY – Year	ss – Seconds	*service* – service indicator	*att* – number of attempts

```
16.01.05-13:58:52,9030399281679,10111,0101,ff,0,1
16.01.05-14:04:06,9030399281679,10111,0101,91,0,1
16.01.05-14:04:15,9,10111,0101,91,0,1
16.01.05-14:04:39,9030399281679,10111,0101,ff,0,1
16.01.05-14:04:50,903039904983,100,0101,ff,0,1
16.01.05-14:05:02,9030399281679,10111,0101,ff,0,1
16.01.05-14:05:03,9,100,0101,ff,0,1
16.01.05-14:05:14,903039904983,100,0101,91,0,1
20.04.05-16:21:10,[4545]981776,2->10200,0101,ff,0,1
20.04.05-16:21:20,[4545]981776,1->10120,0101,ff,0,1
```

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:

```
20.02.05-10:47:52,[0004:01]00491721234567,[0005:01]DLA0307654321,0101,ff,34,1
```

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:

```
12.05.05-10:25:51,40,991783,172.20.25.110:172.20.25.110,0101,ff,8,1
```

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

Example:

```
20.10.05-15:00:06,9004930555555,DLA262121111111,0101,92,24,1
```

11.3 GENERATING ONLINE CDRS

11.3.1 SENDING CDRS VIA TCP/IP

CDRs for connected calls and calls for which no connection was set up can be transmitted online to a database. The CDRs can be sent over a TCP connection from the system to the iCDR or another application. TCP port 6609 is used, and the system sets up the TCP connection automatically. The following configuration entries are required:

In the file `pabx.cfg`:

```
Log=/boot/cdr.log @DB<ip_address>
RRufLog=/boot/failed.log @DB<ip_address>
...
[Mail]
```

The iCDR uses the following format for CDR entries, whereby `nr` represents the number of the CDR, `name` refers to the name of the system. Individual lines end with `LF`:

`S(0,nr,dd.mm.yy-hh:mm:ss:lll,0.LCR,nr,name,0)`

`{A(<CDR as generated by` TELES.VoIPBOX GSM/CDMA`>)}`

Example:

```
S(0,0,04.02.05-13:44:48:000,4,0){
A(04.02.05-13:42:44:000){a=20,0101,,,,siap1612345610,,si66610,;c=0,9;n=;p=255;s=edss1;}
B(04.02.05-13:42:44:000){a=23,0101,,,,siap1612345610,,si66610,;c=0,11;n=;p=255;s=edss1;}
C(04.02.05-13:42:44:000){a=23,0101,,,,siap1612345610,,si66610,;c=0,11;d=124000;n=;p=255;s=edss1;}
D(04.02.05-
13:44:48:000){f=B,CAU_NCC;t=A,CAU_NCC;a=23,0101,,,,siap1612345610,,si66610,;a=20,0101,,,,siap1612345610,
,si66610,;}
E(04.02.05-13:44:48:000);
```

11.3.1.1 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. Each CDR entry generated is sent as e-mail. The following entry in the configuration file activates this function:

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

12 DLA/CALLBACK SERVICES

This chapter describes money-saving features that expand the functionality of your TELES.VoIPBOX GSM/CDMA to include callback capability and DTMF services. It is particularly useful for companies with employees who travel often, because it eliminates expensive roaming fees.

This feature is available as of hardware revision 1.4 (September, 2007).

12.1 CALL CONNECTOR AND CALLBACK SERVER

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.

- Special announcement
- DLA with DTMF
- DLA with fixed destination number
- Callback with DTMF for the second leg number (known OAD or fixed callback number)
- Callback with DTMF and OAD as callback number
- Callback with DTMF and pre-configured callback number
- Callback for a fixed second leg
- DLA with DTMF and PIN for the first leg and callback for the second leg
- Using a PIN in front of the call number
- Callback via SMS
- Callback via HTTP

Numbers transmitted using DTMF tones can be ended by entering a # sign. Otherwise, a 5-second timer is set, after which DTMF transmission will automatically end.

If the callback call is set up from the mobile network, the SIM must be available 24 hours a day. We recommend that you reserve a SIM for this service. Otherwise, another call could block the call initiating callback, which limits the effectiveness of the service.

CDR entries for calls routed as Callback with DTMF include the connection times for the A and B subscribers. The times are separated by a slash (/). If no connection is established to the B subscriber, an entry recording the A subscriber's connection time is generated in the `failed.log` file.

Activating DTMF Tone Recognition

The TELES.VoIPBOX GSM/CDMA can recognize DTMF tones and initiate calls with these tones. In the `pabx.cfg`, enter a virtual DTMF controller, as described in Table 5.20. 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:

```
...
Controller09 = 41 DTMF
...
Subscriber09 = TRANSPARENT ROUTER CHMAX[5]
...
```

The TELES.VoIPBOX GSM/CDMA must be restarted to activate this configuration.

