GTT USER MANUAL

Release 2.2

Software License Agreement

Any software product, including any documentation relating to or describing such software (hereafter collectively called “Software”), provided by GED is furnished to you for installation and use on a single computer. The Software may not be copied, in whole or in part, except for archival purposes, to replace a defective copy, or for program error verification.

You may not reverse engineer, decompile, or disassemble the Software, except to the extent such foregoing restriction is expressly prohibited by applicable law.

Unless earlier terminated by GED as herein provided, the term of each paid-up license shall expire at such time as you discontinue use of the applicable Software on the single processor specified above but otherwise shall be without restriction as to time.

The Software (including any images, applets, photographs, animations, video, audio, music, and text incorporated into the Software) is owned by GED.

Therefore, you must treat the Software like any other copyrighted material (for example, a book or musical recording) except that you may either (a) make one copy of the Software solely for backup or archival purposes, or (b) transfer the Software to a single hard disk provided you keep the original solely for backup or archival purposes. You may not copy the printed materials accompanying the Software.

You may not rent or lease the Software, but you may transfer the Software and accompanying written materials on a permanent basis provided you retain no copies and the recipient agrees to the terms of this Agreement. If the Software is an upgrade, any transfer must include the most recent upgrade and all prior versions.

TABLE OF CONTENTS

About This Book

This book presents information on the system design and operation of the GED GTT media gateway. The purpose of this information is to help you install, use, and maintain the GTT gateway.

Who Should Use This Book

This book is for product distributors, systems integrators, systems analysts, and network administrators who design, install, configure, and maintain wide area networks (WANs) and large-scale communications applications. It contains conceptual and practical information about how to use the GTT gateway within your network.

Conventions Used in This Book

This book uses specific conventions to show the following types of information:

- Number usage
- Replaceable input values
- Messages

Read the following sections to learn more about how this information is shown in the rest of the book.

Notes, cautions, and Warnings

Notes show useful information and/or contain information requiring attention.

Cautions show information requiring extra attention

Warnings are information that, if not followed, could result in injury or equipment damage.

How Numbers Are Used

When numbers are shown in this book, they can appear as descriptive values or as data to be manipulated internally. Decimal values are used frequently; however, alternate number bases are useful when internal data is shown.

Large Decimal Numbers

Numbers greater than 9999 display in SI metric style, where whole numbers that contain more than four digits are broken into groups of three digits that are separated by spaces. For example, the number sixteen thousand three hundred eighty three is shown as 16 383.This avoids confusion between American and European punctuation conventions. However, a number that is internally manipulated by a computer is shown without punctuation or spaces. For example, notice how the value 65 535 appears in the following instruction without a space or a thousands separator within the number:

`Specify 65535 as a maximum value`

Numbers with Different Bases

All numbers shown in this book are decimal values unless the number base is binary or hexadecimal. In these cases, an identifier precedes a binary or hexadecimal number. For example:

- the value of binary 1010
- the value of hex 4F

Replaceable Input Values

In some cases, you can insert user-defined values into commands or you can specify local paths and filenames. These variable values are shown in *italic* typeface.

For example, you might be asked to specify the name of your server in this path:

A:\LOGIN\LOGIN *servername*

The italic typeface shows that you need to replace *servername* with your local server name.

When prompted for variable input represented by lower-case letters, follow these conventions:

When You See This	Substitute This Value
b	Any binary digit
h	Any hexadecimal digit
n	Any decimal digit
x	Any alphabetic value, such as: **_x_:\DOS** where you substitute the correct drive letter for *x*
Multiple letters	A series of digits, such as: **FIRST 2 HEX BYTES: _hhhh_** where you substitute four hexadecimal digits for *hhhh*

When prompted for variable input with embedded decimal points, replace the variable digits and let the decimal points remain to separate32-bit dotted-decimal address segments. For example, you might be prompted to supply a 32-bit, dotted-decimal address in this format:

nnn.nnn.nnn.nnn

where ***nnn*** is a decimal value from 0 through 255. Leading zeros are not required.

GETTING HELP

If, after installing and configuring your GED equipment, you cannot establish Technologies to or from the unit, carefully review the information in this book prior to calling Customer Support (CS).

Before going any further, ensure that you have checked the following:

- Console Port configuration. Verify that the baud rate of your terminal is set to 115200 bps, data bits: 8, parity: none, stop bit: 1, flow control: none.
- Power reset. When cycling power, be sure to leave the power off for a minimum of 30 seconds before reapplying power to the unit.

Checklist

If, after carefully reviewing the information in this book and the GED website knowledge base, your problem persists, contact your product representative or a service representative at GED’s Customer Support (CS). So we can serve you better, make a list of the following items before calling:

- A detailed description of your problem.
- A complete listing of your system components and configuration, including the serial number of your unit and the current software version number.
- A narrative of the actions you performed prior to the problem.
- A list of all system messages posted by your unit.

Contacts

GED Development Co., Ltd.
Address：12F, Building 6, 1369 Dongfang Rd,
Office Phone：86-21-68640585
Fax: 86-21-50892248

1

INTRODUCTION

This chapter presents a high-level introduction to the GTT gateway. The GTT gateway provides voice transmission that enables high-quality, cost-efficient Technologies VoIP service.

Overview

GED's GTT product is designed to bridge the gap between traditional, circuit-based Public Switched Telephone Networks (PSTNs) and the emerging packet-switched networks. The GTT provides an excellent solution for merging digital broadband access networks with the legacy telephone network in a seamless, reliable manner.

Features

This section presents high-level information about the features of the GTT platform. It has been designed to serve smaller, cost efficient deployment environments that require a rich feature set.

One to Four T1/E1 Span Capacity

The GTT’s platform provides one to four T1/E1 spans of capacity (up to 240 voice channels). This allows carriers to identify the ideal size of their deployment and roll out appropriate levels of service.

Scalability

The GTT supports 1, 2 or 4 T1/E1 per chassis. This enables carriers to size the gateway to fit their specific need。

Processing Power

The GTT possesses 4800 MIPS processing power and supports multiple voice codec (G.711, G.729A, G.723, iLBC, GSM) as well as echo cancellation (G.168), DTMF relay (RFC2833), and fax relay (T.30, T.38).

Quick and Easy Installation

The GTT is packaged in a 1U chassis and can be quickly and easily installed using standard tools. It has been designed using industry standards and interoperates with major vendor’s soft-switches.

Simple Configuration

The GTT is configured and monitored via an intuitive built-in web GUI. The GUI provides password protected access from anywhere on the network.

Redundant and Hot Swappable Power Supply Modules

The GTT power supply modules are hot-swappable in the event of a physical failure. These modules require no special tools or training to perform a field replacement.

2

Chassis and Components

This chapter presents the functional architecture of the GTT gateway. It introduces the major gateway components, their functions, and inter-activities.

As shown in Figure 1 and Figure 4, the GTT chassis consists of a control module, T1/E1 module, power supply modules, and two fans. Interconnection is performed via a mid-plane which is functionally equivalent to a backplane. The following sections cover each component in more detail.

Chassis

Dimensions

The GTT chassis is one rack unit (1RU) high, or 1.75 inches (4.4 cm) high x 17.25 inches (43.82 cm) wide x 17.00 inches (43.18 cm) deep. It can be mounted on an Electronics Industry Association (EIA) standard 19inch relay rack or optionally, on a rack shelf or table.

Weight

The GTT chassis weighs ~ 15 lbs. (7 kg).

Front View

Figure 1 shows the front view of an GTT chassis gateway:

Figure 1. GTT Gateway, front View

Control Module

The GTT Control Module (shown in Figure 2) contains the control and processing circuitry and interfaces. The mainboard module includes a daughtercard that provides additional processing power. Voice capacity is controlled via the processing power in the mainboard.

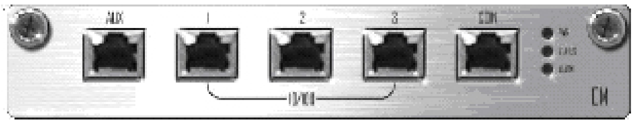


Figure 2. Control Module

AUX Port

The AUX (auxillary) port is not currently supported..

Port 1

Port 1 is used for local management access.

Port 2

Port 2 is not currently supported and will not be displayed in the GUI.

Port 3

Port 3 is an auto-negotiating 10/100Base-T Ethernet port. It is the IP port carrying the VoIP traffic, including SIP protocol and RTP packets. It may also be used to provide remote access for control, management, and maintenance. Pin-outs are shown as follows:

RJ45 Pin-out				LED Status	
1	2	3	6	Orange	Green
TX+	TX-	RX+	RX-	Activity	Link

CON (Console) Port

The Console port is used to provide console access to the GTT during manufacturing and test and is not intended for use in the field.

RJ45 Pin	1	2	3	4	5	6	7	8
Description	NC	NC	TxD	GND	GND	RxD	NC	NC
DB9 Pin			2		5	3		
DB25 Pin			3		7	2		

Indicators

The mainboard indicators have the following meanings:

Table 1: GTT Indicators

Indicator	Color	Description
PWR (Power)	Green	Power is on
	Off	Power is off
Status	Green (flashing)	Normal operation
	Red (flashing)	System is in a diagnostic mode with limited functionality
	Red (on)	System is in a start-up mode, not ready for operation
	Off	System is locked in a non-functional state
Alarm	Green	Normal operation (no alarms)
	Red (flashing)	Alarm condition from an unknown source
	Red (on)	Alarm condition with a known source

T1/E1 Module

The T1/E1 Module (shown in figure 3) always has four active RJ-45 connections. Pinouts are shown as follows:

RJ45 Pin-out	1	2	3	4	5	6	7	8
Description	RX_Ring	RX_Tip	NC	TX_Ring	TX_Tip	NC	NC	NC

Since the Control Module controls the capacity of the system, only the supported numberof T1/E1 lines should be connected.

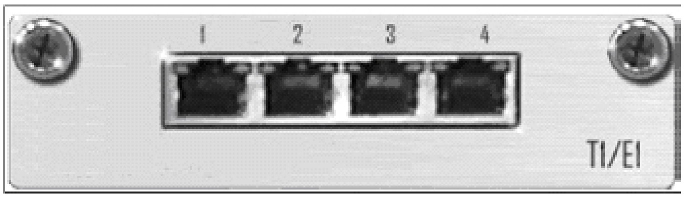


Figure 3. T1/E1 Module

Do not connect T1/E1 lines in excess of the system’s capacity. To determine the capacity of the GTT see the product label.

Rear View

Figure 4 shows the rear view of an GTT chassis.

Figure 4. GTT Gateway, Rear View

Power Supplies

The GTT contains two auto-ranging power supplies. They are capable of 100-240 VAC, 47-63 Hz. The supplies are redundant and hot-swappable. While both supplies are installed, both are running and sharing the power load of the GTT. The power light indicates that the supply is on and in use. The alarm light indicates that the indicated power supply is not functioning properly.

Fans

The GTT chassis contains two fans that provide cooling for the system. Both fans are required to be operational for the system to operate correctly in its specified ambient temperature.

3

INSTALLING THE GTT

The installation of the GTT gateway requires a few simple tasks.

Site Preparation

Power Input

The GTT requires 100-240 VAC, 47-63Hz. Maximum current consumption is 1A.

Redundant Power Feeds

GED recommends redundant power feeds from separate power sources or UPS power backup to ensure the continuous operation of the gateway. This ensures proper operation of the gateway in the event of a power loss from one of the sources.

Airflow and Thermal Cooling Requirements

The GTT gateway is equipped with two fans for cooling the system modules. The air intake is located on the right and left sides with the exhaust vents being located on the rear of the chassis. To prevent overheating, the air intake and exhaust must not be obstructed. After installing the gateway avoid installing other equipment that exhausts hot air directly into the GTT air intake.

