

# SIP VoIP Adaptor

## User Guide

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## Revision Notice

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# 1. Introduction

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The SIP VoIP Adaptor, commonly known as Analog Telephony Adaptor (ATA), is a Public Switched Telephone Network (PSTN) handset-to-Ethernet adaptor software package that enables traditional telephone devices to operate as Internet Protocol (IP) devices. It typically serves as the interface between the PSTN world and the IP world acting as a residential gateway, eventually providing Internet Telephony capability. Cost savings and ease of developing and integrating new services motivate most of the interest in Internet telephony. Internet telephony integrates a variety of services provided by the current Internet and the PSTN infrastructure. Internet telephony employs a variety of protocols, including Real-time Transport Protocol (RTP) for transport of multimedia data and Session Initiation Protocol (SIP) for call signaling and control.

The Conexant® ATA extracts the maximum benefit from Voice over IP (VoIP) and Fax over IP (FoIP) technologies. It interfaces analog telephones with IP-based telephony networks. A typical usage/deployment scenario would be to use the ATA as a separate box behind a home-broadband modem, or integrated into such as a broadband modem.

## 1.1 Scope

This manual describes the configuration and operation of the Conexant ATA software installed in flash memory on a Conexant reference design board along with a Conexant network processor and Conexant VoIP coprocessor devices. The reference board is connected to an analog telephone phone, to the Internet over a wide area network (WAN) connection, and to a host PC over a local area network (LAN) connection.

## 1.2 Audience

The target audience of this document is Conexant's customers that use the Conexant VoIP family of products with the SIP ATA adaptor. It assumes the reader has a basic knowledge of ADSL, networking, and telephony.

## 1.3 Reference Documents

1. CX90412/CX90411/CX90410 Voice Coprocessor (VCP) (Doc. No. 102503)
2. CXP9468/CXP9168 Voice Coprocessor (VCP) Data Sheet (Doc. No. 102090)
3. CX82310 ADSL Bridge/Router Data Sheet (Doc. No. 101719)
4. CX82100 Home Network Processor Data Sheet (Doc. No. 101306)
5. CX8620x Network Processor Data Sheet (Doc. No. 102288)
6. CX82310 ADSL Bridge/Router Firmware Web Interface User Guide (Doc. No. 101690)
7. RFC 3261 – SIP: Session Initiation Protocol

## 1.4 Supported Reference Boards

1. 1 FXS VoIP Router Reference Board with CX82100 Network Processor and CX90410 Voice Coprocessor (TR09-D510-011)
2. 2 FXS + FXO VoIP Wireless Router Reference Board with CX86202 Network Processor and CX90412 Voice Coprocessor (RD02-D320-001)
3. 2 FXS + FXO VoIP Router Reference Board with CX82100 Network Processor and CX90412 Voice Coprocessor (RD02-D380-001)

## 1.5 SIP ATA Key Features

- Supports SIP 3261 (RADVISION SIP stack), TCP/UDP/IP, RTP/RTCP, HTTP, ICMP, ARP, DNS, and DHCP.
- Interoperable with various third party SIP end user devices, e.g., Proxy/Registrar/Server, and gateway products.
- Supports popular voice-coders including G.729, and G.711 (A-law and  $\mu$ -law).
- Supports standard voice features such as Caller ID, Consultation Hold, Call Waiting, Blind Call Transfer, Attended Call Transfer, Call Forwarding, Mid-Call DTMF (only using RTP) and 3-Way Conferencing .
- Supports Silence Suppression, VAD (Voice Activity Detection), CNG (Comfort Noise Generation), and Line Echo Cancellation.
- Supports DIGEST authentication (MD5 only).
- Provides easy configuration through a web interface and limited configuration through the handset dial-pad. Configuration can also be done using update.reg.
- Standards-based implementation (ITU-T, IETF compliant).
- PSTN Backup support through Hardware Relay—applicable when power is down or when WAN interface is down.
- FXO Support which enables the user to make PSTN calls while registered with a VoIP service provider.
- STUN support for NAT traversal.
- Distinctive dial tones (PSTN, IP Dialing, and Normal) are played according to the PSTN/WAN/Registration status.
- Special stutter dial tone (VMI Dial Tone) is provided to indicate voice mail received.



## 2. VoIP Configuration

---

The Conexant ATA is configured using the web interface.