12.1.1 SPECIAL ANNOUNCEMENT

An announcement can be played immediately after the connection has been established. 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 that may pass before the next DTMF tone is entered, `<dir>` refers to the directory, in which the announcement file is saved. `boot` or `data` are possible. The file extension must be 711.

The file's sound format must be PCM!

Example: In this example, a maximum of 5 channels can recognize DTMF tones and change them into dialing data. The announcement is named `DTMF.711` and is saved in the `boot` directory:

```
Subscriber09 = TRANSPARENT ROUTER DTMF[30,/boot/DTMF.711] CHMAX[5]
```

12.1.2 DLA WITH DTMF

The user dials a number in the system that is connected with the DTMF platform. She then enters the number with which she would like to be connected.

Make the following entries in `pabx.cfg` to connect a call directly:

```
MapAll<number>=<DTMFport>DTMF
MapAllDLA=<port>
```

Example: In the following example, the call from the number 123 is connected to the DTMF platform and the call that comes in as DTMF tones is directed to the VoIP port and the VoIP profile `DF`:

```
MapAll123=41DTMF
MapAllDLA=40DF:
```

12.1.3 DLA WITH FIXED DESTINATION NUMBER

The user dials a number in the system that is connected directly with a fixed external number (e.g. international subsidiary number). Make the following entry in the `route.cfg`:

```
MapAll<num>=<port><fixed num>
```

Example: In the following example, the call comes into the number 123456 and is connected to the number 004311111 at the VoIP port and the VoIP profile `DF`:

```
MapAll123456=40DF:004311111
```

12.1.4 CALLBACK WITH DTMF AND OAD AS CALLBACK NUMBER

The user calls a number that is defined so that the user will be called back based on his OAD. An alerting occurs. The user hangs up and is called back. After the user has taken the call, the destination number is entered using DTMF tones. When he has finished dialing, the connection to the destination number is established.

Callback is not possible for VoIP calls.

The following entries in `route.cfg` will initiate callback to the calling party's number:

```
MapAllDTMF=<DTMFport>DTMF
MapAllDLA=<port>
MapAll<number>=CALLB
MapAllCB=<port>
```

Example: In this example, the call with the number 123 is connected with the OAD and the number that comes in as DTMF is directed to the VoIP port and the VoIP profile `DF`:

```
MapAllDTMF=41DTMF
MapAllDLA=40DF:
MapAll123=CALLB
MapAllCB=40DF:
```

12.1.5 CALLBACK WITH DTMF AND PRE-CONFIGURED CALLBACK NUMBER

The user calls a predefined number that is mapped to a defined callback number. An alerting occurs. The user hangs up and is called back at a fixed number. After the user has accepted the call, she must enter the destination number via DTMF. The connection is set up when she finishes dialing.

> **Callback is not possible for VoIP calls.**

Make the following entries in `route.cfg` to initiate callback to a fixed number:

```
MapAllDTMF=<DTMFport>DTMF
MapAllDLA=<port>
MapAll<number>=CALL<callbacknumber>
```

Example: In the following example, the call with the number 123 is connected with the number 03012345. The number that comes in as DTMF is directed to the VoIP port and the VoIP profile `DF`:

```
MapAllDTMF=41DTMF
MAPAllDLA=40DF:
MapAll123=CALL903012345
```

12.1.6 CALLBACK TO OAD AND FIXED SECOND LEG

The user calls a predefined number in the system. An alerting occurs. The user hangs up and is called back based on her OAD. After the user accepts the call, she is connected to a fixed, preconfigured number (e.g. operator or corporate central office.

> **Callback is not possible for VoIP calls.**

Make the following entries in `route.cfg`:

```
MapAllDTMF=<port><num>
MapAll<num>=CALLB
MapAllCB=<port>
```

Example: In the following example, the caller dials 123456 and her OAD is called back through the mobile port’s LAIN 26202. She is then connected with the operator’s number 0 through port 10.

```
MapAllDTMF=10
MAPAll123456=CALLB
MapAllCB=26202
```

12.1.7 DLA WITH DTMF AND PIN FOR FIRST LEG AND CALLBACK FOR SECOND LEG

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 second leg number using DTMF tones.

Make the following entries in `route.cfg`:

```
MapAllDTMF=<DTMFport>DTMF
MapAll<num>=<DTMFport>DTMF VOICE
MapAllDLA<num>=CALL<num> VOICE
MapAllDLA=<port> VOICE
```

Example: The number 123456 is dialed and the PIN 123# is entered. The call is then connected to the number 004930123456. The destination number can now be transmitted through the VoIP port and the VoIP profile **DF** using DTMF tones:

```
MapAllDTMF=41DTMF
MAPAll123456=41DTMF VOICE
MapAllDLA123=CALL9004930123456 VOICE
MapAllDLA=40DF: VOICE
```

The user must enter a # following the PIN. Otherwise the callback to the predefined number will not occur.