The gateway produces heat at a maximum of 256 BTU/hr. The facility air conditioning system must be capable of handling the heat load to reduce the ambient air temperature to an acceptable level for proper operation.

Space Allocation Guidelines

The gateway can be installed in a 19 inch EIA standard open relay rack. An enclosed cabinet is not recommended unless it can provide sufficient cooling capacity for the gateway.

Chassis Mounting Guidelines

The chassis can be flush or mid-mounted. A single gateway requires 1.75 inches (1RU or 4.4cm) of vertical rack space, 17 inches (43cm) of rack depth and weighs approximately 15 lbs (7 kg).

Installing the Chassis

The GTT is shipped in a protective carton to prevent damage during shipping. The shipping carton contains the following items:

- GTT gateway chassis (with modules installed)
- Power cable assemblies
- GTT gateway documentation and software on CD

Standard configuration accessories can be added or removed based on customers' demands.

To install the chassis:

1. Remove the GTT chassis from its shipping carton.
2. The GTT chassis is shipped with its rack mount ears positioned for front (flush) mounting. To change the mount position of the chassis, use a #2 Phillips screwdriver to remove the three #8-32 screws affixing each rack mount ear. Reposition the rack mount ears to the desired mount position.
3. Lift the chassis into the rack so that the chassis mounting ears are aligned with the rack holes. Partially install one rack screw in the bottom corner of each side for the keyhole cutouts. Set the chassis over the partially installed rack screws, then install the remaining rack screws (rack screws not provided).
4. For racks with equipment shelves installed, lift the GTT chassis and position it on the shelf.

Connecting Main Power

The GTT gateway is configured for AC power. GED recommends two independent AC power feeds.

Installation must be performed by a qualified professional who is skilled in the installation and connection of power distribution systems, and is knowledgeable of applicable municipal electrical code requirements.

Applying Power:

1. Connect the supplied power cords to the inlets on the power supplies.
2. The GTT does not have an on/off switch. Plug the GTT. power cord into an outlet. Connect the other end of the power cord to the back of the GTT. The GTT fans will start and the LED indicators will light.
3. To ensure power is not accidentally removed, make sure all plugs are securely connected to the chassis and outlets and that power cables are secured.

Configuration

A brief introduction to using the GTT GUI can be found in Chapter 4 “Getting Started”. Detailed information on configuring the GTT is contained in Chapter 5, “Configuring the GTT”.

4

GETTING STARTED

This chapter explains the steps required to "get started". It also explains the GTT user interface, a web-enabled management tool, which consists of a set of user-friendly menus and screens. Using these menus, the GTT can be configured, monitored, and managed during its deployment.

Starting the GTT

The following procedure explains how to access the GTT:

1. Unpack the GTT (note: keep the packing material so it can easily be shipped again).
2. Power-up the GTT. Before doing so, observe the following precautions:

Always ground the GTT through the protective earth lead of the power cable. Before applying AC power to the GTT, verify that the main plug is inserted into a socket outlet provided with a protective earth contact only. The protective action must not be negated by using an extension cord (power cable) without a protective conductor (grounding). Interrupting the protective (grounding) conductor (inside or outside the unit), or disconnecting the protective earth terminal, can make operation dangerous.

Installation must be performed by a qualified professional who is skilled in the installation and connection of power distribution systems, and is knowledgeable of applicable municipal electrical code requirements.

3. The GTT does not have an on/off switch. Plug the GTT power cord into an outlet. Connect the other end of the power cord to the back of the GTT. The GTT fans will start and the LED indicators will light.
4. Wait until the GTT completes its diagnostics and boot-up sequence (about 2 minutes).
5. Establish a data connection to the GTT. When the GTT is first shipped, only Ethernet port #3 is configured. Connect an Ethernet cable from your PC to Ethernet port 3 on the front panel of the GTT.
6. Open the Microsoft Internet Explorer web browser (version 5.x or later) and type the address: http://192.168.2.240.
7. Continue reading section “Understanding the GTT Web- GUI” on page 23 for login instructions and information about the using the GUI.

Understanding the GTT Web-GUI

The GTT is configured, managed, and monitored using a built-in WebGUI. A screenshot is shown in Figure 3.

Figure 3. WebGUI Main Screen

The WebGUI has been designed to be simple and intuitive. The WebGUI automatically detects the hardware capability of the SGX-100. For example, if the GTT has capacity for two spans of T1/E1, the WebGUI will only show two T1/E1 interfaces (1 and 2) in the appropriate configuration screens. The following sections provide a brief tutorial on the GTT's WebGUI.

Introduction

After the GTT powers up, the GUI may be accessed by opening a standard web browser and entering the default address: **http://192.168.2.240**.

There are two user-levels available with the GTT. Administrator and Operator. The respective default passwords for each are

GTadmin and **operator**.

note

The GTT passwords are case sensitive.

The administrator account has full permissions while the operator can not change network configurations, perform password management, or restore factory default settings. The GTT allows multiple users to log on at the same time. The user with the highest privilege level is able to make changes and perform actions. In the event of two like-privilege users, the first to log-in has control. The remaining users, regardless of their user privileges, will only be able to view configurations and status. For information on your current privilege level see "Login User Info" on page 60.

caution

Gl recommends changing the passwords to prevent unauthorized ac system.

note

The WebGUI will automatically logoff a user after 10 minutes of inactivity. Because of this, it is important to logout using the menu and therefore not prevent access to the system by others (for 10 minutes).

note

To refresh your permissions, you must logout and re-login.

WebGUI Navigation Tree

Figure 4 below shows a high-level navigation tree detailing the different menus and screens found in the WebGUI.

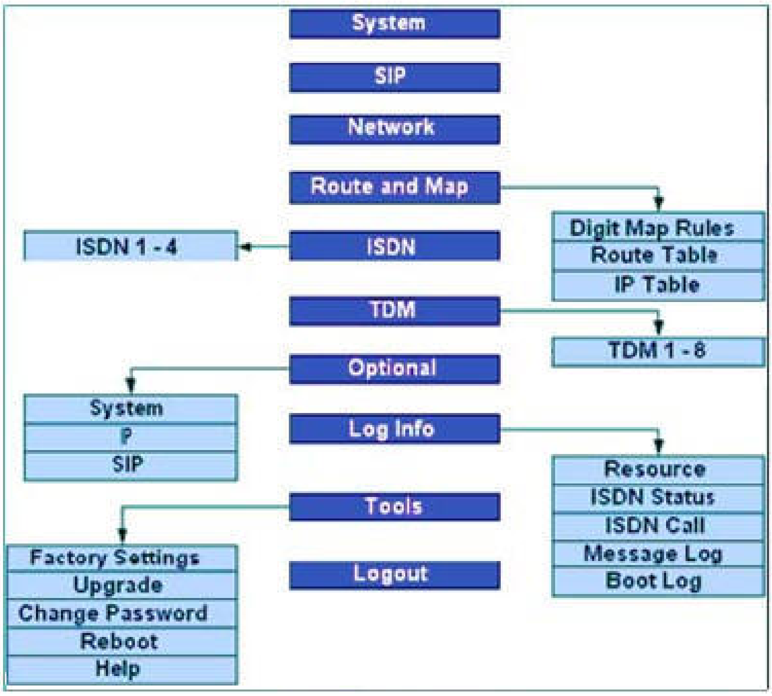


Figure 4. WebGUI Navigation Tree

CONFIGURING THE GTT

This chapter describes the configuration and management of the GTT gateway.

System Configuration

By selecting **system** configuration from the main screen, the **system** configuration screen is shown in Figure 5.

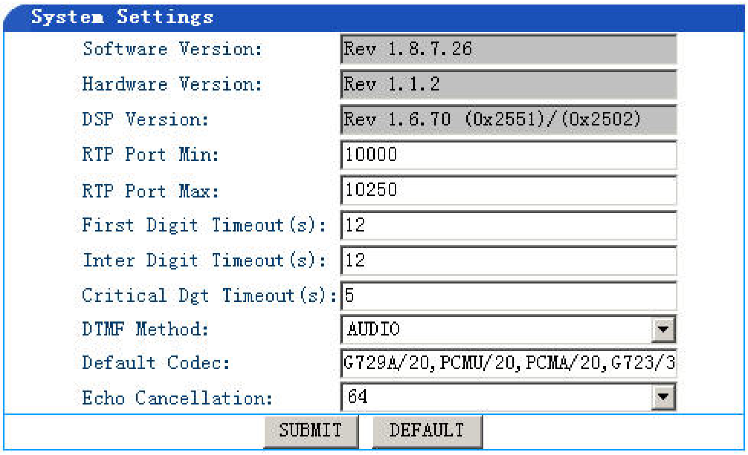


Figure 5. System Settings

Software Version

The **software version** is a read-only field that is automatically updated to show the current version of software running on the system.

Hardware Version

The **hardware version** is a read-only field that is automatically updated to show the current version of hardware. This field is used by GED support person to aid in troubleshooting.

DSP Version

The **DSP version** field is read-only and is updated automatically to show the current version of DSP software.

RTP Port Min and Max

The **RTP Port Min** and **RTP Port Max** settings define a range of available RTP ports to be used for voice traffic. The default settings are **10000** (min) and **10250**.

Since a VoIP call uses two RTP ports (one for RTP and the other for RTCP), one T1/E1 supports 23/30 simultaneous voice calls, and up to four T1/E1 ports are supported, the GTT requires 184/240 RTP ports.

First Digit Timeout

The **First Digit Timeout** value defines how long after SETUP event the GTT will wait before treating the call as abandoned and RELEASE the call. The default value is **12** seconds. This is only applicable for second stage dialing.

Inter Digit Timeout

The **Inter Digit Timeout** value defines how long after a digit has been dialed to wait to send the dialed number out. The default value is **12** seconds.

Critical Digit Timeout

This parameter is used in conjunction with the configured x.T dialing rules. After a specified head of the number (“x”) has been dialed, the GTT will wait the **Critical Digit Timeout** time period prior to sending the dialed number. For example, assuming that dialing rule 021.T has been configured. After the user has dialed 021 and the **Critical Digit Timeout** period has passed, the SGX-100 will send the dialed 021 out. The default value is **5** seconds.

DTMF Method

This parameter is used to set the DTMF signal transmit mode. Available settings are: **Audio mode**, **INFO mode**, and **2833 mode**.
Audio mode is a transparent transmit mode, **Info mode** is an information transmit mode, and **2833 mode** is a RTP event packet transmit mode.

Default Codec

The GTT can support multiple codec simultaneously. Supported codec are shown in Table 2. Enter the value shown in Table 2 to configure the GTT to use a particular codec for calls. To configure the support of multiple codec, enter them on one line separated by a comma (no spaces). The GTT will select the first codec (left-to-right) supported by both sides. The default configuration of codec is: **G729a/20**, **PCMU/20**, **PCMA/20**, **G723/30**, **GSM/20**, **iLBC/20**.

Table 2: Supported Voice Codec

Available Codec	Interval of RTP packet transmission (ms)	GTT Value
G.729A	20,30,40	G729A/20
G.723/30	30,60	G723/30
G.711 mu-law	20	PCMU/20
G.711 a-law	20	PCMA/20
GSM	20	GSM/20
iLBC	20	iLBC/20

Echo Cancellation

The **Echo Cancellation** may be **Enabled** or **Disabled**. Echo cancellation must also be turned on or off on a per-ISDN basis in the ISDN configuration window (see ”Echo Cancellation” on page 43).

Disabling Echo Cancellation overrides the setting of the same name found in the ISDN configuration window. (See “Echo Cancellation” on page 43.)

