

## Public Switched Telephone Network (PSTN) Simulators

(ANALOG FXS AND T1/E1 PORTS)



This user manual is applicable to the following products:

**SIM-2FXS, SIM-2FXS-A, SIM-4FXS, SIM-4FXS-A, CH-4FXS-MOD-2PRI, CH-4FXS-MOD-2PRI-A, CH-8FXS-MOD-2PRI, CH-8FXS-MOD-2PRI-A, CH-32FXS-MOD-2PRI, CH-32FXS-MOD-2PRI-A, SIM-16FXS, SIM-16FXS-A**

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## **1. Chapter 1: Important Information**

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### **1.1 General Disclaimer**

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### **1.4 Patent Information**

The accompanying product is protected by one or more U.S. and foreign patents and patents pending held by Virtual Console, LLC

### **1.5 Warranty**

Virtual Console, LLC warrants its products to be free of defects in materials and factory workmanship for a period of twelve (12) months from date of purchase. This warranty does not apply to damage to products resulting from accident, misuse, service or modification by anyone other than a Virtual Console, LLC authorized service facility/dealer.

The warranty is limited to the original purchaser and is not transferable. Any liability of Virtual Console or its suppliers with respect to the product or the performance thereof under any warranty, negligence, strict liability or other theory will be limited exclusively to product repair or replacement as provided above.

Except for the foregoing, the product is provided "as is" without warranty of any kind including without limitation, any warranty of merchantability or fitness for a particular purpose.

The entire risk of the quality and performance of the software programs contained in the system is with you.

## 1.6 Limitation of Remedies and Damages

Virtual Console, LLC, its agents, employees, suppliers, dealers and other authorized representatives shall not be responsible or liable with respect to the product or any other subject matter related thereto under any contract, negligence, strict liability or other theory for any indirect, incidental, or consequential damages, including, but not limited to loss of information, business, or profits. The law of certain states or nations does not permit limitation or exclusion of implied warranties and consequential damages, so the above limitations, disclaimers, or exclusion may not apply to you. This warranty gives you special legal rights. You may also have other rights that vary by state and nation.

## 1.7 Important Safeguards

Read and understand the following instructions before using the system:

Close supervision is necessary when the system is used by or near children.  
Do not leave unattended while in use.

Always disconnect the system from power before cleaning and servicing and when not in use.

Do not spray liquids directly onto the system when cleaning.  
Always apply the liquid first to a static free cloth.

Do not place this product onto unstable desk, cart or table.  
The product may fall causing serious damage to the product.

Do not immerse the system in any liquid or place any liquids on it.  
Do not disassemble this system (except as instructed in the manufacturer's instructions).  
To reduce the risk of shock and to maintain the warranty on the system, a qualified technician must perform service or repair work.

Connect this appliance to a grounded outlet.  
Connect the system only to surge protected power outlets.  
Keep ventilation openings free of any obstructions.

Unplug this product from the wall outlet and refer servicing to qualified service personnel under the following conditions:

1. When the power supply adapter or plug is damaged or frayed.
2. If liquid has been spilled into the product.
3. If the product has been exposed to rain or water.
4. If the product has been dropped or the enclosure has been damaged.
5. If the product exhibits a distinct change in performance.

**SAVE THESE INSTRUCTIONS**

## 1.8 Regulatory Compliance

This equipment has been tested and found to comply with the limits for a Class A digital device, pursuant to Part 15 of the FCC Rules. These limits are designed to provide reasonable protection against harmful interference when the equipment is operated in a commercial environment. This equipment generates, uses and can radiate radio frequency energy and, if not installed and used in accordance with the instruction manual, may cause harmful interference in which case the user will be required to correct the interference at his own expense.

## 2. Chapter 2: Introduction Overview of PSTN Simulator

### 2.1 Functional Overview

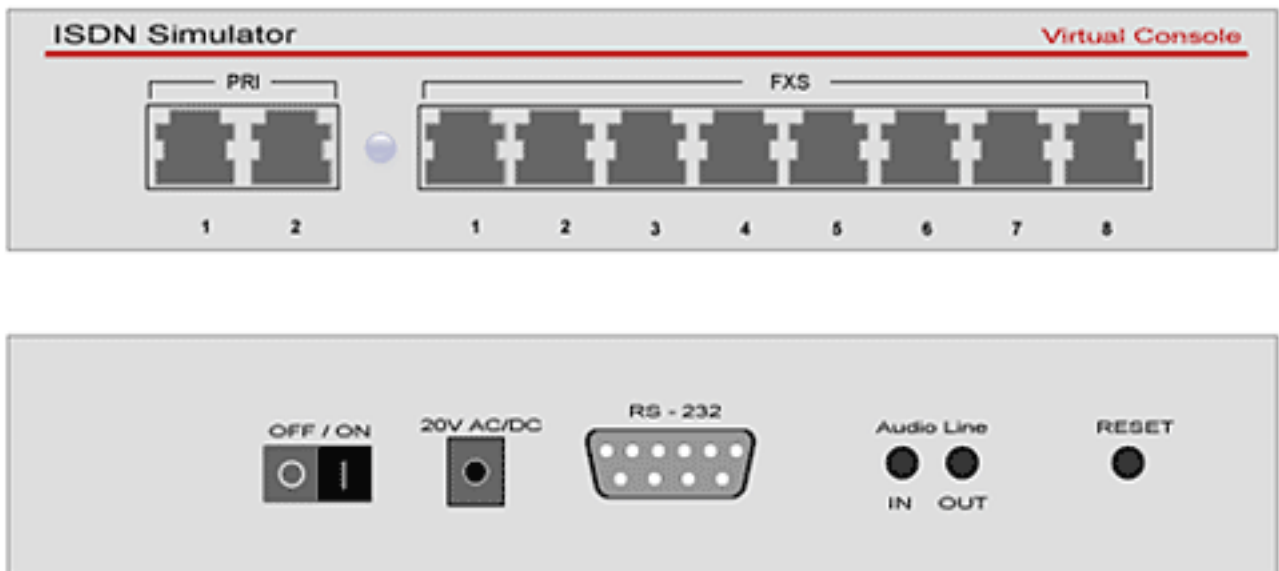
PSTN (Public Switched Telephone Network) Simulator provides an analog and digital\* call switching services eliminating the expense of the real phone lines for testing or demonstrating devices that require analog (FXS) and digital (T1/E1/PRI) interfaces. PSTN Simulator gives access to the rich set of configurable parameters from electrical to line impairments making it a valuable tool for telecom designers and developers.

PSTN simulator comes with fixed number of analog (FXS) loops start ports allowing to connects up to 16 analog telephone sets for full non-blocking switching among any of the ports.

Simulators equipped with T1/E1/PRI module can be used for making calls between analog and T1/E1/PRI ports. Latest versions or simulators can support PRI and CAS signaling on T1/E1 ports.

\* - If equipped with T1/E1/PRI module

### 2.2 Front and Back View





## 2.3 Overview of FXS ports

FXS (Foreign Exchange Station) ports are commonly used to connect analog telephone sets to a CO (Central Office). They can be found as a regular RJ11 wall jacks in every household or offices equipped with analog telephone lines.

RJ11 jacks usually have 6 pins, but only pins 3 and 4 are being used.

PSTN simulator comes with 8-pin RJ45 jacks where only pins 4 and 5 are being used.

PSTN Simulators can accept standard RJ11 connector without any additional rewiring.

Each FXS port on PSTN simulator has two integrated Green LED: Left and Right.

LEDs on FXS ports provide visual indication for call progress states, indication of Connect, Disconnect, on hook and off hook modes.

LED	Legend
Right LED - OFF	Telephone set is ON HOOK or not Connected
LEFT LED - OFF	No active or in progress outgoing calls
Right LED - Solid Green	Telephone set if OFF HOOK
Right LED - Blinking GREEN	Incoming call is in progress
Left LED - Solid GREEN	Call is established
Left LED - Blinking GREEN	Outgoing call is in progress

## 2.4 Audio IN/OUT ports

4 and 8 port PSTN Simulator comes with two standard audio ports: Input and Output.

Audio Input (Audio In) port can be used as a source of Music On Hold (MOH), Audio Output (Audio Out) port can be used for monitoring or recording conversations originated or terminated at any FXS or PRI ports.

Located on the back of the enclosure audio ports are equipped with standard 3.5mm mono jacks with 600-Ohm impedance.

## 2.5 DB9 port

DB9 port provides communication between PSTN simulator and PSTN Manager application which runs on Windows platform. Standard straight-through Serial cable should be used to connect simulator to a PC. Latest version if simulators are being equipped with USB ports.

## 2.6 Power Adapter

PSTN simulator is shipped with a desktop or wall type power adapter with a 2.5mm output jack.

PSTN simulator can accept power source with a range of 12V-20V AC or DC with a current rating of 1.5A.

Switching or Linear adapters can be used. For applications that require low noise levels linear adapters are recommended.

Please be aware that power adapter can get warm during normal operation and require proper ventilation to operate. Do not block airflow to and from power adapter.

Keep it turned off while not in use.

## 2.7 T1/E1/PRI Ports

Some models of PSTN simulator come with two T1/E1/PRI ports.