12.1.8 USING A PIN IN FRONT OF THE CALL NUMBER

To prevent abuse, the following entry can be made to configure a PIN in front of the actual call number:

```
MapAllDLA=$PIN
MapAllPIN<pin>=<port>
```

Example: In the following example, the DTMF tones are analyzed, whereby the first 4 (1111) corresponds with the PIN. The call to subscriber B is initiated when the PIN has been entered correctly. All other DTMF tones are directed to the VoIP port and the VoIP profile **DF**:

```
MapAllDLA=$PIN
MapAllPIN1111=40DF:
```

13 FEATURE PACKAGES

The TELES.VoIPBOX GSM/CDMA feature packages provide additional services. The following feature packages are included:

- SMS Gateway (see Chapter 13.1 on page 13-162)
- Ported Number Screening (see Chapter 13.2 on page 13-167)

13.1 SMS GATEWAY

The SMS Gateway allows you to use your TELES.VoIPBOX GSM/CDMA to send and receive SMS. The following functions are possible with this feature package:

- Sending SMS via e-mail
- Receiving SMS to e-mail, SMS or to a file
- Sending and receiving USSD text messages
- Setting up connections using e-mail
- Sending announcements via e-mail
- Sending automatic SMS for unconnected calls

Bear in mind that the parameters for connection to the SMTP server must be configured in the `pabx.cfg`'s `[Mail]` section (see Chapter 5.2.2).

13.1.1 SENDING SMS VIA E-MAIL

This function makes it possible to send SMS via a TELES.VoIPBOX GSM/CDMA with an ordinary e-mail client. SMS messages are recorded in the CDR log with the service indicator 0500. The destination address with the keyword SMS, the call number, the @ sign and the IP address or the IP name of the TELES.VoIPBOX GSM/CDMA must be entered.

To use this function, you must first set the parameter `<smsc>` in the `pabx.cfg` (see Table 5.22 on page 5-47).

In the following example, an SMS is sent to the mobile number **015553456789**, whereby **sms-mail.server.de** must correspond with the IP address **172.172.172.172**:

Example: **SMS015553456789@172.172.172.172** or
SMS015553456789@sms-mail.server.de

The SMS text must be entered in the text box. The subject box is not used. If the e-mail program supports sending the same e-mail to more than one address, the SMS messages are sent in intervals of one second. The TELES.VoIPBOX GSM/CDMA's algorithms evenly distribute the SMS messages to the mobile modules.

If the TELES.VoIPBOX GSM/CDMA rejects the SMS, an e-mail alerting an aborted SMS will be transmitted to the sender and the attempt will be entered in the corresponding log file (`RRufLog=`). If transmission is successful, a positive response will be sent when the SMS is accepted by the SMS service center. `sent` will then appear in the subject dialog. A corresponding CDR will be entered with a destination address beginning with SMS.

A request to set up a connection with the service 'telephony' and the element 'user-to-user' enables the SMS text to be sent to the TELES.VoIPBOX GSM/CDMA. All TELES.VoIPBOX GSM/CDMAs are supported that allow for SMS messages to be sent by the process described above.

The following entry must appear in the `route.cfg` configuration file for SMS transmission to be possible: `MapAllSMS=<port number>`.

Sent SMS will also be recorded if the call log is active on the system. The formats are described in chapters 11.2.1 and 11.2.2.

Example 1 In the following example, SMS-transmission to the number 01721111111 from the e-mail account j.smith was successful:

```
12.07.06-09:59:10,12.07.06-09:59:11,SMS01721111111,j.smith,0500,1,06,0
```

Example 2 In the following example, SMS-transmission to the number 01721111111 from the e-mail account j.smith was unsuccessful. the output -1 as cause value means that no routing entry was configured:

```
12.07.06-09:04:11,SMS01721111111,j.smith,0500,00,-1,0
```

13.1.2 RECEIVING SMS MESSAGES

This function makes it possible to receive SMS messages via a mobile gateway with an ordinary e-mail client, to forward them to another mobile telephone, or to save them to a file.

Bear in mind that you must set the service type in all identical restrict entries.

Example: In the following example, all incoming voice calls are routed to the operator incoming SMS messages are forwarded to the email account `sysadmin`:

```
Restrict26202=100 01
Restrict26202=@sysadmin 05
```

13.1.2.1 SMS TO E-MAIL

The destination number in the TELES.VoIPBOX GSM/CDMA system must correspond with an e-mail account. The e-mail recipient's name contains the keyword 'sms' and the destination number. The subject box contains the SIM card's IMSI and the caller's number.

