Thomson Gateways

Voice over IP Configuration Guide

R7.4 and higher

Thomson Gateway

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About this Voice over IP Configuration Guide

Used Symbols

A ***note*** provides additional information about a topic.

A ***caution*** warns you about potential problems or specific precautions that need to be taken.

Typographical Conventions

Following typographical conventions are used throughout this manual:

- This sample text indicates a hyperlink to a Web site.

  Example: For more information, visit us at www.thomson-broadband.com.
- This sample text indicates an internal cross-reference.

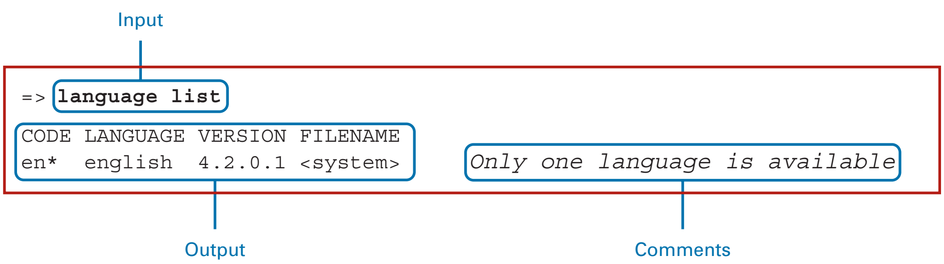
  Example: If you want to know more about guide, see "1 Introduction" on page 7".
- ***This sample text*** indicates an important content-related word.

  Example: To enter the network, you ***must*** authenticate yourself.
- **This sample text** indicates a GUI element (commands on menus and buttons, dialogue box elements, file names, paths and folders).

  Example: On the **File** menu, click **Open** to open a file.
- **`This sample text`** indicates a CLI command to be input after the CLI prompt.

  Example: To obtain a list of all available command groups, type **`help`** at the top level.
- **`This sample text`** indicates input in the CLI interface.
- *`This sample text`* indicates comment explaining output in the CLI interface.

Example:



Documentation and software updates

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1 The VoIP Configuration guide

1.1 Introduction

Goal

The goal of this document is to provide Technical Sales Support Managers (TSSMs) and providers a reference of all of the VoIP Command Line Interface (CLI) commands in order to get a better understanding of the Session Initiation Protocol (SIP) features incorporated in the Thomson Gateway.

Prerequisite

To use this Configuration Guide a basic knowledge of VoIP, SIP in particular, is required. You must be able to set up a voice call starting from a Thomson Gateway running the factory defaults.

Applicability

This version (v2.0) is applicable for R7.4 and higher.

Depending on the model and release of your Thomson Gateway not all of the described parameters and settings might be available.

Content

This document describes all of the applicable CLI commands and their accompanying parameters. You can find information on the configuration of the complete voice command group:

- "1 The VoIP Configuration guide" on page 3.
- "2 When enabling the voice service" on page 9.
- "3 General Voice Configuration" on page 13.
- "4 SIP Configuration" on page 25
- "5 Supplementary services" on page 45.
- "6 Other configurations" on page 63.

1.2 Configuring Voice on your Thomson Gateway

Two ways

You can configure your Thomson Gateway in two ways:

- A basic configuration via the GUI.
- An advanced configuration via CLI commands.

Basic configuration

It is obvious that the basic configuration offers limited settings to configure your Thomson Gateway compared to the advanced configuration. The possible settings in the basic configuration will be treated when they are discussed in the advanced configuration. Basic configuration is done via the Web pages (GUI).

Accessing the Thomson Gateway web pages

To access the Thomson Gateway via the Web pages (GUI):

1. Open a Web browser.
2. In the address bar, type your Thomson Gateway's IP address (factory default is 192.168.1.254) or DNS host name (factory default is http://dsldevice.lan).

> The Domain Name System (DNS) server address must be obtained automatically.
>
> To obtain the DNS server address automatically:
>
> 1. Click Start, Control Panel and double-click the Network Connections icon.
> 2. Open your Local Area Connection.
> 3. Click the General tab and then click Internet Protocol (TCP/IP) in the **This connection uses the following items** list.
> 4. Click the **Properties** button.
> 5. Click the **Obtain DNS server address automatically** option button and then click the **OK** button.

Easy Setup wizard

In the Web pages (GUI) you can launch an Easy Setup wizard that makes the basic voice configuration even more easy.

To launch the wizard:

1 Click **Thomson Gateway** in the left menu.

2 The Thomsom Gateway page appears:

3 In the **Pick a task**... list, click **Set Up**.

The wizard will pop-up in a separate window. The wizard will prompt you to:

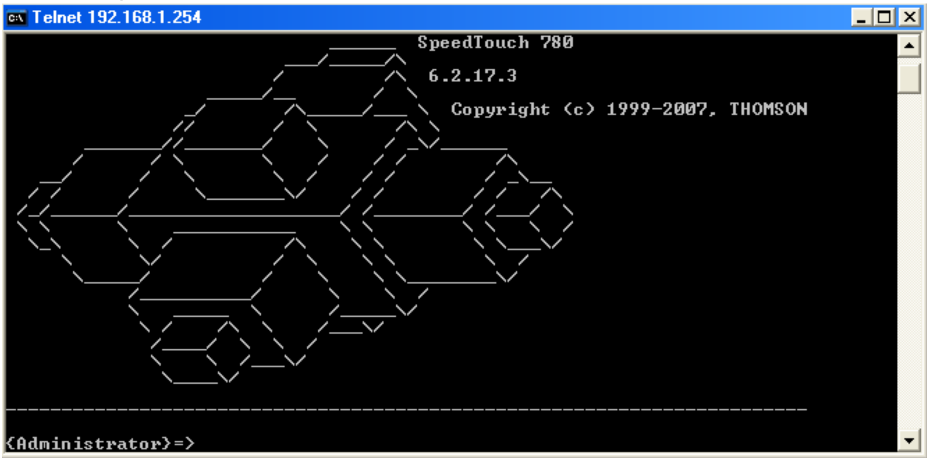
- Select a type of service, for voice select **Voice Router**.
- Specify the Internet connections, i.e. the VPI/VCI of the ATM connection and the type of connection. Your ISP should provide you this information.
- Define the Internet account settings, i.e. user name and password for authentication purposes for your Internet account. Your ISP should provide you this information.
- Define the voice connection settings, i.e. proxy address + port and registrar address + port. Your voice provider should provide you this information.
- Define the voice account settings, i.e. telephone number (URI), user name, password and assign the FXS port to be used. The URI, user name and password should be provided by your voice provider. For more information on these parameters, see “6.1.2 Adding profiles” on page 66.
- Define the Access control settings, i.e. user name and password. This is the access control to the Thomson Gateway management interface (click **Toolbox** and then click **User Management**). The default user name is Administrator and no password is defined.

Advanced configuration

The advanced configuration is done via CLI commands, running a telnet session.

To open a telnet session:

1. Click **Start**, **All Programs**, **Accessories** and then **Command Prompt** or
   Click **Start**, **Run**, type cmd and click OK.
2. At the command prompt type `telnet 192.168.1.254` and press Enter.
3. The default user name is Administrator.
4. Leave the password blank (default value).



The following table gives an overview of basic CLI commands:

Commando	Result
cmd + <Enter>	To go to the specific configuration level.
.. + <Enter>	To go one level up.
: + <Enter>	To go to the root directory.
:cmd + <Enter>	To start the specific command in whatever you are in. This can always be used in front of a command to start from root.
help or ? before cmd + <Enter>	To display more information of the specific command. Example: to display the information of the voice configuration, type :voice help config.
:menu + <Enter>	To display the menu of the available commands at a certain level.
:exit + <Enter>	To exit the session.
:saveall + <Enter>	To save all of the configuration changes.
:env set var=SESSIONTIMEOUT value=0	The session time out is by default set to 120 seconds. You can set a value between 0 and 9999 (seconds). When setting the value to 0, the session time out timer is disabled.

The following tables gives an overview of basic keystrokes in the CLI:

Keystroke	Result
<Tab>	To auto complete a command with the different possibilities.
Arrow up or down	To make a selection between the different parameters.

1.3 Checking the VoIP service status

To check the status of the VoIP service, carry out following command:

```
:voice state
VOIP_SIP-admin. state : enabled
VOIP_SIP-oper.  state : enabled
VOIP_SIP-IP address   : 101.101.101.24
```

The administrative state (VOIP_SIP-admin. state is enabled / disabled when the VoIP service is enabled / disabled).

The operational state (VOIP_SIP-oper. state) is enabled when:

- The administrative state is enabled.
- The IP address and other required parameters (proxy and registrar) is given to the VoIP module.
- There is a route to the outbound proxy.

All three conditions have to be fulfilled.

The VOIP_SIP_IP address is the source IP address.

2 When enabling the voice service

Overview

In order to use VoIP through a UA representing an FXS or DECT port, you must configure your Thomson Gateway as a UA. To do so the following tasks must be performed:

- "2.1 Enabling the VoIP service" on page 10
- "2.2 Binding and enabling the SIP ALG" on page 11
- "2.3 Enabling the RTP_predict_for_term_SIP ALG" on page 12

Chapter 2

2.1 Enabling the VoIP service

By default, the voice service is disabled. To check the status of the VoIP service, carry out following command:

```
:service system list name=VOIP_SIP
Idx Name             Protocol         SrcPort   DstPort      Group        State
-------------------------------------------------------------------------------------
1   VOIP_SIP                                    5060                      disabled
```

To enable the voice service, carry out following command:

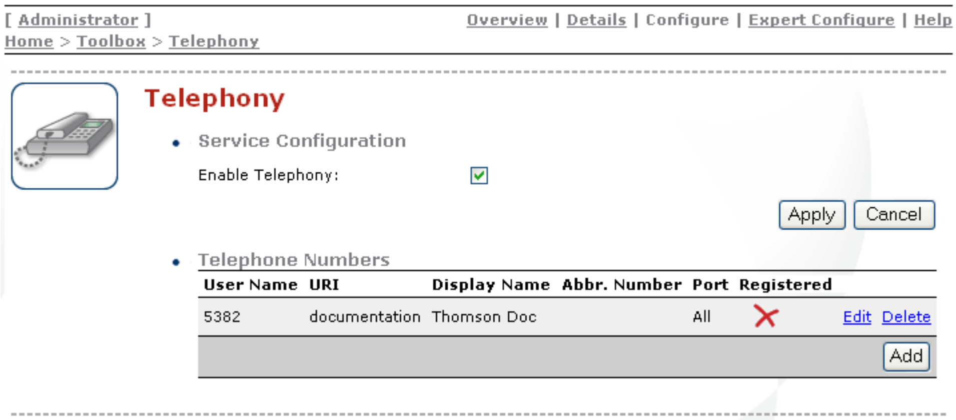
```
:service system modify name VOIP_SIP state=enabled
```

You can also enable the VoIP service via the Web pages (GUI):

1 In **Toolbox menu**, click **Telephony**.

2 Click **Configure**.

3 The Telephony page appears:



4 Select **Enable Telephony**.

Click **Apply**.

Depending on the Thomson Gateway variant, the default listening port (i.e. the port on which the application runs) will differ.

For Thomson Gateways equipped with a SIP UA the default listening port is 5060.

For Thomson Gateways equipped with a SIP Server the default listening port is 5065.

If you want to change the listening port to e.g. 5090, carry out following command:

```
:service system modify name VOIP_SIP port 5090
```

2.2 Binding and enabling the SIP ALG

In order to use the RTP_predict_for_term_SIP ALG, you must first bind and enable the SIP ALG.

The use of the SIP ALG is not allowed when running applications that are using STUN. Only when the use of the SIP ALG is explicitly required, you should enable it.

By default, the SIP ALG is already bound. To check the status of the SIP ALG, carry out following command:

```
:connection bindlist
```

The result is a list of all of the bound or active ALGs.
In this example both, the SIP ALG is bound and both SIP ALG and RTP_predict_for_term_SIP ALG are enabled.

```
Application Proto Portrange Flags
SIP         udp   5060      SIP_ALG:E RTP_predict_for_term_SIP_ALG:E
```

To bind the SIP ALG, carry out following command:

```
:connection bind application=SIP port=5060
```

To enable the SIP ALG, carry out following commands:

```
:connection appconfig application=SIP SIP=enabled
```

2.3 Enabling the RTP_predict_for_term_SIP ALG

The RTP_predict_for_term_SIP ALG is a sort of extension of the SIP ALG. This ALG predicts the connections on which all voice related traffic, as well as signalling (SIP) as data (RTP/RTCP), will arrive. Each voice related IP packet will be processed either by the SIP ALG or forwarded by the RTP_predict_for_term_SIP ALG.

Although the RTP_predict_for_term_SIP ALG can be seen as an extension of the SIP ALG, it can be enabled or disabled separately, independent of the setting of the SIP ALG.

When the SIP ALG is not bound, you can not enable the RTP_predict_for_term_SIP ALG.

When the SIP ALG is bound, you can:

- Enable or disable the SIP ALG.
- Enable or disable the RTP_predict_for_term_SIP ALG.
- NOT disable both, the SIP ALG and RTP_predict_for_term_SIP ALG, simultaneously.

To enable the RTP_predict_for_term_SIP ALG, carry out following commands:

```
:connection appconfig application=SIP RTP_predict_for_term_SIP=enabled
```

3 General Voice Configuration

Overview

This chapter deals with the general voice parameters. The introduction gives an overview of all involved commands and parameters and necessary background information. In the second section, you can find all of the general voice parameters elaborated via CLI commands.

Chapter 3

3.1 Introduction

Overview of the voice config commands

The voice config CLI commands and parameters allows you to configure the general voice parameters.

All voice parameters are preceded with **`:voice config`**.

Following list gives an overview of the involved commands and parameters:

- “ Checking the voice configuration settings” on page 19
- “ Autofxo” on page 19
- “ Digitrelay” on page 19
- “ Click2dial_ports” on page 20
- “ Rtp_portrange” on page 20
- “ Sign_internal” on page 20
- “ Static_intf” on page 21
- “ Intf” on page 21
- “ Secondintf” on page 21
- “ Endofnumber” on page 22
- “ Countrycode” on page 22
- “ Delayeddisconnect” on page 23
- “ Delayeddisconnecttimer” on page 23
- “ Ringmuteduration” on page 23
- “ Feature-mngt” on page 24
- “ Nocallsetupmsg” on page 24
- “ Syslogscope” on page 24

FXO support levels

Foreign eXchange Office (FXO) allows you to call via the PSTN with POTS. FXO support is divided into four levels:

- Level 0: no FXO support (and thus no PSTN backup).
- Level 1: only FXO support for outgoing calls when the power is down.
- Level 2: this level is called reduced FXO. FXO support
  - When the power is down.
  - When there is no VoIP service.
  - When using forced FXO (dialling a prefix to dial via POTS)
  - For incoming FXO calls.
- Level 3: this level is called full FXO. FXO support
  - When the power is down.
  - When there is no VoIP service.
  - When using forced FXO (dialling a prefix to dial via POTS)
  - For incoming FXO calls.
  - For emergency calls without prefix dialling.

Not all Thomson Gateway models support the same FXO feature:

- ***FXO support level 2 model***

  When autofxo is enabled the Thomson Gateway will automatically switch to FXO calls (lifeline) when:
  - The power is down.
  - You are not registered to a VoIP server.

  So, you don't need to dial a prefix when you are not registered or the power is down.

  You will hear the PSTN dialling tone.

  When autofxo is disabled:
  - The Thomson Gateway will automatically switch to FXO calls when the power is down (lifeline).
  - You still need a prefix/dial plan for FXO calls when you are not registered to a VoIP server.

  You need to disable the autofxo function when you want to accomplish signal reinjection.

- ***FXO support level 3 model***

  The device will always have the VoIP service dialling tone, even when not registered to the VoIP server or when no PSTN is connected to the FXO port. The dial plan loaded in the DSP decides where your call be forwarded to (VoIP or FXO).

  When autofxo is enabled your Thomson Gateway will automatically switch to FXO calls when the power is down (lifeline).

The following table gives an overview of how calls are treated depending on the status and settings.