SIP Configuration

By selecting **SIP** from the navigation menu the **SIP** configuration screen will appear. (See Figure 6.)

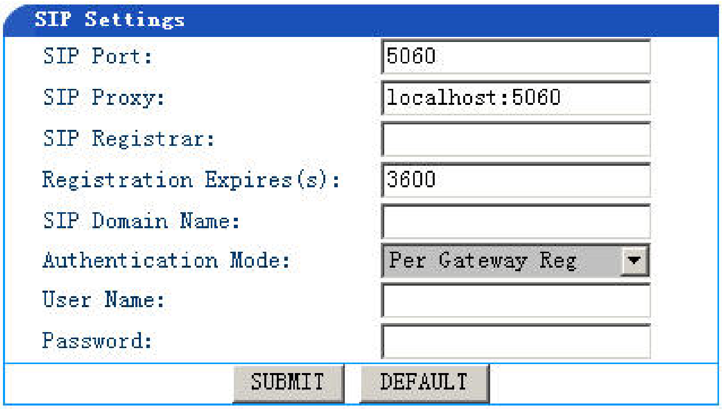


Figure 6. SIP Settings

SIP Port

The **SIP Port** parameter is used to set the SIP local port. The default value is **5060**. The value can be any setting so long as it isn’t used elsewhere.

SIP Proxy

SIP Proxy allows a SIP Proxy to be designated for the GTT.

The format of the **SIP Proxy** is the address or domain name separated from the port number by a colon. Two examples of valid settings are:

- 202.202.2.202:5060
- softswitch.com:5060

When using the domain name format of an address, you must configure and activate the DNS service in the “Network Setting” screen.

SIP Registrar

SIP Registrar allows SIP registrar server to be identified.

The format of the **SIP Registrar** is the address or domain name separated from the port number by a colon. Two examples of valid settings are:

- 202.202.2.202:5060
- softswitch.com:5060

When using the domain name format of an address, you must configure and activate the DNS service in the "Network Setting" screen.

Registration Expires

The **Registration Expires** parameter defines how often the GTT re-registers with the SIP server. The default value is **3600** seconds.

SIP Domain Name

SIP Domain Name defines the domain name. If the **SIP Domain Name** is not configured, the GTT will use the address of the **SIP Proxy**.

The format of the **SIP Registrar** is the address or domain name separated from the port number by a colon. Two examples of valid settings are:

- 202.202.2.202:5060
- softswitch.com:5060

When using the domain name format of an address, you must configure and activate the DNS service in the "Network Setting" screen..

Authentication Mode

The **Authentication Mode** parameter defines how the gateway will register with the proxy/server. It may be configured to **Per Gateway Registration**, **Per Endpoint**, or **Per Gateway Authorization**.

User Name

The **User Name** parameter is the user name that will be used to register with the proxy.

Password

The **Password** parameter is the password that will be used to login to the proxy.

Network Configuration

By selecting **SIP** from the main screen, the **SIP** configuration screen is shown in shown in Figure 7.

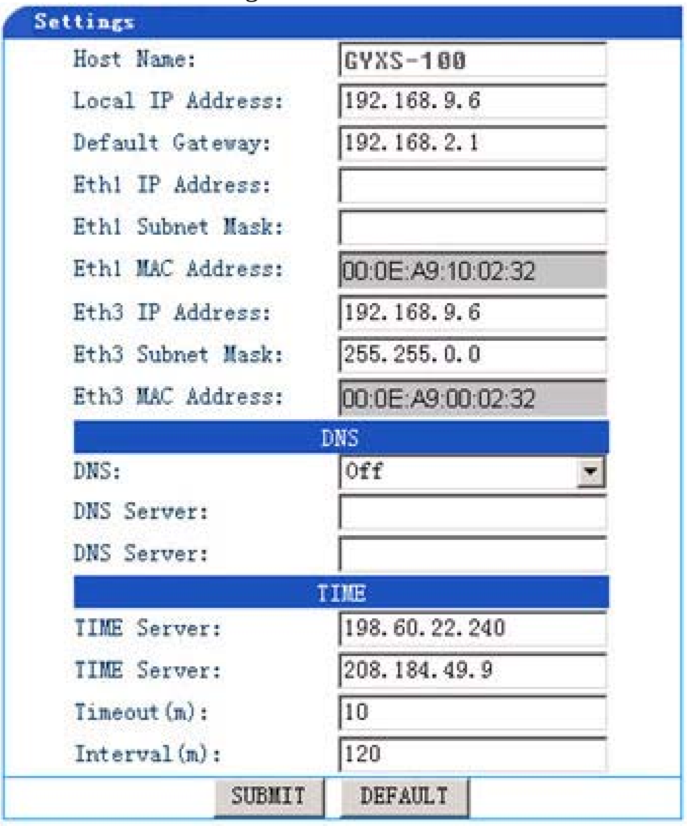


Figure 7. Network Settings

Host Name

The **Host Name** parameter allows giving the GTT a name. For example, it could be set to “GTT-3rdSt-NYNY”. Valid domain name characters and symbols are allowed.

Local IP Address

The **Local IP Address** parameter reports the address that is currently being used. This value is useful to make sure management connectivity with the system is not lost when making configuration changes.

Default Gateway

The **Default Gateway** parameter allows configuring the default IP route for the GTT.

Ethernet Configuration

While there are three Ethernet ports on the GTT only Ethernet ports one and three may be configured.

IP Address

The **IP Address** parameter configures the IP address of the Ethernet port. The standard IP address format is accepted: **aaa.bbb.ccc.ddd**. The default setting is: **<u>192.168.2.240</u>**.

Make sure this address is valid and accessible from where you are. An invalid setting will require a site-visit to recover remote access to the GTT.

Subnet Mask

This configures the **subnet mask** of the Ethernet port. The format of the mask is: **aaa.bbb.ccc.ddd**. The default setting is: **<u>255.255.255.0</u>**.

MAC Address

The **MAC Address** is the hard-coded read-only hardware address of the Ethernet port. This value can be used for detailed debugging and troubleshooting.

DNS

The DNS parameter configures the DNS functionality **Enable** or **Disable**. When DNS is **Enabled**, up to two DNS servers may be entered. The first server is the primary and the second is used as a backup. The standard IP address format is accepted: **aaa.bbb.ccc.ddd**.

EMS Server

Not currently supported.

TIME

The GTT allows the entry of up to two TIME servers. The first entry will be used as the primary server and the second entry as the backup.

TIME Server

The **Time Server** parameter allows the configuration of the IP addresses of two time servers (primary and backup). The standard IP address format is accepted: **aaa.bbb.ccc.ddd**.

Timeout

If the GTT can not find the TIME server after the **Timeout** has expired then it will look for the TIME server at the backup location. The default **Timeout** period is **10** minutes.

Interval

The **Interval** parameter designates how often the GTT will query the TIME server to update and synchronize its time. The default **Interval** is **120** minutes.

Route and Map

The following section details the Digit Map, Route Table, and IP Table capabilities of the GTT.

Digit Map

As shown in Figure 8, the digit map may optionally be configured with various dialing rules to determine if the dialed number sequence is correct or not.

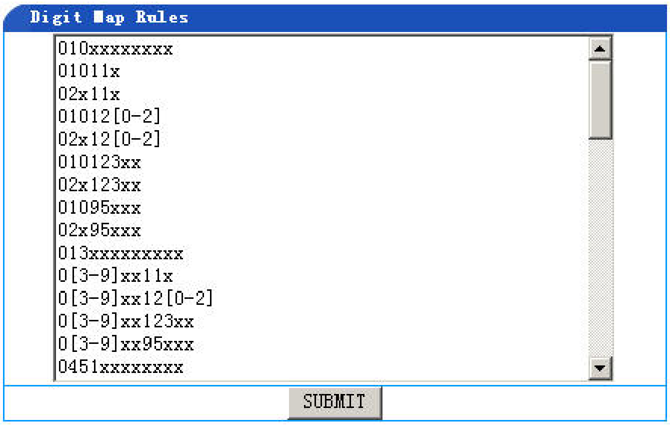


Figure 8. Digit Map Table

Usage

The following table shows generic examples for the usage of the Digit Map feature.

Table 3: Digit Map Usage

Setting	Description
x	Any number ranged from 0- 9
“.”	Represents any multidigit number ranged from 0- 9
x.T	The GTT allow a phone number of random length comprised of digits 0-9. If within the time specified (see "Critical Digit Timeout" on page 27) no new digit has been received, the gateway will send out the dialed number
x.#	If after the entry of a phone number of random length comprised of digits 0-9 the ”#” key is pressed, the number will be sent.
[2-8]xxxxxx	A seven digit number starting with any number 2-8 will be sent

Table 3: Digit Map Usage

Setting	Description
91x	A three digit number starting with 91 which is used to finish an emergency call
911	Send the number 911 immediately after being received.

Route Table

The Route Table (shown in Figure 9) has two functions in the GTT, digit translation and routing. Digit translation and routing can be applied to both directions of a call, either from IP or from the PSTN. If the Route Table is empty:

- Calls from IP addresses will select the first available (nonfull) T1/E1. (1, 2, 3, then 4)
- Calls from the PSTN will be sent to the IP address configured in "SIP Proxy" on page 29.

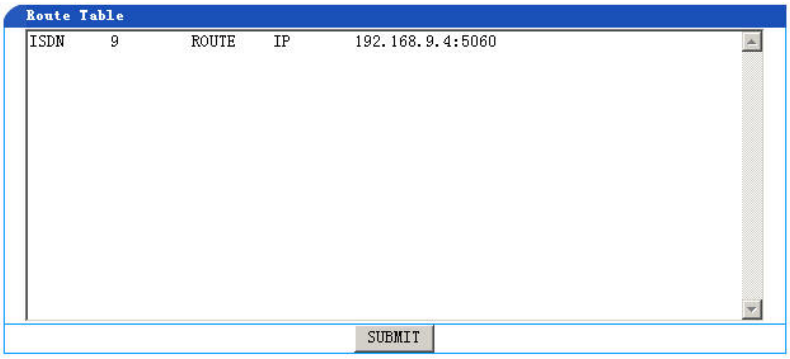


Figure 9. Route Table

Route Table Syntax

Route Table statements use the following syntax:

Format: **Source Number Handle [Parameter]**

or,

Format: **Source Number** ROUTE **Destination [Parameter]**

Source

Source can be **ISDN** or **IP**. When **source** is **IP**, an address can optionally be specified, e.g., [xxx.xxx.xxx.xxx] or [xxx.xxx.xxx.xxx:port]

Number

The default number is called party number. If the calling party number is being used, add **CPN** in front of the number. The **Number** can be:

- A specific number like (114, 61202700)
- A number prefix like (61., 61x5 or 61)
- A number range like (268[0-1,3-9])

Handle

The **Handle** parameter can be:

- **KEEP** digits from left or right
- **REMOVE** digits from the left or right
- **ADD** prefix or suffix number
- **REPLACE** prefix or the whole number
- **END** digit manipulation
- **SEND180** force send 180 on ring back
- **SEND183** force send 183 on ring back
- **HIDE** calling party number presentation
- **CODEC** for the call
- **RELAY -** First connect to a configured number, then, after connected, out pulse the real number.

Destination

The **Destination** parameter can be:

- **NONE** (call restricted)
- **IP -** The address and optional port number are specified. To specify local IP, the following entries are valid: “localhost: 5060” or “127.0.0.1:5060”
- **ISDN -** ISDN span number can be specified.

Route and Map Examples

The following examples are designed to show how route and digit mappings can be used effectively.