The **ATA Configuration** page can be reached as follows:

- Open your web browser.
- If the PC is connected to the LAN interface, type <http://10.0.0.2/> and press the **Enter** key to get the router **Home Page**.
- On the router **Home Page**, click the **VoIP** link on the left frame to view the **ATA Configuration** page.

In general, configuration changes made using the web interface will be activated only upon clicking Save & Reboot button on the Save Savings / Reboot page.

*Note: Certain Voice Parameters do not require a **Save & Reboot** to take effect. These Voice Parameters will take effect on the next voice call after the Voice Parameter is entered and submitted. If **Save & Reboot** is **not** done, then these Voice Parameters will not be saved over a power cycle. The Voice Parameters that can be changed “on the fly” are noted in the respective sections.*

## 2.1 ATA Configuration Page

The **ATA Configuration** page sets parameters for the VoIP application.

**ATA Configuration**

PTM Software Version: 2.36.11 built on Jul 20 2005, 11:41:02

**Line Based Config**

Line to Configure :

Enable SIP Registration: ☒ Yes ☐ No

Transport Type:

Service Provider To Use:  [Update Service Provider](#)

Login Account To Use:  [Update User Login Account](#)

Current Registration Status: **REGISTERED**

Current Voice Call Status: **NO VOICE CALL IN PROGRESS**

[Timer Parameters](#)

[Call Feature Parameters](#)

[Address Book Configuration](#)

[Advanced Telephony Settings](#)

Settings need to be saved to Flash and the system needs to be rebooted for the changes to take effect.

[Save Configuration](#)

**Non-Line Config**

[Country Specific Ring & Tone Parameters](#)

[General Parameters](#)

[Advanced System Telephony Parameters](#)

[FXO Parameters](#)

**Navigation Links:**

- Status**
  - Home
  - PPP
  - ADSL
- Configuration**
  - WAN
  - LAN
  - VoIP
  - PPP
  - NAT
  - Virtual Server
  - Bridge Filtering
  - DNS
  - Dynamic DNS
  - User Level Username/Password
  - Save Settings / Reboot
- Admin Privilege**
  - WAN Status
  - ATM Status
  - ADSL Configuration
  - Route Table
  - Learned MAC Table
  - RIP Configuration
  - Misc Configuration
  - TCP Status
  - Admin Level
  - Username/Password
  - Reset to Factory Default
  - Diagnostic Test
  - System Log
  - Local Code Image Update
  - Network Code Image Update
  - Firmware
  - Boot Code

The **ATA Configuration** page is divided into three general categories: **Version Details**, **Line Based Config**, and **Non-Line Config**.

**Version Details:** This section displays the current versions of the ATAA and PTM software.

**Line Based Config:** This section configures parameters for the selected line.

- **Line to Configure:** Select **0** (default) to configure Line 1 or select **1** to configure Line 2. When a line is selected, the other fields in this section are refreshed with the currently configured values. For example, if 0 (Line 1) is selected, the web page is refreshed to reflect the current value for **Enable SIP Registration**, **Service Provider To Use**, **Login Account To Use**, and the **Current Registration Status**.
- **Enable SIP Registration:** Click **Yes** to enable SIP Registration (default) or click **No** to disable SIP Registration for the selected line.

*Note: Disabling SIP Registration means incoming calls cannot be received and outgoing calls through the proxy also cannot be made; this is done to enable back-to-back direct User Agent (UA) calling using the SIP directory in the sipcfg.ini. Refer to Appendix C.1 for details on how to configure sipcfg.ini.*

- **Transport Type:** Select the transport to use for SIP signaling, UDP or TCP.
- **Service Provider To Use:** Select the service provider to work with the ATA for the selected line. Different service provider-specific details can be configured by clicking the **Update Service Provider** link to display the **SIP Service Provider** configuration page and by following the instructions in Section 2.1.3.

*Note: When a different service provider is chosen from the drop-down list, the **Login Account To Use** drop-down list is updated to reflect the login details available and configured for the selected service provider.*

- **Login Account To Use:** Select the Login Account to work with the ATA for the selected line and the selected service provider. Different Login Account details can be configured by clicking the **Update User Login Account** link to display the **User Login Account** page and by following the instructions in Section 2.1.4. This permits multiple logins to be created for each service provider.
- **Current Registration Status:** This field displays the current registration status of the selected line as defined in the following table:

Current Registration Status Shown	Condition
VOIP SERVICES DISABLED	SIP is disabled
REGISTRATION DISABLED	Registration is disabled on web page
REGISTERING.....	WAN is up and Registration is being attempted (assuming Registration is enabled)
REGISTERED	Registration succeeded
REGISTRATION FAILED	Registration failed
UNREGISTERED	Board is powered on, but WAN is still down.