Status and settings					Full FXO	Reduced FXO
Power down			Call		FXO connected to FXS (via relais)	FXO connected to FXS (via relai
			Incoming FXO			
				incfxodest=fxs	Ringing to FXS	Ringing to FXS
				incfxodest=dect	Ringing to FXS	Ringing to FXS
				incfxodest=all	Ringing to FXS	Ringing to FXS
				incfxodest=none	Ringing to FXS	Ringing to FXS
			Incoming VoIP		NO	NO
			Outgoing FXO		YES	YES
			Outgoing VoIP		NO	NO
Power up						
	SIP REG OK				FXO disconnected from FXS	FXO disconnected from FXS
			Incoming FXO			
				incfxodest=fxs	Ringing to FXS	Ringing to FXS (via relais)
				incfxodest=dect	Ringing to DECT handset	No ringing to FXS nor DECT har
				incfxodest=all	Ringing to FXS and DECT handset	Ringing to FXS (via relais)
				incfxodest=none	No ringing to FXS nor DECT handset	No ringing to FXS nor DECT har
			Incoming VoIP		Ringing to destination port(s)	Ringing to destination port(s)
			Outgoing FXO		YES	YES (via relais)
			Outgoing VoIP		YES	YES
	NO SIP REG					
		autofxo=disabled			FXO disconnected from FXS	FXO disconnected from FXS
			Incoming FXO			
				incfxodest=fxs	Ringing to FXS	Ringing to FXS (via relais)
				incfxodest=dect	Ringing to DECT handset	No ringing to FXS nor DECT har
				incfxodest=all	Ringing to FXS and DECT handset	Ringing to FXS (via relais)
				incfxodest=none	No ringing to FXS nor DECT handset	No ringing to FXS nor DECT har
			Incoming VoIP		NO	NO
			Outgoing FXO		YES	YES (via relais)
			Outgoing VoIP		NO	NO
		autofxo=enabled			FXO disconnected from FXS	FXO connected to FXS (via relai
			Incoming FXO			
				incfxodest=fxs	Ringing to FXS	Ringing to FXS ports (via relais)
				incfxodest=dect	Ringing to DECT handset	Ringing to FXS ports (via relais)
				incfxodest=all	Ringing to FXS and DECT handset	Ringing to FXS ports (via relais)
				incfxodest=none	No ringing to FXS nor DECT handset	Ringing to FXS ports (via relais)
			Incoming VoIP		NO	NO
			Outgoing FXO		YES	YES (via relais)
			Outgoing VoIP		NO	NO

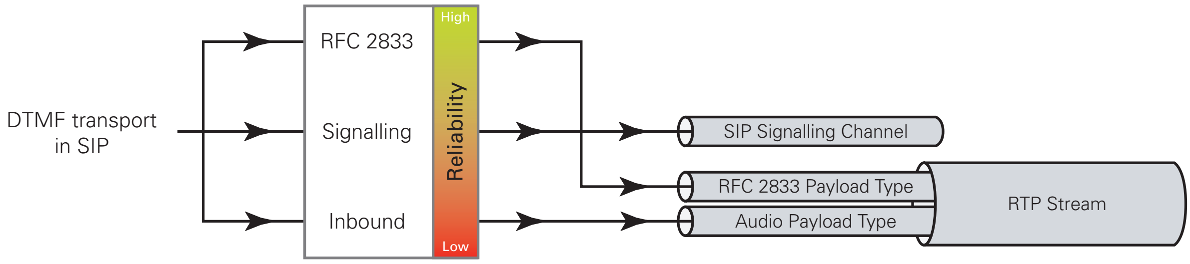
DTMF tones

In VoIP, Dual-Tone Multi-Frequency (DTMF) tones can be sent according to three methods:

- In-band, the DTMF information is sent together with the voice RTP stream. This method is the least reliable. It is advised to use this method only in combination with high bit-rate codecs (A-law and μ-law codecs, such as G.711).
- Signalling or via SIP INFO messages, the DTMF information is sent along the signalling path of the call. More information can be found in RFC 2976.
- RFC 2833, the DTMF information is sent via separate RTP payload packets. This method guarantees the highest quality and reliability.

The latter two methods are also called out-of-band.

Which method that can be used is determined by the VoIP server and the applied VoIP tool (e.g. Microsoft NetMeeting only supports in-band).



VoIP syslog messages

Syslog messages for VoIP have the following format:

- A header part containing a timestamp and hostname (IP address).
- A message part containing:
  - An indication that this syslog message is used for for a VoIP application (TAG field).
  - The product number of the Thomson Gateway.
  - An identification of the VOIP account (for SIP this is the AOR).
  - The phone line (e.g. FXS1, FXS2, DECT1,...).
  - The supported/configured/applicable syslog scope:
    - VoIP messages
    - Dial manager events
    - Statistics

VoIP messages

A syslog message is constructed at reception and sending of a VoIP message:

- An outgoing call involving a INVITE message, includes, at sending of the INVITE, in the syslog mesage the information to who the call is made. This corresponds with the SIP To header, and also reflects the possible number translation applied in the dial plan.
- An incoming call involving a INVITE message, includes, at reception of the INVITE, in the syslog message the information from who the call is made. This corresponds with the SIP From header.

Dial manager events

A syslog message will be constructed at occurrence of a dial-manager event. Dial manager events are events occurring on the local phones, and include:

- Onhook, offhook, digits 0-9, *, #.
- Hookflash.
- All tones that are generated in the RGW towards the phone.

- All local actions for supplementary services.

Statistics

A syslog message will be constructed at the end of a call, containing the following call statistics:

- Duration (number of seconds) of received/transmitted RTP stream.
- Number of received/transmitted RTP packets.
- Number of received/transmitted RTP bytes.
- Number of lost packets in received RTP stream.
- Average jitter (number of milliseconds) in received RTP stream.

3.2 Commands overview

Checking the voice configuration settings

To check the current voice configuration settings, carry out following command:

```
:voice list
autofxo          : enabled
digitrelay       : auto
click2dial_port : all
rtp_portrange    : 1024-49151
sign_internal    : external
static_interface: disabled
interface        : LocalNetwork
sec interface    :
endofnumber      : #
country code     : 0
delayed disc     : disabled
delayed disc tmr: 60
ringmuteduration: 0
feature-mngt     : internal
nocallsetupmsg   : never
syslog-scope     : none
```

The examples used in the following overview result in the above settings.

Autofxo

Syntax: autofxo=<{disabled|enabled}>

Default setting: enabled

Example:

```
:voice config autofxo=enabled
```

Explanation:

Depending on the Thomson Gateway model, the autofxo function behaves differently. The distinction between the models is based on the level of FXO support. For more information see “ FXO support levels” on page 14.

Digitrelay

Syntax: digitrelay=<{auto|inband|rfc2833|signalling}>

Default setting: auto

Example:

```
:voice config digitrelay=auto
```

Explanation:

This parameter sets how your Thomson Gateway sends DTMF tones: inband, rfc2833, signalling or auto. When auto is selected, the Thomson Gateway detects which method can be used, based on the capacity of the receiving party, starting with the highest reliability (rfc2833) and decreasing to the lowest reliability (in-band). For more information see “ DTMF tones” on page 17.

Click2dial_ports

Syntax: click2dial_ports=<{FXS1|FXS2|BT|DECT|all}>

Default setting: all

Example:

```
:voice config click2dial_port=all
```

Explanation:

The click2dial_ports parameter is used for two independent features:

- As a GUI element. With the click2dial_ports parameter you can also determine on which port you make the call when you click on a URL or on an entree in the address book. You can only select one device or all.
- For Call Completion on Busy Subscriber (CCBS). When the callee is busy and the CCBS feature is enabled, the callee will be called back when he becomes free. This parameter defines which ports will start to ring when the callee becomes available. See also " CCBS" on page 47.

Rtp_portrange

Syntax: rtp_portrange=<start port-end port> with port range: 1024-49151

Default setting: 1024-49151

Example:

```
:voice config rtp_portrange=1024-49151
```

Explanation:

The rtp_portrange parameter determines the start and the end port that can be used for the RTP and RTCP streams.
Typically, four ports are used for a call:

- Two ports for outgoing RTP and RTCP streams. The RTCP port = RTP port + 1.
- Two ports for incoming RTP and RTCP streams.

This defined port range is not reserved for RTP and RTCP streams only. Those ports can also be used by all NAPT applications. It is advised to set a port range of at least (successive) 200 ports to ensure uninterrupted VoIP services.

Sign_internal

Syntax: sign_internal=<{internal|external}>

Default setting: external

Example:

```
:voice config sign_internal=external
```

Explanation:

With this parameter you can keep the signalling for local calls local, so they are not sent via the proxy. However, you must also be registered to make internal calls.

Static_intf

Syntax: static_intf=<{enabled|disabled}>

Default setting: disabled

Example:

```
:voice config static_intf=disabled
```

Explanation:

When you want to define a dedicated interface for voice only, you must enable the static_intf. It defines the source IP selection for the voice traffic. The configuration of the interface is done with the intf parameter.

Intf

Syntax: intf=<interfacename>

Select one of the available IP interfaces.

You must enable the static_intf parameter to carry out this command. When the static_intf is disabled the intf is ignored.

Example:

```
:voice config intf=Internet
```

Explanation:

With the intf parameter, you define which IP interface must be selected for voice. The IP address of that interface will be used as the source address of the voice messages.

To view all interfaces offering IP connectivity, carry out following command

```
:ip iflist
```

It does not make sense to connect to the LOOP interface.

Secondintf

Syntax: secondintf=<interfacename>

Select one of the available IP interfaces.

You must enable the static_intf parameter to carry out this command.

Example:

```
:voice config secondintf=Internet
```

Explanation:

In case the first interface fails, a second interface can be defined. This does only make sense if a second interface is available and is different from the one defined in the parameter intf.

This parameter is only used in H.323.

Endofnumber

Syntax: endofnumber=<{#|*|none}>

Default setting: #

Example:

```
:voice config endofnumber=#
```

Explanation:

The time available between entering two digits is set by the interdigittimer parameter (see “ Interdigit” on page 80). By default this parameter is set to five seconds. This implies that the dialling of the number starts after the interdigittimer value.

When entering the endofnumber symbol the dialling starts immediately.

Countrycode

Syntax: countrycode=<number {0-900}>

You must enable the globalnumbpostprocess parameter to enable this parameter.

To enable the globalnumbpostprocess parameter, carry out following command:

```
:voice numbtransl config globalnumbpostprocess enabled
```

Default setting: 0

Example;

```
:voice config countrycode=0
```

Explanation:

The number you enter is the local country code. This parameter enables local country code detection and translation. This enhances following process:

- When the number is preceded by the + symbol, the + symbol is removed.
- When the first three digits of the number match the local country code, then these digits will be removed and a 0 will be added before the first digit.
  Examples:
  - The number +445612345678 becomes 05612345678 in the UK.
  - The number +335612345678 becomes 05612345678 in France.
- When the first three digits don’t match the country code, then the prefix for international calls will be added.
  Examples:
  - The number +445612345678 becomes 00445612345678 in France.
  - The number +335612345678 becomes 00335612345678 in the UK.

When the countrycode is set to 0, this feature is disabled.

Delayeddisconnect

Syntax: delayeddisconnect=<{enabled|disabled}>

Default setting: disabled

Example:

```
:voice config delayed_disc=disabled
```

Explanation:

This parameter allows you to hang up the phone and continue the call on another phone that is connected to the same profile. This feature only works when you have receive a call and not when you have initiated the call

You can set the time how long it takes to finish the call when hanging up with the delayeddisconnecttimer parameter.

Delayeddisconnecttimer

Syntax: delayeddisconnecttimer=<number{1-600}> expressed in seconds

Default setting: 60

Example:

```
:voice config delayeddisconnecttimer=60
```

Explanation:

This timer delays the BYE signal with the value in seconds stated in the configuration.

This parameter is recommended when you want to accomplish signal reinjection.

Ringmuteduration

Syntax: ringmuteduration=<number{0-60000}> expressed in milliseconds

Default setting: 0

Example:

```
:voice config ringmuteduration=0
```

Explanation:

This parameter sets the time between an incoming 18x message and the local ring-back signal to be activated.
When the ringmuteduration time expires:

- and the remote party has sent a ring-back tone (via the RTP stream), that ring-back tone will be heard.
- and the remote party has NOT sent a ring-back tone, your user agent will generate a ring-back tone.

When ringmuteduration is set to 0, this feature is disabled.

Feature-mngt

Syntax: feature-mngt=<{internal|external}>

Default setting: internal

Example:

```
:voice config feature-mgnt=internal
```

Explanation:

With this parameter you can choose to run the supplementary services logic locally in the Residential GateWay (RGW) or in the VoIP network:

- Internal: in the RGW
- External: in the VoIP network

Nocallsetupmsg

Syntax: nocallsetupmsg=<{unav-numb|never}>

Default setting: never

Example:

```
:voice config nocallsetupmsg=never
```

Explanation:

If the parameter nocallsetupmsg is set to

- unav-numb, then no call setup message will be sent (on FXS, DECT interface, ...) in case the number is unavailable.
- never, then there will always be sent a call setup message, provided CLIP is activated locally.

Syslogscope

Syntax: syslogscope=<{none|all|only-stats|only-dm-events|only-msgs}>

Default setting: none

Example:

```
:voice config syslogscope=none
```

Explanation:

With the syslogscope parameter you can set which Volp syslog messages should be constructed:

- none: no syslog messages will be constructed for VoIP
- only-stats: a syslog message will be constructed at the end of a call, containing the call statistics as defined above
- only-dm-events: a syslog message will be constructed at occurrence of a dial-manager event
- only-msgs: a syslog message will be constructed at reception and sending of a VOIP message
- all: syslog messages will be constructed at all occurences of only-stats, only-dm-events and only-msgs

For more information, see “ VoIP syslog messages” on page 17.

4 SIP Configuration

Overview

This chapter deals with the typical SIP parameters. The introduction gieves an overview of all involved commands and parameters and the necessary background information. A second section describes how basic SIP configuration parameters can be configured via the Web pages. Finally, in the third and fourth section, you can find all of the SIP related parameters elaborated via CLI commands.

- “4.1 Introduction” on page 26.
- “4.2 Basic SIP configuration” on page 28.
- “4.3 SIP configuration commands overview” on page 29.
- “4.4 SIP response map” on page 42

Chapter 4

4.1 Introduction

Overview of the SIP configuration commands

The SIP config CLI commands and parameters allows you to configure the typical SIP parameters.

All SIP configuration parameters are preceded with **`:voice sip config`**.

Following list gives an overview of all the involved commands and parameters:

Address of Record and SIP URI

The Address of Record (AoR) or SIP URI is the public address of a SIP user agent. The AoR is text based and uses http and is similar to a typical e-mail address. The structure of an AoR is ***user name@domain name***. The user name is either a name, number or a combination of both. The domain name is either the IP address and port number or host name.

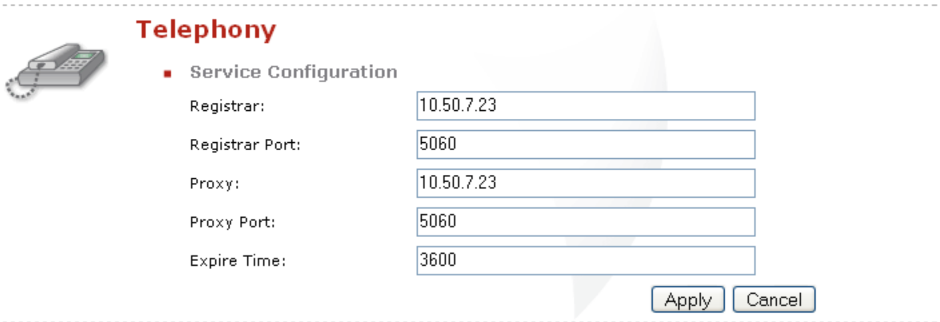
When the term SIP URI is used in the GUI or CLI, it only refers to the user name of the AoR and NOT to the complete AoR or SIP URI.

4.2 Basic SIP configuration

Web pages

You can configure some of the SIP parameters via the Web pages (GUI). Proceed as follows:

1 On the Thomson Gateway home page, click **Toolbox**
2 Click **Telephony.**
3 Click **Expert Configure.**
4 In **Pick a task**, click **View Telephony Services.**
5 Type the parameters (for information about the parameters, see “4.3 SIP configuration commands overview” on page 29):



6 Click **Apply.**

4.3 SIP configuration commands overview

Checking the configuration

To check the voice sip configuration, carry out following command:

```
:voice sip list
UserAgent domain              :
Primary proxy address         : 10.50.2.204:5060
Secondary proxy address       : 0.0.0.0:5060
Primary registrar address     : 10.50.2.204:5060
Secondary registrar address   : 0.0.0.0:5060
Expire time                   : 1800
Expire time delta             : 5
Notifier address              : 10.50.2.200:5060
Subscribe expire time         : 1800
Call Waiting reply            : 182
Transport                     : UDP
rtpmapstaticPT                : Disabled
reinvite_stop_audio           : Disabled
PRACK                         : Disabled
Clir format                   : standard
DTMF */# in INFO method       : 1011
Clip consider displayname     : yes
SDP packet time               : 20
Replace #                     : Enabled
Symmetric codec               : Enabled
Reinvite at calling fax detect : Disabled
SIPURI port                   : Enabled
rport                         : Disabled
SDP username                  : THOMSON_GATEWAY
ringtoneat183                 : Disabled
T38 Port increment            : 0
Ping timer                    : 0
Min SE timer                  : 0
Session expires timer         : 0
Expires timer                 : 0
Register backoff timeout      : 0
Sticky outbound proxy         : Enabled
Privacy Header                : ignore
SDP username per ua           : Disabled
Stop register on 403          : Disabled
```

The examples used in the following overview result in the above settings.

