SIM-2FXS-A

Phone Line Simulator

User Manual 2.1 Updated March 2006

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1. Chapter 1: Important information

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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.

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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.
- 8. If the product has been dropped or the enclosure has been damaged.
- 9. 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 analogue PSTN call switching service eliminating the expense of the real phone lines for testing or demonstrating devices that require analogueue (FXS). 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 2 analogueue (FXS) loop start ports.

2.2. Overview of FXS ports

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

Depending on country of use, UK BT style adaptors including ring capacitors are provided.

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	Indication/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

2.3. DB9 port (COM 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.

2.4. 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. PSTN simulator is not designed for non-stop 24x7 operations. Keep it turned off while not in use.

3. Chapter 3: Getting started with PSTN Simulator

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 labelled "Power" Turn power switch into an ON position

3.3. Installing PSTN Manager software

Install PSTN Manager application 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 any Windows platform.

3.4. Connecting analogue telephones to simulator

To connect analogue telephones to FXS ports you can use regular RJ11 cables which are usually supplied with a telephones. Thought 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.

4. Chapter 4: Configuring simulator with PSTN Manager

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

Some of the features in PSTN Manager do not apply to this version of the Simulator. These features are greyed out if they do not apply.

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.

The CDROM supplied with the simulator has suggested configuration files for various countries.

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:

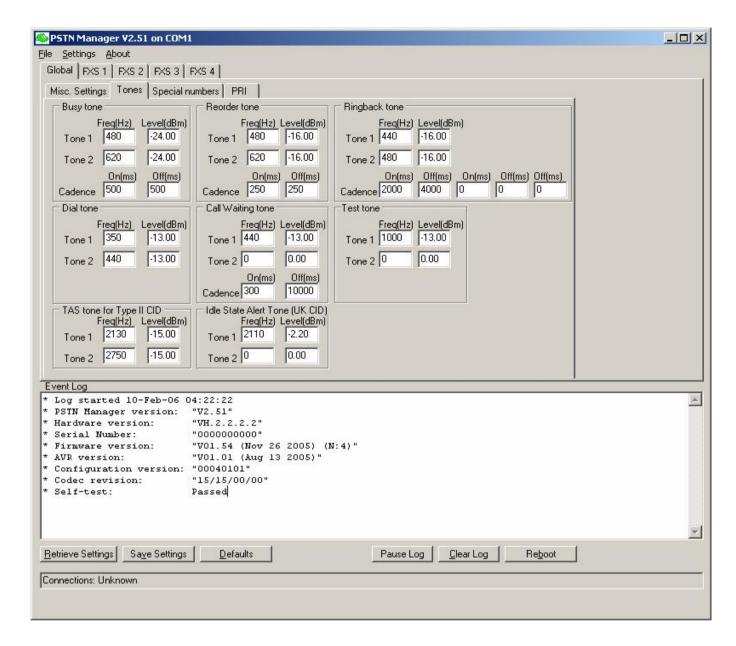
Single tone: Tone consist of a single tone at a given frequency

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 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

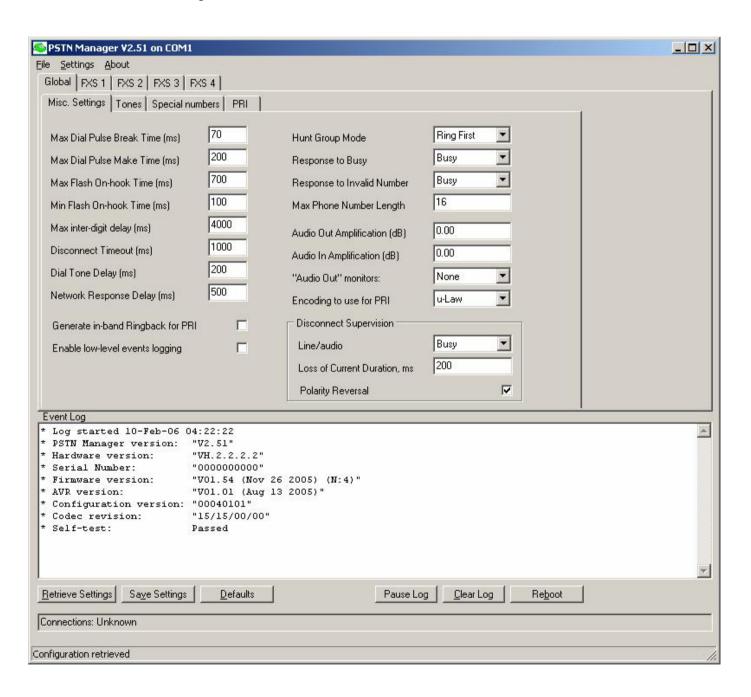
Tone 2:

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.