Example:

From: sms262124915553230618@gsm.teles.de

Subject: SMS 262123203500514 4915553230618

A mapping entry must indicate an e-mail account with a prefixed @. The following syntax is used:

`Restrict<port>=@<addressee> 05`

As an alternative, the @ sign can be substituted with a colon (:) in the recipient's address. If only a destination account is given, the configured domain name is used.

13.1.2.2 SMS TO SMS

This configuration makes it possible to forward SMS messages via a mobile gateway to another mobile telephone.

`Restrict<port>=@-><port><mobile number> 05`

Example:

```
Restrict20=@->200155512345678 05
```

13.1.2.3 SMS TO FILE

Using this configuration, you can save SMS messages via a mobile gateway to a file.

Make the following entry in the `pabx.cfg`:

`MsgLog=/boot/msg.log`

The following entry in the `Route.cfg` is also required:

`Restrict<port>=@FILE 05`

13.1.3 SETTING UP CONNECTIONS VIA E-MAIL

This function sets up a connection between subscriber A and subscriber B via e-mail. Subscriber A is identified by an e-mail address and is dialed first. Subscriber B is called when the connection to subscriber A has been set up.

A connection can be set up via e-mail with the keyword 'CALL,' the destination number, the @ sign, and the IP address or the TELES.VoIPBOX GSM/CDMA system's IP name.

The following example shows a connection with the destination number **0123456789**, whereby **msg-mail.server.de** must correspond with the IP address **172.172.172.172**.

Example: **CALL0123456789@172.172.172.172** or
CALL0123456789@msg-mail.server.de

Any text contained in the text box will be sent to subscriber A as user-to-user information. The subject box is not used.

Subscriber A is identified by an e-mail address and must be activated in the TELES.VoIPBOX GSM/CDMA system. The subscriber's name must appear before the @ sign. This name must be assigned a corresponding MapOut command.

Example: Subscriber's e-mail address is meier@server.de. Subscriber's extension is 555. Configure `MapAll@555=meier.`

In addition to CTI capability, this function allows for callback via e-mail.

13.1.4 SENDING ANOUNCEMENTS VIA E-MAIL

It is possible to send announcements using e-mail. An audio file with Teles G.711 A-law encoding is simply sent as an attachment. The destination address begins with the keyword `play`, followed by the telephone number, the @ sign and finally the TELES.System's IP address or name.

Example: the e-mail address will look like this:

```
play123456@192.168.0.1
```

or

```
play123456@anouncement.server.de
```

Make the following entry in the `pabx.cfg`:

`MapAllplay<number>=<port><number>`

Example: In the following example, a connection to 123456 is set up through mobile controller's LAIN 26202:

```
MapAllplay123456=26202123456
```

After the call has been successfully established, the system generates an e-mail that contains the keyword `play<num>` in the from line. The keyword `connected` appears in the subject line.

If an error occurs, the keyword `error` appears in the subject line. For example, errors may occur when the called number is occupied. Bear in mind that the system will attempt to resend the message as often as is defined in the parameter `MailToHostRetries`.

To distinguish between voice calls and announcement calls in the CDRs, the keyword `play` appears in front of the DAD in the CDRs.

13.1.5 DISPLAYING INCOMING CALLS

With this function, you can use e-mail to signal incoming calls. Two signaling types are possible:

- Display all incoming calls that receive a busy or ringing signal. Enter the keyword `CTI[001.000.000.000]` in the VoIP controller's `Subscriber` line of the TELES.VoIPBOX GSM/CDMA system's `pabx.cfg` configuration file.
- Display all unsuccessful incoming calls (callback list) that receive a busy signal or remain unanswered. Enter the keyword `CTI[002.000.000.000]` in the VoIP controller's subscriber line of the TELES.VoIPBOX GSM/CDMA system's `pabx.cfg` configuration file.

The destination is the address of the called subscriber configured in a corresponding map entry. A callback can be initiated when the recipient responds to the e-mail.

13.1.6 SENDING AUTOMATIC SMS FOR UNCONNECTED CALLS

When the TELES.VoIPBOX GSM/CDMA is implemented in a corporate network and connected to a PBX or between a PBX and the outside line, the following configuration entry in the `pabx.cfg` activates a feature, whereby the system automatically sends an SMS message to dialed mobile numbers that are unreachable or not answering.

A configurable text containing the callers OAD is sent in the SMS message, so that the mobile user knows who called him through the TELES.VoIPBOX GSM/CDMA's interface and can return the call.

The parameter `SMSInfo` activates this feature. The text can be configured on an individual basis, and the caller's number is automatically generated when you enter `%s`. You must enter the text that is to be sent in quotation marks:

```
SMSInfo="<text>%s<text>"
```

No SMS will be generated for unconnected calls if the service code `VOICE` or DATA appears in the mapping entry.
The SMS center number must be defined (see Table 5.22), and the routing entry for sending SMS must be configured.
At least two SIM cards must be activated in the TELES.VoIPBOX GSM/CDMA for this feature to work.