Useragentdomain

Syntax: useragentdomain=<string>

Example:

```
:voice sip config useragentdomain=
```

Explanation:

Typically, the domain name of the AoR is the primary registrar address. Because of logical reasons, e.g. to keep an overview, a user agent domain can be defined. When you define the user agent domain it will be used as the domain name of your AoR instead of the primary registrar address.

When the user agent domain is not set, the domain name of the AoR address will be filled in with the primary registrar address.

Primproxyaddr

Syntax: primproxyaddr=<string>

Example:

```
:voice sip config primproxyaddr=10.50.2.204
```

Explanation:

The primary proxy address defines the (outbound) proxy server. The string must be an IP address or Fully Qualified Domain Name (FQDN) string.

The parameters primproxyaddr and proxyport are defined at the same time.

Secproxyaddr

Syntax: secproxyaddr=<string>

Example:

```
:voice sip config secproxyaddr=10.50.3.126
```

Explanation:

In case the primary proxy address fails, a second proxy address can be defined. This does only make sense if a second proxy address is available and is different from the one defined in the parameter primproxyaddr.

Proxyport

Syntax: proxyport=<{SIP|...} or number>

Default setting: 5060

Example:

```
:voice sip config proxyport=5060
```

Explanation:

The proxy port is the outbound proxy port for the primary proxy address.

The parameters primproxyaddr and proxyport are defined at the same time.

Secproxyport

Syntax: proxyport=<{SIP|...} or number>

Default setting: 5060

Example:

```
:voice sip config secproxyport=5060
```

Explanation:

The second proxy port is the outbound proxy port for the secondary proxy address.

Primregaddr

Syntax: primregaddr=<string>

Example:

```
:voice sip config primregaddr=10.50.2.204
```

Explanation:

The primary registrar address is the server to which the user agents register. The string must be an IP address or FQDN string. When no user agent domain is defined the primary registrar address is used as domain name of the AoR.

The parameters primregaddr and regport are defined at the same time.

Secregaddr

Syntax: secregaddr=<string>

Secregaddr is available as parameter, but up till now it is not supported in any configuration.

Regport

Syntax: regport=<{SIP|...} or number>

Default setting: 5060

Example:

```
:voice sip config regport=5060
```

Explanation:

The registrar port is outbound registrar port for the primary registrar address.

The parameters primregaddr and regport are defined at the same time.

Regexpire

Syntax: regexpire=<number{60-65535}> expressed in seconds

Default setting: 3600

Example:

```
:voice sip config regexpire=1800
```

Explanation:

The expire time is the time after which the user agent will register again with the registrar.

Regexpire_Tbefore

Syntax: regexpire_Tbefore=<number{1-60}> expressed in seconds

Default setting: 1

Example:

```
:voice sip config regexpire_Tbefore=5
```

Explanation:

The expire time delta is the time to send a new registration message before the registration time expires (as defined in the “ Regexpire” on page 32).

Notifier_ addr

Syntax: notifier_addr=<string>

Default setting: the primary proxy address

Example:

```
:voice sip config notifier_addr=10.50.204.200
```

Explanation:

The string must be an IP address or FQDN string. The notifier address is the address of the server generating the MWI tone or the voice-mail server.

When the Message Waiting Indicator (MWI) service is provisioned and activated and the mwi_network parameter is set to sollicited you must define the notifier address and port. For more information, see “5 Supplementary services” on page 45.

Notifier_port

Syntax: notifier_port=<{SIP|...} or number>

Default setting: 5060

Example:

```
:voice sip config notifier_port=5060
```

Explanation:

The notifier port is the outbound port used by the the server generating the MWI tone or the voice-mail server. For more information, see “ Secregaddr” on page 31.

Subscribe_expire

Syntax: subscribe_expire=<number{60-65535}> expressed in seconds

Default setting: 3600

Example:

```
:voice sip config subscribe_expire=1800
```

Explanation:

The subscription expire time is the time after which the subscription to the notifier address expires and no more NOTIFY messages will be sent to the SIP user agent (in case of sollicited MWI).

CWreply

Syntax: cwreply=<180|182>

Default setting: 180

Example:

```
:voice sip config cwreply=182
```

Explanation:

Assume an active call between A and B, and A is called by C:

When user agent A has activated the call waiting service:

- UA A is informed of the incoming call
- UA C will receive a 180 or 182 response. 180 indicates the ring tone, 182 indicates the queued tone.

Transport

Syntax: transport=<{UDP|TCP}>

Default setting: UDP

Example:

```
:voice sip config transport=udp
```

Explanation:

The used transport protocol.

The option TCP is not supported yet in the current Thomson Gateway releases.

RtpmapstaticPT

Syntax: rtpmapstaticPT = <{disabled|enabled}>

Default setting: disabled

Example:

```
:voice sip config rtpmapstaticPT=disabled
```

Explanation:

Enable or disable the adding of RTP mappings (mapping PT number with codec) in the Session Description Protocol (SDP) body.
Static payload types are designated to specific RTP mappings. Although not required, they can also be mapped to RTP encodings using the rtpmapstatic PT parameter.

For dynamic payload types the mapping is always sent because the payload values are predefined.

Reinvite_stop_audio

Syntax: reinvite_stop_audio=<{enabled|disabled}>

Default setting: disabled

Example:

```
:voice sip config reinvite_stop_audio=disabled
```

Explanation:

When this parameter is enabled, the RTP audio stream is stopped when a reinvite message is sent.

PRACK

Syntax: PRACK=<{enabled|disabled}>

Default setting: disabled

Example:

```
:voice sip config PRACK=disabled
```

Explanation:

The provisional acknowledge insures reliability of provisional 1XX responses (when the UAS offers them).

For more information on PRACK, see RFC 3262.

Clirformat

Syntax: clirformat=<{standard|nonstandard}>

Default setting: standard

Example:

```
:voice sip config clirformat=standard
```

Explanation:

When the Calling Line Identification Restriction (CLIR) format is set to standard following address appears in the From: anonymous<SIP:anonymous@domain name>.

When the CLIR format is set to non-standard following address appears in the From: anonymous<SIP:user name@domain name>.

The non-standard setting might be required for billing reasons.

DTMF*/#inINFO

Syntax: DTMF*#inINFO=<{*#|1011}>

Default setting: 1011

Example:

```
:voice sip config DTMF*#inINFO=1011
```

Explanation:

When a sending DTMF event is detected, the SIP INFO message that will be sent depends on the setting:

DTMF event	SIP INFO signal	
	setting=*#	setting=1011
*	*	10
#	#	11

Clip_consider_displayname

Syntax: clip_consider_displayname={no|yes}>

Default setting: yes

Example:

```
:voice sip config clip_consider_name=yes
```

Explanation:

When yes is selected the Caller ID is private when the displayname, user or host is anymous.

When no is selected the Caller ID is private when the user or host is anynomous. the displayname is not taken into account.

Sdp_ptime

Syntax: sdp_ptime=<{10|20|30|notsent}> expressed in milliseconds

Default setting: 20

Example:

```
:voice sip config sdp_ptime=20
```

Explanation:

The SDP packet time defines the length of an RTP packet, which is typically 20 ms. This parameter is sent in a SDP message.

Replace#

Syntax: replace#=<{enabled|disabled}>

Default setting: enabled

Example:

```
:voice sip config replace#=enabled
```

Explanation:

When enabled the # in an INVITE message will be replaced with %23.

Symmetriccodec

Syntax: symmetriccodec=<{enabled|disabled}>

Default setting: enabled

Example:

```
:voice sip config symmetriccodec=enabled
```

Explanation:

When this parameter is enabled the first codec in the priority list of the caller which is shared with the callee will be used as codec.

When this parameter is disabled the codec defined in **`:voice codec list`** will be used.

Mind that most gateways do not support asymmetrical use of codec. For more information about codecs, see "6.5.2 Configuring codecs" on page 77.

Reinvite_at_cgfax_detect

Syntax: reinvite_at_cgfax_detect=<{enabled|disabled}>

Default setting: disabled

Example:

```
:voice sip config reinvite_at_cgfax_detect=disabled
```

Explanation:

The reinvite_at_cgfax_detect parameter is by default disabled because the reinvite should be sent by the callee to make sure that both sides are faxes. After detection of a fax there will always be sent a reinvite to G.711 or T.38.

When the fax transport is set to T38 or inband_reneg, reinvite when calling fax is detected must be enabled.

SIPURI_port

Syntax: SIPURI_port=<{enabled|disabled}>

Default setting: enabled

Example:

```
:voice sip config SIPURI_port=enabled
```

Explanation

When the SIP URI port is enabled the port number will be included in the request line for outbound messages. This is especially advised when the SIP URI port is different from 5060, because this is the default port that will be used when the parameter is disabled.

Rport

Syntax: rport=<{enabled|disabled}>

Default setting: disabled

Example:

```
:voice sip config rport=disabled
```

Explanation:

The receive port should be enabled when no SIP ALG is applied.

SDP_username

Syntax: SDP_username=<string>

Default setting: 780

Example:

```
:voice sip config SDP_username=THOMSON_GATEWAY
```

Explanation

The SDP username is applied on all SIP UA's. By default the SDP username is based on the product number (e.g. 716v5, 780, ...). In case no product number is available, the – is used.

Ringtoneat183

Syntax: ringtoneat183=<{enabled|disabled}>

Default setting: disabled

Example:

```
:voice sip config ringtoneat183=disabled
```

Explanation

This parameter enables or disables local ringing when 183 SESSION PROGRESS is received. To prevent that early media is heard together with the local ringing tone this parameter must be disabled.

T38portincrement

Syntax: t38portincrement=<number{0-65535}>

Default setting: 0

Example:

```
:voice sip config t38portincrement=0
```

Explanation

This parameter defines the offset of the T.38 port number compared to the RTP port number. When the parameter is set to 0 a random port will be used.

Ping

Syntax: ping=<number{0-86400}> expressed in seconds

Default setting: 0

Example:

```
:voice sip config ping=0
```

Explanation

The ping timer defines the time between two keep-alive ping requests. When the parameter is set to 0 no SIP ping will be sent.

Min-se

Syntax: min-se=<number{0-604800}> expressed in seconds

Default setting: 0

Example:

```
:voice sip config min-se=0
```

Explanation

The minimum session expires timer defines the minimum negotiation time before a session expires from UA's. For more information on session timers, see RFC 4028.

Session-expires

Syntax: session-expires=<number{0-604800}> expressed in seconds

Default setting: 0

Example:

```
:voice sip config session-expires=0
```

Explanation

The session expires timer defines the time after which reinvite messages has to be sent to keep a session alive. For more information on session timers, see RFC 4028.

Expires timer

Syntax: expires=<number{0-3600}> expressed in seconds

Default setting: 0

Example:

```
:voice sip config expires=0
```

Explanation

The INVITE expires timer to final 200 OK message. When the expire time is reached the call will be cancelled (CANCEL). When the timer is set to 0, no expire is used.

Register-backoff-timer

Syntax: register-backoff-timer=<number{0-86400}>

Default setting: 0

Example:

```
:voice sip config register-backoff-timer=0
```

Explanation

Fixed backoff timer in SIP registration procedure.

Stickyoutbproxy

Syntax: stickyoutbproxy=<{disabled|enabled}>

Default setting: enabled

Example:

```
:voice sip config stickyoutbproxy=enabled
```

Explanation

The parameter stickyoutbproxy sets how to handle outband proxy lock over transactions:

- When sticky outband proxy is enabled, the INVITE is sent to the same IP address as used for REGISTER. On other NON-REGISTER messages, if the configured outbound proxy is a Fully Qualified Domain Name (FQDN), DNS is used for resolving purposes, and thus these messages can be sent to another IP address.
- With sticky outband proxy disabled, if the configured outbound proxy is a FQDN, DNS is used for resolving purposes. The IP address out of the received DNS reply is used used as proxy address to send the SIP message to and can differ from the one used for REGISTER .

Privacy

Syntax: privacy=<{ignore|strict|loose}>

Default setting: ignore

Example:

```
:voice sip config privacy=ignore
```

Explanation

If the privacy is set to

- ignore, then the privacy header is ignored.
- strict, then the privacy header is applied in a strict way:
  - If “P-Asserted-Id” is taken into account, it is only checked if privacy header has value = ID. If indeed privacy header = ID, then caller ID and caller name is private (i.e. restricted).
  - If “From” is taken into account, it is only checked if privacy header has value = user. If indeed privacy header = user, then caller ID and caller name is private (i.e. restricted).
- loose, then the Privacy header is applied in a loose way :
  - Regardless if “P-Asserted-Id” or “From” is taken into account, it is checked if privacy header has value = ID or user. If indeed privacy header = ID or user, then caller ID and caller name is private (i.e. restricted).

SDP_username_per_UA

Syntax: SDP_username_per_UA=<{disabled|enabled}>

Default setting: disabled

Example:

```
:voice sip config SDP_username_per_UA=disabled
```

Explanation

When the parameter SDP_username_per_UA is enabled, the global configuration for the SDP username is ignored. In this case the SDP username is populated with the username configured in the SIP UA profile. If the username is empty, the username becomes “–“.

Stop_register_on_403

Syntax: stop_register_on_403=<{disabled|enabled}>

Default setting: disabled

Example:

```
:voice sip config stop_register_on_403=disabled
```

Explanation

With this parameter you can enable/disable the stopping with registering when a SIP message 403 is returned by the SIP registrar server. SIP message 403 (forbidden) means the server understood the request but is refusing to fulfill it.

4.4 SIP response map

Overview

On reception of a specific SIP response code a specific tone and/or textmessage (for DECT handsets) can be given to the local user. These mappings are configured in the SIP response map.

Textmessages are not supported yet in the current Thomson Gateway software.

The tone is specified in the tone description table and can consist of a tone pattern, a tone file and a tone text. For more information about tones, see “6.11 Tones” on page 90.

The wildcard x is supported. E.g. SIP response code 5xx identifies all response codes in the 5xxx series.

Viewing the SIP response map

To view the SIP response map, carry out the following command:

```
:voice sip responsemap list
Resp. Code | Tone                                | Text message
-------------------------------------------------------------------------------------------
       1xx | none                                |
       180 | ringback                            |
       181 | ringback                            |
       182 | ringback                            |
       183 | ringback                            |
       2xx | none                                |
       3xx | none                                |
       4xx | warning                             |
       486 | busy                                |
       5xx | warning                             |
       6xx | warning                             |
```

Adding a SIP response

To add a SIP response, carry out the following command:

```
:voice sip responsemap add
responsecode = <{string}>
     The SIP response code.
tone = <{dial|noen|remotecallhold|callhold|remotecallwaiting|callwaiting|rejection|confirmat
ion|release|warning|congestion|busy|ringback|mwi|specialdial|stutterdial}>
     The tone to be played.
textmessage = <{quoted string}>
     The text message to display. Only applicable for DECT handsets.
```

Textmessages are not supported yet in the current Thomson Gateway software.

Modifying SIP response

To modify a SIP response, in this example a textmessage is added to the response 486, carry out the following command;

```
:voice sip responsemap modify
responsecode = 486
[tone] =
[textmessage] = Busy
:voice sip responsemap modify responsecode=486 textmessage=busy
```

Deleting a SIP response

To delete a SIP response, carry out the following command:

```
:voice sip responsemap delete
resonsecode= <{string}>
      The SIP response code.
```

The responsecode must be listed in the SIP response map.

Flushing the SIP responsemap

To flush all SIP responses, carry out the following command:

```
:voice sip responsemap flush
```

5 Supplementary services

Overview

This chapter deals with the supplementary services. The introduction starts with an overview of all of the supplementary services and background information on them. The second section describes how the supplementary services are configured via CLI commands and, where applicable, via the Web pages (GUI).