Following simulator models have two PRI port:

CH-8FXS-MOD-2PRI, CH-8FXS-MOD-2PRI-A  
CH-4FXS-MOD-2PRI, CH-4FXS-MOD-2PRI-A

Following simulator models do not have PRI ports:

SIM-2FXS, SIM-2FXS-A  
SIM-4FXS, SIM-4FXS-A  
SIM-16FXS, SIM-16FXS-A

## 2.8 ISDN Manager

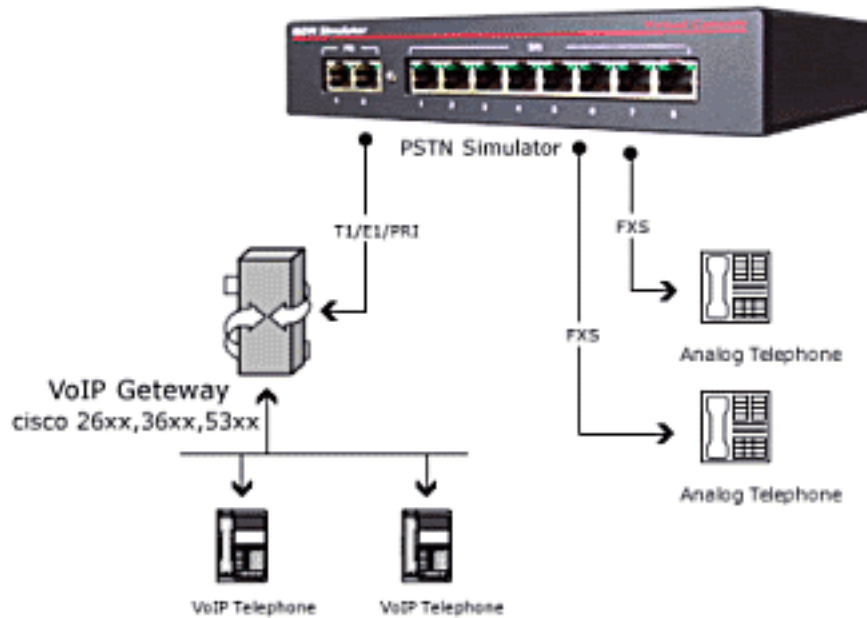
Standard PSTN simulator kit includes a floppy or CD with PSTN Manager application. PSTN Manager is the only way of configuring PSTN simulator.

PSTN Manager is a Windows application that can be installed on the following OS:  
Windows 2000 and XP

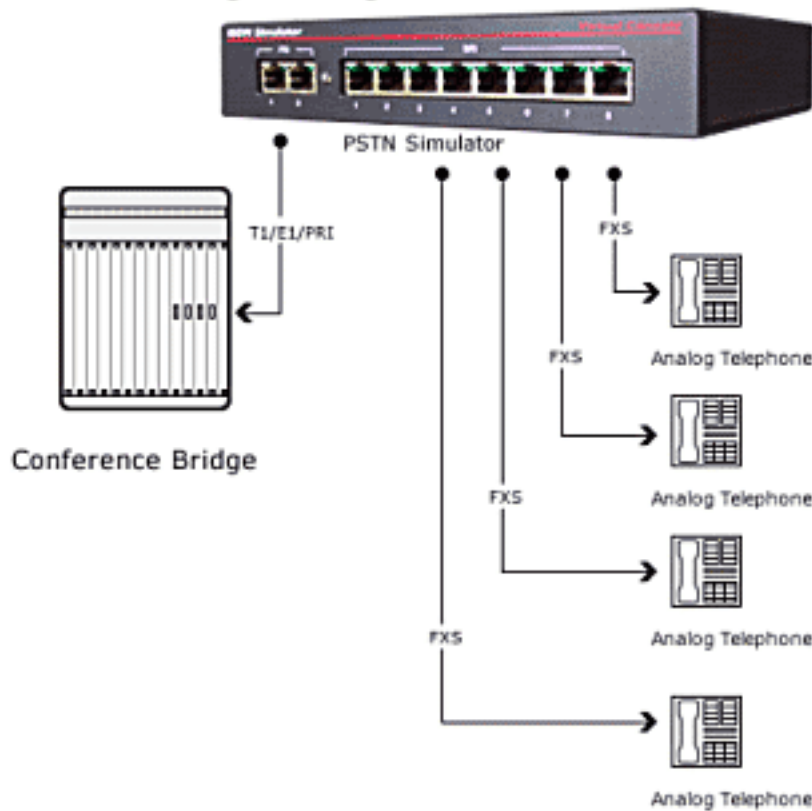
## 2.9 Typical applications for PSTN simulator.

Following diagrams demonstrate typical applications for PSTN simulator

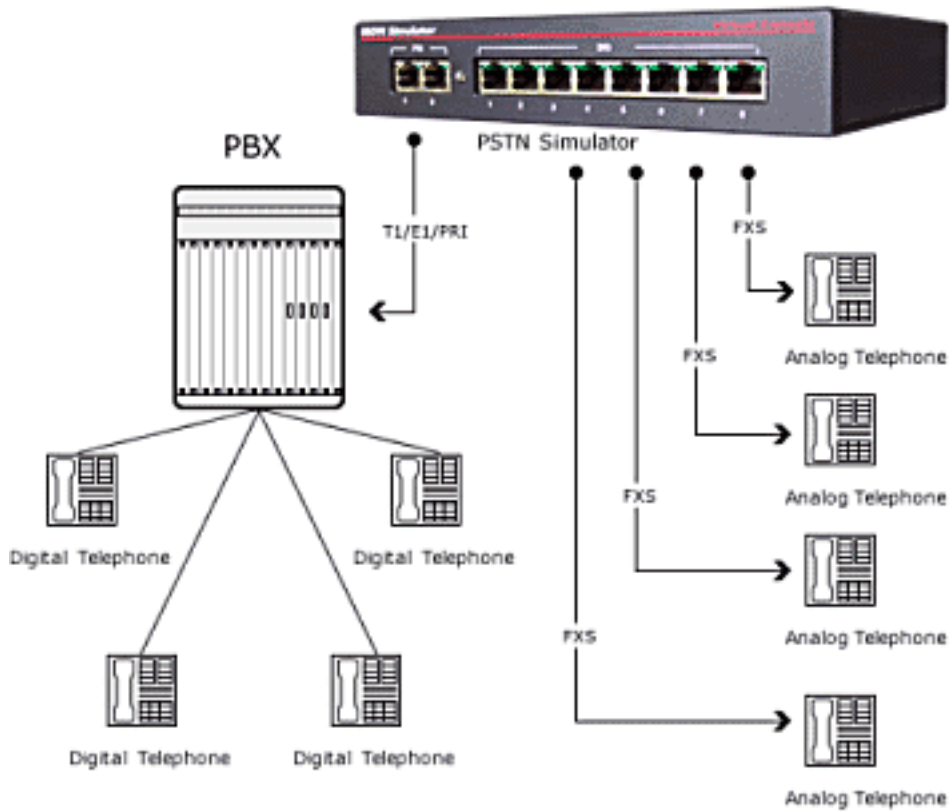
### 1. VoIP Gateway Testing



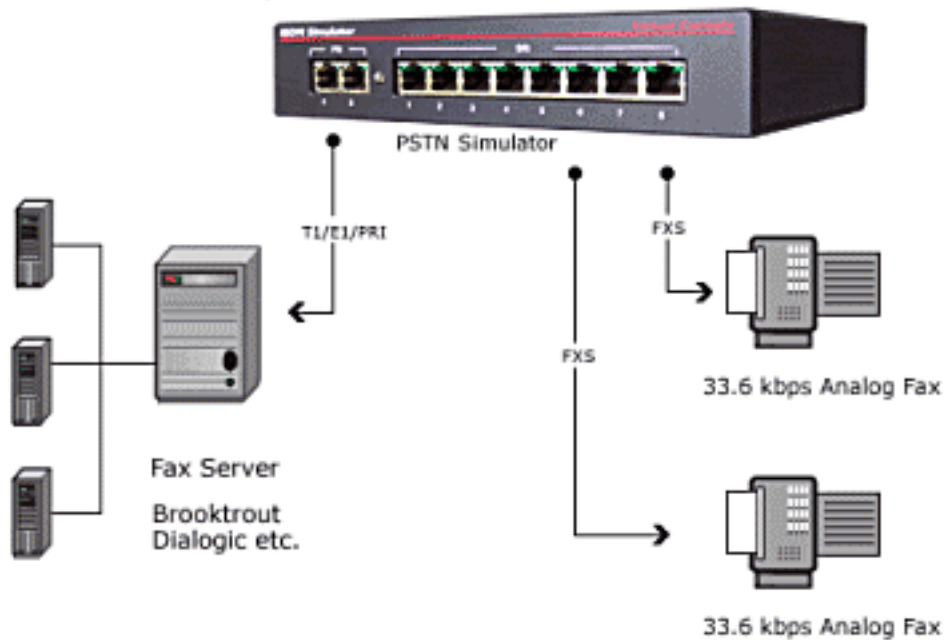
### 2. Conference Bridge Testing



### 3. PBX Testing



### 4. Fax Server Development



## **3. Chapter 3: Getting started with PSTN Simulator**

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### **3.1 Connecting PSTN simulator to a PC**

After unpacking your PSTN simulator, connect DB9 port on a simulator to a COM port on your PC with a supplied RS-232 cable. RS232 cable is included with every simulator sold.  
RS-232 cable is a standard straight-through serial cable with 9 pins Male/Female connectors.

### **3.2 Powering up PSTN simulator**

Make sure that Power switch is in OFF position.  
Connect output power jack from power adapter to a DC jack on a simulator labeled "Power"  
Turn power switch into an ON position

### **3.3 Installing PSTN Manager software**

Install PSTN Manager from the floppy or CD which came with a simulator kit.  
PSTN Manager requires 2Mb of free space on your hard drive and can be installed on the following OS:  
Windows 2000 and XP.

### **3.4 Connecting analog telephones to simulator**

To connect analog telephones to FXS ports you can use regular RJ11 cables which are usually supplied with a telephones. Though FXS ports on simulator are equipped with RJ45 jacks they can accept RJ45 connectors as well.  
You can test the connection by lifting a handset and verifying a presence of the dial tone.

### **3.5 Connecting T1/E1/PRI devices to simulator**

To connect devices with T1/E1/PRI interfaces to PSTN simulator you need to use straight-through CAT5 or CAT3 cable with RJ-45 connectors. RJ45 Jacks on simulator are wired for T1/E1 Network side.