Examples - Digit Maps

IP 02161202700 KEEP -8 ;61202700

IP 021 REMOVE 3 ;Any number start with 021, the 021 prefix is removed

IP CPN6120 ADD 021 ;CPN number start with 6120, prefix 021 is added

IP CPN6120 ADD -8888 ;CPN number start with 6120, 8888 is appended

ISDN CPN88 REPLACE 2682000 ;CPN number started with 88, the prefix “88” is replaced with 2682000

ISDN CPN88. REPLACE 2682000 ;CPN number started with 88, the whole number is replaced with 2682000

IP CPN2 SEND180 ;CPN number start with 2, always send 180 on ring back

IP CPN3 SEND183 ;CPN number start with 3, always send 183 on ring back (voice cut through)

IP[61.2.44.53:5060] CPNX. HIDE ;Any call from 61.2.44.53:5060, calling party number presentation restriction is applied

IP 6120 CODEC PCMU/20/64 ;Any number start with 6120, use 20ms pTime PCMU codec with 64ms echo cancellation

IP 010 RELAY 17909 ;Calls to 010x, first connect to 17909, after the call connected, then pulse out the called party number

Examples - Digit Map - End Handle

The END handle is used to terminate digit manipulation for certain condition. After the END statement, no digit manipulation will be applied on further matches. Refer to the example below:

IP 12345 ADD -8001 ;add suffix 8001

IP 12345 REMOVE 4 ;then remove first 4 digits

IP 12345 END ;stop digit manipulation, no change on later matching condition

IP 12345 REPLACE 777 ;won't take effect for 12345x

IP 12 KEEP -3 ;won't take effect, but any number start with 12 other than 12345 will take effect

IP 123456 ADD -8002 ;won't take effect, if this rule is needed it should be moved before the 12345 rule and put an END statement after it

Examples - Route Functionality

The following examples cover the **ROUTE** functionality:
IP 8621 ROUTE ISDN 1 ;call has 8621 prefix, route to ISDN span 1

IP CPN8620 ROUTE ISDN 2 ;calling party number started with 8620, route to ISDN span 2

ISDN 021 ROUTE IP 228.167.22.34:5060 ;call has 021 prefix, route to 228.167.22.34:5060

ISDN 020 ROUTE IP 61.234.67.89:5060 ;call has 020 prefix, route to 61.234.67.89:5060

IP CPN[1,3-5] ROUTE NONE ;calls from CPN starting with 1, 3, 4 and 5 will be rejected

Example - Using Route Instead of End

The following example shows how the **ROUTE** statement can be used instead of the **END** statement to stop digit manipulation. In this case, further matching on the same condition won't take place.

IP 12345 ADD -8001 ;number started with 12345 add suffix 8001

IP 12345 REMOVE 4 ;then remove first 4 digits

IP 12345 ROUTE ISDN 2 ;then stop further matching digit manipulation and route to ISND span 2

IP[222.34.55.1] CPNX. REPLACE 2680000 ;Calls from 222.34.55.1, Calling party number is replace with 2680000

IP[222.34.55.1] CPNX. HIDE ;then calling party number presentation restriction is applied

IP[222.34.55.1] CPNX. ROUTE ISDN 2 ;then route to ISDN span 2.

Example - Relay Traffic to Another VoIP Platform

In the following example, traffic will be moved to another VoIP platform
that has a common access number (e.g. 17909).

IP[222.34.55.1] CPNX. REPLACE 2680000

IP[222.34.55.1] CPNX. RELAY 17909

All calls from 222.34.55.1 will have their calling party number replaced with 2680000. Next, after 17909 cut-through, the GTT will pulse out the CDPN (in-band DTMF) and let the 17909 platform make the final connection.

Example - Traffic Redirection

Sometimes the IP traffic to the GTT has a CDPN prefix that tells us where to relay the traffic.

IP 17909 REMOVE 5
IP 17909 RELAY 17909
IP 17909 ROUTE ISDN 2,3

First the GTT removes the CDPN prefix that is used for routing (17909 is removed). Next the GTT calls the 17909 access number. After connecting to the 17909 platform, the rest of the CDPN is pulsed out (in-band DTMF) and the connection is made by the 17909 platform.

IP Table

The IP Table (shown in figure) is used to identify which SIP Server IP address(es) may use the GTT. Addresses appearing in the table are allowed to use the GTT. If the table is left **empty** then all addresses are allowed.

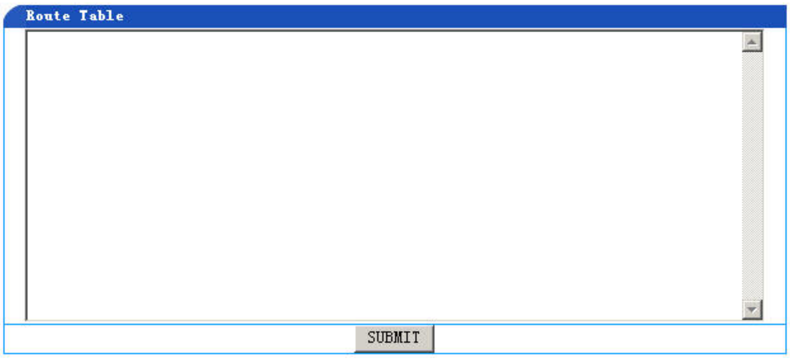


Figure 10. IP Table

ISDN Configuration

The GTT always has four T1/E1 interfaces. Capacity is determined by the processing power purchased and supplied by the control module. A specific T1/E1 interface is selected by choosing **ISDN Config** from the left menu pane on the **main screen**. The specific ISDN span can then be selected and configured. Configurable for the ISDN spans are shown in Figure 11.

Do not connect T1/E1 lines in excess of the system's capacity. To determine the capacity of the GTT see the product label.

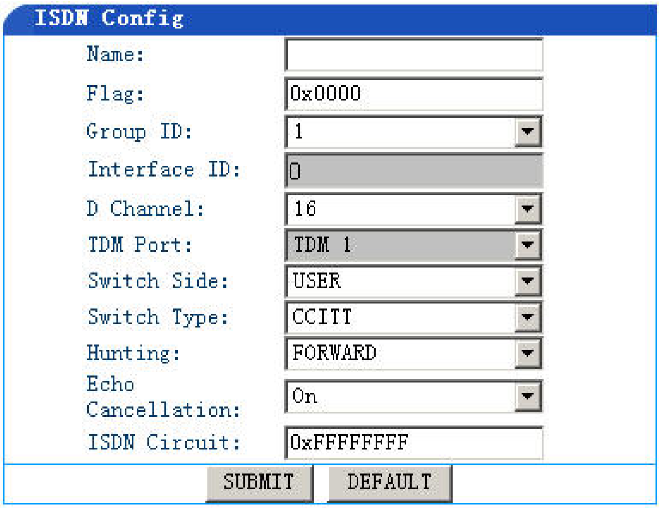


Figure 11. ISDN Configuration

Name

The **Name** parameter reports which ISDN span is currently being configured.

Flag

The **Flag** parameter is used to enable multiple special handling functions on a particular ISDN. ISDN flag is bit oriented. Each bit may be set to 1 to enable special handling. Multiple bits can be set to enable multiple functions on particular ISDN. The default setting of the **Flag** parameter is **0x0000**. Various settings and their behavior are explained in the following table.

Table4: ISDN Flag Settings

Setting	Behavior
0x0000	Default
0x0001	The GTT will send SERVICE message when ISDN layer 2 is up, and before layer 3 is ready. Some PBXs require this message in order to function properly。
0x0002	Allow cut-through without called party number (CDPN). (Normally ISDN call without a CDPN will be rejected.) A typical application is for large company PBX expansion: PBX1 => GTT1 => IP => GTT1002 => PBX2.
0x0004	Enable second stage dialing. When an ISDN call comes in, the GTT will cut-through and play dial tone while collecting digits for further routing. This is used in calling card or PSTN-VoIP number bonding applications.

Table4: ISDN Flag Settings

Setting	Behavior
0x0008	This setting is used in conjunction with second stage dialing (0x0004) and plays an announcement instead of dial-tone for digit collection. The GTT has *.pcm files (audio format) that can be customized and loaded for this application. Contact GED support for more details.
0x0010	This setting is used in conjunction with second stage dialing (0x0004). If configured, after digit collection, the GTT will use the original CPN for

	the CPN of the second stage dialing of the call. (Otherwise the original CDPN will be used as the CPN of the second stage.)
0x0020	This setting is used in conjunction with second stage dialing (0x0004). If configured, after digit collection the collected digits are attached to the original CPN and sent to the softswitch as the CDPN for further routing. (Otherwise the collected digits are sent to the softswitch as the CDPN for further routing.)
0x0100	Any CPN not starting with 0 will be set to type: SUBSCRIBER. (Otherwise, type will be unknown.)
0x0200	Any CPN starting with 0 will be set to type: NATIONAL. (Otherwise, type will be unknown.)
0x0400	Any CPN starting with 00 will be set to type: INTERNATIONAL. (Otherwise, type will be unknown.)
0x0800	Restrict CPN presentation for the entire ISDN.

Group

The **Group** parameter defines the group ID of the selected ISDN line. It is not recommended to change the value. T1/E1 interface to group mapping is interface# : group #. For example, T1/E1 interface #3 is assigned ISDN group #3.

Interface ID

The **Interface ID** parameter default setting is **0**.

D Channel

The **D Channel** parameter defines the signaling channel. Typically the signaling channel for a T1 interface is **24**, and **16** for an E1 interface.

SPAN

The **SPAN** drop-down menu reports which interface is being configured.

Switch Side

Switch Side defines the ISDN behavior. The setting of **Switch Side** for the other side of the T1/E1 line must be opposite that of the GTT. The settings for **Switch Side** are **User** and **Network**.

Switch Type

The default settings for **Switch Type** are:

- T1 - **NI2**
- E1 - **CCITT**

Hunting

The **Hunting** parameter is used to set how the GTT searches an idle time-slot. **Hunting** can be set to **Forward** or **Backward**.

Echo Cancellation

Using the **Echo Cancellation** drop-down menu, echo cancellation can be turned **Enable** or **Disable** for an individual ISDN line.

Setting the system parameter Echo Cancellation (see “Echo Cancellation Length” on page 13) to **Disable** will turn off Echo Cancellation at the system level.

ISDN Circuit

The **ISDN Circuit** is a bit representation of all circuits for a TDM. Each bit is set to “1” to enable a voice channel. The default is: **0xFFFFFFFF** (all on **-** E1 30 channels, T1 23 channels).

For example **0xF0FFFFFF** means circuit 0-23 and 28-31 are enabled and circuit 24-27 are Disabled. This configuration will be combined with T1/E1 settings as well. The E1 application uses time slot 0 for synchronization and time slot 16 for DCH, so for the above example, the actual available voice channels are 1-15, 17-23 and 28-31. A T1 application has only 24 channels and the last one is for DCH, so for the above example, the actual available voice channels are 0-22 (or 1-23 T1 timeslot).

TDM Configuration

The GTT always has four T1/E1 interfaces. Capacity is determined by the processing power purchased and supplied by the mainboard. A specific T1/E1 interface is selected by choosing **TDM Config** from the left menu pane on the **main screen**. As shown in Figure 12, TDM settings can then be selected and configured.

Do not connect T1/E1 lines in excess of the system's capacity. To determine the capacity of the GTT see the product label.