- "5.1 Introduction" on page 46.
- "5.2 Commands Overview" on page 49.

5.1 Introduction

Summary of the supplementary services

Following list summarises the supplementary services available:

- Acr: Anonymous call rejection
- 3pty: three party call
- Callreturn: to call back the last missed call
- Ccbs: call completion on busy subscriber (automatic callback)
- Cfbs: call forwarding on busy
- Cfnr: call forwarding on no reply
- Cfu: call forwarding unconditional
- Clip: calling line identification presentation
- Clir: calling line identification restriction
- Clironcall: clir for only one call
- ForcedFXO: switch to FXO (PSTN)
- Hold: put an active call on hold
- Mwi: message waiting indication
- Transfer: call transfer between local ports
- Waiting: incoming call while active call indication
- Waitingoncall: call waiting active for only one call

In order to use the services they must be provisioned by the provider. Only the provisioned services can be activated or deactivated locally using the Web pages (GUI) or activation and deactivation code. Withdrawn service can not be activated.

Default settings

Following services are by default provisioned and activated:

- 3pty
- Clip
- Hold
- Transfer
- Waiting
- Waitingoncall

Transparent and local

When supplementary services are transparent they are managed at provider level. When they are local they can be activated by an individual user.

Example:
When the CLIR function is activated locally, the provider receives a request from "anonymous<sip:anonymous@domain_name" and does not know who to charge. For this reason the clir function is made transparent. To use the clir function the caller has to use a predefined code (by the provider) implying that he wants to use the clir function. The provider will enable to clir function towards the callee. So, towards the provider the clir function can not be used, only requested to be applied towards the caller.

Provisioning

When a service is:

- provisioned by the provider, it is local and can be activated/deactivated by an individual user.
- withdrawn by the provider, it is transparent and can not be activated/deactivated by an individual user.

To manage all supplementary services at provider level, services must be provisioned/withdrawn as follows:

- Must be provisioned:
  - Transfer (default)
  - Hold (default)
  - Waiting (default)
  - Clip (default)
- Can be provisioned:
  - 3pty (default)
  - waitingoncall (default)
  - mwi
  - ccbs
- Must be withdrawn:
  - Acr
  - Clir
  - ForcedFXO
  - Clironcall
  - Cfu
  - Cfnr
  - Cfbs

CCBS

Call Completion on Busy Subscriber (CCBS) is also referred to as automatic callback. Is enables user A calling user B, who is busy, to get a callback when user B becomes available. When user B goes on-hook, and does not initiate a call within the idle-guard timer, user B is said to become available. The idle-guard timer allows user B to initiate a call before receiving a CCBS. The idle-guard-timer is not configurable and is set to 10 seconds.

When user B becomes available, user A will be notified. The free phones at user A will ring as configured by the CLI (not by the GUI). The CLI command to define which ports must ring is described in “ Click2dial_ports” on page 20. The phone that goes off hook first will generate a ringing tone and a call is set up to user B.

When user A is involved in a call when user B becomes available (user A is notified by a SIP NOTIFY message), user A will not be informed at the time user B becomes available, even when call waiting is activated (see “ Call Waiting” on page 48).

The CCBS request is removed when no off-hook at user A is detected within the CCBS-recall-timer. The CCBS-recall-timer is not configurable and is set to 10 seconds.

Message Waiting Indication

To configure the Message Waiting Indication(MWI) feature, proceed as follows:

1 Provision the MWI feature, see “5.2.2 Provisioning and withdrawing services” on page 51.

2 Activate the MWI feature, see “5.2.3 Activating and deactivating services” on page 52.

3 Select the MWI network variant, see “ Mwi_network” on page 58.

   If you select:

   - The unsollicited variant proceed with step 7 in this procedure.
   - The sollicited variant proceed with step 4. in this procedure.

4 Define the notifier address, see “ Notifier_ addr” on page 33.

5 Define the notifier port, see “ Notifier_port” on page 33.

6 Define the subscription expire time, see “ Subscribe_expire” on page 33.

7 Select how the phone handles the MWI message, see “ Mwi_phone” on page 58.

Call Transfer

Two variants of call transfer exist:

- Blind call transfer: the caller will get forwarded to the destination number directly, without offering him the opportunity to talk to the person at the destination.
- Attended call transfer or call transfer with consultation: before a caller gets forwarded to the destination number, he will have the opportunity to talk to the person at the destination.

The blind call transfer variant is not supported.

Call Waiting

When you activate call waiting two channels per profile will be assigned. This also implies that when:

- The profile is mapped to a single physical port for voice device (i.e. FXS1, DECT1),
- Cfbs is activated,
- You have an active calling (or call on hold) and
- A new call is coming in,

the call will not be forwarded because there is still a channel available. Only when you are involved in a conference call, the call will be forwarded to the call forward destination.

5.2 Commands Overview

Overview

In this section all of the commands concerning supplementary services will be elaborated. You can find information on:

- "5.2.1 Viewing the supplementary services" on page 50.
- "5.2.2 Provisioning and withdrawing services" on page 51.
- "5.2.3 Activating and deactivating services" on page 52.
- "5.2.4 Assigning Service Codes" on page 57.
- "5.2.5 Configuring supplementary services" on page 58.

- Flushing all supplementary services:

```
:voice services flush
```

When you provision or activate a supplementary service, that service is provisioned or activated for all profiles.

5.2.1 Viewing the supplementary services

Viewing the supplementary services

When you list the supplementary services via the CLI all of the services are displayed, whether they are provisioned or not, activated or not. To view the status of the supplementary services via CLI, type:

```
:voice services servicelist
Service              Status   ActCode  DeactCode  RegCode  ActAndRegCode  IntCode  Provisioned
-----------------------------------------------------------------------------------------------
transfer          activated      *96       #96                             *#96          yes
hold              activated      *94       #94                             *#94          yes
waiting           activated      *43       #43                             *#43          yes
mwi             deactivated      *98       #98                             *#98           no
clip              activated      *30       #30                             *#30          yes
clir            deactivated      *31       #31                             *#31           no
acr             deactivated      *15       #15                             *#15           no
3pty              activated      *95       #95                             *#95          yes
forcedFXO       deactivated     *01*                                                      no
cfu             deactivated      *21       #21     **21*          *21*     *#21           no
cfnr            deactivated      *61       #61     **61*          *61*     *#61           no
cfbs            deactivated      *67       #67     **67*          *67*     *#67           no
callreturn      deactivated      *69                                                      no
ccbs            deactivated      *37       #37                             *#37           no
clironcall      deactivated     *31*      #31*                                            no
waitingoncall     activated     *43*      #43*                                           yes
```

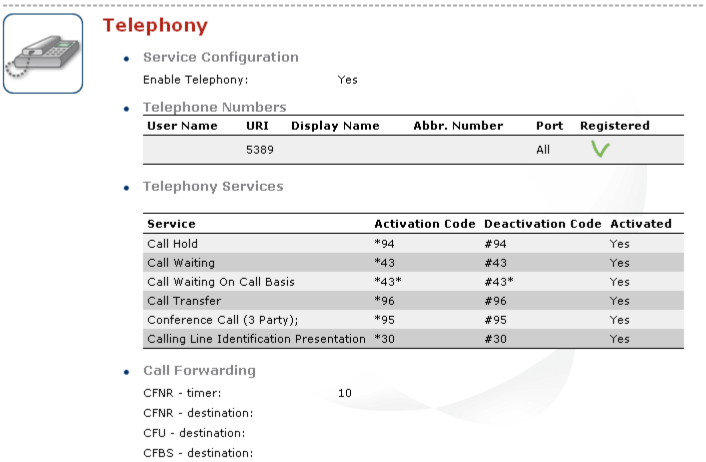
You can also view the supplementary services via the Web pages (GUI). Only the services that are provisioned are displayed. To view the status of the supplementary services via the Web pages (GUI):

1 On the Thomson Gateway home page, click **Toolbox**.

2 Click **Telephony**.

3 Click **Details**.

4 In **Pick a task**, click **View Telephony Services**.


Telephony

- Service Configuration

Enable Telephony: Yes

- Telephone Numbers

User Name	URI	Display Name	Abbr. Number	Port	Registered
	5389			All	

- Telephony Services

Service	Activation Code	Deactivation Code	Activated
Call Hold	*94	#94	Yes
Call Waiting	*43	#43	Yes
Call Waiting On Call Basis	*43*	#43*	Yes
Call Transfer	*96	#96	Yes
Conference Call (3 Party);	*95	#95	Yes
Calling Line Identification Presentation	*30	#30	Yes

- Call Forwarding

CFNR - timer: 10

CFNR - destination:

CFU - destination:

CFBS - destination:

You can enable Web pages (GUI) via MLP.

5.2.2 Provisioning and withdrawing services

Provisioning services

To provision a supplementary service, in this example clir, type following command:

```
:voice services provision type=clir
```

Following supplementary services are dependent on other supplementary services:

- Provisioning 3pty requires hold to be provisioned.
- Provisioning transfer requires hold to be provisioned.
- Provisioning waiting requires hold to be provisioned.
- To provision clironcall, you must provision clir. Clironcall can not be provisioned directly.
- To provision waitingoncall, you must provision waiting. Waitingoncall can not be provisioned directly.

Withdrawing services

To withdraw a supplementary service, in this example clir, type following command:

```
:voice services withdraw type=clir
```

If you withdraw clir, clironcall is also withdrawn.

If you withdraw waiting, waitingoncall is also withdrawn.

5.2.3 Activating and deactivating services

Various ways

Supplementary services must be provisioned in order to activate them.

You can activate or deactivate supplementary services in four ways:

- " Using CLI commands".
- " Using the Web pages".
- " Using Service Code Commands".
- " Using Switching Order Commands"

You can only apply the services callreturn, clironcall, hold and transfer on an active call.

Dependent supplementary services

Following supplementary services are dependent on other supplementary services:

- Activating 3pty requires hold to be activated.
- Activating transfer requires hold to be activated.
- Activating waiting requires hold to be activated.
- To activate/deactivate clironcall, you must provision clir. Clironcall can not be activated/deactivated directly.
- To activate/deactivate waitingoncall, you must provision waiting. Waitingoncall can not be activated/deactivated directly.

Using CLI commands

To activate a supplementary service, in this example hold, carry out following command:

```
:voice services activate type=hold
```

Before you can activate the supplementary services cfu, cfbs you must configure the destination. For cfnr you must configure the destination and timer. To do so, see "5.2.5 Configuring supplementary services" on page 58.

To deactivate a supplementary service, in this example hold, carry out following command:

```
:voice services deactivate type=hold
```

Using the Web pages

Only supplementary services that are provisioned are shown in the Web pages (GUI).

To activate or deactivate a supplementary service, proceed as follows:

1. On the Thomson Gateway home page, click **Toolbox**
2. Click **Telephony**.
3. Click **Details**.
4. In **Pick a task**, click **View Telephony Services.**
5. Click **Configure**.

6 Select or clear the check box of the supplementary service you would like to activate or deactivate.

- Telephony Services

Service	Activation Code	Deactivation Code	Activated
Call Hold	*94	#94	☑
Call Waiting	*43	#43	☐
Call Transfer	*96	#96	☑
Conference Call (3 Party);	*95	#95	☐
Calling Line Identification Presentation	*30	#30	☑
Calling Line Identification Restriction	*31	#31	☐
Message Waiting Indicator	*98	#98	☐
Call Forwarding on Busy Subscriber	*67	#67	☐

7 Click **Apply**.

Before you can activate the supplementary services cfu, cfbs you must configure the destination. For cfnr you must configure the destination and timer. To do so, see “5.2.5 Configuring supplementary services” on page 58.

Using Service Code Commands

To activate or deactivate a supplementary service, in this example hold, proceed as follows:

1 Take the phone of the hook.

2 Dial *94 (activate) or #94 (deactivate).

3 Wait for the confirmation tone (or rejection tone).

4 Hang up.

To view all of the activation codes and deactivation codes see “5.2.1 Viewing the supplementary services” on page 50.

To assign activation and deactivation codes see “5.2.4 Assigning Service Codes” on page 57.

You can check whether of not the service, in this example hold, is successfully activated by dialing *# 94 afterwards.

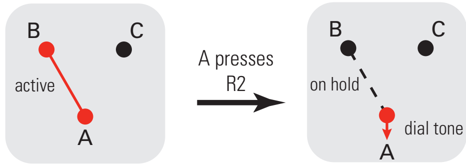
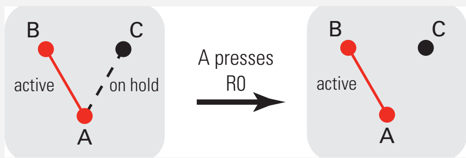
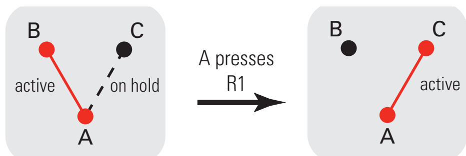
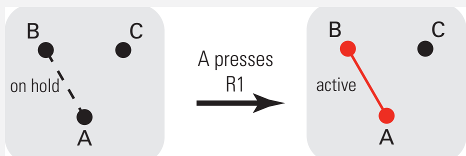
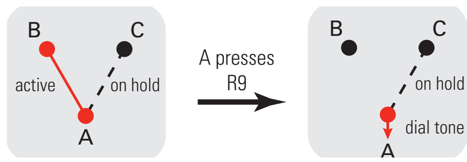
Using Switching Order Commands

A Switching Order Command (SOC) consists of a Hook Flash (HF) or Register recall (R) followed by a digit. In order to use SOCs, certain supplementary services have to be activated. These are described in the following tables using a three party service with:

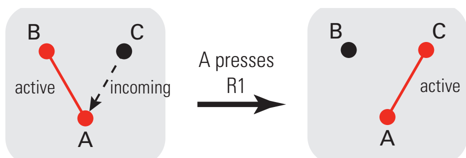
- User A the original caller.
- User B the first involved in the conference.
- User C the last involved in the conference.

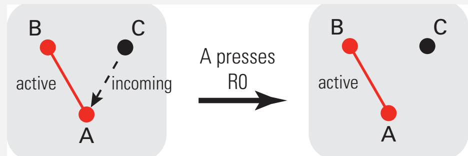
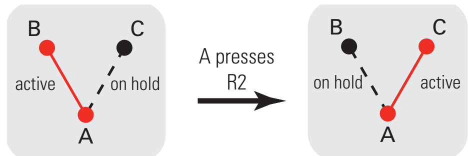
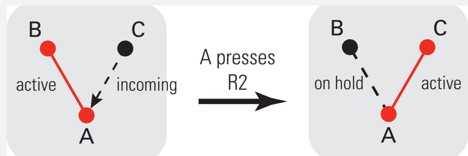
When B is disconnected, C becomes the new user B

- Activate **Hold** when you want

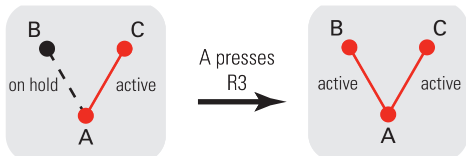
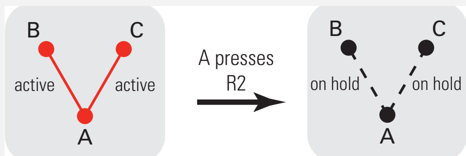
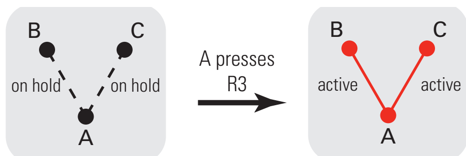
Action	Press	Illustration
To put an active call on hold and enable a call set up (the dial tone is generated)	R, 2	
To terminate the call on hold	R, 0	
To terminate an active call and switch to the call on hold	R, 1	
To retrieve the call on hold (when there is no active call)	R, 1	
To terminate an active call and enables a call set up (the dial tone is generated)	R, 9	

- Activate **Waiting** when you want to:

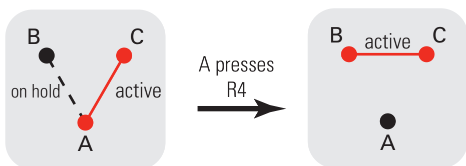
Action	Press	Illustration
To terminate an active call and switch to an incoming call	R, 1	

Action	Press	Illustration
To reject an incoming call	R, 0	
To switch between an active call and a call on hold	R, 2	
To switch between an active call and an incoming call	R, 2	

- Activate **3 PTY** when you want to:

Action	Press	Illustration
To establish a conference call (or 3 party connection)	R, 3	
During 3 pty: put B and C on hold	R, 2	
During 3 pty: retrieve B and C	R, 3	

- Activate **Transfer** when you want to:

Action	Press	Illustration
Transfer a call	R, 4	

- Activate **CCBS** when you want to set up a call to a busy subscriber (Press R, 5).