- **Current Voice Call Status:** This field displays the current registration status of the selected line as defined in the following table:

Current Voice Call Status Shown	Condition
NO VOICE CALL IN PROGRESS	No voice calls on the selected endpoint currently
VOICE CALL IN PROGRESS... CODEC USED IS G711U	Voice call in progress with PCMU or G711U as the selected codec
VOICE CALL IN PROGRESS... CODEC USED IS G711A	Voice call in progress with PCMA or G711A as the selected codec
VOICE CALL IN PROGRESS... CODEC USED IS G729	Voice call in progress with G729 as the selected codec
VOICE CALL HELD BY REMOTE END	Remote endpoint has held the voice call
VOICE CALL ON HOLD BY LOCAL END	Local endpoint has put the voice call on hold

- **Timer Parameters:** Click this link to open the **Timer Parameters** page to configure timer values for the selected line. See configuration details in Section 2.1.5.
- **Call Feature Parameters:** Click this link to open the **Call Feature** page to configure parameters related to call features for the selected line. See configuration details in Section 2.1.6.
- **Address Book Configuration:** Click this link to open the Address Book Configuration web page to configure address book entries. See configuration details in Section 2.1.7.
- **Advanced Telephony Settings:** Click this link to open the **Advanced Telephony Settings** page to configure advanced telephony parameters for the selected line. See configuration details in Section [2.1.8](#)
- **Submit Changes:** Click **Submit Changes** to save the configured settings in this section to system RAM for the selected line.

*Note:* After clicking **Submit Changes** to save section settings to system RAM, you must permanently save the configuration and reinitialize the system as follows.

- **Save Configuration:** Click this link to go to the **Save Settings / Reboot** page. On the **Save Settings / Reboot** page, click **Save & Reboot** to permanently save the settings to system flash memory and to reinitialize the system to the new settings for the selected line.

**Non-Line Config:** This section configures system-level parameters independently of the selected line.

- **Country Specific Ring & Tone Parameters:** Click this link to display the **Country Specific Ring & Tones** configuration page to define parameters for various tones. See configuration details in Section 2.1.1.
- **General Parameters:** Click this link to display the **General** configuration page to configure non-line system-level parameters. See configuration details in Section 2.1.2.
- **Advanced System Telephony Parameters:** Click this link to display the web page for configuring system level telephony settings. See configuration details in Section [2.1.9](#).
- **FXO Parameters:** Click this link to display the **FXO Configuration** page to configure FXO settings. See configuration details in Section [2.1.10](#).

### 2.1.1 Country Specific Ring & Tones Configuration

The **Country Specific Ring & Tones** configuration page defines parameters for the various tones (ring, dial, busy, ringback, etc.) generated by the ATA application.

ATA provides default ring and tone parameters configured for operation in the USA. Flexibility is provided to change the existing ring-tone parameters, and to add new countries as well as to edit/delete existing countries.

**Country Specific Ring & Tones**

Working Country:

New Country:

**Ring Parameters**

Ring:

**Tone Parameters**

PSTN Dial Tone: <input type="text" value="454,-200,554,-200"/>	Peer2Peer Dial Tone: <input type="text" value="300,-200,500,-200"/>
Normal Dial Tone: <input type="text" value="350,-200,440,-200"/>	Busy Tone: <input type="text" value="480,-200,620,-200"/>
RingBack Tone: <input type="text" value="440,-200,480,-200"/>	Call Waiting Tone: <input type="text" value="440,-200,440,-200"/>
Alerting Tone: <input type="text" value="350,-200,0,0,100,-"/>	Congestion Tone: <input type="text" value="480,-200,620,-200"/>
Fast Busy Tone: <input type="text" value="480,-200,620,-200"/>	Confirm Tone: <input type="text" value="800,-200,0,0,500,-"/>
Warble Tone: <input type="text" value="1400,-120,2040,-1"/>	Unobtainable Tone: <input type="text" value="480,-200,620,-200"/>
Recall Tone: <input type="text" value="350,-200,440,-200"/>	Stutter Dial Tone: <input type="text" value="350,-200,440,-200"/>
VMI Dial Tone: <input type="text" value="650,-200,740,-200"/>	

RingTone Action:  Country

Note: Settings need to be saved to Flash and the system needs to be rebooted for the changes to take effect.