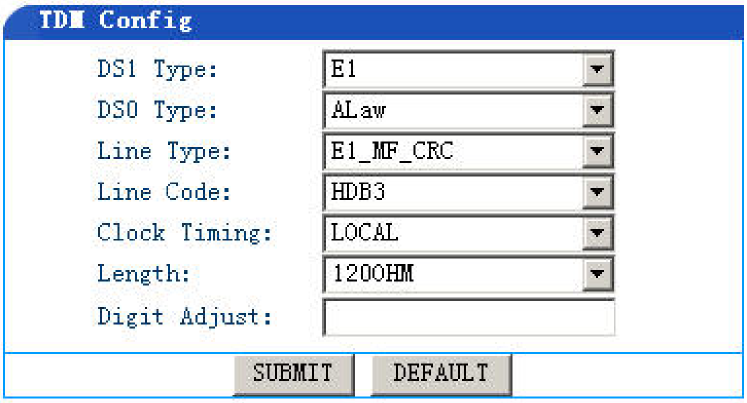


Figure 12. TDM Configuration

DS1 Type

DS1 Type configures if the T1/E1 interface operates as a **T1** or **E1** interface. Select the desired setting from the **DS1 Type** dropdown box.

DS0 Type

The **DS0 Type** allows configuring the PCM encoding type. Allowed settings are **ULaw** and **ALaw**.

Line Type

If the GTT **DS1 Type** is set to **T1** then **Line Type** can be set to **D4**, **SF** (Superframe), **ESF** (Extended Superframe), or **T1_UNFRAMED** mode.

If the GTT **DS1 Type** is set to **E1** then **Line Type** can be set to **E1_CRC**, **E1**, **E1_UNFRAMED**, or **E1_MF** mode.

Line Code

If the GTT **DS1 Type** is set to **T1** then **Line Code** can be set to **B8ZS** or **AMI**.

If the GTT **DS1 Type** is set to **E1** then **Line Code** can be set to **HDB3 .**

Clock Timing

The **Clock Timing** parameter can be configured as **Local** (use local clock)or **SPAN** (use recovered clock from the interface).

Note: If one interface is configured to use SPAN clock, all other interfaces will use the same recovered clock. When more than one interface is configured to use SPAN clock, GTT will adopt the recovered clock from the first upped interface.

Length

The **Length** setting configures the line build out (LBO) of the T1/E1 line. The default setting for T1 is **Shorthaul / 110 FT**. The default for E1 is **120 Ohm**.

Digit Adjust

This parameter is not currently supported.

Optional Settings

The **Optional Settings** screens can be accessed by selecting **Optional** from the navigation area on the left side of the WebGUI screen.

Optional System Settings

Figure 13 shows a screenshot of the optional system settings.

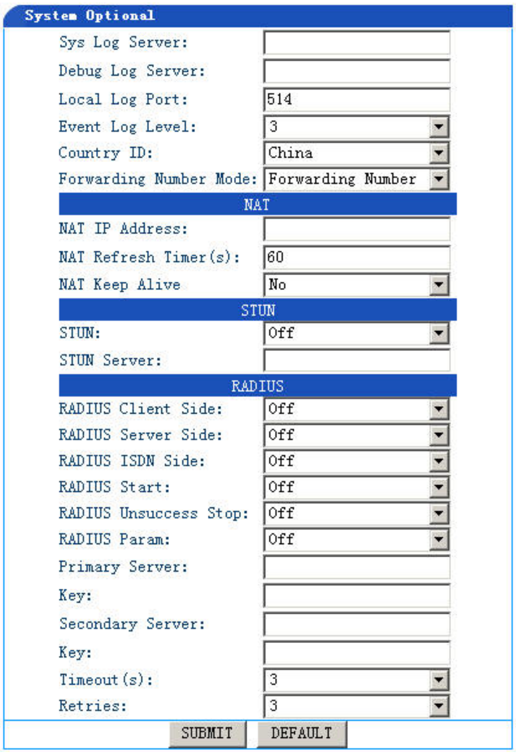


Figure 13. Optional System Settings

Event Log Level

The **Event Log Level** defines the significance of an event that will be logged. Allowed values are **1-5**. The default setting is **3**. The higher the number the greater number of event types will be recorded.

It is not recommended that the Event Log Level be set to a number greater than 3 for a normally functioning system. Settings greater than 3 will affect system performance.

Country ID

The **Country ID** is used to allow the GTT to customize various information presented to the user. The default is **US**.

Forward Number Mode

When call forwarding is used, this parameter defines which number is indicated as the originating telephone number. **Calling Party Number** designates that the calling party number is indicated while **Forwarding Number** indicates the number responsible for forwarding the call.

NAT

IP Address

The **IP Address** field configures the IP address of the GTT's public IP address and port number. The format of the entry is **aaa.bbb.ccc.ddd:eeee** .

Refresh Timer

The **Refresh Timer** configures the frequency with which NAT information is refreshed. The default is **60** seconds.

Keep Alive

NAT Keep Alive may be **Enabled** or **<u>Disabled</u>**.

RADIUS Settings

The GTT supports RADIUS for accounting purposes. RADIUS parameters are covered in the following sections.

RADIUS Client Side

The **RADIUS Client Side** parameter defines whether or not the charging function of the RADIUS client is used. **Radius Client Side** can be set to **Enable** or **<u>Disable</u>**.

RADIUS Server Side

The **RADIUS Server Side** parameter defines whether or not the charging function of the RADIUS server is used. **Radius Server Side** can be set to **Enable** or **Disable**.

RADIUS ISDN Side

The **RADIUS ISDN** parameter defines whether or not the charging function of the RADIUS ISDN is used. **Radius ISDN** can be set to **Enable** or **<u>Disable</u>**.

RADIUS Start

If **RADIUS Start** is set to **Enable**, the GTT will transmit the RADIUS record of the call start and end. If **RADIUS Start** is set to **Disable**, the GTT will not transmit the RADIUS record of the call start or end.

RADIUS Unsuccessful Stop

The **RADIUS Unsuccessful Stop** parameter defines whether the GTT transmits a RADIUS record of an abandoned call when the charging function of the RADIUS client or server is turned on. This feature may be configured to **Enable** or **Disable**.

RADIUS Parameter

If the **RADIUS Parameter** is set to **Enable**, then the GTT transmits H323 CONF_ID. If it is set to **Disable**, then is H323 CONF_ID doesn't exist, it is replaced with CALL ID.

Primary and Secondary Server

The GTT allows the configuration of up to two RADIUS servers. To configure a RADIUS server, its IP address and port must be entered. If the port number is left off, the default port of **1813** will be used. The format of the entry is **aaa.bbb.ccc.ddd:nnnn**.

Key

A **key** must also be entered to protect communication between the RADIUS client and server. The settings of the **key** between the client and server must match.

Timeout

The **Timeout** parameter defines the timeout period for the RADIUS functionality of the GTT. The default is **3** seconds.

Retries

The **Retries** parameter designates the number of times information will be retransmitted when an acknowledgement is not received. The default setting is **3** times.

Optional IP Settings

Figure 14 shows a screenshot of the optional IP settings.

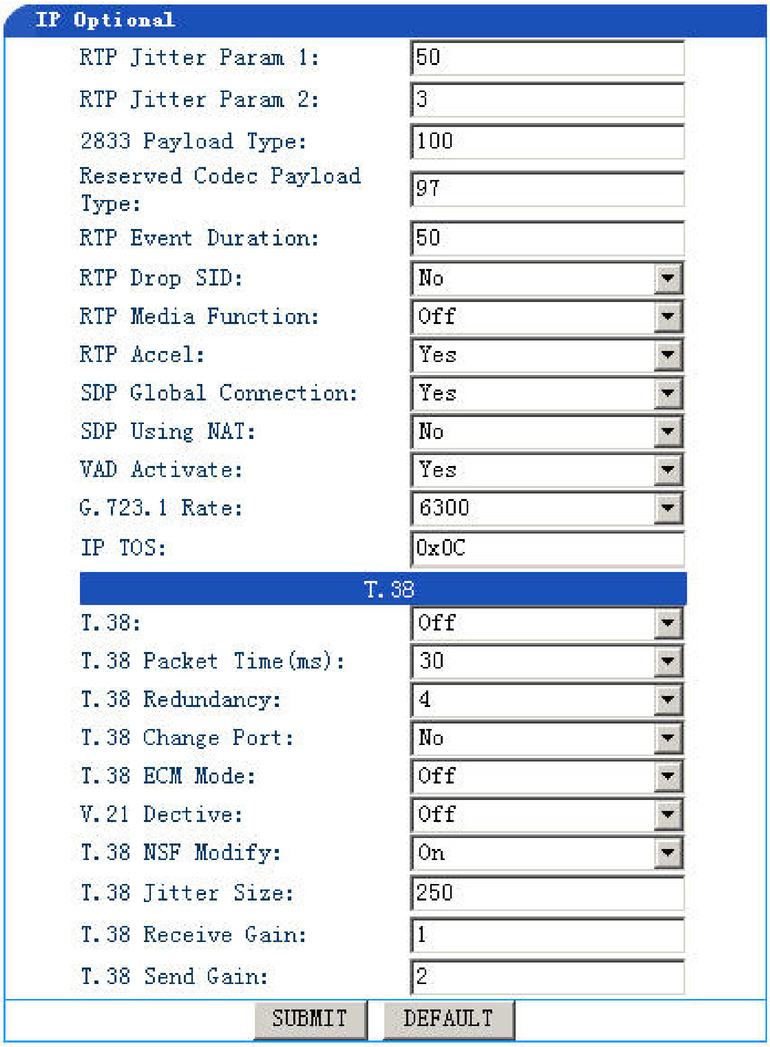


Figure 14. Optional IP Settings

RTP Jitter Param

The **RTP Jitter Parameter 1** and **RTP Jitter Parameter 2** parameters define the maximum and minimum number of frames in the jitter buffer. The **RTP Jitter Parameter 1** has a default of **50** and **RTP Jitter Parameter 2** has a default of **3**. Under normal situations, GED recommends these settings not be modified.

2833 Payload Type

The **2833 Payload Type** parameter sets the payload type used for transmitting 2833 packets. The allowable range is **97 - 127**.

RTP Event Duration

The default setting is **50**. This entry is only applicable when DTMF mode is set to 2833. It set time interval in millisecond to send 2833 RTP event. When a valid in-band signal is detected, the 2833 RTP event will be sent out in specified interval in the whole duration of the signal. When the in-band signal is detected OFF, the 2833 RTP event will be sent out 3 times at specified interval.

RTP Drop SID

The **RTP Drop SID** parameter may be set to **Enable** or **Disable**. This entry is used to specify if received RTP silent packet will be dropped. If Enabled, GTT will drop RTP silent packet (for CODEC G.729, G.723).

RTP Media Function

The **RTP Media Function** parameter may be set to **Enable** or **Disable**. RTP Media Function is used to support IAD under NAT without soft switch or Media Server assistance. When an IAD under NAT uses GTT as its proxy, enable RTP Media Function will allow GTT to cross NAT to setup media path. We suggest you enable this function in GTT.

SDP Global Connection

To allow **SDP Global Connections**, configure this parameter to **Enable**. When set to **Disable**, global SDP connections will not be allowed.

SDP Using NAT

This parameter may be set to **Enable** or **Disable**.

VAD Activate

The **VAD Activate** parameter may be set to **yes** or **no**. When set to **yes**, the GTT will not send speech packets of silence and instead begin CNG (comfort noise generation) to substitute for the unsent speech packets.

G.723.1 Rate

The G.723.1 codec can be set to two encoding rates: **5300** and **6300** bps.

IP TOS

The **IP TOS** parameter is used to set the TOS value on packets. This parameter is set using HEX settings. The default value is **0x0c**. The following table represents other settings.

Table 5: IP TOS HEX Settings

HEX	Description
0x00	Normal service (default)
0x02	Minimize cost
0x04	Maximize reliability
0x08	Maximize throughput
0x10	Minimize delay

It is also possible to set multiple bits. For example a setting of **0x18** would set **IP TOS** to maximize throughput and minimize delay.