- An empty switching order is also possible, i.e. a hook flash without a switching order digit. To differentiate a hook flash + switching order and a hook flash + called number, a HookFlash or RegisterRecall timer is used. When the RegisterRecall timer expires, the SOC has an ‘empty’ switching order. The HookFlash timer is defined in the country custom file.
  The result of this function is holding an active call and enabling a call set-up (the dial tone is heard) when the HookFlash timer has expired.

Switching order commands R + 5 up to R + 9 are NOT supported.

5.2.4 Assigning Service Codes

Two categories

By default, the ETSI service codes are configured for the supplementary services, but you can assign other service codes. Depending on the way you assign service codes there are two categories:

- Standard: this implies that the activation code and deactivation code are the same, except for the prefix.
- Non-standard: this implies that the activation code and deactivation code are different.

Both categories can be configured. You can select a category with the prefix_serv_code command. For more information see “ Prefix_servcode” on page 59.

Standard service codes

To assign the standard service codes for supplementary services, in this example we assign 50 to hold, carry out following commands:

```
:voice services assign_sc type=hold servicecode=50
```

A standard service code is preceded by:

- The prefix * for activation of a supplementary service.
- The prefix # for deactivation of a supplementary service.

So, in this example the activation code for hold is *50, the deactivation code is #50.

Non-standard service codes

When assigning non-standard service codes, you must also define the action (activate, deactivate or register) that has to be executed, as well as include the prefix in the service command.

A non-standard code:

- Can be composed of maximum five digits.
- Only digits 0 - 9, * and # are allowed.
- Must start with * or #.

To assign the non-standard service codes for supplementary services, in this example we assign *5#74 to activate the hold service:

```
:voice services assign_pxsc type=hold action=activate servicecode=*5#74
```

Non-standard service codes are very rarely used.

5.2.5 Configuring supplementary services

Overview

To check the configuration of the supplementary services, carry out following command:

```
:voice services list
Phone MWI type          : both
Network MWI type        : sollicited
Prefix service code     : standard
Service code command    : standard
CFU destination         : 789123456
CFNR timer              : 15
CFNR destination        : 8456
CFBS destination        : 8456
HFHF feature            : disabled
CW reject persistence   : disabled
CW timer                : 30
```

The examples used in the following overview result in the above settings.

Mwi_phone

Syntax: mwi_phone=<{immediate|deferred|both}>

Default setting: deferred

Example:

```
:voice services config mwi_phone=both
```

Explanation:

The mwi_phone parameter defines how the phone handles a message waiting indication. Following settings are possible:

- Immediate: a message waiting indication is displayed on the phone.
- Deferred: a message waiting indication tone is generated as you take the phone of the hook.
- Both: both of the previous message waiting indications are activated.

Mwi_network

Syntax: mwi_network=<{sollicited|unsollicited}>

Default setting: sollicited

Example:

```
:voice services config mwi_network=sollicited
```

Explanation:

Two variant networks for MWI are possible:

- Sollicited: in this case a subscription message is sent to a server with Notify response, so you must define the notifier address and port as well as the subscription expire time. For more information see “ Notifier_addr” on page 33 and “ Notifier_port” on page 33.
- Unsollicited: the provider will generate an MWI message automatically.

Prefix_servcode

Syntax: prefix_servcode=<{standard|nonstandard}>

Default setting: standard

Example:

```
:voice services config prefix_servcode=standard
```

Explanation:

When this parameter is set to:

- Standard: The standard service codes are operational.
- Non-standard: The non-standard service codes are operational. When no non-standard service codes are assigned, the standard service codes are operational. To check which service code is operational see " Viewing the supplementary services" on page 50.

servcode_cmd

Syntax: servcode_cmd=<{standard|nonstandard}

Default setting: standard

Example:

```
:voice services config servcode_cmd=standard
```

Explanation:

When the parameter is set to:

- Standard: Only standard service codes can be used. No non-standard service codes can be assigned.
- Non-standard: The creation and use of non-standard service codes is allowed.

Cfu_dest

Syntax: cfu_dest=<string>

Default setting: empty

Example:

```
:voice services config cfu_dest=789123456
```

Explanation:

All incoming calls are forwarded to an external number defined by this parameter (e.g. your mobile number).

Cfnr_timer

Syntax: cfnr_timer=<number{0 - 600}> expressed in seconds

Default setting: 10

Example:

```
:voice services config cfnr_timer=15
```

Explanation:

When an incoming call is not answered within the period defined by this parameter, the call will be directed to the destination defined by the parameter cfnr_dest (see below).

Cfnr_dest

Syntax: cfnr_dest=<{string}>

Default setting: empty

Example:

```
:voice services config cfnr_dest=8456
```

Explanation:

Incoming calls are forwarded to the number (internal or external) defined by this parameter when the call is not taken.

Cfbs_dest

Syntax: cfbs_dest=<{string}>

Default setting: empty

Example:

```
:voice services config cfbs_dest=8456
```

Explanation:

Incoming calls are forwarded to the number (internal or external) defined by this parameter when the callee is busy.

Hfhf_feature

Syntax: hfhf_feature=<{disabled|enabled}>

Default setting: disabled

Example:

```
:voice services config hfhf_feature=disabled
```

Explanation:

When this parameter is disabled the normal call state machine is used. When enabled the US variant of the call state machine is applied.

Cw_reject_persistence

Syntax: cw_reject_persistence=<{disabled|enabled}>

Default setting: disabled

Example:

```
:voice services config cw_reject_persistence=disabled
```

Explanation:

When this parameter is set to enabled the rejection of incoming waiting calls is persistent for the entire duration of a call once you have rejected an incoming waiting call during that call. All new incoming waiting calls are autonoumously and immediately rejected.

Cw_timer

Syntax: cw_timer=<number{0-120}>

Default setting: 30

Example:

```
:voice services config cw_timer=30
```

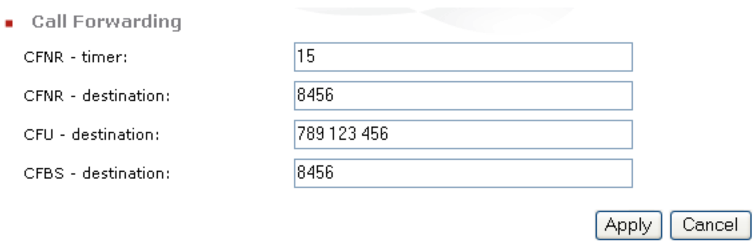
Explanation:

When you ignore an incoming call, nothing is done until this timer expires. After the expiration the waiting call is no longer offered and a busy indication is sent to the remote party.

Using the Web pages

You can also configure the call forwarding service via the Web pages (GUI). To do so, proceed as follows:

1. On the Thomson Gateway home page, click **Toolbox**
2. Click **Telephony**.
3. Click **Details**.
4. In **Pick a task**, click **View Telephony Services.**
5. Click **Configure**.
6. Type:
   - The CFNR Timer.
   - The CFNR-destination.
   - CFU-destination.
   - CFBS-destination.



7. Click **Apply**.

6 Other configurations

Overview

This chapter deals with the description of a variety of configurations. Because they each contains only a very few commands and parameters they are grouped in one chapter.

6.1 Configuring Profiles

Overview

Per SIP User Agent (UA) a profile is supported. This section describes how these profiles are configured and managed, as well as via CLI commands and, where applicable, via the Web pages (GUI).

You can find information on:

- "6.1.1 Checking the profiles" on page 65.
- "6.1.2 Adding profiles" on page 66.
- "6.1.3 Deleting profiles" on page 68.
- "6.1.4 Modifying profiles" on page 69.

- Flushing all profiles:

```
:voice profile flush
```


- "6.5 Codec" on page 75.

6.1.1 Checking the profiles

Using the CLI

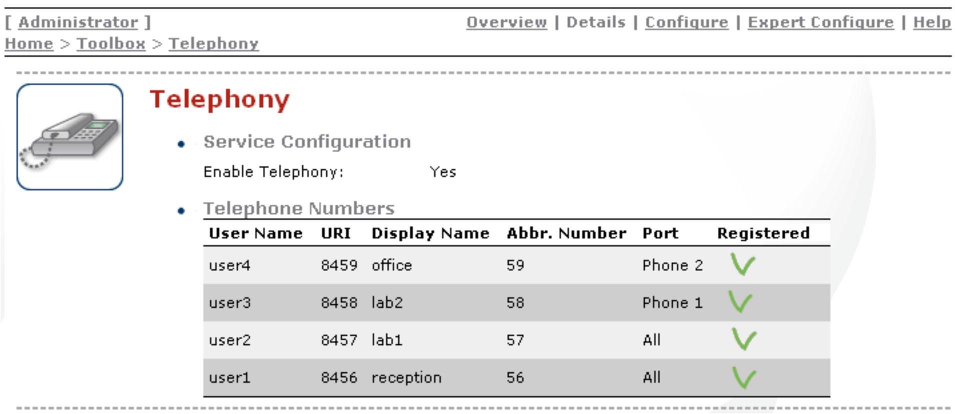
To check all of the defined user profiles, carry out following command:

```
:voice profile list
Idx  Port    Uri    DisplayName     Username     Abbr Nbr  Status   RegStatus    Msg Waiting
-------------------------------------------------------------------------------------------
0    COMMON  8456   reception       user1        56        Enabled  Registered   No
1    COMMOM  8457   lab1            user2        57        Enabled  Registered   No
2    FXS1    8458   lab2            user3        58        Enabled  Registered   No
3    FXS2    8459   office          user4        59        Enabled  Registered   No
```

Using the Web pages (GUI)

To check all of the defined user profiles, proceed as follows:

1. On the Thomson Gateway home page, click **Toolbox**
2. Click **Telephony**.
3. Click **Details**.



6.1.2 Adding profiles

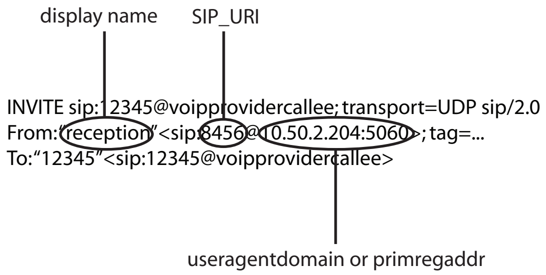
Using the CLI

To create a VoIP user profile, carry out following command:

```
:voice profile add
SIP_URI = <string>
     The SIP_URI is the username of the AoR. Typically, the SIP_URI is a telephone number, but
     it can be a mixture of both, numerals and letters.
[username] = <string>
     The user name is for authentication of this specific profile. The user name is optional.
[password] = <password>
     The password is for authentication of this specific profile. The password is optional
Please retype password for verification.
[password] = <password>
[displayname] = <string>
     The display name is used for the CLIP info and is an alias for the SIP_URI. The display
     name is typically a text string and is optional.
voiceport = <{FXS1|FXS2|DECT|COMMON}>
     The voice port is the analogue line port linked to this specific profile.
[abbr] = <string>
     An abbreviated number that will be mapped to this specific profile. This function is
     ONLY supported when the SIP_URI only contains numerals and no letters. This function is
     only for internal use and is optional.
[enable] = <{disabled|enabled}>
     Enable or disable this profile.
```

Structure of a SIP message

You can find the SIP_URI and display name in a SIP message:



Example

Following example illustrates the previous command:

```
:voice profile add
SIP_URI = 8456
[username] = user1
[password] = *******
Please retype password for verification.
[password] = *******
[displayname] = reception
voiceport = COMMON
[abbr] = 56
[enable] = enabled
:voice profile add SIP_URI=8456 username=user1 password=_DEV_D1299F4419D8F37E
displayname=reception voiceport=COMMON abbr=56 enable=enabled
```

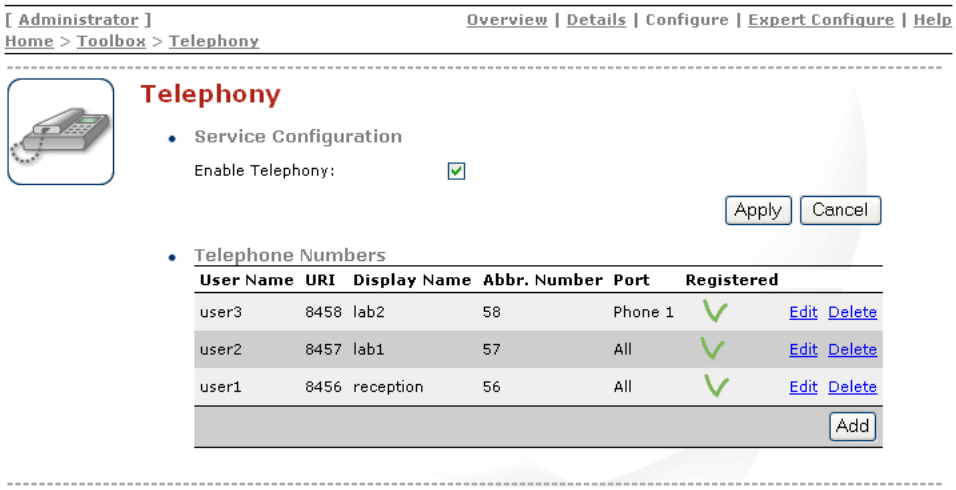
Using the Web pages (GUI)

To create a VoIP user profile, proceed as follows:

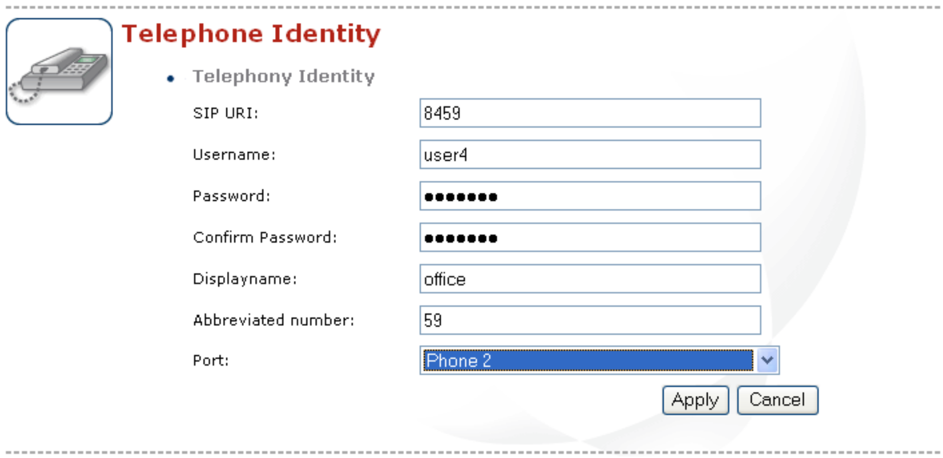
1 On the Thomson Gateway home page, click **Toolbox.**

2 Click **Telephony**.

3 Click **Configure.**



4 Click **Add** and complete the parameters.



5 Click **Apply**.

6.1.3 Deleting profiles

Using the CLI

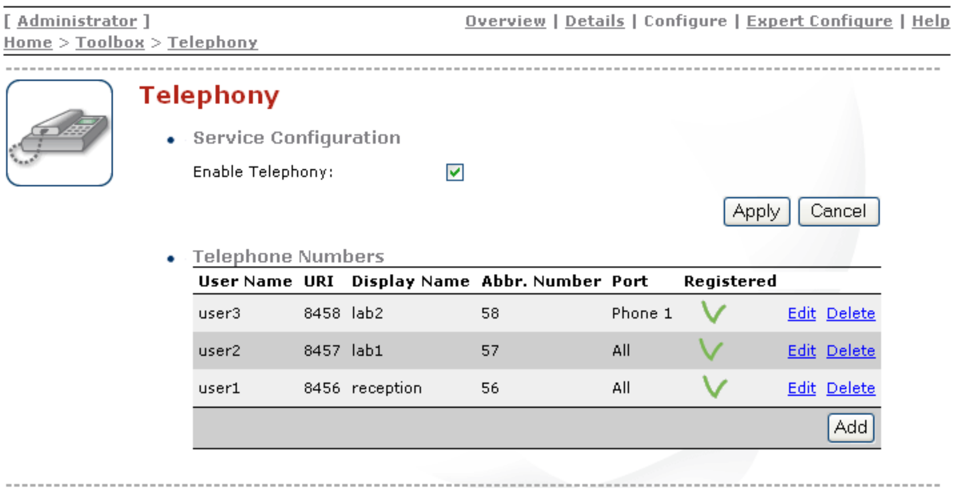
To delete a user profile, in this example 8459, carry out following command:

```
:voice profile delete SIP_URI=8459
```

Using the Web pages (GUI)

To delete a user profile, proceed as follows:

1. On the Thomson Gateway home page, click **Toolbox.**
2. Click **Telephony**.
3. Click **Configure.**
4. Click **Delete** at the end of the row of the profile you want to delete.