**Note:** If your CPE device is also wired as network side you need to use T1/E1 crossover cable.

### **3.6 Monitoring/recording calls with Audio out port**

You can configure PSTN simulator to duplicate the audio stream associated with any of the FXS ports and route it to "Audio Out" port. By connecting the "Audio Out" port to a speakers or audio amplifier you can broadcast, listen or record Rx or Tx audio streams associated with a particular FXS port.

### **3.7 Configuring Music on Hold with Audio In port**

You can configure PSTN simulator to play a "Music on Hold" after calling or called party is placed on Hold. Connect a source of music (CD player or FM radio) to an "Audio In" port and verify it by calling a special test phone number "09". You should hear music in your handset.

**Note:** "Audio in" and "Audio out" ports available only on following models

CH-8FXS-MOD-2PRI, CH-8FXS-MOD-2PRI-A  
SIM-16FXS, SIM-16FXS-A  
CH-4FXS-MOD-2PRI, CH-4FXS-MOD-2PRI-A

## **4. Chapter 4: Configuring simulator with PSTN Manager**

Using PSTN Manager is the only way of configuring the simulator.

Any changes that you make in PSTN Manager do not take an effect until you save the configuration to PSTN simulator. Some of the changes can be applied "on the fly" and some requires a reboot of the simulator. PSTN Manger will determine if the Reboot is required and automatically reboot the simulator if needed.

Please note that all established calls will be lost if simulator is rebooted.

You can also save your current configuration to a file on the disk by accessing a

"Main Menu -> File-> Save Config" interface.

PSTN Manager is a set of configuration Tabs for Global (applicable to all ports) and Local (port specific) parameters. The actual set of parameters is based on firmware version and firmware feature set installed and activated on a particular PSTN simulator.

### **4.1 Replicating configuration among FXS ports**

You can easily replicate changes made on a single FXS port To: or From: particular FXS port by using a "Copy configuration" button on each FXS port configuration tab.

You can copy configuration from the current port to all FXS ports or a particular FXS port number or you can copy configuration from any FXS port number to a current FXS port.

Please note that **Phone numbers and Caller ID** information will not be copied as a part of this process.

### **4.2 Configuring Global Parameters**

#### **4.2.1 Configuring call progress tones**

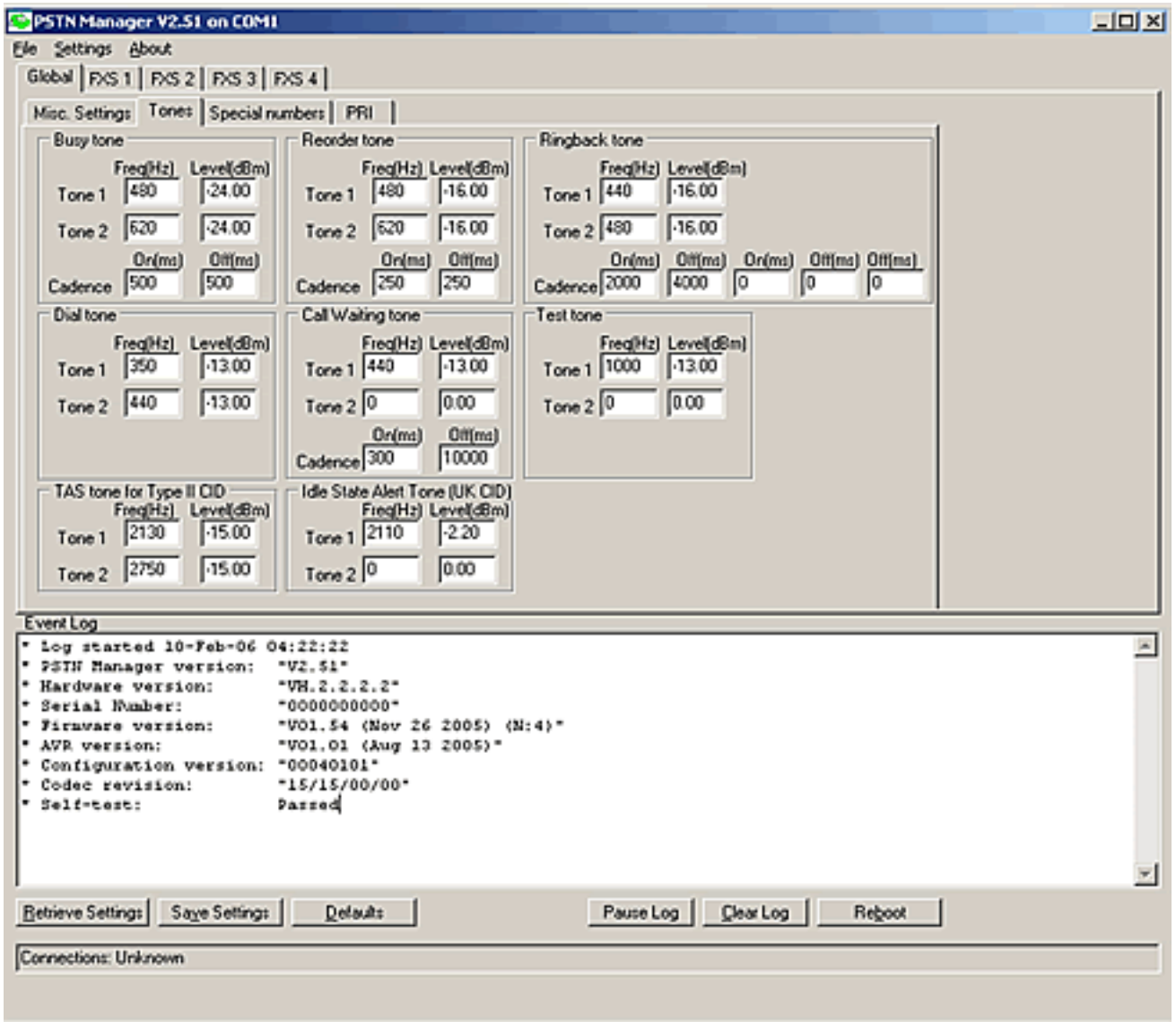
Call progress tones can be configured as a single or dual tone bands with a specific frequency, cadence and signal power. Different countries has deployed different standards for call progress tones. Refer to an ITU E.180 standard for further details on country specific call progress tones.

Three primary standards are:

**Single tone:** Tone consist of a single tone at a given frequency

**Dual tone:** **(Tone 1 + Tone 2):** Tone is the result of a sum of two tones.  
Each tone has its own frequency.

**Dual tone:** **(Tone 1 x Tone 2):** Tone is the result of Tone1 modulated by Tone 2.  
Each tone has its own frequency



#### 4.2.1.1 Dial tone

Dial tone is a tone advising that the exchange is ready to receive a call information and inviting the user to start sending call information. In North America, the most common dial tone consist of the sum of two tones: Tone 1 = 350 Hz and Tone 2 = 440Hz at -13 dBm.

PSTN Manager allows following values for the dial tone:

Tone 1:

Frequency: from 0 Hz to 3999 Hz with 1 Hz increments.  
Power: from -100.00 dBm to 3.00 dBm with 0.01 dBm increments

Tone 2:

Frequency: from 0 Hz to 3999 Hz with 1 Hz increments.  
Power: from -100.00 dBm to 3.00 dBm with 0.01 dBm increments

To hear the configured Dial tone, call special test number 01.

#### 4.2.1.2 Busy tone

Busy tone is tone advising the caller that the called party is engaged in another call.

In North America, the most common busy tone consists of the sum of two tones: Tone 1 = 480 Hz and Tone 2 = 620 Hz at -24 dBm with cadence of 500ms ON, 500ms OFF

PSTN Manager allows following values for the busy tone:

Tone 1:	Frequency:	from 0 Hz to 3999 Hz with 1 Hz increments.
	Power:	from -100.00 dBm to 3.00 dBm with 0.01 dBm increments
Tone 2:	Frequency:	from 0 Hz to 3999 Hz with 1 Hz increments.
	Power:	from -100.00 dBm to 3.00 dBm with 0.01 dBm increments
Cadence:		0- 9999ms ON, 0-9999ms OFF with 1ms increments

To hear the configured busy tone, call special test number 02.

#### 4.2.1.3 Ringback Tone

Ringback tone is a tone advising the caller that a connection has been made and that a calling signal is being applied to a telephone number or service point.

In North America, the most common ringback tone consists of the sum of two tones: Tone 1 = 440 Hz and Tone 2 = 480 Hz at -16 dBm with a cadence of 2000ms ON, 4000ms OFF.

PSTN Manager allows following values for the ringback tone:

Tone 1:	Frequency:	from 0 Hz to 3999 Hz with 1 Hz increments.
	Power:	from -100.00 dBm to 3.00 dBm with 0.01 dBm increments
Tone 2:	Frequency:	from 0 Hz to 3999 Hz with 1 Hz increments.
	Power:	from -100.00 dBm to 3.00 dBm with 0.01 dBm increments
Cadence:		0- 9999ms ON, 0-9999ms OFF with 1ms increments

To hear the configured ringback tone, call special test number 04

#### 4.2.1.4 Reorder Tone

Reorder tone is a tone advising the caller that the groups of lines or switching equipment necessary for the setting-up of the required call or for the use of a specific service are temporarily engaged. In North America, the most common reorder tone consists of the sum of two tones: Tone 1 = 480 Hz and Tone 2 = 620 Hz at -16 dBm with a cadence of 2500ms ON, 250ms OFF

PSTN Manager allows following values for reorder tone:



- Tone 1:
  - Frequency: from 0 Hz to 3999 Hz with 1 Hz increments.
  - Power: from -100.00 dBm to 3.00 dBm with 0.01 dBm increments
  
- Tone 2:
  - Frequency: from 0 Hz to 3999 Hz with 1 Hz increments.
  - Power: from -100.00 dBm to 3.00 dBm with 0.01 dBm increments
  
- Cadence: 0- 9999ms ON, 0-9999ms OFF with 1ms increments

To hear the current reorder tone, call special test number 05.