T.38

The **T.38** setting **Enables** or **Disables** the T.38 fax function. If **T.38** is enabled it is necessary to configure the following T.38 parameters.

T.38 Packet Time

The **T.38 Packet Time** parameter is used to set the packing interval of the T.38 data frame. The value can be set to **10**, **20**, **30**, **40**, **50**, or **60** ms.

T.38 Redundancy

The **T.38 Redundancy** parameter configures the number of T.38 data frames in each T.38 data packet. **T.38 Redundancy** may be set from **1 - 6**. The default setting is **4**.

T.38 Change Port

When configured to **Enable**, the GTT will change the UDP port when switching to T.38 mode. If **T.38 Change Port** is set to **Disable** the GTT will reuse the RTP port number that was used in the creation of the connection.

T.38 ECM Mode

The **T.38 ECM Mode** may be set to **Enable** or **Disable**.

V.21 Detect

The **V.21 Detect** parameter may be set to **Enable** or **Disable**.

T.38 NSF Modify

The **T.38 NSF Modify** parameter may be set to **Enable** or **Disable**.

T.38 Jitter Size

The **T.38 Jitter Size** parameter defines the jitter size. The default setting is **250**.

T.38 Receive Gain

The **T.38 Receive Gain** parameter defines the receive gain. Set between 1 and 4, while set to 1 will adjust signal to one half of its original amplitude, set to 2 will keep the original signal amplitude, set to 3 will adjust signal to 2 times of its original amplitude, set to 4 will adjust signal to 4 times of its original amplitude. The default setting is **1**.

T.38 Send Gain

The **T.38 Send Gain** parameter defines the send gain. Set between 1 and 4, while set to 1 will adjust signal to one half of its original amplitude, set to 2 will keep the original signal amplitude, set to 3 will adjust signal to 2 times of its original amplitude, set to 4 will adjust signal to 4 times of its original amplitude . The default setting is **2**.

Optional SIP Settings

Figure 15 shows a screenshot of the optional SIP settings.

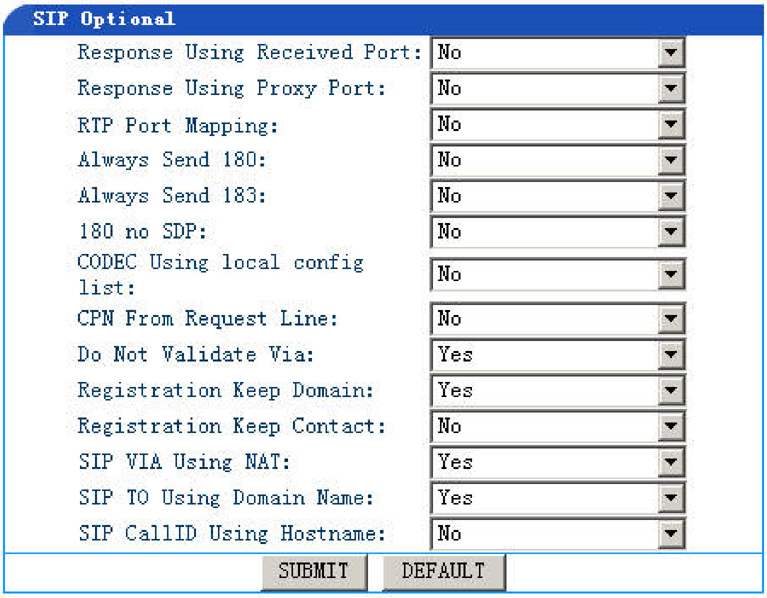


Figure 15. Optional SIP Settings

Response Using Received Port

This parameter is used to set whether the GTT uses the receive port as the reply port. If **Response Using Received Port** is set to **Enable** this feature is enabled. The default setting is **Disable** and the GTT will use the default port 5060.

Response Using Proxy Port

This parameter is used to set whether the GTT uses the proxy port as the reply port. If **Use Proxy Port as Reply Port** is set to **Enable** this feature is enabled. The default setting is **Disable** and the GTT will use the default port 5060.

RTP Port Mapping

This parameter is used to set whether to invoke RTP port mapping function. **Enable**: invoke RTP port mapping function, and adopt local SIP port and RTP port; **Disable**: close RTP port mapping function, and adopt port which is requested by STUN.

Always Send 180

If this parameter is set to **Enable**, the GTT will map all alerting messages (ALERTING with and without in-band indicator) to 180. An example of when this parameter would be **Enabled** is when an IAD does not support a 183 message.

If **Always Send 180** is **Disabled**, no mapping will occur and the 18x message will be sent.

Always Send 183

If this parameter is set to **Enable**, the GTT will map all alerting messages (ALERTING with and without in-band indicator) to 183. An example of when this parameter would be **Enabled** is when an ISDN switch is configured to handle the GTT as an end-user, and always provide ring-back or announcement but still uses the regular ALERTING message. In this case, a 180 message will cause the other side to hear ring-back, but the ISDN switch may announce “Destination Busy”.

If **Always Send 183** is **Disabled**, no mapping will occur and the 18x message will be sent.

180 with SDP

If **180 with SDP** is set to **Enable** then 180 messages are sent with SDP. With a **Disable** setting the GTT will send 180 messages without SDP.

CODEC Using Local Config List

If this parameter is set to **Enable**, the GTT will use its local CODEC preference order. By default, GTT will choose CODEC based on IAD’s preference order. Enable this flag in applications that require GTT to choose CODEC based on its local preference order.

CDPN From Request Line

If this parameter is set to **Enable**, the GTT will obtain (Called Party Number) from SIP <Request Line>. By default GTT gets CDPN from SIP <To> field. Enable this flag if GTT is required to use CDPN in SIP <Request Line> that was modified (add/remove prefix, etc) by soft switch for certain application.
Use config file to view or set these parameters:

Do Not Validate Via

This parameter defines if the via field in the SIP response is neglected (**Enable**) or not (**<u>Disable</u>**).

Registration Keep Domain

When this parameter is set to **Enable** the full domain name information is used to register. When set to **Disable** only the common part is used to register.

Registration Keep Contact

This parameter is used when the GTT is registering across a private network. If set to **Enable** the GTT will register with the original contact information. A **Disable** setting allows the contact information to change.

SIP VIA Using NAT

This parameter controls whether the SIP <VIA>will use the public network address information supplied by NAT or the private network address information. If this parameter is set to **Enable** then the SIP <VIA> will use the public network information supplied by NAT. A **Disable** setting will cause the SIP <VIA> to use the private network address information.

SIP TO Using Domain Name

This parameter controls whether the proxy information or domain name information (set in SIP Setting section) will be used by SIP <To>. If set to **Enable,** the GTT will use the domain name information. If set to **Disable,** the GTT will use the proxy information.

SIP CallID Using Hostname

This parameter controls whether the proxy information or hostname (set in SIP Setting section) will be used by SIP <Call-ID>. If set to **Enable,** and the GTT will use the hostname information. If set to **Disable,** the GTT will use the proxy information.

Max Forward

The **Maximum Forwarding Times of Signaling** parameter controls the maximum number of times a signaling packet will be forwarded. If the limit is exceeded, the signaling packet will be discarded. This parameter can be set from **1 - 70**. The default setting is **60**.

Times (A-B, D-K)

These parameters are read-only and are used for troubleshooting purposes.

Optional ISDN Settings

Figure 16 shows a screenshot of the optional SIP settings.

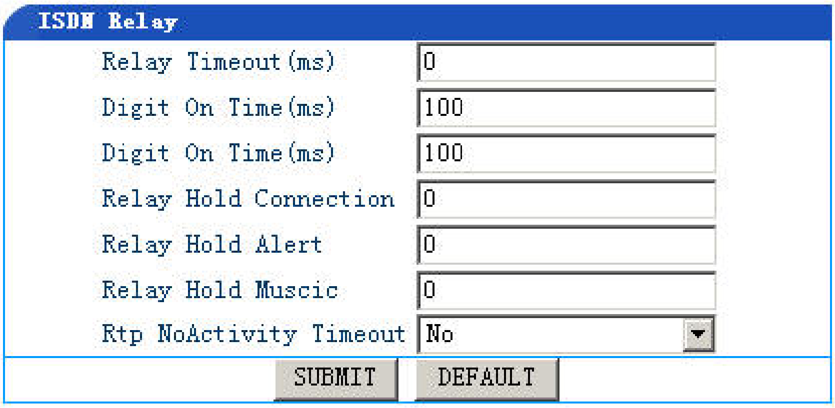


Figure 16. Optional ISDN Settings

Relay Timeout

The **Relay Timeout** parameter sets the delay in milliseconds from the time the ISDN relay connection is made to the time the CDPN is pulsed out using in-band DTMF.

Digit On Time

This parameter in millisecond is used to set pulsing digits on time.

Digit Off Time

This parameter in millisecond is used to set pulsing digits off time.

Relay Hold Connection

The **Relay Hold Connection** parameter defines the delay time in seconds from the time the ISDN relay connection is made to the time the SIP connection message is sent out.

The reason for delay is because the final connection is made by another platform using in-band DTMF and it may take time to make the final connection. **Relay Hold Connection** should be set according to the maximum connecting time of the relayed platform. For example, in the 17909 platform application in China, the maximum connection time for a call to Europe is approximately 30 seconds.

If media activity is detected before the time specified, the GTT will immediately send out the SIP connection message.

Relay Hold Alert

The **Relay Hold Alert** time defines the delay in seconds from when a receiving alert on the ISDN relay connection is received to when the SIP alert message is sent.

The **Relay Hold Alert** time should be less than that of **Relay Hold Connection**.

If media activity is detected before the time specified, the GTT will immediately send out the SIP connection message.

Relay Hold Music

The **Relay Hold Music** defines the .pcm file sequence number for the hold music. A setting of **0** indicates no hold music. If it is configured with a number other than **0**, the specified file will be played during the **Relay Hold Connection** time.

NoActivity Timeout

The **NoActivity Timeout** parameter defines what the GTT does if no media activity is detected after the **Relay Hold Connection** time has expired. If set to **Enable**, the GTT will disconnect the call (treated as a connection fail). If set to **<u>Disable</u>**, the GTT will cut through.

6

MONITORING

This chapter discusses the monitoring screens and features of the GTT Web-GUI. Monitoring information is available on the WebGUI by selecting **Logs** from the left navigation pane. The SGX-100 provides the following information screens and logs:

Resource

The **Resource Information** screen is shown in Figure 17. This screen shows the IP address and level of all WebGUI users, SIP register information, and relevant information for telephone and RTP.

Figure 17. Resource Information Screen

Login User Info

The **Login User Info** reports the address from which you are managing the GTT. The single digit to the right of the IP address indicates your current permission level:

- **1** = Read only
- **2** = Read/Write

SIP Registration Info

Messages under this title show GTT registration information. If GTT is not required to register, message “not enabled” will be shown. If registration is enabled, it will show <Contact> info of the registration message from GTT and the response from the registration server.