6.1.4 Modifying profiles

Using the CLI

To modify a user profile, in this example the voice port of SIP_URI 8458 is modified to common, carry out following command:

```
:voice profile modify
Index = 1
     The index of the profile you wish to modify.
SIP_URI = 8458
NEW_URI = 8458
[username] = user3
[password] =
[displayname] = sales
voiceport = FXS1
[abbr] = 58
[enable] = enabled
:voice profile modify SIP_URI=8458 NEW_URI=8458 voiceport=COMMON enable=enabled
```

Using the Web pages (GUI)

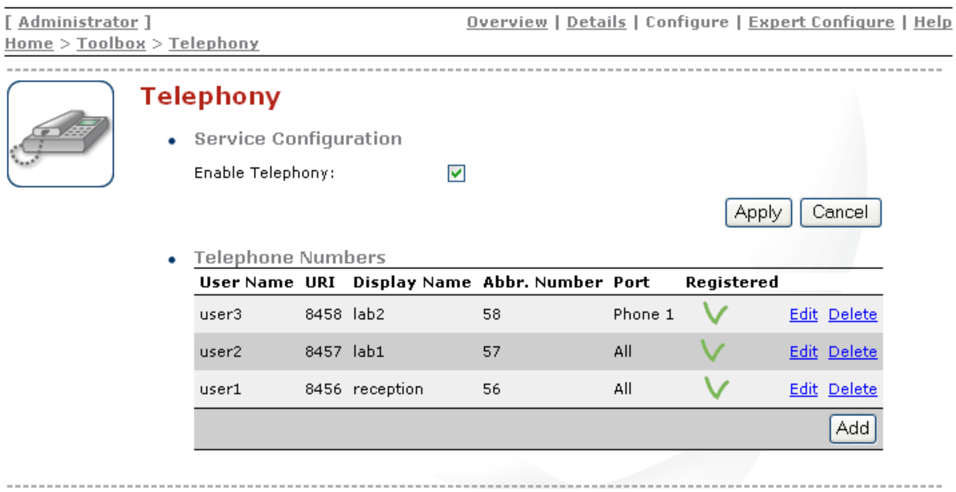
To modify a user profile, proceed as follows:

1 On the Thomson Gateway home page, click **Toolbox**.

2 Click **Telephony**.

3 Click **Configure**.

4 Click **Edit** at the end of the row of the profile you want to modify.



5 Modify the parameter(s).

6 Click **Apply**.

6.2 Call Admission Control

Introduction

The Call Admission Control (CAC) command defines the maximum number of ports that can be used simultaneously by a profile.

Checking the CAC configuration for voice

To check the setting of the cac parameter, carry out following command:

```
:voice cac list
max#portsperprofile=all
```

Configuring CAC for voice

To configure CAC for voice, carry out the following command:

```
:voice cac config max#portsperprofile= <{one|all}>
```

Explanation:

When the parameter is set to:

- One: only one call per profile can be set up simultaneously.
- All: multiple calls (depending on the applied codec) can be set up simultaneously.

6.3 Voice Country Settings

Country settings

Country-specific settings, such as dialling tone, hook flash timer, DTMF tones, polarity,...are pre-loaded in the dslxcfg.bin file in the dl-directory.

You can load the settings for the following countries:

- Australia
- Belgium (Belgacom)
- Denmark
- ETSI
- France1 (France Telecom)
- France2 (Tiscali)
- France3 (Tele2)
- Germany
- Italy
- Netherlands (KPN)
- Norway
- Spain (Telefonica)
- Sweden (Breadband)
- UK (BT, Bulldog)
- US (North America)

Configuring country settings

To load the country settings for e.g. Belgium, carry out following command:

```
:voice country config country=belgium
```

The default country setting is etsi.

To check the current country setting, carry out the following command:

```
:voice country list
Current selected country: belgium
```

Viewing available country settings

To check which country settings can be loaded, carry out following command:

```
:voice country countrylist
Available countries:
etsi
chile
france1
netherlands
northamerica
spain
uk
```

6.4 Fax

Introduction

To send faxes, there are two options:

- Using lossless codecs of G.711.
- Using the real-time fax over IP protocol T.38.

When using SIP, an INVITE message is sent to the called party at call set-up, requesting a voice connection. A voice connection is then established. The sender sends the fax tone, upon detection of the fax signal the receiver can sent a RE-INVITE to define which codec that has to be used. If not, the receiver assumes that the sender will use the G.711 codec.

Checking the configuration

To check the fax configuration, carry out following command:

```
:voice fax list
detect_timeout  : 0
early-detect-faxmodem: enabled
transport            : inband_reneg
udptl_redun          : 0
```

Detect_timeout

Syntax: detect_timeout=<number{0-120}> expressed in seconds

Default setting: 60 seconds

Example:

```
:voice fax config detect_timeout=0
```

Explanation:

A fax generates a tone of 1100 Hz and a modem a tone of 1300 Hz. This continuous tone is detected by the DSP. In order to avoid that during a voice call a modem or fax call initiation is detected (e.g. a person whistling unintentionally the modem or fax tone), you can limit the modem and fax detection by the DSP in time. This is the time-out in seconds used to detect a modem or fax call. 0 means that no time-out is defined. The timer starts to run as soon as the voice call is in the connected state.

Early -detect-faxmodem

Syntax: early-detect-faxmodem=<{enabled|disabled}>

Default setting: enabled

Example:

```
:voice fax config early-detect-faxmodem=enabled
```

Explanation:

This parameter enables an early detection whether the call is a modem or fax call. The DSP generates a RE-INVITE in order to negotiate the codec or protocol to be used so that no traffic from the modem or fax is missed during set-up.

Transport

Syntax: transport=<{inband_auto|inband_reneg|t38}>

Default setting: T38

If the board or release does not support T.38, the default setting will inband_reneg.

Example:

```
:voice fax config transport=inband_reneg
```

Explanation:

When the T.38 method is selected, the receiver will detect that a fax call is initiated and will renegotiate to use the T.38 protocol for the fax call.

When the inband autonomous is selected, both the sender and receiver will detect that a fax call is initiated and will send the fax autonomously. This means without involving VoIP signalling for capability detection, to inband VDB (Voice Band Data) codec, G.711, so that the fax T.30 protocol can be transmitted.

When the inband renegotiation method is selected, the receiver will detect that a fax call is initiated and will renegotiate to a VBD call.

Udptl_redun

Syntax: udptl_redun=<number{0-3}>

Default setting: 0

Example:

```
:voice fax config udptl_redun=0
```

Explanation:

This parameter defines the number of redundant IFP packets to be applied in transmit direction. You can find more information on the principle of operation of redundancy in ITU-T T.38 (chapter 9.1.4.1).

6.5 Codec

6.5.1 Introduction

Mean Opinion Score

The quality of speech of a codec is expressed in Mean Opinion Score (MOS). With MOS, a wide range of listeners judge the quality of a voice sample on a scale of 1 (bad) to 5 (excellent), so it is a subjective rating. The scores are averaged to provide the MOS for that sample. The general rule for an accepted quality says that the MOS value of a codec should at least be 3.7.

The table below shows the relationship between the different codecs and their MOS scores.

Audio Codec	Technology	Bitrate (Kbps)	MOS	Compression delay (ms)
G.711	PCM	64	4.1	0.75
G.723.1	MP-MLQ	6.3	3.9	30
G.723.1	ACELP	5.3	3.65	30
G.726	ADPCM	32	3.85	1
G.728	LD-CELP	16	3.61	3 to 5
G.729	CS-ACELP	8	3.92	10
G.729a	CS-ACELP	8	3.7	10

Difference between A-law and µ-law

The A-law or µ-law is the companding (compression - expanding) algorithm used in the G.711 standard. The µ-law is applied in North-America and Japan, the A-law is applied in the rest of the world.

Overview of the supported codecs

The following table gives an overview of the contemporary applied codecs:

Codec	Supported since
G.711 audio at 64 kbps, A-law	R 5.3
G.711 audio at 64 kbps, A-law, truncated to 7 bits	Not
G.711 audio at 64 kbps, µ-law	R 5.3
G.711 audio at 64 kbps, µ-law, truncated to 7 bits	Not
G.722 7 kHz audio	R 6.1
G.723.1 at either 5.3 or 6.3 kbps	R 5.3
G.723.1 at either 5.3 or 6.3 kbps with silence suppression as in Annex A	R 5.3
G.726: ADPCM at 16 kbps	R 5.3
G.726: ADPCM at 24 kbps	R 5.3

Codec	Supported since
G.726: ADPCM at 32 kbps	R 5.3
G.726: ADPCM at 40 kbps	R 5.3
G.728 audio at 16 kbps	Not
G.729 Annex A audio at 8 kbps	R 5.3
G.729 Annex A audio at 8 kbps with silence suppression as in Annex B	R 5.3

Depending on the board and/or release codecs are available. The G.711 codec is always enabled and can not be disabled. However, if required, you can change the priority of the applied codecs.

Packetisation time

The packetisation time is the length of a VoIP packet in seconds. Since release 6.1 the packetisation time is 20 ms. Before (release 5.4)it was 30 ms. The higher the packetisation time is, the lower the overhead is and the higher the delay is.

Voice Activity Detection

The Voice Activity Detection (VAD) detects silence and speech in a conversation. Only speech has to be transmitted, silence can be omitted.

Comfort Noise Generation

Comfort Noise Generation (CNG) is artificial background noise to fill the silence in a conversation resulting from VAD.

6.5.2 Configuring codecs

Checking the configuration

To check the codec configuration, carry out the following command:

```
:voice codec list
Codec Type   Packet Delay   Voice Act. Detection   Priority   Status
----------------------------------------------------------------------
     g711u           30            enabled           1     enabled
     g711a           30            enabled           2     enabled
      g722           30            enabled           3    disabled
    g723_1           30            enabled           4    disabled
   g726_16           30            enabled           5    disabled
   g726_24           30            enabled           6    disabled
   g726_32           30            enabled           7    disabled
   g726_40           30            enabled           8    disabled
      g729           30            enabled           9    disabled
```

Configuring a codec

To configure a codec, carry out the following command:

```
:voice codec config
type = <{g711u|g711a|g722|g723.1|g726_16|g726_24|g726_32|g726_40|g729}>
     The codec type.
ptime = <{10|30|20}>
     The packet time.
ptime_g723 = <{30}>
     The packet time.
vad = <{disabled|enabled}>
     Enable or disable VAD.
priority = <number{1-9}>
     The codec capability priority.
status = <{disabled|enabled}>
     Enable or disable this capability.
```

6.5.3 Codecs with a dynamic payload type

Introduction

Dynamic payload types are by definition dynamic, meaning that the payload type numbers used by codecs with a dynamic payload type ("dynamic codecs") are free to be chosen in the range reserved for dynamic codecs. It is however possible to enforce the mapping of codecs with dynamic payload type to a specific payload type number.

Checking

To check the dynamic codec configuration, carry out the following command:

```
:voice codec dynamic list
Codec Type  Payload Type
--------------------------
   rfc2833      97
   g726_16      98
   g726_24      99
   g726_32      100
   g726_40      101
```

Configuring

To map a dynamic codec to a specific payload type number, carry out the following command:

```
:voice codec dynamic config
type = <{rfc2833|g723_1|g726_16|g726_24|g726_32|g726_40}>
     the codec type.
payloadtype = <number{96-128}>
     the payload type.
```

6.6 Basic call handling for FXS and FXO ports

FXS and FXO ports

When you connect the Thomson Gateway to analog phones, you plug phone cables into Foreign eXchange Subscriber (FXS) ports on the Thomson Gateway. These ports provide POTS servicem including battery currentm ring voltage and dial tones to the phones.

When you connect the Thomson Gateway to the TELCO, you plug the lines from the Foreign eXchange Office (FXO) network side into the FXO port on the Thomson Gateway. This port provides onhook/offhook indication (loop closure) to the FXO network side.

Basic call handling

The following call topologies, involving FXS and FXO ports, are supported on the Thomson Gateway:

		To							
		FXO	Specific FXS	All FXS	Specific DECT HS	All DECT HS's	All FXS and DECT HS's	VoIP	None
From	FXO			Y		Y	Y		Y
	Specific FXS	Y	Y		Y	Y	Y	Y	
	Specific DECT HS	Y	Y		Y	Y	Y	Y	
	VoIP		Y		Y	Y	Y		

Below is explained how the call handling parameters for the FXS and FXO ports can be checked and configured.

Checking the configuration of the FXS and FXO ports

To check the configuration of the FXS and FXO ports, carry out the following commands:

```
:voice fxoport list
FXO disconnect timer      : 1000
Incoming fxo destination : all

:voice fxsport list
Standard inter digit timer (closed) : 5000
Open number interdigit timer        : 5000
```

Fxodisconnect

Syntax: fxodisconnect=<number{500-5000}> expressed in milliseconds

Default setting: 1000 ms

Example:

```
:voice fxoport config fxodisconnect=1000
```

Explanation:

When the dial plan (see “6.8 Dial plan” on page 84) determines that an existing call on FXO must be cleared, the FXO line must go onhook to clear the existing call, followed by going offhook to initiate a new call. The period between going onhook and offhook is determined by the parameter fxodisconnect.

Incfxodest

Syntax: incfxodest=<{fxs|dect|all|none}>

Default setting: all

Example:

```
:voice fxoport config incfxodest=all
```

Explanation:

This parameter defines to which destination an incoming FXO call is diverted. When you divert to FXS and/or DECT, all FXS and/or DECT devices will ring.

This function is only for models equipped with full FXO.

Interdigit

Syntax: interdigit=<number{10-30000}> expressed in milliseconds

Default setting: 5000 ms

Example:

```
:voice fxsport config interdigit=5000
```

Explanation:

This timer is the maximum allowable time between the dialing of digits. This timer is restarted every time a digit is dialed. Expiration of this timer indicates "End of Dialing".

InterdigitOpen

Syntax: fxodisconnect=<number{10-30000}> expressed in milliseconds

Default setting: 5000 ms

Example:

```
:voice fxsport config interdigitOpen=5000
```

Explanation:

This timer is the maximum allowable time between the dialing of digits once the minimum number of digits defined on a prefix based has been reached. This timer is only applicable to "open numbering", where the exact number of digits for a prefix is not known.

6.7 DECT

Introduction

For the DECT port and the DECT handset the following parameters can be configured for updating and subscrption purposes.

Checking the DECT handset configuration

To check your DECT handset configuration, perform the following action:

```
:voice decthandset list
username     : inventel
machine      : developers.inventel.com
port         : 80
directory    : inventel/dect_handset/generic_handset
state        : Disabled
```

Username

Syntax: username=<string>

Example:

```
:voice decthandset config username= inventel
```

Explanation:

The (new) username to be used on the DECT handset.

Password

Syntax: password=<string>

Example:

```
:voice decthandset config password= password
```

Explanation:

The (new) password to be used on the DECT handset.

Machine

Syntax: machine=<string>

Example:

```
:voice decthandset config machine= developers.inventel.com
```

Explanation:

The IP address or FQDN where the software builds can be found to update your DECT handset.

Port

Syntax: port=<string>

Default setting: 0

Example:

```
:voice decthandset config port= 80
```

Explanation:

The port the DECT handset has to listen to when searching for new software builds.

Directory

Syntax: directory=<string>

Example:

```
:voice decthandset config directory= inventel/dect_handset/generic_handset
```

Explanation:

The directory where the software builds can be found to update your DECT handset.

State

Syntax: state=<{disabled|enabled}>

Default setting: disabled

Example:

```
:voice decthandset config state= disabled
```

Explanation:

Enable or disable updating of the software.

Checking the DECT port configuration

To check your DECT port configuration, perform the following action:

```
:voice dectport list
pin           : ----
substimeout : 90
```

Pin

Syntax: pin=<string>

Default setting: ----

Example:

```
:voice dectport config pin= 1956
```

Explanation:

The PIN used by the DECT base station.