4.2.2. 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 have one type of in-band audio signal such as 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 short 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 dialling 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. 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)

4.2.9. 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)

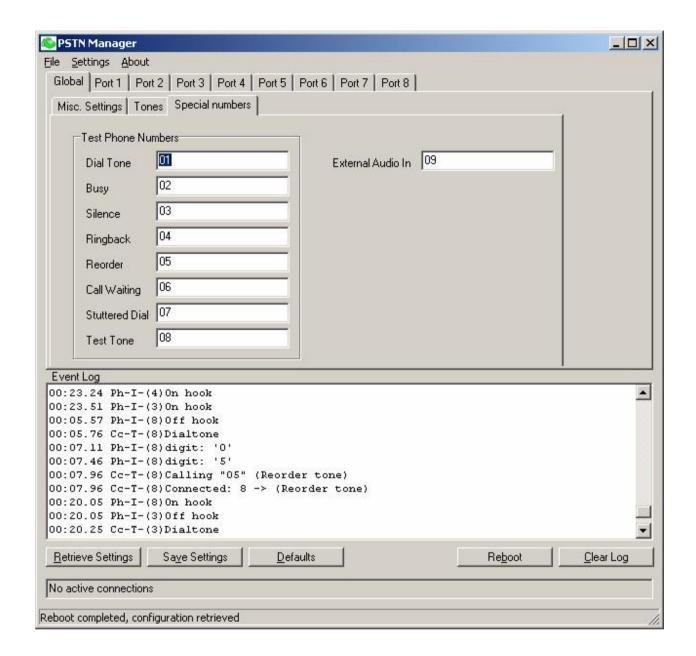
Silence (Calling party will hear nothing)

4.3. Special test phone numbers

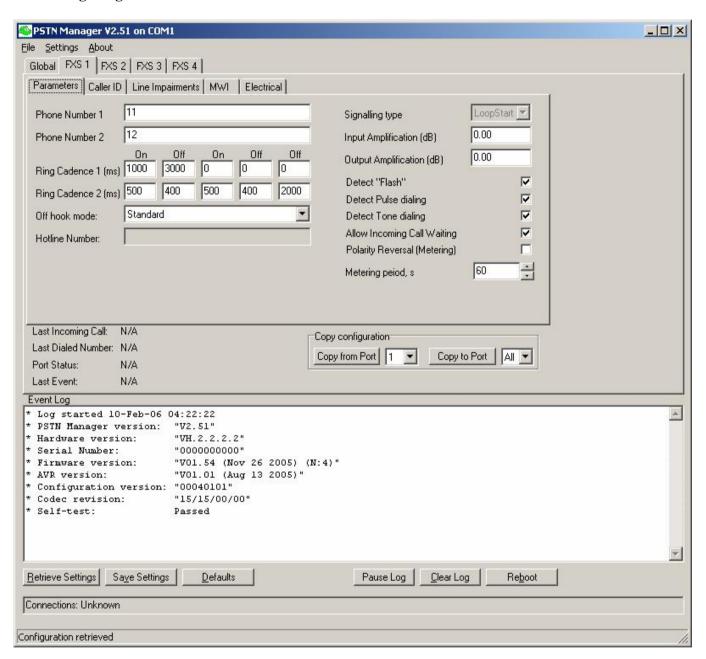
Special test phone numbers are the virtual phone lines by calling to which you can hear standard call progress tones. You can come up with your own phone numbers using up to 16 digits for each.

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



4.4. Configuring FXS interfaces



4.4.1. Line impedance

600 Ohm or 900 Ohm selectable impedance is available for each FXS port

4.4.2. 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

4.4.3. 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

4.4.4. 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 dialled manually.