[Save Configuration](#)

**Working Country:** Select the country for which the ring-tone parameters will apply. The default country is USA. When a country is selected from this drop-down list, the ring-tone parameters entered for that country are automatically displayed.

The maximum number of countries that can be configured is 32.

New countries can be defined and added manually.

**New Country:** Enter the name of the country to be added if the field is blank, or change the name displayed in the **Working Country** field as desired.

*Notes:* 1. For the following tone parameters, time is specified in milliseconds and frequency is specified in Hertz.

2. Amplitude values, specified in dBm multiplied by a factor of 10.

### Ring Parameters

**Ring:** Enter five consecutive fields separated by commas: **Frequency**, **OnTime1**, **OffTime1**, **OnTime2**, and **OffTime2**.

### Tone Parameters

**PSTN Dial Tone:** Enter six consecutive fields separated by commas: **Freq1**, **Amp1**, **Freq2**, **Amp2**, **OnTime1**, **OffTime1**, **OnTime2**, and **OffTime2**. The PSTN is used when WAN is disconnected or not-operative.

**Peer2Peer (IP Dialing) Dial Tone:** Enter six consecutive fields separated by commas: **Freq1**, **Amp1**, **Freq2**, **Amp2**, **OnTime1**, **OffTime1**, **OnTime2**, and **OffTime2**. These parameters are used when Registration is disabled.

**Normal Dial Tone:** Enter six consecutive fields separated by commas: **Freq1**, **Amp1**, **Freq2**, **Amp2**, **OnTime1**, **OffTime1**, **OnTime2**, and **OffTime2**. These parameters are used when Registration is enabled and up.

**Busy Tone:** Enter six consecutive fields separated by commas: **Freq1**, **Amp1**, **Freq2**, **Amp2**, **OnTime1**, **OffTime1**, **OnTime2**, and **OffTime2**.

**RingBack Tone:** Enter six consecutive fields separated by commas: **Freq1**, **Amp1**, **Freq2**, **Amp2**, **OnTime1**, **OffTime1**, **OnTime2**, and **OffTime2**.

**Call Waiting Tone:** Enter six consecutive fields separated by commas: **Freq1**, **Amp1**, **Freq2**, **Amp2**, **OnTime1**, **OffTime1**, **OnTime2**, and **OffTime2**.

**Alerting Tone:** Enter six consecutive fields separated by commas: **Freq1**, **Amp1**, **Freq2**, **Amp2**, **OnTime1**, **OffTime1**, **OnTime2**, and **OffTime2**.

**Congestion Tone:** Enter six consecutive fields separated by commas: **Freq1**, **Amp1**, **Freq2**, **Amp2**, **OnTime1**, **OffTime1**, **OnTime2**, and **OffTime2**.

**Fast Busy Tone:** Enter six consecutive fields separated by commas: **Freq1**, **Amp1**, **Freq2**, **Amp2**, **OnTime1**, **OffTime1**, **OnTime2**, and **OffTime2**.

**Confirm Tone:** Enter six consecutive fields separated by commas: **Freq1**, **Amp1**, **Freq2**, **Amp2**, **OnTime1**, **OffTime1**, **OnTime2**, and **OffTime2**.

**Warble Tone:** Enter six consecutive fields separated by commas: **Freq1**, **Amp1**, **Freq2**, **Amp2**, **OnTime1**, **OffTime1**, **OnTime2**, and **OffTime2**.

**Unobtainable Tone:** Enter six consecutive fields separated by commas: **Freq1**, **Amp1**, **Freq2**, **Amp2**, **OnTime1**, **OffTime1**, **OnTime2**, and **OffTime2**.

**Recall Tone:** Enter five consecutive fields separated by commas: **Freq1**, **Amp1**, **Freq2**, **Amp2**, and **Stutter Duration**.

**Stutter Dial Tone:** Enter five consecutive fields separated by commas: **Freq1**, **Amp1**, **Freq2**, **Amp2**, and **Stutter Duration**.

**VMI Dial Tone:** Enter five consecutive fields separated by commas: **Freq1, Amp1, Freq2, Amp2, and Stutter Duration**. This plays a distinctive stutter dial tone when Off Hook and there is Voice Mail waiting.