#### 4.2.1.5 Call Waiting Tone

Call waiting tone is a tone advising the user of the call waiting supplementary service who is engaged on a call that someone is attempting to call his number. In North America, the most common Call Waiting tone consists of single tone: Tone 1 = 440 Hz at -13 dBm with a cadence of 300ms ON, 10000ms OFF.

PSTN Manager allows following values for Call Waiting tone:

- Tone 1:
  - Frequency: from 0 Hz to 3999 Hz with 1 Hz increments.
  - Power: from -100.00 dBm to 3.00 dBm with 0.01 dBm increments
  
- Tone 2:
  - Frequency: from 0 Hz to 3999 Hz with 1 Hz increments.
  - Power: from -100.00 dBm to 3.00 dBm with 0.01 dBm increments
  
- Cadence: 0- 9999ms ON, 0-9999ms OFF with 1ms increments

To hear the configured call waiting tone, call special test number 06.

#### 4.2.1.6 Test Tone

Test tone is a non-standard tone of the user's choice.

PSTN Manager allows following values for the test tone:

- Tone 1:
  - Frequency: from 0 Hz to 3999 Hz with 1 Hz increments.
  - Power: from -100.00 dBm to 3.00 dBm with 0.01 dBm increments
  
- Tone 2:
  - Frequency: from 0 Hz to 3999 Hz with 1 Hz increments.
  - Power: from -100.00 dBm to 3.00 dBm with 0.01 dBm increments

To hear the configured test tone, call special test number 08.

#### 4.2.1.7 TAS Tone for Type II Caller ID

TAS (TE Alerting Signal) is being used in Caller ID Type II transmissions.

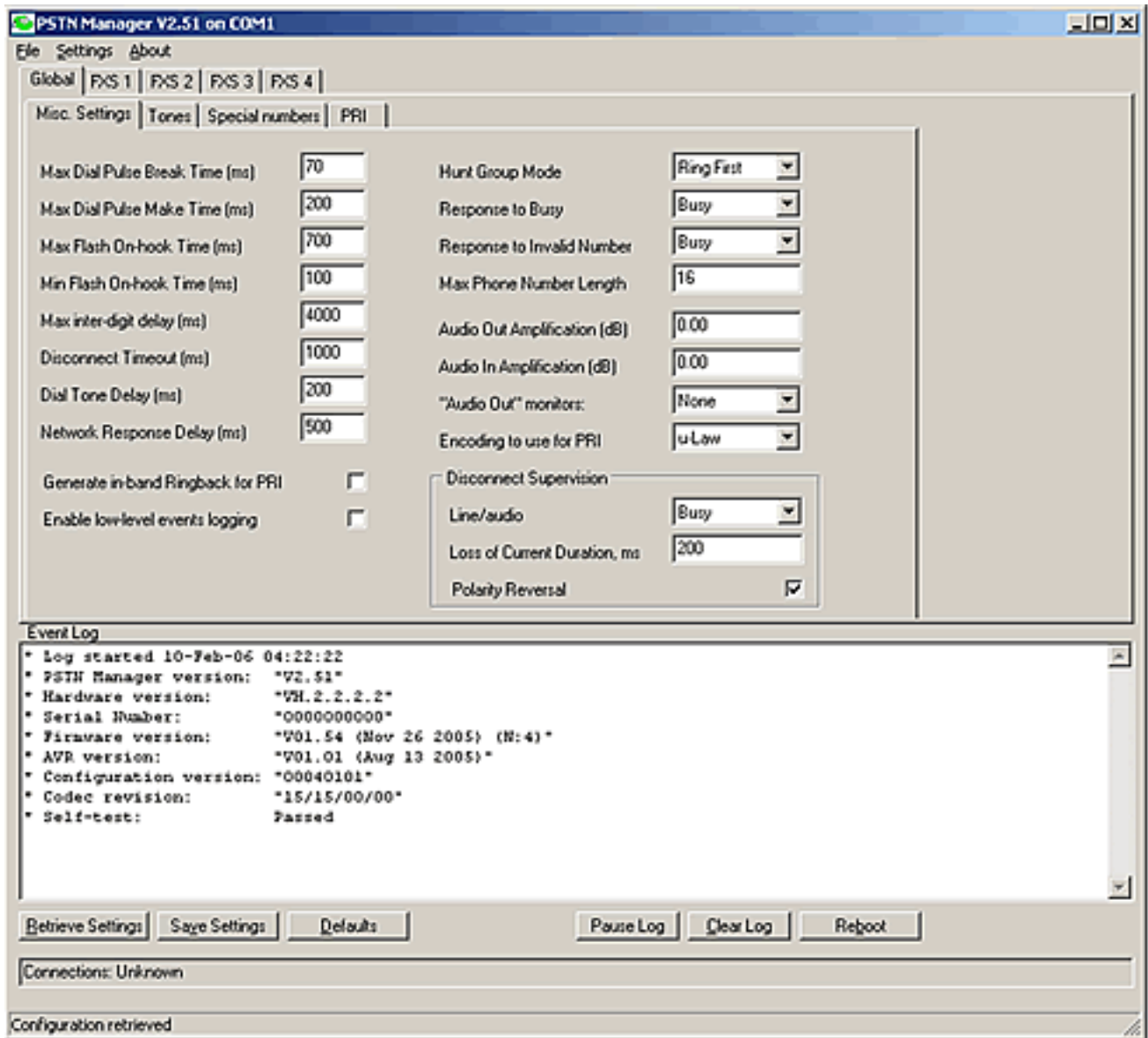
Please refer to a GR-30-CORE, LSSGR standard for details on utilizing TAS signal and its effect to Caller ID transmission.

#### 4.2.1.8 Idle State Alert Tone for UK Caller ID transmission

This tone is specific to a UK BT Caller ID transmission and being use only when UK Caller ID Type is selected on FXS ports.

Please refer to a European Telecommunications Standard ETS 300 001 for details on using this tone.

#### 4.2.2. Disconnect Supervision



"Disconnect Supervision" defines how PSTN Simulator treat an event when Called part disconnects. In the event when Called party disconnects Calling party can hear one of the following types of in-band audio signal: Busy or Dialtone or Reorder or Silence.

Those in-band signals can be used by DSP based systems for correct detection of disconnect event.

In addition to in-band signal Simulator can advise Calling party by Polarity Reversal (Battery Reversal) or brief loss of current supplied to a line.

Options "Loss of Current" and "Polarity Reversal" can be used at the same.

### **4.2.3 Maximum Flash on-hook time**

"Maximum Flash on-hook time" defines the border line between "Flash" and "On Hook" events. Pressing a hook for less than this time will result in a "Flash", pressing a hook for a longer than this time will result in an "On Hook" event.

Default timer value is 700ms.

PSTN Manager allows following values for the "Maximum Flash on-hook time":  
From 0ms to 65535ms with 1ms increment.

### **4.2.4 Minimum Flash on-hook time**

"Minimum Flash on-hook time" defines a minimum time hook has to be pressed to generate a "Flash". Pressing a hook for less than this time will not have any result, pressing a hook for a longer than this time will result in "Flash" event.

Default timer value is 100ms.

PSTN Manager allows following values for the "Minimum Flash on-hook time":  
From 0ms to 65535ms with 1ms increment.

### **4.2.5 Disconnect timeout**

"Disconnect timeout" defines a minimum time that the hook has to be pressed to generate an "On Hook" event.

Default timer value is 1000ms.

PSTN Manager allows following values for the "Disconnect timeout":  
From 0ms to 65535ms with 1ms increment.

### **4.2.6 Dial tone delay**

"Dial tone delay" is the timer that defines how soon the user will hear a dial tone after lifting a handset.

Default timer value is 200ms.

PSTN Manager allows following values for the "Dial tone delay":  
From 0ms to 65535ms with 1ms increment.

#### 4.2.7 Network response delay

“Network response delay” is a timer that defines how soon the user will hear Ringback after dialing the destination phone number.

Default timer value is 500ms.

PSTN Manager allows following values for the “Network response delay”:From 0ms to 65535ms with 1ms increment.

#### 4.2.8 Hunt group mode

PSTN simulator supports different modes of operation and logic for the hunt groups. Hunt group is a subset of phone lines that have been assigned an identical phone number. The easiest way to create a Hunt Group is to assign an identical secondary phone number to several phone lines while keeping primary phone number unique. This way you can still dial particular phone line and have a hunt group with the secondary phone number.

When PSTN simulator identifies that the dialed phone number has been assigned to more than one phone line it will use a "Hunt group mode" parameter to determine which of the phone lines Ring signal has to be applied to.

Default value is: Ring first

PSTN Manager allows following values for the Hunt group mode:

**Ring first** (Rings the first phone line in a hunt group)

**Ring next** (Rings the first available phone line in a hunt group)

**Ring all** (Rings all phone lines in a hunt group)

#### 4.2.9 Response to busy

“Response to Busy” defines which tone calling party will hear when the called party is engaged in another call.

Default value is: Busy

PSTN Manager allows following values for the “Response to busy”

Busy (Calling party will hear a busy tone)

Reorder (Calling party will hear a reorder tone)

#### 4.2.10 Response to invalid number

“Response to invalid number” defines which tone the calling party will hear when the called number does not exist.

Default value is: Busy

PSTN Manager allows following values for the “Response to invalid number”:

Busy (Calling party will hear a busy tone)

Reorder (Calling party will hear a reorder tone)

Silence (Calling party will hear nothing)

#### **4.2.11 Maximum phone number length**

“Maximum phone number length” defines how many digits PSTN simulator has to collect before forwarding call to a PRI module.