The parameter pin is only visible when voice is activated on the Thomson Gateway. When the Thomson Gateway is configured with the factory default settings, the parameter pin is not visible.

Substimeout

Syntax: substimeout=<number{0-600}> in seconds

Default setting: 60 seconds

Example:

```
:voice dectport config substimeout= 60
```

Explanation:

The DECT subscription time window in seconds (default: 60 sec). the VoIP/voice LEd will flash during this time.

6.8 Dial plan

Introduction

The dial plan is used for outgoing calls and supplementary service actions.

The following entries in the dial plan exist by default and can only be viewed:

- The entries dealing with configured phone numbers assigned to local voice ports.
- The entries dealing with actions on the provisioned supplementary services.
- If forced FXO is activated, an entry is automatically added in the dialplan. This entry contains the prefix as defined by the supplementary services and the default port FXO. Other parameters of this entry can be configured.

All other entries in the dial plan must be created an can be modified and deleted.

Overview

In this section all of the commands concerning the dial plan are explained. You can find information on:

- “6.8.1 Viewing the dial plan” on page 85
- “6.8.2 Adding a user defined entry in the dial plan” on page 86
- “6.8.3 Modifying a user defined entry in the dial plan” on page 87
- Deleting a user defined entry in the dial plan:

```
:voice dialplan delete prefix= <string>
     The prefix which identifies this entry.
```

- Flushing all user defined entrees of the dial plan:

```
:voice dialplan flush
```

6.8.1 Viewing the dial plan

To view the dial plan of your Thomson Gateway, perform the following action:

```
:voice dialplan list
Prefix  DefPort  FallbPort  Prior  Fallback  Min.Dig  Max.Dig  PosMod  Rem.Dig
-------------------------------------------------------------------------------
                         Insert  Rescan  Data                            Action
-------------------------------------------------------------------------------
    *96        NA          NA      NA          No         3       30        0         0
                                          No     No            Activate_Call_transfer
    #96        NA          NA      NA          No         3       30        0         0
                                          No     No          Deactivate_Call_transfer
   *#96        NA          NA      NA          No         4       30        0         0
                                          No     No         Interrogate_Call_transfer
    *94        NA          NA      NA          No         3       30        0         0
                                          No     No                Activate_Call_hold
    #94        NA          NA      NA          No         3       30        0         0
                                          No     No              Deactivate_Call_hold
   *#94        NA          NA      NA          No         4       30        0         0
                                          No     No             Interrogate_Call_hold
   *43*        NA          NA      NA          No         0       30        1         4
                                         Yes     No     Activate_Call_waiting_on_call
   #43*        NA          NA      NA          No         0       30        1         4
                                         Yes     No   Deactivate_Call_waiting_on_call
    *43        NA          NA      NA          No         3       30        0         0
                                          No     No             Activate_Call_waiting
    #43        NA          NA      NA          No         3       30        0         0
                                          No     No           Deactivate_Call_waiting
   *#43        NA          NA      NA          No         4       30        0         0
                                          No     No          Interrogate_Call_waiting
    *30        NA          NA      NA          No         3       30        0         0
                                          No     No                     Activate_CLIP
    #30        NA          NA      NA          No         3       30        0         0
                                          No     No                   Deactivate_CLIP
   *#30        NA          NA      NA          No         4       30        0         0
                                          No     No                  Interrogate_CLIP
    *95        NA          NA      NA          No         3       30        0         0
                                          No     No                     Activate_3PTY
    #95        NA          NA      NA          No         3       30        0         0
                                          No     No                   Deactivate_3PTY
   *#95        NA          NA      NA          No         4       30        0         0
                                          No     No                  Interrogate_3PTY
  53640                    NA     Low          No         5        5        0         0
                              NA          No     No                         Huntgroup
  53643                    NA     Low          No         5        5        0         0
                              NA          No     No                         Huntgroup
  53642       FXS          NA     Low          No         5        5        0         0
                              NA          No     No                    ROUTE_excl_eon
  53641       FXS          NA     Low          No         5        5        0         0
                              NA          No     No                    ROUTE_excl_eon
   *31*        NA          NA      NA          No         4       19        1         4
                                         Yes     No       Activate_CLIR_on_call_basis
   #31*        NA          NA      NA          No         4       19        1         4
                                         Yes     No     Deactivate_CLIR_on_call_basis
    *31        NA          NA      NA          No         3       30        0         0
                                          No     No                     Activate_CLIR
    #31        NA          NA      NA          No         3       30        0         0
                                          No     No                   Deactivate_CLIR
   *#31        NA          NA      NA          No         4       30        0         0
                                          No     No                  Interrogate_CLIR
```

6.8.2 Adding a user defined entry in the dial plan

Syntax

To add a user defined entry in the dial plan, carry out the following command:

```
:voice dialplan add
prefix = <string>
      The prefix which identifies this entry.
defaultport = <{FXS1|FXS2|DECT|FXO|VoIP|NA}>
      The default outgoing port.
fallbackport = <{FXS1|FXS2|DECT|FXO|VoIP|NA}>
      The fallback outgoing port.
priority = <{NA|Low|High}>
      The priority of the entry.
fallback = <{disabled|enabled}>
      The fallback mechanism status of this entry.
minimumdigits = <number{1-31}>
      If the minimum number of digits is reached and the called number is ended, an outgoing
      call is initiated.
maximumdigits = <number{1-31}>
      If the maximum number of digits has been reached an outgoing call is initiated.
      posofmodify = <number{0-31}>
      Startposition at which a number of digits must be removed and/or inserted.
renumdigits = <number{0-31}>
      The number of digits that need to be removed from the complete number, starting at
      posofmodify.
[insert] = <string>
      The string which must be inserted at posofmodify after removing renumdigits.
rescan = <{no|yes}>
      Rescanning of the dial plan with the result of the new entry needed or not.
data = <{no|yes}>
      This entry is used for data calls or not.
action = <{none|ROUTE_exl_eon|ROUTE_incl_EON}>
      The action parameter.
            ROUTE_exl_eon: A call will be set up (routed). The end-of-number indication will
            not be sent out.
            ROUTE_incl_eon: A call will be set up (routed). The end-of-number indication will
            be sent out.
```

6.8.3 Modifying a user defined entry in the dial plan

Syntax

To modify a user defined entry in the dial plan, carry out the following command:

```
:voice dialplan modify
prefix = <string>
newprefix = <string>
     In case of update prefix, the new prefix which identifies this entry.
     In case of same prefix, the same value as for prefix.
defaultport = <{FXS1|FXS2|DECT|FXO|VoIP|NA}>
fallbackport = <{FXS1|FXS2|DECT|FXO|VoIP|NA}>
priority = <{NA|Low|High}>
fallback = <{disabled|enabled}>
minimumdigits = <number{1-31}>
maximumdigits = <number{1-31}>
posofmodify = <number{0-31}>
renumdigits = <number{0-31}>
[insert] = <string>
rescan = <{no|yes}>
data = <{no|yes}>
action = <{none|ROUTE_exl_eon|ROUTE_incl_EON}>
```

6.9 QoS

Introduction

For voice the following Quality of Service (QoS) parameters can be set.

Checking the QoS settings

To view the QoS settings for voice, carry out the following command:

```
:voice qos list
Traffic Type                QOS field                 Value
-------------------------------------------------------------
  Signalling                     DSCP                  af42
    Realtime                     DSCP                    ef
```

Configuring QoS

To configure the QoS settings for voice, carry out the following command:

```
:voice qos config
type = <{Signaling|Realtime}>
     The type of traffic that needs QoS: signaling for SIP and realtime for RTP.
qosfield = <{DSCP|precedence}>
     The QoS field to be used.
dscp = <{ef|af11|af12|af13|af21|af22|af23|af31|af32|af33|af41|af42|af43|cs0|cs1|cs2|cs3|cs4|
cs5|cs6|cs7} or number>
     The DSCP value.
or
precedence = <{routine|priority|immediate|flash|flash-override|CRITIC-ECP|internetwork-
control|network-control} or number>
     The precedence value.
```

6.10 DNS

Introduction

This section describes the DNS settings needed for the state machine related to SIP registration.

Checking the DNS settings

To view the DNS settings for voice, carry out the following command:

```
:voice dns list
DNS start entry        : first
Nbr of DNS entries     : 5
```

Startentry

Syntax: startentry=<{first|random}>

Default setting: first

Example:

```
:voice dns config startentry=first
```

Explanation:

This parameter is the entry to start with in the registration procedure. This can be either the first entry or a random entry.

Maxentries

Syntax: interdigit=<number{1-24}>

Default setting: 5

Example:

```
:voice dns config maxentries=5
```

Explanation:

This parameter represents the maximum number of used entries in a DNS response.

6.11 Tones

Introduction

The following tables are used for tones generation:

- *Tone event table*
  This table contains a list of events for which a tone is defined. The tone itself is specified in the tone description table.
- *Tone description table*
  This table defines for every tone what must be generated: a tone specified by a pattern in the tone pattern table, a file to be played and tone text.
- *Tone pattern table*
  This table specifies the phases of a tone. Per tone phase (entry) the distinct tones are specified by means of frequency and power level, tone on/off, duration and an entry to the next tone phase. The table may be set up such that entries form loops or may end after a finite sequence.

Overview

In this section all of the commands concerning tones are explained. You can find information on:

6.11.1 Tone event table

Viewing the tone event table

To view the tone event table, carry out the following command:

```
:voice tone eventtable list
Event ID | Tone
---------------------------------------------
       1 | specialdial
       2 | congestion
       3 | specialdial
       4 | congestion
       5 | specialdial
       6 | congestion
       7 | specialdial
       8 | congestion
       9 | specialdial
      10 | congestion
      11 | specialdial
      12 | congestion
      13 | specialdial
      14 | congestion
      15 | congestion
      16 | dial
      17 | stutterdial
      18 | mwi
     101 | warning
     102 | warning
     103 | warning
     104 | stutterdial
     105 | confirmation
     106 | rejection
     107 | confirmation
     108 | rejection
     201 | congestion
     202 | congestion
     203 | congestion
     204 | congestion
     205 | congestion
     305 | none
     401 | callwaiting
     402 | remotecallwaiting
     403 | none
     404 | dial
     405 | callhold
     406 | remotecallhold
     407 | release
     501 | none
     502 | none
     503 | none
     999 | warning
```

Tones

The table below gives a description of all available tones.

Tone	Description
Dial tone	A tone indicating the readiness to receive call information and inviting the user to start sending call information. This tone is sent after off-hook action or after register recall (in conversation).
Special dial tone	Same description as stutter dial tone.
Stutter dial tone	A tone indicating the readiness to receive call information and inviting the user to start sending call information. This tone is sent after off-hook action (at call set-up) when supplementary info must be given in an interactive Service Code Command (SCC) dialogue.
Ringing tone	A tone indicating to the caller that ringing or alerting is applied to the called line.
Busy tone	A tone indicating that the called telephone is engaged in another call and that the call waiting supplementary service is not active.
Congestion tone	A tone indicating to the caller that: - An outgoing call is not possible. This tone is given instead of the dial tone. - The equipment/bandwith for the setting up the required call or for the use of a specific services are temporarily engaged or not present (this includes CAC check).
Special information tone	Not used in the current Thomson Gateway software.
Call waiting tone	A tone indicating to a user, with the call waiting supplementary service active, who is engaged in a call that someone is attempting to call his number.
Call hold tone	Not used in the current Thomson Gateway software.
Message waiting indication tone	A tone indicating to the user that at least one new message is waiting in a voice mailbox service. This tone is sent when the user goes off-hook (if MWI phone variant is defered).
Warning tone	A tone indicating: - To the caller that he is trying to call his own number. - To the caller that a call setup via VoIP is not successful but it is not because the callee was busy. - To the caller and callee that, when a conversion took place, the other party has hooked on and that the connection has been released. - To the caller that the called number does not have the minimum number of digits as defined in the dial plan. - To the caller that a call initation is rejected by the dial plan.
Service confirmation tone	A tone indicating to the user that a supplementary service action (SCC) was completed successfull. The user must then go on-hook.

Tone	Description
Service rejection tone	A tone indicating to the user that: ■ A supplementary service action (SCC) was not completed successfull. The user must then go on-hook. ■ An outgoing call screening resulted in rejection of the outgoing call.

Modifying a tone event

To modify a tone event, carry out the following command:

```
:voice tone eventtable modify
eventid = <{1|2|3|4|5|6|7|8|9|10|11|12|13|14|15|16|17|18|101|102|103|104|105|106|107|108|201
|202|203|204|205|305|401|402|403|404|405|406|407|501|502|503|999}>
     The event ID.
tone = <{dial|none|remotecallhold|callhold|remotecallwaiting|callwaiting|rejection|confirmat
ion|release|warning|congestion|busy|ringback|mwi|specialdial|stutterdial}>
     Tone to play for this event.
```

Flushing the tone event table

To flush all tone events, carry out the following command:

```
:voice tone eventtable flush
```

6.11.2 Tone description table

Viewing the tone description table

To view the tone description table, carry out the following command:

By default the ETSI defined tones are applied.

```
:voice tone descrtable list
Tone                 | Status    | Delay | Pattern ID | File         | File rep. | Text             |
 Max. Dur. | Next tone | Repeat after
------------------------------------------------------------------------------------------
-------------------------------------
dial                 |  enabled |     0 |         51 |              |         0 |                  |
         0 |           |          0
none                 |  enabled |  0 |         0 |              |         0 |                |
      0 |           |          0
remotecallhold       | disabled |     0 |         91 |              |         0 |                  |
         0 |           |          0
callhold             |  enabled |     0 |         31 |              |         0 |                  |
         0 |           |          0
remotecallwaiting    | disabled |     0 |        101 |              |         0 |                  |
         0 |           |          0
callwaiting          |  enabled |  2500 |         11 |              |         0 |                  |
         0 |           |          0
rejection            |  enabled |     0 |         71 |              |         0 |                  |
         0 |           |          0
confirmation         |  enabled |     0 |         21 |              |         0 |                  |
         0 |           |          0
release              |  enabled |     0 |         81 |              |         0 |                  |
         0 |           |          0
warning              |  enabled |     0 |        141 |              |         0 |                  |
         0 |           |          0
congestion           |  enabled |     0 |         41 |              |         0 |                  |
         0 |           |          0
busy                 |  enabled |     0 |          1 |              |         0 |                  |
         0 |           |          0
ringback             |  enabled |     0 |        111 |              |         0 |                  |
         0 |           |          0
mwi                  |  enabled |     0 |         61 |              |         0 |                  |
         0 |           |          0
specialdial          |  enabled |     0 |        121 |              |         0 |                  |
         0 |           |          0
stutterdial          |  enabled |     0 |        131 |              |         0 |                  |
         0 |           |          0
```

Adding a tone description

To add a tone description, carry out the following command:

```
:voice tone descrtable add
tone = <string>
     The name of the tone.Typical values are : busy | callwaiting | callhold | confirmation |
     congestion | dial | mwi | none | rejection | release | remotecallwaiting |
     remotecallhold | ringback | specialdial | stutterdial | warning | extra1 | extra2. The
     extra tones are introduced to allow additional tones to be defined.
status = <{disabled|enabled}>
     Enable or disable this tone. If a disabled tone entry is referenced, the result is that
     no tone is played.
delay = <number>
     The delay in milliseconds.
[patternentryid] = <number>
     this is a pointer to an entry ID in the tone pattern table, indicating the start of the
     tone pattern for this tone. If the tone is also specified by a tone file, the tone
     pattern is only applied in case the tone file cannot be applied.
     If the referenced patternentryid does not exist, this tone can not be played.
[file] = <string>
     This is the file name of a tone/announcement file that has been downloaded.
     The download may have occurred via the TR 069 Download mechanism or by some other means.
     In case the tone is also specified by a tone pattern, the file will always have higher
     priority than the tone pattern.
[filerepeat] = <number>
     The default number of times the data in the file should be repeated. If the value 0
     (zero) is specified then the tone should be played indefinitely.
text = <quoted string>
     The text to be displayed on the screen of the VoIP device when the tone is played and no
     specific error message has been provided.
[maxduration] = <number>
     Maximum duration of the tone generation of the patttern in milliseconds.
[nexttone] = <string>
     Name of the next tone. when the current tone has reached the maximum duration or the file
     generation has ended, the nexttone (if specified) will be generated.
repeatafter = <number>
     Time after which the tone pattern must be repeated in milliseconds. This period defines
     the interval between two instances of the pattern.
```

Modifying a tone description

To modify a tone description, in this example the delay of the tone callwaiting is changed to 3000 ms, carry out the following command:

```
:voice tone descrtable modify
tone = callwaiting
[status] =
[delay] = 3000
[patternentryid] =
[file] =
[filerepeat] =
[text] =
[maxduration] =
[nexttone] =
[repeatafter] =
:voice tone descrtable modify tone=callwaiting  delay=3000
```

Deleting a tone description

To delete a tone description, carry out the following command:

```
:voice tone descrtable delete
tone = <string>
```

Only user defined tone description can be deleted?