PSTN Manager allows following values for the "Off Hook mode":

Standard (Default)

Hotline aka PLAR (Simulator automatically dials to a pre-configured phone number from

the "Hotline" filed)

Do nothing (no dial tone will be provided)

4.4.5. 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

4.4.6. 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.

4.4.7. Detect Pulse dialling

Turns ON and OFF detection of Pulse dialling

Default is: ON

4.4.8. Detect Tone dialling

Turns ON and OFF detection of Tone dialling

Default is: ON

4.4.9. 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.

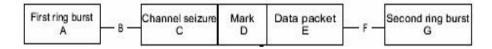
You can enable Metering polarity Reversal on each FXS port and Polarity Reversal will occur every 60 seconds (60 seconds is default). It can be adjusted to a lesser or greater time interval.

4.5. Configuring Caller ID

4.5.1. Caller ID Overview

Caller ID is the feature of the 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.

Fig: Caller ID Type I transmission flow without line reversal



4.5.2. Caller ID Data

Sets the Caller ID information for calls originating from the individual FXS port

4.5.2.1. Date and Time

Allows setting of Caller ID date and time.

4.5.2.2 Calling Number

A custom number can be set, one of the ports numbers, or absent.

4.5.2.3 Calling Name

A custom name or absent can be set.

4.5.3 Caller ID Localisation

Allows customisation of Caller ID to various International Standards. Supported localisations are US, UK FSK with line reversal, ETSI FSK, and ETSI DTMF.

4.5.3.1 Advanced Caller ID Settings

Depending on the localisation selected, various parameters are available for customisation.

4.5.3.2 Type

Only Type 1 on hook Caller ID is supported.

4.5.3.3 Modulation

For FSK based Caller ID, the modulation can be set to Bell 202 or V.23.

4.5.3.4 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.

4.5.3.5 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.

4.5 3.6 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.

4.5.3.7 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.

4.5.3.8 Signal Level

Allows the FSK level to be set. Default value is -3 dBm.

4.5.3.9 Number of Seizure bits

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.

4.5.3.10 Number of Mark bits

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 200 bits.

4.5.3.11 UK Caller ID

Defines delay after line reversal and alert duration.

4.6 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

4.7 MWI. Message Waiting Indicator

PSTN Simulator provides three types of Message Waiting Indicators

- a. FSK signal based (V.23 or Bell)
- 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.

4.8 Electrical parameters

Impedance: 600 and 800 Ohm

Off hook loop DC current: 10, 20 and 32mA. Default 32mA

Ring Frequency: 16, 20, 25, 35, 50 65, 75Hz. Default 25Hz

Ring Voltage: 30, 35, 40, 45, 50, 55 V (RMS). Default 55V (RMS)

5. Chapter 5: 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.

To determine the current version of FXS 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"

6. Chapter 6: PSTN Simulator Technical Specifications

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: 10, 20, 35 mA

Battery Source: 48 VDC

Ringing Source: 16, 20, 25, 35, 50 65, 75Hz. Default 25Hz

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: 16, 20, 25, 35, 50 65, 75Hz. Default 25Hz
Impedance: 600 Ohm + 2.2 uf or 800 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 incrementsCadence: 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:programmableMinimum inter-digit time:programmableRing 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 dBNegative twist accept: < 8 dBMin tone accept duration: 40 msMax tone reject duration: 25 msMin inter-digit pause duration: 40 msMax tone drop-out duration: 20 ms

Signaling velocity: >= 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: 80 ms (default)

Range: 70 - 90 ms in 5 ms increments

Wait for ACK: 160 ms (default)

Range: 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: $\pm 1 \text{ dB}$

Satellite Delay

0 - 1000 ms in 1 ms increments

White Noise

Range: 300 - 3400 Hz

Level Accuracy $\pm 2 \text{ dB}$

7. Chapter 7: Glossary of Terms

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.

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.

DTMF

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

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 analogue 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.

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.

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 signalling 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 signalling 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.

Pav Tone

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

PBX (Private Branch Exchange)

A private telephone exchange.

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

Tone

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

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