Example: In the following example, SMS messages for mobile users are generated only when calls cannot be connected. The network prefix is 0155 and the LAIN is 26212. The company's mobile prefix is 57777.
No other mobile targets for mobile carriers with the LAIN 26212 and 26213 receive SMS, since the parameter `VOICE` has been defined in the mapping entry:

`pabx.cfg`:

```
SMSInfo="You got a call from %s . Please call back."
```

`route.cfg`:

```
MapAllSMS=26212

MapAll015557777=|26212015557777<<17
MapAll01555=|2621201555<<17 VOICE
MapAll01556=|2621201556<<17 VOICE

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

13.2 PORTED NUMBER SCREENING

Ported Number LCR Extension is a function that enables you to map defined destination call numbers to other destination numbers or networks (number portability). This function is used to allow telecommunications subscribers to change carriers without having to change their telephone numbers.

Number portability is used in the fixed network, as well as in the mobile network. Usually the numbers are mapped in their respective networks. Implementation of this information and the corresponding routing processes result in significant cost savings, as tariff differences between calls to 'normal' and ported subscribers are eliminated.

The database of ported numbers runs on the TELES.iMNP, which provides the data online for the entire network. You can also choose an external provieder.

The TELES.VoIPBOX GSM/CDMA automatically routes calls through specific ports, so that all calls through the same carrier (including ported numbers) are routed through the port containing that carrier's SIM card.

13.2.1 SYSTEM REQUIREMENTS

Ported number screening requires the following:

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

13.2.2 ROUTING AND CONFIGURATION

To connect to the number portability database, you must set the entries described in Chapter 5.2.3 ⇨.

An appropriate routing entry in the `route.cfg` file is required to activate Ported Number LCR Extension. This includes activation of digit collection and the following mapping configuration:

```
...
DTMFWaitDial=<sec>
MapAll<num>=|$ph<<<count>
MapAllph=|D@<num><<01
```

The routing entries for the TELES.iMNP results contain the keyword QN, followed by the query result, an equal sign and the controller:

```
MapAllQN<query>=<controller>
...
```

Example: The following example uses digit collection (11 digits plus `$ph`). Every incoming call with a leading digit of 0 results in an TELES.iMNP query. The SIM-card LAINs are used instead of controller numbers. All numbers that come back from the TELES.iMNP with the LAIN for Carrier_1 (26211) are then routed through Carrier_1's SIM card with CLIR. The same applies for Carrier_2 (26212), Carrier_3 (26213) and Carrier_4 (26214). Numbers that the TELES.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 TELES.iMNP does not respond within two seconds

(D@0), the call is routed as it comes in, whether it is ported or not:

```
DTMFWaitDial=5
MapAll0=|$ph<<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
```

14 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.

14.1 SIGNALING PARAMETERS

Table 14.101 Customized Parameters: Protocol-Independent VoIP Signaling

Protocol-Independent VoIP Signaling Parameters
VoipDad=<num> The digits/numbers defined here will appear in front of the original DAD. If the parameter is to be valid in only one direction, you must set another profile without this parameter for the other direction.
VoipOad=<num> The digits/numbers defined here will be transmitted in front of the original OAD. If a minus (-) is entered, the original OAD will not appear. Only the digits entered in front of the minus sign will be displayed. If the parameter is to be valid in only one direction, you must set another profile without this parameter for the other direction. To limit this feature to OADs consisting of a certain number of digits, enter a !, followed by the number of digits, at the end of the entry. In the following example, the digits `567` will appear only if the OAD has at least 6 digits: EXAMPLE: `VoipOad=567!6` To modify the original OAD, enter `randomx`, whereby x represents a number of random digits that will appear in the OAD. EXAMPLE: `VoipOad=567random2-`
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.
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.

Table 14.101 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.**
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.
VoipIgnoreDADType=<mode> Enter `yes` to change the DAD type to unknown, e.g. from international. The type is lost, e.g. the leading 00 bit is removed. Default `no`.
VoipSuppressInbandInfoAvailableIndicatorInCallProceeding=<mode> Enter `yes` to send or receive the Progress Indicator in the Q.931 Call Proceeding message. Default `no`.
VoipG72616PayloadType=<num> Changes the SIP payload type for G.726 16 b/s. Default is 35. A common value is 102.
VoipG72624PayloadType=<num> Changes the SIP payload type for G.726 24 b/s. Default is 36. A common value is 99.
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.
VoipDataBypassPayloadType=<num> Defines the payload type for the RTP packets when the call is sent as a data call. Default 96.
VoipMinInterDigitTime=<ms> Defines a number of milliseconds the system waits between transmission of DTMF tones.

Table 14.102 Customized Parameters: H.323 Signaling

H.323 Signaling Parameters
VoipService=0x<service indicator> This parameter sets the barrier capability. For example, it can be used for calls coming from VoIP with the barrier 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`
VoIPOwnIpAddress=<ip addr> If the system is behind a NAT firewall that does not translate H.323, the NAT firewall's public IP address is transmitted as own IP address in the H.323 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 TELES.VoIPBOX GSM/CDMA's private IP address.
VoipMapOadType=<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.
VoipSetupAck=<int> 1=setup acknowledge is transmitted; 0= setup acknowledge is not transmitted; 2 (default) =transmitted with H.323 information.
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.
VoipCanOverlapSend=<mode> Enter off to deactivate overlap sending during setup (default on).

Table 14.103 Customized Parameters: SIP Signaling

SIP Signaling Parameters
VoipOwnAddress=<account@domain> Used for the `From` field in Sip-Invite and Sip-Response messages. If only the domain is entered, the origination address (e.g. from ISDN) followed by an @ sign will automatically be set at the beginning.
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. Beispiel: From: "John" <sip:493011111@teles.de>
VoipContact=<account@domain> Used for the `Contact` field in Sip-Invite and Sip-Response messages.
VoipP-Preferred-Identity=<string> Sets the P-Preferred-Identity field in the SIP invite message. The following settings are possible toward SIP: `*` The OAD coming from ISDN/POTS is transmitted. `<string>` The defined string is transmitted A combination of both is possible. Examples: 030* or tel:* or sip:user@carrier.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
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 **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.

Table 14.103 Customized Parameters: SIP Signaling *(continued)*

SIP Signaling Parameters
VoipDadSource=<int> SIP only: defines the field from which field the called party number coming from SIP is to be taken: 0 = URL 1 = To: field 2 = Remote-Party-ID with party = called
VoipUseMaxPTime=<mode> SIP only. Enter `yes` to set the field `mptime` (max packet time) with the values set in `VoipTxm` (`ptime`). Default `no`.
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.
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`.
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/POTS with overlap sending. When this parameter is active, `VoipPrack` is automatically set to `yes`. Default is `no`.
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.
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` It may be necessary to remove some of these entries for some peers.

Table 14.103 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`.**

14.2 LOCATION SERVER PARAMETERS

The following parameters can be used in the VoIP profile when the SIP agent wants to register with the TELES.VoIPBOX GSM/CDMA.

Table 14.104 Customized Parameters: Location Server

Location Server Parameters
VoipOwnUser=<string> Defines the username the agent uses to register.
VoipOwnPwd=<string> Defines the password the agent uses to register.
VoipExpires=<sec> Defines the maximum number of seconds the agent's registration applies (default 3600).
VoipAuth=<mode> Defines the authentication procedure `www` (default) or `proxy`.

Example: The following example creates an account for a user agent with the username 130 and password test130. Authentication occurs with the procedure www:

```
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
```

14.3 ROUTING PARAMETERS

Table 14.105 Customized Parameters: VoIP Routing

VoIP Basic Parameters

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 (e.g. other codec, `WaitForConnect`, etc.). 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.

VoipUseIpStack=<mode> Enter `Yes` to facilitate direct use of an xDSL or dial-up connection if the corresponding profile is defined. Default is `No`.
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.**
VoipEnumDomain=<name> Enter a domain name for the E.164 number query. The default domain is arpa.
VoipUseStun=<mode> Enter yes (default yes) to use the STUN values for the VoIP profile.

14.4 QUALITY PARAMETERS

Table 14.106 Customized Parameters: VoIP Quality

VoIP Quality Parameters
VoipSilenceSuppression=<mode> Activates silence suppression (see Table 5.28).

Table 14.106 Customized Parameters: VoIP Quality *(continued)*

VoIP Quality Parameters
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 5.24.
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.**
VoipRtpTos=<num> Enter a value between 0 and 255 to set the TOS (type of service) field in the RTP packet IP header. Possible values are described in Table 14.107. If your IP network uses diferentiated 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. **NOTE: `VoipUseIpStack` must be 0 (default).**
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 14.107. 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).**
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: 0 = 10ms (default)) 1 = 5 ms 2 = 10 ms 3 = 20 ms

Table 14.106 Customized Parameters: VoIP Quality *(continued)*

VoIP Quality Parameters
VoipCallGroup=<name> All outgoing VoIP calls for VoIP profiles with the same `VoipCallGroup` name are distributed cyclically to these profiles.
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.
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`.**
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`.**
VoipDJBufOptFactor=<count> Enter a value between 0 and 13 to set the balance between low frame erasure rates and low delay (default 7).
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 90). If `0` is entered, no entry will be generated and the connection will not be terminated.
VoipTcpKeepAlive=<mode> Enter `yes` (default) to send the `RoundTripDelayRequest` message every 10 seconds (necessary for long calls with firewalls using TCP aging).
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.
VoipBrokenDetectionTimeout=<ms> When this parameter is set, the system recognizes an interruption in the transmission of RTP/RTCP data in the VoIP connection following the set number of milliseconds. 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.

Table 14.106 Customized Parameters: VoIP Quality *(continued)*

VoIP Quality Parameters
VoipAutoRtpAddr=<mode> Some application scenarios require automatic RTP IP address and port recognition for VoIP calls, for example if a firewall or NAT changes the IP address of incoming RTP data. Enter `Yes` to activate automatic recognition (default `No`).

Table 14.106 Customized Parameters: VoIP Quality *(continued)*

VoIP Quality Parameters
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

Table 14.106 Customized Parameters: VoIP Quality *(continued)*

VoIP Quality Parameters
VoipVoiceVolume=<num> The volume of VoIP calls coming from the Ethernet. The range is 0-63. The default value of 32 is 0 dB.
VoipInputGain=<num> The volume of VoIP calls coming from ISDN,POTS or mobile. The range is 0-63. The default value of 32 is 0 dB.

Table 14.106 Customized Parameters: VoIP Quality *(continued)*

VoIP Quality Parameters

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.

VoipECE=<mode> Enter `yes` (default) to set echo cancellation. Enter `no` to disable echo cancellation.

Table 14.106 Customized Parameters: VoIP Quality *(continued)*

VoIP Quality Parameters
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.

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. 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.

The following specifications for Quality of Service correspond with RFC791 and RFC1349.

Table 14.107 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							

Table 14.107 Quality of Service Values *(continued)*

101	CRITIC/ECP
100	Flash override
011	Flash
010	Immediate
001	Priority
000	Routine

14.5 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 14.108 Customized Parameters: VoIP Compression

VoIP Compression Parameters
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.
VoipAggRemoteDataPort=<port> `VoipAggRemoteDataPort=29500` Enter the port for the VoIP peer that is used for aggregated packets (compressed data). Default: 29500.
VoipAggOwnDataPort=<port> `VoipAggOwnDataPort=29500` Enter the own port number used for aggregated packets. Default: 29500.
VoipAggRemoteRtpPortSpacing=<count> Defines the space between the ports used for the peer's individual RTP streams (default 2).

14.6 FAX/MODEM PARAMETERS

Table 14.109 Customized Parameters: VoIP Fax

VoIP Fax/Modem Parameters
VoipFaxTransport=<int> Enter 2 and signaling will switch to G.711a (framesize 40ms) when the peer cannot handle fax transmission with T.38. The codec will change when the system detects a fax or modem connection on the channel. `0` = disabled (default); `1` = relay. T.38 is always used. **NOTE: Bear in mind that if T.38 is defined in the `VoipCompression=` line of the VoIP profile, the system will switch only when it detects a modem connection. Fax calls will still be transmitted using T.38.**
VoipFaxBypassPayloadType=<num> Defined the payload type for a fax's RTP packets when T.38 is not used (default 102).
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 4 - 12000 5 - 14400 (default) EXAMPLE: `VoipFaxMaxRate=5`
VoipFaxECM=<mode> You can use this parameter to disable the error correction mode for fax transmission: `yes`=enabled (default), `no`=disabled.
VoipFaxProtocol=<int> Defines the protocol used: 0 = TCP 1 = FRF.11 2 = UDP, datarate management 1 3 = UDP, datarate management 2 (default)
VoipT38ErrorCorrectionMode=<int> Sets the error-correction mode: 0 = Redundancy (default) 1 = Forward error correction

Table 14.109 Customized Parameters: VoIP Fax *(continued)*

VoIP Fax/Modem Parameters