Flushing the tone description table

To flush all tone descriptions, carry out the following command:

```
:voice tone descrtable flush
```

6.11.3 Tone pattern table

Viewing the tone pattern table

To view the tone pattern table, carry out the following command:

```
:voice tone patterntable list
Id | Tone | Freq1 | Power1 | Freq2 | Power2 | Freq3 | Power3 | Freq4 | Power4 | Duration | Next Entry | Max Loops | Next En
try After Max Loops
---------------------------------------------------------------------------------------------------------------------------
  1 |  on |  425 |  -19 |    0 |    0 |    0 |    0 |    0 |    0 |    500 |    2 |    0 | 0
  2 |  on |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    500 |    1 |    0 | 0
 11 |  on |  425 |  -20 |    0 |    0 |    0 |    0 |    0 |    0 |    200 |   12 |    0 | 0
 12 |  on |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    200 |   13 |    0 | 0
 13 |  on |  425 |  -20 |    0 |    0 |    0 |    0 |    0 |    0 |    200 |   14 |    0 | 0
 14 |  on |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    0 |   9000 |   11 |    0 | 0
 21 |  on |  765 |  -21 |  850 |  -21 |    0 |    0 |    0 |    0 |   1000 |   22 |    0 | 0
 22 |  on |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    0 |   5000 |   21 |    0 | 0
 31 |  on | 1400 |  -20 |    0 |    0 |    0 |    0 |    0 |    0 |    400 |   32 |    0 | 0
 32 |  on |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    0 |  15000 |   31 |    0 | 0
 41 |  on |  425 |  -20 |    0 |    0 |    0 |    0 |    0 |    0 |    250 |   42 |    0 | 0
 42 |  on |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    250 |   41 |    0 | 0
 51 |  on |  425 |  -19 |    0 |    0 |    0 |    0 |    0 |    0 |      0 |    0 |    0 | 0
 61 |  on |  425 |  -19 |    0 |    0 |    0 |    0 |    0 |    0 |   1200 |   62 |    0 | 0
 62 |  on |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    0 |     40 |   63 |    0 | 0
 63 |  on |  425 |  -19 |    0 |    0 |    0 |    0 |    0 |    0 |     40 |   62 |    4 | 51
 81 |  on | 1400 |  -20 |    0 |    0 |    0 |    0 |    0 |    0 |    500 |   82 |    0 | 0
 82 |  on |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    0 |   5000 |   83 |    0 | 0
 83 |  on | 1400 |  -20 |    0 |    0 |    0 |    0 |    0 |    0 |    500 |   84 |    0 | 0
 84 |  on |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    0 |   5000 |   81 |    0 | 0
 91 |  on | 1400 |  -20 |    0 |    0 |    0 |    0 |    0 |    0 |    400 |   92 |    0 | 0
 92 |  on |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    0 |  15000 |   91 |    0 | 0
101 |  on |  425 |  -20 |    0 |    0 |    0 |    0 |    0 |    0 |    200 |  102 |    0 | 0
102 |  on |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    200 |  103 |    0 | 0
103 |  on |  425 |  -20 |    0 |    0 |    0 |    0 |    0 |    0 |    200 |  104 |    0 | 0
104 |  on |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    0 |   9000 |  101 |    0 | 0
111 |  on |  425 |  -19 |    0 |    0 |    0 |    0 |    0 |    0 |   1000 |  112 |    0 | 0
112 |  on |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    0 |   4000 |  111 |    0 | 0
121 |  on |  425 |  -19 |    0 |    0 |    0 |    0 |    0 |    0 |      0 |    0 |    0 | 0
131 |  on |  425 |  -19 |    0 |    0 |    0 |    0 |    0 |    0 |    500 |  132 |    0 | 0
132 |  on |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    0 |     50 |  131 |    0 | 0
141 |  on | 1400 |  -20 |    0 |    0 |    0 |    0 |    0 |    0 |    500 |  142 |    0 | 0
142 |  on |    0 |    0 |    0 |    0 |    0 |    0 |    0 |    0 |   5000 |  141 |    0 | 0
151 |  on | 2130 |  -16 | 2750 |  -16 |    0 |    0 |    0 |    0 |    100 |    0 |    0 | 0
```

Adding a tone pattern

To add a tone pattern, carry out the following command:

```
:voice tone patterntable add
id = <number>
     ID of the tone pattern entry. The value must be unique within this table.
tone = <{off|on}>
     Defines whether or not a tone is on during this phase of the pattern. If the value is
     off, the frequency and power parameters in this entry are ignored.
[freq1] = <number>
     First tone frequency in hertz.  A value of zero indicates this tone component is not
     used.
[power1] = <number>
     First tone power level in units of dBm.
[freq2] = <number>
     Second tone frequency in hertz.  A value of zero indicates this tone component is not
     used.
[power2] = <number>
     Second tone power level in units of dBm.
[freq3] = <number>
     Third tone frequency in hertz.  A value of zero indicates this tone component is not
     used.
[power3] = <number>
     Third tone power level in units of dBm.
[freq4] = <number>
     Fourth tone frequency in hertz.  A value of zero indicates this tone component is not
     used.
[power4] = <number>
     Fourth tone power level in units of dBm.
[duration] = <number>
     Duration of this phase of the tone pattern in milliseconds.
[nextentry] = <number>
     The entry ID for the next phase of the tone pattern, after the specified duration of this
     phase has completed. A value of zero indicates that the tone pattern is to terminate
     after the current phase is completed.
[maxloops] = <number>
     Maximum number of loops.
[nextentryafterloops] = <number>
     Next entry after loops.
```

Modifying a tone pattern

To modify a tone pattern, in this example the duration of ID91 is changed to 300 ms, carry out the following command:

```
:voice tone patterntable modify
id = 91
tone =
[freq1] =
[power1] =
[freq2] =
[power2] =
[freq3] =
[power3] =
[freq4] =
[power4] =
[duration] = 300
[nextentry] =
[maxloops] =
[nextentryafterloops] =
:voice tone patterntable modify id=91 tone=on duration=300
```

Deleting a tone pattern

To delete a tone pattern, carry out the following command:

```
:voice tone patterntable delete
id = <number>
     First entry ID of the tone pattern to delete.
[endentryid] = <number>
     Last entry ID of the tone pattern to delete.
```

The parameters id and endentryid must exist in the tone pattern table.

Flushing the tone pattern table

To flush all tone patterns, carry out the following commands:

```
:voice tone patterntable flush
```

6.12 Ringing

Introduction

Ringing is an alternative name for ringing current, such that a POTS phone connected to the FXS port rings. The same term is used for ringing of all types of phones : POTS phone, DECT handset.

There is a difference between ringing and ringing tone. For more information about ringing tone see “6.11 Tones” on page 90.

Configuration and usage on a per voice profile (e.g. SIP UA) basis is possible. However also a generic applicable configuration is defined, i.e. the configuration is applicable on all voice profiles. Currently only this generic configuration is supported.

- Ringing is currently only applicable towards POTS phones attached to the FXS interface, and not to the DECT handsets.
- Ringing is currently only supported for voice SIP

Following tables are used for the ringing configuration:

- ***Ringing event table***
  This table contains a list of events for which a ring profile is defined. The ring profile itself is specified in the ringing description table.
- ***Ringing description table***
  This table defines for every ring profile what must be generated: a ringing specified by a pattern in the ringing pattern table.
- ***Ringing pattern table***
  This table specifies the phases of the ringing. Per ringing phase (entry) it is specified if ringing must be on/off, duration and an entry to the next ringing phase.

Overview

In this section all of the commands concerning ringing are explained. You can find information on:

- “6.12.1 Ringing event table” on page 101
- “6.12.2 Ringing description table” on page 102
- “6.12.3 Ringing patttern table” on page 104

6.12.1 Ringing event table

Viewing the ringing event table

To view all ringing events, carry out the following command:

```
:voice ringing eventtable list
profile = <string>
     The profile on which future ringing operations must be done. Currently only the value
     all is supported.
```

Setting a ringing profile

To set a specific ringing profile for future ringing modification, carry out the following command:

```
:voice ringing eventtable set
profile = <string>
     The profile on which future ringing operations must be done. Currently only the value
     all is supported.
```

Modifying a ringing event

To modify a ringing event, carry out the following command:

```
:voice ringing eventtable modify
eventid = <number>
     Event ID.
ringid = <number>
     Ring ID. A ring ID of zero indicates ringing is disabled for this event.
```

Events

The table below gives a description of all available event IDs for ringing:

Event	Event ID	Default ring ID
Incoming call from VoIP.	001	1
CCBS: remote party came free.	011	1
Phone goes on-hook/ terminates active call while there was an incoming waiting call.	021	1
Phone goes on-hook/terminates active call while there was a call on hold	022	1

Flushing the ringing event table

To flush all ringing events, carry out the following command:

```
:voice ringing eventtable flush
```

6.12.2 Ringing description table

Viewing the ringing description table

To view all ringing descriptions, carry out the following command:

```
:voice ringing descrtable list
profile = <string>
     The profile on which future ringing operations must be done. Currently only the value
     all is supported.
```

Setting a ringing profile

To set a specific ringing profile for future ringing modification, carry out the following command:

```
:voice ringing descrtable set
profile = <string>
     The profile on which future ringing operations must be done. Currently only the value
     all is supported.
```

Adding a ringing description

To add a ringing description, carry out the following command:

```
:voice ringing descrtable add
ringid = <number>
     Ring ID.
status = <{disabled|enabled}>
     Enable or disable this ring description entry. If a disabled ring description is
     referenced, the result is that no ring is played.
ringname = <quoted string>
     Name of the ringing. By default the ring name is based on the entry ID: ring1, ring 2,...
[patternentryid]= <number>
     Entry ID in the ringing pattern table.
maxduration = <number>
     The ringing generation of the pattern stops when the maximum duration is reached (in
     seconds). A maximum duration of 0 seconds means there is no limitation.
```

Modifying a ringing description

To modify a ringing description, carry out the following command:

```
:voice ringing descrtable modify
ringid = <number>
status = <{disabled|enabled}>
ringname = <quoted string>
[patternentryid]= <number>
maxduration = <number>
```

Deleting a ringing description

To delete a ringing description, carry out the following command:

```
:voice ringing descrtable delete
ringid = <number>
     Ring ID.
```

Flushing the ringing description table

To flush all ringing descriptions, carry out the following command:

```
:voice ringing descrtable flush
```

6.12.3 Ringing patttern table

Viewing the ringing pattern table

To view all ringing patterns, carry out the following command:

```
:voice ringing patterntable list
profile = <string>
     The profile on which future ringing operations must be done. Currently only the value
     all is supported.
```

Setting a ringing profile

To set a specific ringing profile for future ringing modification, carry out the following command:

```
:voice ringing patterntable set
profile = <string>
     The profile on which future ringing operations must be done. Currently only the value
     all is supported.
```

Adding a ringing pattern

To add a ringing pattern, carry out the following command:

```
:voice ringing patterntable add
id = <number>
     ID of the ringing pattern.
ringing = <{off|on}>
     Enable or disable the ringer to be on/off for the specified period.
duration = <number>
     Duration of this phase of in the ring pattern (in milliseconds). A value of 0 indicates
an unlimited duration.
[nextentry]= <number>
     Entry ID for the next phase of the ring pattern, after the specfied duration of this
phase is reached. A value of 0 indicates that the ring pattern terminates after the current
phase.
```

Modifying a ringing pattern

To modify a ringing pattern, carry out the following command:

```
:voice ringing patterntable modify
id = <number>
[ringing] = <{off|on}>
[duration] = <number>
[nextentry]= <number>
```

Deleting a ringing pattern

To delete a ringing pattern, carry out the following command:

```
:voice ringing patterntable delete
id = <number>
     ID of the ringing pattern.
```

Flushing the ringing pattern table

To flush all ringing patterns, carry out the following command:

```
:voice ringing patterntable flush
```

6.13 Telephony statistics

Introduction

This section describes how the generic and detailed telephony statistics for each voice port can be retrieved.

The following definitions are applicable:

- An outgoing call is successful when the voice application detects the called number to be complete and results in bothway communication. For VoIP this means that two unidirectional RTP streams exist.
- An incoming call is successful when the voice application detects a setup message (INVITE message) with a called number recognized as belonging to one of the terminations within the Thomson Gateway, resulting in bothway communication.

Managing the telephony statistics using the Web pages (GUI)

To view the telephony statistics (call data statistics), proceed as follows:

1. On the Thomson Gateway home page, click **Toolbox**.
2. Click **Telephony**.
   Information about the 10 last calls is displayed in the section **Last Calls** (start time, local number, remote number, duration, port).
3. Click **View telephony statistics and logs** in the **Pick a task list**.
   The section **Call Statistics** shows the following parameters per voice port:
   - Incoming successful calls
   - Incoming missed calls
   - Outgoing successful calls
   - Outgoing failed calls

   The section **Call Log** shows the same information as **Last Calls**.

To reset the telephony statistics, proceed as follows:

1. On the Thomson Gateway home page, click **Toolbox**.
2. Click **Telephony**.
3. Click **View telephony statistics and logs** in the **Pick a task list**.
4. Click **Reset telephony statistics** in the **Pick a task list**.

Viewing the telephony statistics using CLI

To view the telephony statistics of a voice port, carry out the following command:

```
:voice stats list
voiceport = <{FXS1|FXS2|DECT|all}>
type = <{detailed|generic|all}>
```

The following types of telephony statistics can be consulted per voice port:

- Generic
  Number of incoming calls, number of successful incoming calls, number of missed incoming calls, number of successful outgoing calls, number of unsuccessful outgoing calls.
- Detailed
  - Call data statistics: Calling ID, calling port, called ID, called port, timestamp indicating start of connection, call duration. These statistics are shown for each successful and unsuccessful call via VoIP or FXO.

- RTP statistics: End-to-end delay, remote packets lost, remote jitter. These statistics are shown for each successful call involving VoIP.

Missed and unsuccessful calls will have a call duration of 0 seconds when shown.

- All
  Detailed and generic statistics.

The example below shows a full statistics report for all voice ports:

```
:voice stats list voiceport=all type=all
Generic statistics:

Total nbr of incoming calls      : 1
Successful nbr of incoming calls : 1
Missed nbr of incoming calls     : 0
Total nbr of outgoing calls      : 2
Successful nbr of outgoing calls : 2
Failed nbr of outgoing calls     : 0

Detailed statistics:

Outgoing call from 53671 at FXS1 to 53672 via VoIP registered at 2000-01-01T02:58:09Z (call
duration 2 seconds)
Mean end-to-end delay             : 0 ms
Worst end-to-end delay            : 0 ms
Remote packets lost (cumulative)  : 0
Remote packets lost (ratio)       : 0.0 packets/s
Remote packets lost (fraction)    : 0.0 %
Remote mean jitter                : 0 ms
Remote worst jitter               : 0 ms

Outgoing call from 53671 at FXS1 to 53660 via VoIP registered at 2000-01-01T02:58:29Z (call
duration 4 seconds)
Mean end-to-end delay             : 6 ms
Worst end-to-end delay            : 7 ms
Remote packets lost (cumulative)  : 0
Remote packets lost (ratio)       : 0.0 packets/s
Remote packets lost (fraction)    : 0.0 %
Remote mean jitter                : 0 ms
Remote worst jitter               : 0 ms

Incoming call from 53671 via VoIP to 53672 at FXS2 registered at 2000-01-01T02:58:09Z (call
duration 2 seconds)
Mean end-to-end delay             : 0 ms
Worst end-to-end delay            : 0 ms
Remote packets lost (cumulative)  : 0
Remote packets lost (ratio)       : 0.0 packets/s
Remote packets lost (fraction)    : 0.0 %
Remote mean jitter                : 0 ms
Remote worst jitter               : 0 ms
```

Resetting the voice statistics using CLI

To reset the statistics of a voice port, carry out the following command:

```
:voice stats reset
voiceport = <{FXS1|FXS2|DECT|all}>
type = <{detailed|generic|all}>
```

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