Default value is: 10

PSTN Manager allows following values for the “Maximum phone number length”:From 1 to 16 with increment of 1

#### **4.2.12 “Audio Out” amplification**

“Audio Out amplification” defines amplification in dB that will be applied to the original signal before sending it out of the “Audio Out” port.

Default value is: 0.00 dB (no amplification)

PSTN Manager allows following values for the Audio Out amplification:  
From -100 dB to + 30dB with 0.01 dB increment

#### **4.2.13 “Audio In” amplification**

“Audio In amplification” defines amplification in dB that will be applied to the original signal before feeding this signal into any of the FXS or PRI.

Default value is: 0.00 dB (no amplification)

PSTN Manager allows following values for the “Audio In amplification”:From -100 dB to + 30dB with 0.01 dB increment

#### **4.2.14 “Audio Out” monitors**

“Audio Out monitors” defines which FXS port will be monitored by the “Audio Out” port. Example: You want to record or broadcast to speakers everything that a telephone set connected to FXS port 1 is receiving or transmitting. You would select Port 1 as a value for this parameter and connect “Audio out” port to a recorder or speakers.

Default value is: None

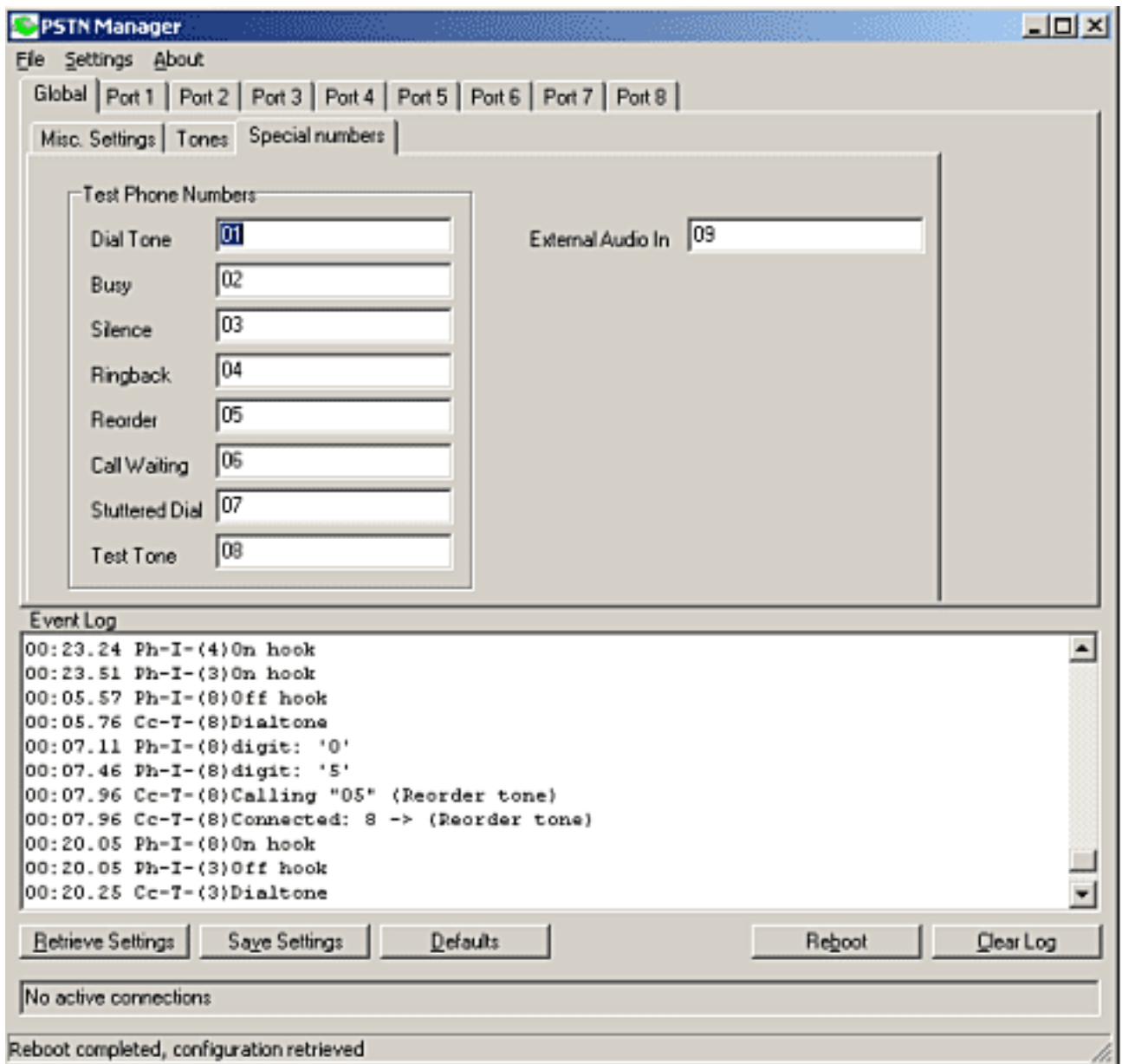
PSTN Manager allows following values for the “Audio Out monitors”:  
From: Port 1 to Port 8

### **4.3 Special test phone numbers**

Special test phone numbers are the virtual phone lines by calling to which you can hear standard call progress tones or "Audio in" line. You can come up with your own phone numbers using up to 16 digits for each.

Following test phone numbers are supported in firmware 1.0

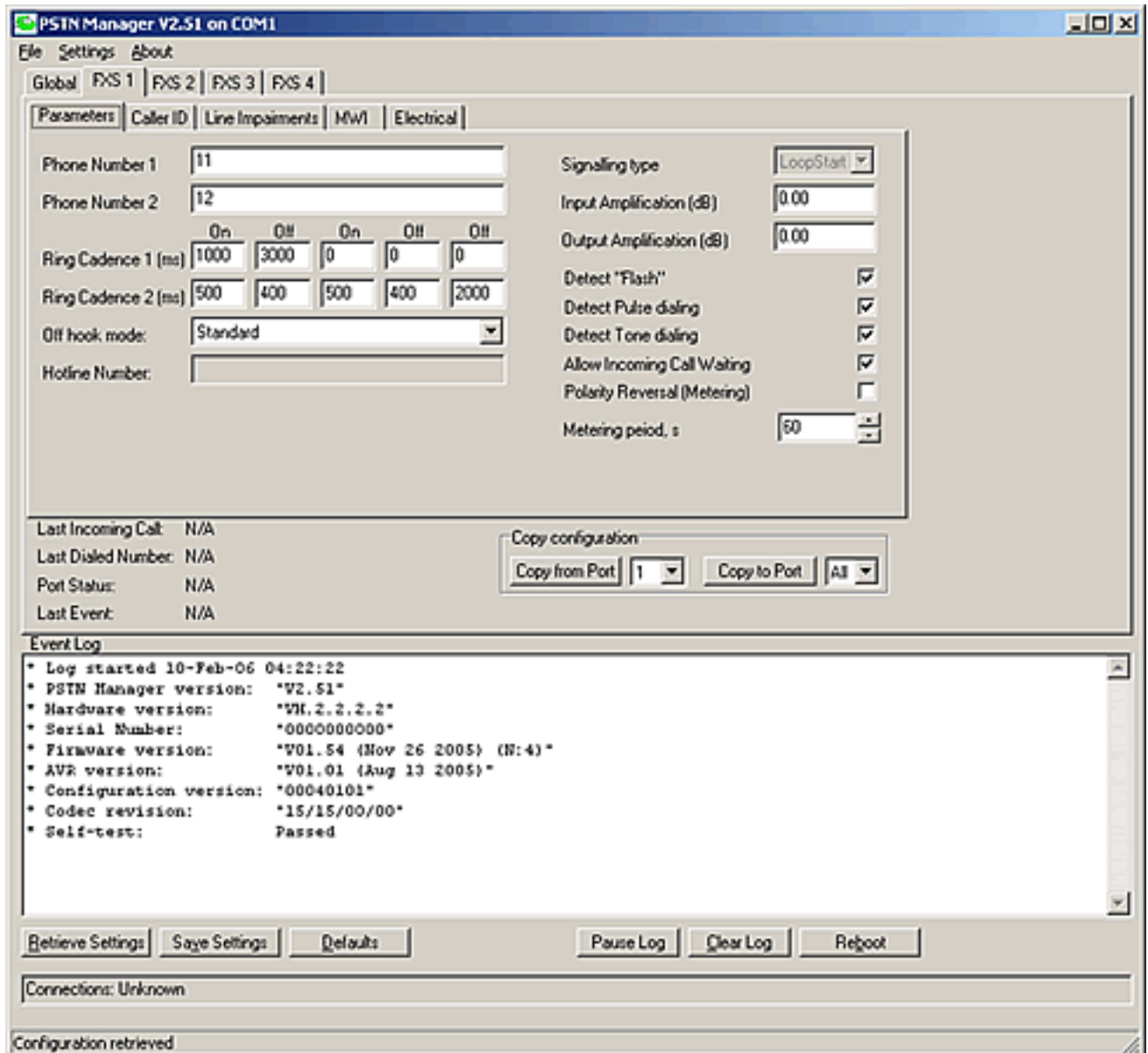
Event	Test Phone
Dial tone	01
Busy	02
Silence	03
Ringback	04
Reorder	05
Call Waiting	06
Stuttered Dial	07
Test Tone	08
Audio In	09



## 4.4 Configuring Hunt Groups

You can configure a Hunt Group by assigning an identical phone number to two or more phone lines. Each FXS port can have up to two phone numbers assigned and it is easier to use second phone number for creating a Hunt Groups while still having a unique primary phone number so you can call particular phone line even if it is still a member of the Hunt Group.

## 4.5 Configuring FXS interfaces



### 4.5.1 Signaling type

There are two types of signaling available to the FXS ports: Loop Start or Ground Start.