Call Context Info

Messages under this title are listed for all calls in GTT. Each message contains the fields in the following order:
1) internal call context index;

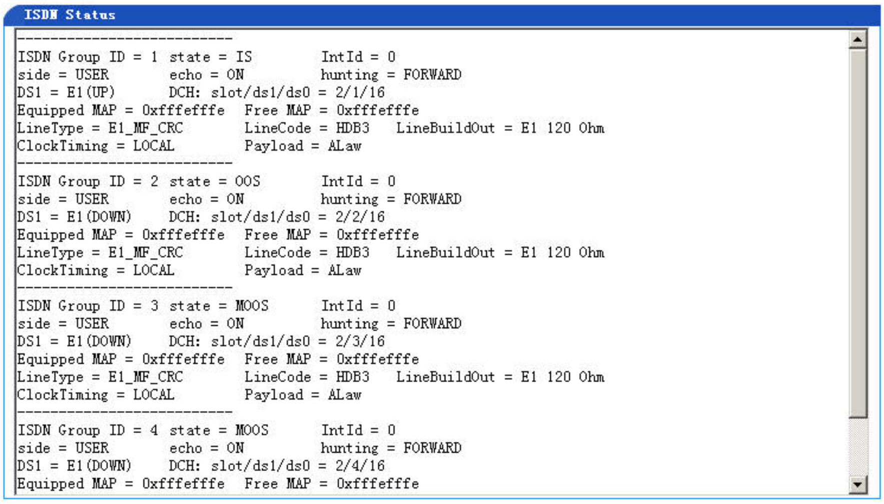
2) internal call ID for the call;
3) internal RTP context index for the call;
4) call originating side -> call terminating side;
5) remote SIP IAD's IP address and port;
6) local RTP port for the call;
7) selected CODEC for the call;
8) ISDN circuit selected for the call: slot number / span number / circuit number;
9) ISDN call state;
10) CPN -> CDPN for the call.

RTP Context Info

Messages under this title are listed for all active RTP channels. Each message contains two parts: the first part is the internal RTP context index created, and the second part is the internal call ID corresponding to this RTP channel.

ISDN Status

The **ISDN Status Information** screen is shown in Figure 18. The following sections provide additional details on the **ISDN Status Information** screen.



```
ISDN Status
--------------------------
ISDN Group ID = 1  state = IS        IntId = 0
side = USER        echo = ON         hunting = FORWARD
DS1 = E1(UP)       DCH: slot/ds1/ds0 = 2/1/16
Equipped MAP = 0xfffefffe  Free MAP = 0xfffefffe
LineType = E1_MF_CRC       LineCode = HDB3   LineBuildOut = E1 120 Ohm
ClockTiming = LOCAL        Payload = ALaw
--------------------------
ISDN Group ID = 2  state = OOS       IntId = 0
side = USER        echo = ON         hunting = FORWARD
DS1 = E1(DOWN)     DCH: slot/ds1/ds0 = 2/2/16
Equipped MAP = 0xfffefffe  Free MAP = 0xfffefffe
LineType = E1_MF_CRC       LineCode = HDB3   LineBuildOut = E1 120 Ohm
ClockTiming = LOCAL        Payload = ALaw
--------------------------
ISDN Group ID = 3  state = MOOS      IntId = 0
side = USER        echo = ON         hunting = FORWARD
DS1 = E1(DOWN)     DCH: slot/ds1/ds0 = 2/3/16
Equipped MAP = 0xfffefffe  Free MAP = 0xfffefffe
LineType = E1_MF_CRC       LineCode = HDB3   LineBuildOut = E1 120 Ohm
ClockTiming = LOCAL        Payload = ALaw
--------------------------
ISDN Group ID = 4  state = MOOS      IntId = 0
side = USER        echo = ON         hunting = FORWARD
DS1 = E1(DOWN)     DCH: slot/ds1/ds0 = 2/4/16
Equipped MAP = 0xfffefffe  Free MAP = 0xfffefffe
```

Figure 18 . ISDN Status Information Screen

ISDN Group ID

This field reflects the current setting of the ISDN **Group** setting.
For more information see "Group" on page 42.

State

There are three possible values for **state**:

Table 4: ISDN States

State	Definition	Description
IS	In Service	In service / currently used
OOS	Out of Service	Not currently used
MOOS	Manual Out of Service	Continuously not invoked and backup signaling channel is not invoked (can't be used)

Int ID

This field reflects the current setting of the **Interface ID** parameter.
For more information see "Interface ID" on page 43.

Side

This field reflects the current setting of the **Switch Side** parameter.
For more information see "Switch Side" on page 43.

Echo

This field reflects the current setting of the **Echo Cancellation** parameter.
For more information see "Echo Cancellation" on page 43.

Hunting

This field reflects the current setting of the **Hunting** parameter.
For more information see "Hunting" on page 43.

DS1

This field reflects the current setting of the **DS1 Type** parameter.

For more information see "DS1 Type" on page 45.

Slot / ds1 / ds0

This parameter identifies the hardware identification of the T1/E1 port in the system. **Slot** is the location of the T1/E1 Interface card and is always **1**. **ds1** is the T1/E1 port number on the T1/E1 Interface card. **ds0** denotes the channel which is configured as the **D Channel**. For more information see "D Channel" on page 43

Equipped MAP

This field shows the available states of the 30 time-slots in the E1 card (except the 0 and 16th slot, which are reserved).

See ISDN Circuit configurable for more information. 0xfffefe is expressed in the binary system. If a certain number is 1, it indicates the corresponding time-slot is available.

LineType

This field reflects the current setting of the **Line Type** parameter. For more information see "Line Type" on page 45.

LineCode

This field reflects the current setting of the **Line Code** parameter. For more information see "Line Code" on page 45.

LineBuildOut

This field reflects the current setting of the **Length** parameter. For more information see "Length" on page 45.

Clock Timing

This field reflects the current setting of the **Clock Timing** parameter. For more information see "Clock Timing" on page 45.

Payload

This field reflects the current setting of the **DS0 Type** parameter. For more information see "DS0 Type" on page 45.

ISDN Call

The **ISDN Call Information** screen is shown in Figure 19. The following sections provide additional details on the **ISDN Call Information** screen.

Figure 19. ISDN Call Information Screen

Group ID

This field reflects the current setting of the ISDN **Group** setting. For more information see "Group" on page 42.

TS (Time-Slot)

This field denotes the internal TDM time-slot being used by the ISDN time-slot. The internal TDM time-slot is 8M, and is distributed to four E1 interfaces (2M each). Each E1 has 32 time-slots, as a result the internal TDM time-slot has a total of 128 (4 x 32) timeslots.

DSP

The GTT Control Module has up to eighteen DSP chips which are identified from DSP0 to DSP18. The **DSP** field indicates which DSP chip is being used by the indicated time-slot of the T1/E1.

Remote

This parameter reports the current remote IP address which could be the address of the RTP Media Server or access gateway, which comes before the RTP port number.

Local

The **local** parameter shows the locally used RTP port number.

Codec

The **codec** value reports the currently used codec. For more information see "Default Codec" on page 28.

Ref (Call Reference)

This parameter is the call reference which is used to recognize the information involved with call or equipment book/cancel requests on local user and network interfaces. The call reference doesn't include the end-to-end function of ISDN. In other words, this parameter is valid only in current segment (user and network side) and in the next segment the same call reference value can also be used.

State

This is the call state which denotes the current state of a call. See the following table for detailed definitions for various call states.

Table 5: Call States

Serial Number	State	User Side Call Status	Network Side Call Status
U0 / N0	Null	No Call Present	No Call Present

Table 5: Call States

Serial Number	State	User Side Call Status	Network Side Call Status
U1 / N1	Call Initiation	This state exists when a number is being called out, at the same time the user send call setup request to the network	This state exists when a number is being called out, at the same time the network has received the call set-up request, but no response has been received.
U2 / N2	Overlap Sending	This state exists when a number is being called out; at the same time the user has received the confirmation information of call set-up request, which denotes that the user is allowed to send call information to the network with overlap mode.	This state exists when a number is being called out, at the same time the network has confirmed the call setup request, and prepare to receive call information with overlap mode.

Table 5: Call States

Serial Number	State	User Side Call Status	Network Side Call Status
U3 / N3	Out Pro-cess(OUT -PROC)	This state exists when a number is being called out, at the same time the user has received the confirmation information of call set-up request, which denotes that the network has received all the call information which would be used to set up a call.	This state exists when a number is being called out, at the same time the network has sent forth the confirmation information of call set-up request, which denotes that the network has received all the call information which would be used to set up a call.
U4 / N4	Call Deliver-ed	This state exists when a number is being called out, at the same time the network has received the remote indication for beginning to send forth reminder signal.	This state exists when a number is being called out, at the same time the network has instructed the remote user beginning to send forth reminder signal.
U6 / N6	Call Present	This state exists when a number is being called in, at the same time the network has received the call set-up request, but no response to it.	This state exists when a number is being called in, at the same time the network has sent forth the call setup request, but no satisfied response received.

Table 5: Call States

Serial Number	State	User Side Call Status	Network Side Call Status
U7 / N7	Call Received	This state exists when a number is being called in, at the same time the user has send forth reminder indication, but no response is received.	This state exists when a number is being called in, at the same time the user has received the reminding indication, but no response is received.
U8 / N8	Connect Request	This state exists when a number is being called in, at the same time the user has answered the call and is waiting for the response of the call.	This state exists when a number is being called in, at the same time the network has received response, but no call has been sent forth.
U9 / N9	Incoming Call Process	This state exists when a number is being called in, at the same time the user has sent forth the confirmation information, which denotes that the user has received all the call information which would be used to set up a call.	This state exists when a number is being called in, at the same time the network has received the confirmation information, which denotes that the user has received all the call information which would be used to set up a call.

Table 5: Call States

Serial Number	State	User Side Call Status	Network Side Call Status
U10 /N10	Active	When a number is being called in, this state denotes that the user has received the confirmation from the network user that has received the call. When a number is being called in, this state denotes that the user has received the indication from the remote user's response to the call.	When a number is being called in, this state denotes that the network has given the call to the called-user. When a number is being called out, this state denotes that the network has sent forth the user an indication from the remote user's response to the call
U11/ N11	Disconnect Request	This state denotes that the user has sent forth a request to the network for disconnecting the end-to-end connection, and is waiting for response.	This state denotes that the network has received the request from the user for disconnecting the end-toned connection.
U12 / N12	Disconnect Indication	This state denotes that the user has received the disconnect request, because the network has disconnected the end-toned connection.	This state denotes that the network has disconnected the end-to-end connection, and has sent forth a request for disconnecting the user to network connection.

Table 5: Call States

Serial Number	State	User Side Call Status	Network Side Call Status
U15 / N15	Pause Request	This state denotes that the user has requested the network to suspend the call and is waiting for response.	This state denotes that the network has received the pause request, but no response is received.
U17 / N17	Resume Request	This state denotes that the user has requested the network to resume the suspended call and is waiting for response.	This state denotes that the network has received the request to resume the suspended call, but no response is received.
U19 /N19	Release Request	This state denotes that the user has requested the network to release the call and is waiting for response.	This state denotes that the user has requested the user to release the call and is waiting for response.
N22	Call Terminate		This state exists when the call is point-to-multipoint connection, before any user can get the call, the call is disconnecting.

Table 5: Call States

Serial Number	State	User Side Call Status	Network Side Call Status
U25 / N25	Overlap Reception	This state exists when a number is being called in, at the same time the user has confirmed the call setup request, and prepare to receive the call information with overlap mode(if any).	This state exists when a number is being called in, at the same time the user has received the confirmation information of call set-up request, which denotes that the network is allowed to send forth call information to the user with overlap mode.

CPN

This is the calling party number.

CDPN

This is the called party number.

Timestamp

The **timestamp** value reports the setup time (which is always **0**) and the connection time. As shown on the screen, the format is setup time / connection time. **Timestamp** values are measured in seconds.

CallID

The **CallID** is a number that is used to identify a call when SIP is switching information. The length and value of the **CallID** are randomly generated.

Message Log

The **Message Log** is shown in Figure 20.
This page shows most recent SIP and ISDN messages received or sent out in GTT. The size of this page is approximately 100-150 lines depending on the content in each message. This page is useful to debug problems related to SIP or ISDN signaling.
If you need longer history of the messages, the complete message log can be viewed at </var/tmp/message.log>. The size of the log file is 500KB. You may see three message log files in </var/tmp>: “message.log”, “message.log.1” and “message.log.2” with total 1500KB size limitation. “message.log” always contains most recent messages. When “message.log” is full, “message.log.1” will be copied to “message.log.2”, “message.log” will be copied to “message.log.1”, and “message.log” will be emptied to store new messages.