VoipT38CtrlDataRedundancy=<int> Defines the redundancy level for control packets: 0 = Disable (default) 1-7 = Sets level
VoipT38ImageDataRedundancy=<int> Defines the redundancy level for fax content: 0 = Disable (default) 1-3 = Sets level
The following parameters are responsible to set the modem transport method if a modem connection is detected.
VoipV21Transport=<mode> 0=disabled (must be set to 0).
VoipV22Transport=<mode> 0=disabled, 2=bypass (default).
VoipV23Transport=<mode> 0=disabled, 2=bypass (default).
VoipV32Transport=<mode> 0=disabled, 1=relay (default), 2=bypass .
VoipV34Transport=<mode> 0=disabled, 1=fallback to v32, 2= bypass (default).

14.7 DTMF PARAMETERS

Table 14.110 Customized Parameters: VoIP DTMF

VoIP DTMF Parameters
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, `VoipDtmfTransport` should be `1` or `3`.
VoipDtmfTransport=<int> 0 (H323) = DTMF relayed with H.225 signaling information. 0 (SIP) = DTMF relayed with SIP INFO. 1 = DTMF and MF taken from audio stream and relayed to remote. 2 (default) = DTMF and MF kept in audio stream and not relayed. 3 = DTMF and MF taken from audio stream and relayed to remote as per RFC2833. 4 (SIP only) = SIP INFO messages will be relayed as DTMF and MF.
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)
VoipRFC2833PayloadType=<num> This parameter changes the DTMF payload type. The default value is 96, a common value is 101.

15 OPTIONAL FUNCTION MODULES

This chapter contains a description of modules that expand the functionality of the TELES.VoIPBOX GSM/CDMA, such as:

- HTTP User Interface
- TELES.iPBX
- SNMP agent
- DNS forwarder
- ipupdate - DynDNS client

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.

15.1 OVERVIEW

The modules can be downloaded using FTP. The access data for each module is as follows:

- Http User Interface
  ftp://195.4.12.80
  user: httpd
  password: httpd
- TELES.iPBX
  ftp://195.4.12.80
  user: ipbx
  password: ipbx
- DNS Forwarder
  ftp://195.4.12.80
  user: dnsmasq
  password: dnsmasq
- snmp agent
  ftp://195.4.12.80
  user: snmp
  password: snmp
- ipupdate
  ftp://195.4.12.80
  user: ipupdate
  password: ipupdate

Install the respective software package on the TELES.VoIPBOX GSM/CDMA using TELES.GATE Manager. For a description of how to update the software, please refer to Chapter 7.3 ⇨. Make sure the module's file ending is correct before installation. The number in the file ending shows the starting order of the modules. Do NOT change this number if it is 0! All other modules can simply be numbered in ascending order.

For instance, the ending for the optional function module will be `tz2` or higher:

- `tz2`
- `tz3`

Following completion of transmission, you must adjust the module's configuration and restart the TELES.VoIPBOX GSM/CDMA. Once you have restarted the system, you can use the required features.

15.2 HTTP USER INTERFACE

The HTTP user interface is a user-friendly tool that can be used by carriers, administrators and individual users to configure the TELES.VoIPBOX GSM/CDMA. For a detailed description of the HTTP user interface, please see Chapter 4.11.2 ⇨.

15.3 TELES.IPBX

The TELES.iPBX is a soft PBX that runs as an add-on application on TELES CPE devices. These include TELES.VoIPBOX GSM/CDMAs (BRI or analog). It is used to connect local IP telephones and soft phones, as well as traditional line-based PBX extensions and telephones. The connection to the public telephone network can occur via VoIP, through the traditional PSTN network, or a combination of both. Both analog and ISDN (BRI) lines can be connected as PSTN. Connection to the carrier can occur using SIP, H.323, or a combination of both. Multiple VoIP destinations can be can be mapped through the routing process. The TELES.iPBX can be used to add local or remote IP extensions (work@home) to an existing PBX without requiring changes to the existing PBX, or you can implement the TELES.iPBX to completely replace your old PBX. For further information, please refer to the TELES.iPBX Systems Manual, which can also be found on the FTP server.

Features

- Caller ID
- Call forward/transfer
- Call parking/retrieve
- Conference calling
- DND (Do Not Disturb)
- Music on hold
- Direct inward dial access
- Direct outward dial
- Different dial plans
- Hunt groups
- Push to talk
- Dial by name
- Fax support
- Voicemail
- IVR

15.4 SNMP AGENT

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 (E.g. Layer 1 errors on E1 lines) and error recovery messages are sent via SNMP trap.

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 15.111 Traps for IP Interfaces

Trap Number	Interface
0	Ethernet 1
1	Ethernet 2
2	Loopback
3	xppp= (if used)
4	pppoe= (if used)

Bear in mind that the keyword `ALARM` must be entered in the appropriate POTS 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 section `[snmpd]`:

Table 15.112 Settings in the Section `[snmpd]`

Parameter	Definition
Port=<port>	Defines the target port for the trap server (default 161).
`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`.

15.5 DNS FORWARDER

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 TELES.VoIPBOX GSM/CDMA, 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 TELES.VoIPBOX GSM/CDMA's address as DNS server address, so that no public DNS server address is required. The TELES.VoIPBOX GSM/CDMA 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 TELES.VoIPBOX GSM/CDMA 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.

15.6 IPUPDATE - DYNDNS CLIENT

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 `ip.cfg`, in the `[DynDNS]` section:

Table 15.113 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.	

Table 15.113 pabx.cfg: DynDNS *(continued)*

DynDNS Parameters
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.

Example: 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 TELES.VoIPBOX GSM/CDMA works as DSL router and the dynamically allocated IP address of the PPPoE interface is used:

```
[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 TELES.VoIPBOX GSM/CDMA when the IP connection does not have a fixed IP address. In this case, you can access the system, for example with the TELES.GATE Manager, if the host name is used in the Remote Number dialog. Example entry in the Remote Number dialog: `IP:host.domain.de`

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