**Loop Start** is a two-way, two-wire (tip and ring) signaling method in which the current used for the signaling flows in a circuit (loop) between a telephone and PBX or a telephone and a central office. Seizure of the line is accomplished by going off-hook which causes current to flow in a circuit (loop).

**Ground Start** is a method of signaling from a terminal or subscriber loop to a switch, in which method one side of a cable pair is temporarily grounded.

When used as a trunk, loop start lines are more susceptible to event of glare when both sides try to seize the line simultaneously. Ground start is less affected by this problem and is preferred way to establish a trunk with FXS interface. Support for ground start signaling will be added in the next versions of firmware.

**Note:** Ground Start Signaling is not available on all simulator models. Please consult manufacturer before purchasing if this feature is important to you.

### 4.5.2 Line impedance

600 Ohm or 800 Ohm selectable impedance is available for each FXS port

### 4.5.3 Assigning Phone numbers

There are two phone numbers that can be assigned to each FXS port.

Each number can be from 1 to 16 digits long. You can dial either one to call a particular FXS port. You can use secondary phone numbers to create a Hunt Groups or to test custom ring cadences.

Default numbers for each port are two digit numbers:

FXS 1	First number 11, Second number 12
FXS 2	First number 21, Second number 22
FXS 3	First number 31, Second number 32
FXS 4	First number 41, Second number 42
FXS 5	First number 51, Second number 52
FXS 6	First number 61, Second number 62
FXS 7	First number 71, Second number 72
FXS 8	First number 81, Second number 82

### 4.5.4 Configuring ring cadence

Cadence is the rhythm of the ring voltage applied to a telephone connected to FXS port.

It is defined as a sequence of ON and OFF timers.

You can configure two ring cadences for each FXS port. One cadence for each phone number.

Default value for Phone 1 is: 1000ms ON, 3000ms OFF

Default value for Phone 2 is: 500ms ON, 400ms OFF, 500ms ON, 400ms OFF, 2000ms OFF

PSTN Manager allows following values for the Cadence timers:

From 0ms to 9999ms with 1ms increment

**Note:** Using a 0 value for the first ON timer will prevent a ring on that FXS port. This can be useful for maintaining quiet environment, when the ring noise is not desirable, however work still has to be done.



#### 4.5.5 Off hook mode

“Off hook mode” defines how PSTN simulator must treat an OFF hook even on particular FXS port. Default value is “Standard”. PSTN simulator will provide a Dial tone and will wait for the called number to be dialed manually.

PSTN Manager allows following values for the “Off Hook mode”:

<b>Standard</b>	Default
<b>Hotline (PLAR)</b>	Automatically dials pre-configured phone number assigned in the "Hotline" filed
<b>Do nothing</b>	No dial tone will be provided

#### 4.5.6 Input/Output amplification

“Input and Output amplification” defines amplification in dB that will be applied to a signal which comes from/to a connected telephone set to a particular FXS port.

Default value is: 0.00 dB (no amplification)

PSTN Manager allows following values for the “Input” and “Output” amplification: From -100 dB to + 30dB with 0.01 dB increment

#### 4.5.7 Detect “Flash”

Check button “Detect Flash” turns “ON” detection of the “Flash” event.

“Flash” is an event of pressing a telephone’s hook for a time between “Min and Max Flash on hook time” defined in Global parameter tab.

#### 4.5.8 Detect Pulse dialing

Turns ON and OFF detection of Pulse dialing

Default is: ON

#### 4.5.9 Detect Tone dialing

Turns ON and OFF detection of Tone dialing

Default is: ON

#### 4.5.10 Allow Incoming Call Waiting

Turns ON and OFF an incoming call notification when called party is engaged in another call.

Called party will hear a Call waiting tone either every 10 seconds or according to a configured cadence if different value was defined.

Default is: ON

#### 4.5.11 Maximum Interdigit Delay

Maximum interdigit delay defines maximum time allowed between two consecutive dialed digits.

After this timer is expired, simulator considers that destination number was dialed completely and it will perform a phone table lookup or forward it to a PRI module.

Default value is: 3 seconds (3000ms)

#### **4.5.12 Polarity Reversal for Metering**

Some applications such as pay phones and billing systems require polarity reversal to occur in a predefined interval for correct billing calculation.

You can enable Metering polarity Reversal on each FXS port and Polarity Reversal will occur every 60 seconds (60 seconds is default). It can be adjusted to a lesser or greater time interval.

#### **4.5.13 Configuring Caller ID**

##### **4.5.13.1 Caller ID Overview**

Caller ID is the CLASS service of PSTN simulator which delivers the calling party phone number and name to the telephone set of the called party. This service gives the called party ability to identify caller by telephone number or name even before answering the phone or when is on call with a third party.

North American Caller ID services were defined by BellCore. The documents GR-30-CORE and SR-TSV-002476 specify the voice-band data transmission which governs the interface for Caller ID (Type I) and Caller ID on Call Waiting (CIDCW). CIDCW is also called Type II application where the data transmission is under the off-hook condition.

GR-30-CORE, Voice-band Data Transmission Interface, can be used for references for Caller ID and its protocol. It describes the standard modem-based technology from an SPCS (Stored Program Controlled Switching System) to a CPE (Customer Premises Equipment). It provides some of the lower-layer requirements of the CID protocol. SR-TSV-002476, CPE Compatibility Considerations for the Voice-band Data Transmission Interface, defines requirements for CPE compatibility with GR-30-CORE. It addresses signaling protocol, data transmission, signaling detection and generation, and design consideration.

##### **4.5.13.2 Caller ID Type I and Caller ID Type II**

Two types of Caller ID are available for configuration. Caller ID Type I is the service when caller information is delivered when called telephone is on hook. Caller ID Type II is the service when caller identity is delivery to the called party while called party is engaged in a call with a third party. This type of Caller ID also know as "Caller ID on Call Waiting" or CIDCW.

Caller ID type I is a subset of Type 2.

##### **4.5.13.3 Caller ID multiple or single message**

Single Data Message Format (SDMF) or Multiple Data Message Format (MDMF) defines which information delivered to the called party as a part of Caller ID Data message.

SDMF delivers date, time and a calling number.

MDMF delivers date, time, calling number and a caller's name.

##### **4.5.13.4 Caller ID Data**

Caller ID data consists of date, time, caller's phone number and a caller's name

##### **4.5.13.5 Reasons for absence of Caller ID information**

Caller ID information is not always can be presented to the called party. Due to a lack of SS7 signaling in some areas, calling number cannot be exchanged among telephone switches. In this case called party will have an "Out of Area" message instead of the calling party phone number.

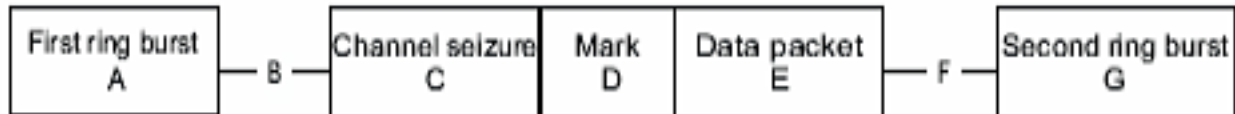
Calling party can also elect an option to make its identity invisible to the called party. In that case called party will have a "Private" message instead of the calling party phone number.

PSTN simulator allows you to generate all possible scenarios with a missing Caller ID data and to choose a particular reason for Caller ID data absence.

#### 4.5.13.6 Caller ID transmission parameters

Caller ID data consists of date, time, callers phone number and a callers name

Fig: **Caller ID Type transmission flow**



PSTN simulator allows precise manipulation of the Caller ID transmission parameters.

**"First Ring On"** defines the time for the duration of the first ring burst. In the Caller ID transmission flow diagram it corresponds to the Event A.

Default value is 1.0 second.

**"Pre CID Delay"** defines the timeout between the end of the first ring and the start of the channel seizure block. In the Caller ID transmission flow diagram it corresponds to the Event B.

Default value is 0.6 second.

**"Number of Seizure bits"** parameter defines the block of continuous bits of alternating 0 and 1. The first bit to be transmitted is 0 while the last bit is 1. It is used for Caller ID Type I only. In the Caller ID transmission flow diagram it corresponds to the Event C.

Default value is 300 bits.

**"Number of Mark bits"** parameter defines the number of bits of continuous high transmitted after the seizure bits block. In the Caller ID transmission flow diagram it corresponds to the Event D.

Default value is 180 bits.

**"Post CID Delay"** defines the timeout between the end of the Caller ID data transmission and the second ring. In the Caller ID transmission flow diagram it corresponds to the Event F.

Default value is 1.0 second.

#### 4.5.14 MWI. Message Waiting Indicator

PSTN Simulator support three types of Message Waiting Indicators

- a. FSK signal based
- b. Stutter Dialtone based
- c. DC voltage based (neon lamp)

Each of the types can be tested by pressing a Start/Stop buttons in PSTM Manager MWI tab.

#### 4.5.15 Electrical parameters

Physical level electrical parameters supported by PSTN Manager located in "Electrical" port specific tab.

Impedance:	600 and 800 Ohm
Off hook loop DC current:	10, 20 and 32mA
Ring Frequency:	16 to 75Hz
Ring Voltage:	30 to 55 VRMS

#### **4.5.16 Line Impairments**

Line impairments allow software developers and QA testers to simulate real world conditions on telephone lines such as noise, delays or echo. Delays can be configured for Transmit and Receive path separately for up to 2000 ms. Noise is a White Gaussian distribution noise. Echo can be configured for up to 2000ms

## **5. Chapter 5: Chapter 5: Using CLASS or PBX Features**

### **5.1 Call hold**

When you need to place a remote party on hold, you can do so by pressing a “Flash” button on your telephone. After pressing a “Flash” button you will hear a secondary (stuttered) dialtone which will allow you to make another call while a first party is still on hold. After you finished and would like to revert to a holding party you need to put your handset on hook and wait till you hear a ring. After you pick up handset you will be connected back to a party that was on hold.