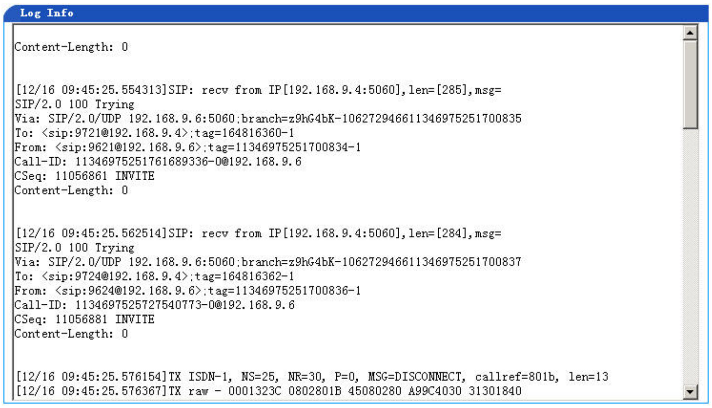
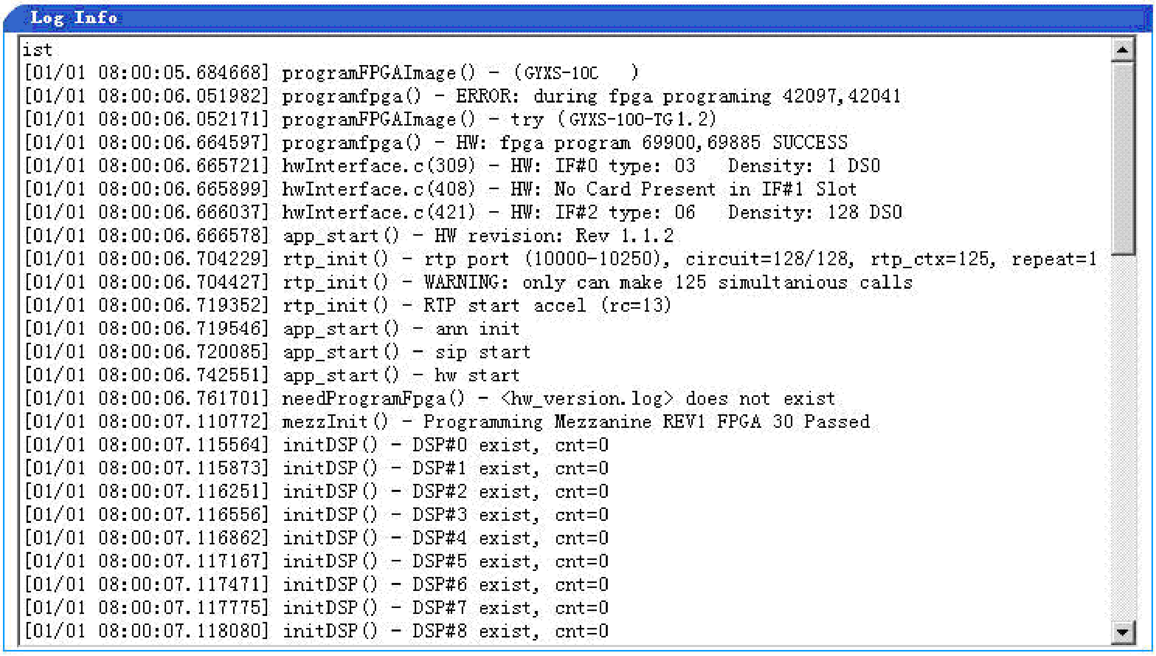


Figure 20. Call Information Log

Boot Log

The **Boot Log** is shown in Figure 21. This log is used by GED to aid in troubleshooting the GTT hardware and software.



```
Log Info
ist
[01/01 08:00:05.684668] programFPGAImage() - (GYXS-10C    )
[01/01 08:00:06.051982] programfpga() - ERROR: during fpga programing 42097,42041
[01/01 08:00:06.052171] programFPGAImage() - try (GYXS-100-TG1.2)
[01/01 08:00:06.664597] programfpga() - HW: fpga program 69900,69885 SUCCESS
[01/01 08:00:06.665721] hwInterface.c(309) - HW: IF#0 type: 03   Density: 1 DS0
[01/01 08:00:06.665899] hwInterface.c(408) - HW: No Card Present in IF#1 Slot
[01/01 08:00:06.666037] hwInterface.c(421) - HW: IF#2 type: 06   Density: 128 DS0
[01/01 08:00:06.666578] app_start() - HW revision: Rev 1.1.2
[01/01 08:00:06.704229] rtp_init() - rtp port (10000-10250), circuit=128/128, rtp_ctx=125, repeat=1
[01/01 08:00:06.704427] rtp_init() - WARNING: only can make 125 simultanious calls
[01/01 08:00:06.719352] rtp_init() - RTP start accel (rc=13)
[01/01 08:00:06.719546] app_start() - ann init
[01/01 08:00:06.720085] app_start() - sip start
[01/01 08:00:06.742551] app_start() - hw start
[01/01 08:00:06.761701] needProgramFpga() - <hw_version.log> does not exist
[01/01 08:00:07.110772] mezzInit() - Programming Mezzanine REV1 FPGA 30 Passed
[01/01 08:00:07.115564] initDSP() - DSP#0 exist, cnt=0
[01/01 08:00:07.115873] initDSP() - DSP#1 exist, cnt=0
[01/01 08:00:07.116251] initDSP() - DSP#2 exist, cnt=0
[01/01 08:00:07.116556] initDSP() - DSP#3 exist, cnt=0
[01/01 08:00:07.116862] initDSP() - DSP#4 exist, cnt=0
[01/01 08:00:07.117167] initDSP() - DSP#5 exist, cnt=0
[01/01 08:00:07.117471] initDSP() - DSP#6 exist, cnt=0
[01/01 08:00:07.117775] initDSP() - DSP#7 exist, cnt=0
[01/01 08:00:07.118080] initDSP() - DSP#8 exist, cnt=0
```

Figure 21. Startup Information Log

7

SYSTEM TOOLS

This chapter discusses the system tools available on the GTT. Monitoring information is available on the WebGUI by selecting **Tools** from the left navigation pane. The GTT provides the following system tools:

Factory Settings

As shown in Figure 22, the **Factory Settings** selection will ask for confirmation to restore the system to its factory defaults.

caution

Restoring factory settings to a remote GTT will cause the IP Address used to access the system to revert to its default. This will likely cause remote access to be lost.

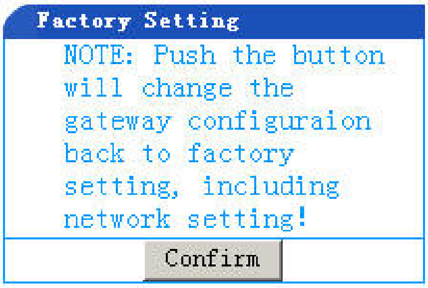


Figure 22. Factory Settings

Upgrade

The **Upgrade** screen is shown in Figure 23. After entering the required information and selecting **Upgrade**, the GTT will download the software. After the download is complete, the GTT will prompt you to restart the gateway to implement the new software.

After selecting Upgrade do not perform any operations on the GTT. Allow the software update process to proceed in the background.

The following sections provide additional details on the **Upgrade** screen.

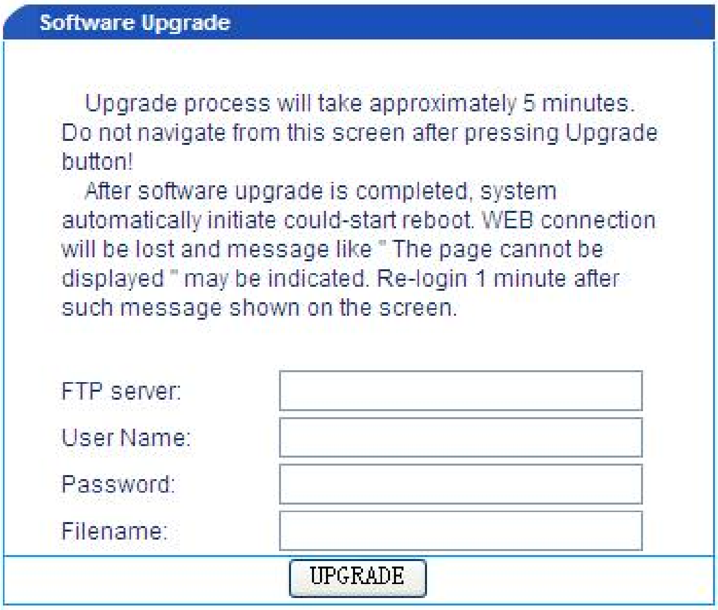


Figure 23. Upgrade Screen

FTP Server

The **FTP Server** input box identifies the IP address (**aaa.bbb.ccc.ddd**) or domain name of the ftp server.
Example
http://220.248.100.68 /Autoprovision/
ftp://220.248.100.68 /Autoprovision/
ftp://username:password@220.248.100.68 /Autoprovision/

User Name

Input the user name of log in ftp server.

Password

Input the password of log in ftp server.

Filename

Input the software version to be downloaded. If left blank the software will be updated to the latest version.

Change Password

The **Change Password** screen is shown in Figure 24. Only the **administrator** account has the permissions to change passwords.

The current **operator** password is displayed in plain text and can be changed independently of the **administrator** password.

Figure 24. Change Password Screen

Reboot

The **Reboot** screen is shown in Figure 25.

Figure 25. Gateway Restart Screen

8

TROUBLESHOOTING

This chapter contains information to help you solve problems that may occur while installing and using the GTT gateway.

GETTING HELP

If, after installing and configuring your GED equipment, you cannot establish Technologies to or from the unit, carefully review the information in this book and in the other books prior to calling Customer Support (CS).

Before going any further, ensure that you have checked the following:

- Console Port configuration. Verify that the baud rate of your terminal is set to 9600 bps.
- Power reset. When cycling power, be sure to leave the power off for a minimum of 30 seconds before reapplying power to the unit.
- Command review. Review the GTT gateway commands in the Configuration chapter.
- Software version. To ensure that you have the latest enhancements and product features, GED ships every unit with the latest software version. Therefore, when you are installing or reinstalling units into your system, verify that each unit is equipped with identical software versions.

Checklist

If, after carefully reviewing the information in this manual, your problem persists, contact your product representative or a service representative at GED’s Customer Support (CS). So we can serve you better, make a list of the following items before calling:

- A detailed description of your problem.
- A complete listing of your system components and configuration, including the serial number of your unit and the software version number it is running.
- A narrative of the actions you performed prior to the problem.
- A list of all system messages posted by your unit.

Contacts

GED (Shanghai) Co., Ltd.
Address：12F, Building 6, 1369 Dongfang Rd,
Office Phone：86-21-68640585
Fax: 86-21-50892248

A

Parts and Accessories

GTT gateway parts and accessories can be purchased from your GED authorized service provider. The tables in this appendix list the part numbers for field-replaceable items. For the latest information (including prices) on the parts described in this appendix, consult your GED representative.

GTT Cables

Cable Description
T1/E1 120 Ohm Balance Cable RJ48 (P) to RJ48 (P)
T1/E1 75 ohm BNC Cable
75 / 120 ohm Impedance Converter

Spares

Description
Chassis (includes 2 P/S, T1/E1 module)
P/S module
T1/E1 module (4 ports)
CM module (1-span capacity)
CM module (2-span capacity)
CM module (4-span capacity)