### **5.2 Three-way-calling**

To use a there-way-calling, follow this procedure:

- Call a first party by dialing its phone number

- After first party has answered a call, press the Flash button on your telephone

- This will place a first party on hold and you will hear a secondary (stuttered) dial tone

- Once you have a secondary dial tone, dial a second party’s telephone number

- After a second party has answered a call, press again the Flash button on your telephone

- This will interconnect two calls into a three-way calling conference

## **6. Chapter 6: Optional T1/E1/PRI module**

### **6.1 Overview of T1/E1/PRI module**

Optional T1/E1/PRI module support switching audio/video or data calls among ant B-channels or FXS ports.

With a module installed you can place a call from any of the analog telephones to any of the PRI B-channels by dialing a one of the 30 phone numbers configured for each of the PRI ports and vise versa. T1/E1/PRI module supports PRI call switching over T1 or E1 and supports following PRI switch types:

**National NI-2, 5ESS, DMS-100, NET3 and Q.SIG**

T1 or E1 selection is done on the fly and does not require any hardware upgrades.  
PRI ports are equipped with a standard RJ45 jack for T1 or balanced E1 connection.

### 6.1.1 RJ48/45 pin-out for T1/E1/PRI module

PSTN simulator is equipped with a two standard RJ45 jacks for PRI ports and connects to a PRI end-devices with straight shielded CAT3 or CAT5 twisted pair cable.

Following cable pin-out is being used at the simulator side:

Pin	Description
1	RX Tip
2	RX Ring
3	RX SHIELD
4	TX TIP
5	TX RING
6	TX SHIELD
7	Not used
8	Not used

**Note:** For connecting PSTN simulator to a Cisco router with a balanced 120-ohm E1/PRI interface with a DB15 connector you need to use Cisco cable P/N: CAB-E1-PRI/NT

### 6.1.2 LEDs description for T1/E1/PRI module

Each PRI port have two integrated LEDs: Left LED is Yellow and Right LED is Green.

When Yellow (left) LED is ON, it indicates that a Remote (Yellow) alarm is present.

When Green (right) LED is ON it indicates that the interface in activated.

When Green (right) LED is blinking, it indicates that one or more calls are active on that port.

## 6.2 Configuring T1/E1/PRI module

When PRI module is installed your ISDN Manager will display two additional tabs for configuring T1, E1 and PRI parameters.

### 6.2.1 Interface type: T1 or E1

First step in configuring PRI call switching is interface type selection. PRI module supports full T1 or full E1 operation. Default is T1.

### 6.2.2 Line Code and Framing

There are several different linecodes and framings available for T1 or E1 interfaces

**Linecode for T1:** AMI, B8ZS  
**Framing for T1:** SF, ESF

**Linecode for E1:**        **AMI, HDB3**  
**Framing for E1:**       **CRC4, no-CRC4**

**Linecode and framing must match on CPE and simulator. If you experience a large number of CRC errors, packet loss or connectivity problems, always check linecode and framing first and ensure that it matches on both ends.**

### **6.2.3 Line build-out**

Line build-out ensures signal attenuation compensation for long cable runs. If you connect to a simulator with a cable greater than 100 ft. long, select the appropriate line build-out.

### **6.2.4 Clock source**

Clock source is the one of the most important parameters in ensuring flawless operation of T1 or E1 lines. PSTN simulator's PRI module requires that module itself (network) should provide a clock to connected devices. If you connect a router or another CPE device with a T1/E1 interface to a simulator PRI interface, configure it to use a network clock provided by simulator for synchronization purposes.

### **6.2.5 Configuring PRI Switch type**

PSTN simulator PRI module conforms to a different PRI switch signaling options:

- National ISDN 2
- ATT/Lucent 5ESS
- Nortel DMS-100
- Q.931 Q.SIG

### **6.2.6 Scan Order**

"Scan Order" parameter defines the B-channel selection order for outbound calls. When simulator places a call to an end device it has to select an outgoing B-channel. If the "ascending" order is configured simulator will select the lowest B-channel available. It will select a highest B-channel if the "descending" order is configured.

With a high call rate it is statistically possible that the end device and the simulator will try to seize the same B-channel resulting in the event called glare. To prevent glare from occurring it is advisable that the CPE and the Simulator use different B-channel selection orders.

### **6.2.7 Configuring PRI Phone numbers**

You can configure up to 30 phone numbers on the "Phone numbers" tab in PSTN Manager. Phone numbers are not associated with any particular B-channels which are being allocated based on "Scan Order" parameter value.

### **6.2.8 Simulating Network Delays**

To add transmission delays to B-channels, similar to the ones existing in satellite based links, select the desired delay timer from 1ms to 500ms on Tools tab.

Delays apply to B-channels only, and not to D-channel. Default value is: 0 ms (no delays)

## 6.2.9 Generating Yellow alarms

It is possible to generate Yellow (remote) alarm on each T1/E1/PRI port by pressing a Start button on Tools sub-tab.

## 7. Chapter 7: Upgrading PSTN simulator firmware

Virtual console is constantly working on adding new features to its products. Please check <http://www.vconsole.com> for the latest firmware updates for your PSTN simulator. There is a separate firmware file for 8 port FXS chassis and for PRI module. If your simulator has a PRI module installed, make sure that you update PRI firmware as well. After upgrading a firmware for FXS and/or PRI, always use a latest Manager application which is available for downloading from [www.vconsole.com](http://www.vconsole.com).

To determine the current version of FXS or PRI module firmware use a PSTN Manager log window output:

```

* Log started 1-May-04 19:21:55
* Hardware version:      "VH.2.2.1.1"
* Serial Number:        "0000000000"
* Firmware version:     "V01.03 (Apr 30 2004)"      <-Chassis firmware version
* AVR version:          "V01.01 (Apr 14 2004)"
* Configuration version: "00040100"
* Codec revision:       15/15/15/15
* Module TMS version:   "V01.16 (Apr 24 2004)"      <-PRI module firmware version
* Module cfg version:   "00030104"
* Module AVR version:   "V01.01 (Apr 17 2004)"
* Module Hardware version: "TH.02.03.01.03, BbkVer=2"

```

### Upgrading FXS chassis firmware

### Upgrading PRI module firmware

## 8. Chapter 8: PSTN Simulator Technical Specification

### Power

External power supply	
Input:	110-240 VAC, 50/60 Hz
Output:	20 VDC 1.5A

### Telephone Interface (POTS-FXS)

Connector:	RJ11/RJ45 Modular Jack
Loop Current:	20 mA
Battery Source:	48 VDC
Ringing Source:	20 - 80 VAC with 10 V increments. (Default is 60 VAC)
Ring Cadence:	

Up to two distinctive cadences can be assigned per port, one per each of the phone numbers.

“Double” cycles support (ON, OFF, ON, OFF, PAUSE)“ON” and “OFF” cycles: 0-65 seconds in 1 ms increments

Ring Frequency :	17 - 70 Hz in 1 Hz increments
Impedance:	600 Ohm + 2.2 uf or 900 Ohm + 2.2 uf
Attenuation/Amplification:	-90 to +30 dB in 0.01 dB increments.
On-hook Disconnect Timeout:	1 - 65535 ms in 1 ms increments
On/Off-hook Detect Flash:	1 - 65535 ms in 1 ms increments
Line Reversal:	50 - 1000 ms in 5 ms increments

### Call Progress Tones

Ringback Tone (Dual or Single tone)	
Frequency:	440 Hz and 480 Hz (default)
Range:	0 - 4000 Hz in 1 Hz increments
Level:	-16 dBm per tone (default)
Range:	-90 to +3 dBm in 0.01 dBm increments
Cadence:	2000 ms (ON) - 4000 ms (OFF) (default)
Range:	0 - 65 sec. in 1 ms increments
Dial Tone (Dual or Single tone)	
Frequency:	350 Hz and 440 Hz (default)
Range:	0 - 4000 Hz in 1 Hz increments
Level:	-13 dBm per tone (default)
Range:	-90 to +3 dBm in 0.01 dBm increments
Busy Tone (Dual or Single tone)	
Frequency:	440 Hz and 620 Hz (default)
Range:	0 - 4000 Hz in 1 Hz increments
Level:	-24 per tone dBm (default)
Range:	-90 to + 3 dBm in 0.01 dBm increments
Cadence:	500 ms (ON) - 500 ms (OFF) (default)
Range:	0 - 65 sec. in 1 ms increments
Reorder Tone (Dual or Single tone)	
Frequency:	480 Hz and 620 Hz (default)
Range:	0 - 4000 Hz in 1 Hz increments
Level:	-16 dBm per tone (default)
Range:	-90 to +3 dBm in 0.01 dBm increments
Cadence:	250 ms (ON) - 250 ms (OFF) (default)
Range:	0 - 65 sec. in 1 ms increments
Test Tone	
Frequency:	1004 Hz (default)
Range:	0 to 4000 Hz in 1 Hz increments
Level:	-90 to + 3 dBm in 1 dBm increments
Call Waiting Tone (Dual or Single tone)	
Frequency:	440 Hz single (default)
Range:	0 - 4000 Hz in 1 Hz increments
Level:	-13 dBm (default)
Range:	-90 to +3 dBm in 0.01 dBm increments
Cadence:	300 ms (ON) - 10000 ms (OFF) (default)
Range:	0 - 65 sec. in 1 ms increments



Number Unobtainable Tone (Dual or Single tone)	
Frequency:	950 Hz and 1400 Hz $\pm$ 1 Hz (Default)
Range:	0 - 4000 Hz in 1 Hz increments
Level:	-13 dBm (default)
Range:	-90 to + 3 dBm in 1 dBm increments
Cadence:	330 ms (ON) - 5 ms (OFF) - 330 ms (ON) - 5 ms (OFF) (default)
Range:	0 - 10 sec. in 5 ms increments

### Dialing Characteristics

Rotary Detection	8 - 30 PPS
Percent Break range:	40 ms / 60 ms Make / Break ratio (US, Germany) 33 ms / 67 ms Make / Break ratio (UK, France, Ireland)
Minimum break time:	6 ms
Maximum break time:	programmable
Minimum make time:	6 ms
Maximum make time:	programmable
Minimum inter-digit time:	programmable
Ring trip detection time:	2 periods

### DTMF Detection

Complies with the requirements of ITU-T  
Q.24, BellCore GR-30-CORE (TR-NWT-000506)  
Deutsche Telecom network (BAPT 223 ZV 5)

Frequency accept:	$< \pm 1.5\%$ (+4 Hz) and $< \pm 1.8\%$
Frequency reject:	$> \pm 3\%$
Positive twist accept:	$< 8$ dB
Negative twist accept:	$< 8$ dB
Min tone accept duration:	40 ms
Max tone reject duration:	25 ms
Min inter-digit pause duration:	40 ms
Max tone drop-out duration:	20 ms
Signaling velocity:	$\geq 93$ ms/digit
Signal detection level:	-48 to 0 dBm
DTMF noise tolerance:	-12dB to lowest amplitude tone

### Caller ID

FSK Tone Level:	-15 dBm per tone (default)
Range:	-10 to -60 dBm in 1 dBm increments
FSK Low Frequency (Tone 1):	1200 Hz (default)
Range:	1100 - 1400 Hz in 1 Hz increments
FSK High Frequency (Tone 2):	2200 Hz (default)
Range:	2000 - 2300 Hz in 1 Hz increments
CAS Tone Level:	-15 dBm per tone (default)
Range:	- 6 to 60 dBm in 1 dBm increments
CAS Tone 1:	2130 Hz (default)
Range:	10 - 3000 Hz in 1 Hz increments
CAS Tone 2:	2750 Hz (default)
Range:	10 - 3000 Hz in 1 Hz increments

CAS Tone Duration: Range:	80 ms (default) 70 - 90 ms in 5 ms increments
Wait for ACK: Range:	160 ms (default) 100 - 300 ms in 5 ms increments

---

**Line Impairments**

Echo Time:	Selectable from 0 ms to 100 ms
Level:	60 dB (default)
Configurable Range:	4 - 60 dB in 1 dB increments
Level Accuracy:	± 1 dB

Satellite Delay  
0 - 1000 ms in 1 ms increments

White Noise Range:	300 - 3400 Hz
Level Accuracy	± 2 dB

---

**Audio Output Port (Monitoring)**

Connector:	3.5 mm mono audio plug
Output Impedance:	600 Ohm
Frequency response:	300 - 3400 Hz
Amplification:	-90 to +30 dB in 0.01 dB increments

---

**Audio Input Port (Music on Hold)**

Connector:	3.5 mm mono audio plug
Output Impedance:	600 Ohm
Amplification:	-90 to +30 dB in 0.01 dB increments

---

## 9. Chapter 9: Glossary of Terms

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**AMI (Alternate Mark Inversion)**

Alternate "1" bits have their polarity reversed (between + and - 3 volts). Also, between two successive "1"s, the voltage briefly pauses at 0. Most often used if framing mode is SF/D4.

**B8ZS (Binary 8 Zero Substitution)**

Method of maintaining density of "1"s on the line by deliberately inserting "bipolar violations". Most often used if framing mode is ESF.

**Busy Tone**

A tone advising the caller that the telephone number is busy.

**Call Waiting**

A service feature that provides an indication to a terminal already engaged in an established call that one or more calls are awaiting connection.

**Call Waiting Tone**

A tone advising the user of the call waiting supplementary service who is engaged on a call that someone is attempting to call his number.

**Caller ID**

A network service feature that permits the recipient of an incoming call to determine, even before answering, the number from which the incoming call is being placed.

**CO (Central Office)**

Local telephone company office where all local loops in a given area connect and circuit switching of subscriber lines occurs.

**Codec**

Coder-Decoder. Device that typically uses pulse code modulation (PCM) to transform analog signals into a digital bit stream and digital signals back to analog.

**Congestion Tone**

A tone advising the caller that the groups of lines or switching equipment necessary for the setting-up of the required call or for the use of a specific service are temporarily engaged.

**CPE (Customer Premises Equipment)**

Communications equipment that resides on the customer's premises.

**dB**

Abbreviation for decibel. One tenth of the common logarithm of the ratio of relative powers, equal to 0.1 B (bel).

**dBm**

Abbreviation. dB referenced to one milliwatt. dBm is used in communication work as a measure of absolute power values. Zero dBm equals one milliwatt.

**Dial Tone**

A tone advising that the exchange is ready to receive call information and inviting the user to start sending call information.

**DID**

(Direct Inward Dial) DID trunks are one-way trunks that allow you to dial into a PBX without operator intervention.

**DOD**

(Direct Outward Dial) DOD trunks are also one-way trunks that allow you to connect directly to the CO.

**DS0**

Digital Signal Level 0. The base for the digital signal X series; transmits digital signals over a single channel at 64 Kbps.

**DS1**

Digital Signal Level 1. Framing specification used in transmitting digital signals at 1.544-Mbps on a T1 facility (United States) or at 2.108-Mbps on an E1 facility (Europe).

**DTMF**

Dual Tone Multi Frequency. The signal a telephone company receives when a telephone's touch keys are pressed. Also known as Touchtone.

**E1**

European digital trunk standard for 2.048Mbps transmission; an E1 trunk carries 30 64Kbps voice or data channels, plus two 64Kbps signaling and framing channels.

**E&M**

RecEive and TransMit. A trunking arrangement used for two-way switch-to-switch or switch-to-network connections. It is used with E1 and T1 interfaces.

**Echo Cancellation**

A technique used in high speed modems and voice circuits to isolate and filter out unwanted signal energy caused by echoes from the main transmitted signal.

**Flash**

While the phone is off-hook, quickly pressing and releasing the flash hook to signal the central office or PBX that you are requesting special processing, for example, call waiting.

**Flash hook**

The plunger the phone's handset rests on while on-hook.

**FSK (frequency-shift keying)**

Frequency modulation in which the modulating signal shifts the output frequency between predetermined values.

**FXO (Foreign Exchange Office)**

FXO interface connects to the Public Switched Telephone Network (PSTN) central office and is the interface offered on a standard telephone. FXO interface is an RJ-11 connector that allows an analog connection at the PSTN's central office or to a station interface on a PBX.

**FXS (Foreign Exchange Station)**

FXS interface connects directly to a standard telephone and supplies ring, voltage, and dial tone. FXS interface is an RJ-11 connector that allows connections to keysets, and PBX.

**Ground Start**

A method of signaling from a terminal or subscriber loop to a switch, in which method one side of a cable pair is temporarily grounded.

**HDB3**

A modified AMI signaling code that only applies to E-1 and is used to preserve one's density on the line.

**Hunt group**

A hunt group is a set of phone lines that you can access by using a single phone number.

**ITU-T**

International Telegraph and Telephone Consultative Committee. International body that develops worldwide standards for telecommunications technologies.

**Impedance**

The combined effect of resistance, inductance and capacitance on a transmitted signal. Impedance varies at different frequencies.

**Jitter**

The deviation of a transmission signal in time or phase. It can introduce errors and loss of synchronization in high speed synchronous communications.

**Loop Start**

A two-way, two-wire (tip and ring) signaling method in which the current used for signaling flows in a circuit (loop) between a telephone and PBX or a telephone and central office. Seizure of the line is accomplished by going off-hook which causes current to flow in a circuit (loop).

**Loop Current**

The current that flows through the circuit from the telephone switch to the voice board when the channel is off-hook

**On-hook**

The signaling state that occurs when a handset is sitting on the phone (the phone's inactive state) and the flash hook is depressed.

**Off-hook**

The signaling state that occurs when the handset is removed from the phone and the flash hook is released. When a phone is taken off-hook it signals the central office or PBX that it needs attention, for example, to make a call or to answering an incoming call.

**Pay Tone**

A tone advising users of a payphone that a payment is required.

**PBX (Private Branch Exchange)**

A private telephone exchange.

**PCM (Pulse Code Modulation)**

A procedure for adapting an analog signal (such as voice) into a 64 kbps digital bit stream for transmission.

**PRI**

An application that uses a single channel to carry the signaling for all other channels on a line. On a T-1 line, the application uses channels 1 through 23 (B channels) to carry data, digital voice, and compressed video. Channel 24 (D channel) carries the signaling for all 23 B channels. On an E-1 line, the application uses channels 1 through 15 and 17 through 31 (B channels) to carry data, digital voice, and compressed video. Channel 16 (D channel) carries the signaling for all 30 B channels.

**PSTN (Public Switched Telephone Network)**

The telecommunications network commonly accessed by ordinary telephones, key systems, PBX trunks and data equipment.

**Ring**

A signal of specific duration and character that indicates to a user that a calling party is engaged in an access attempt.

**Ring Detect**

The act of sensing that an incoming call is present by determining that the telephone switch is providing a ringing signal to the voice board.

**Ringback Tone**

A tone advising the caller that a connection has been made and that a calling signal is being applied to a telephone number or service point.

**Special Information Tone**

A tone advising the caller that the called number cannot be reached for reasons other than "subscriber busy" or "congestion".

The tone may also be used in conjunction with recorded announcements to signify that what the caller is about to hear is a recording. It should always be used to precede all call failure announcements

**Three Way Calling**

A facility that allows a simultaneous call to be established between three different numbers.

**Tone**

An audible signal consisting of one or more superimposed frequencies with a defined cadence and duration.

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