**RingTone Action:** Select the drop-down option (**DISPLAY, ADD, EDIT** or **DELETE**) to display or manipulate the ring-tone parameters for the selected working country.

- **DISPLAY:** Select **DISPLAY** to enable display of the ring-tone parameters when a country is selected in the **Working Country** field. This is the default selection.
- **ADD:** Select **ADD** to add the updated ring-tone parameters for the new country appearing in the **New Country** field, upon clicking **Submit Changes**. The **New Country** field must not be empty. After **Submit Changes** is clicked, **DISPLAY** will be displayed.
- **EDIT:** Select **EDIT** to overwrite the ring-tone parameters for the selected country in **Working Country** field with the updated ring-tone parameters and to overwrite the name in **Working Country** field if a name has been entered in the **New Country** field, upon clicking **Submit Changes**. After **Submit Changes** is clicked, **DISPLAY** will be displayed.
- **DELETE:** Select **DELETE** to delete the selected **Working Country** from the country list, upon clicking **Submit Changes**. After **Submit Changes** is clicked, **DISPLAY** will be displayed.

*Note: All Ring and Tone parameters, edited for the current chosen country, do not require a Save & Reboot to take effect. Selection of a different country does not take effect immediately and will need a save and reboot.*

*Note: See Appendix D for ring tone descriptions.*

**Submit Changes:** Click **Submit Changes** to save the settings on this page to system RAM.

*Note: After clicking **Submit Changes** to save page settings to system RAM, you must permanently save the configuration and reinitialize the system as follows.*

**Save Configuration:** Click this link to go to the **Save Settings / Reboot** page. On the **Save Settings / Reboot** page, click **Save & Reboot** to permanently save the settings to system flash memory and to reinitialize the system to the new settings.



## 2.1.2 General Configuration Page

The **General** configuration page configures system-level parameters not related to the selected line. This has five sections: **SIP Device**, **VoIP General**, **BLAM Server**, **STUN Parameters**, and **Default Dial Plan Parameters**.

**General**

**SIP Device**

Enable SIP: ☒ Yes ☐ No

Local SIP Port:  Media Base Port:

**VoIP General**

Enable Auto Login: ☐ Yes ☒ No User Agent String:

Append UserId: ☐ Yes ☒ No Pin Number:

Interface To Use:

**BLAM Server**

RTrace fifo size:  KB RDump fifo size:  KB

Port:

Blam Current State - **not running**

**NAT Traversal**

NAT Traversal Technique:

STUN Server:  STUN Port:

Force Keep Alive: ☐ Yes ☒ No Keep Alive Period:

**Default Dial Plan**

Default Dial Plan String:

Settings need to be saved to Flash and the system needs to be rebooted for the changes to take effect.

[Save Configuration](#)

**SIP Device:** This section configures the following information:

- **Enable SIP:** Click **Yes** to specify SIP is to be used for signaling (default), or **No** to specify SIP is not to be used for signaling.
- **Local SIP Port:** Enter the local SIP Port number on which ATA should listen for messages. The range is 1 to 65535. The default port is 5060.
- **Media Base Port:** Enter the Media Base Port (also known as RTP port) number. This parameter provides the base value from the media (RTP) ports that are assigned for various lines and the different call-sessions that may exist within an end-point. Odd port values are not recommended. If an Odd Value is entered, the next higher even value is used as the Media Base Port. This is to conform to the RFC specifications. The range is 1 to 65500. The default port is 5000.

**VoIP General:** This section configures the following information:

- **Enable Auto Login:** If enabled, the system will obtain the login information from the service provider using the MAC address.  
*Note: When enabled and successful in obtaining the login information, no VoIP web pages will be displayed.*
- **User Agent string:** Enter the User Agent string to be sent in the User Agent/Server Header of the SIP requests. Usually customer software version is provided.
- **Append User ID:** Enable this to append the local registration number or PIN Number entered in the following field to the TO header in the SIP requests. If PIN Number field is left empty, the User ID (see Section 2.1.4) is appended to the TO Header. If filled, the PIN Number is appended to the TO Header.
- **Pin Number:** Enter PIN number to be appended to the TO Header after enabling Append User ID field.
- **Interface To Use:** Select the VoIP interface to be used for calls. The default is the first available and enabled WAN interface.

*Note: The user can make calls only when the specified or default (in case specified interface is not available) interface has come up successfully.*

For a non-ADSL system, the available interface can be Ethernet 1. Ethernet 1 is the default MAC interface available on the 1-Line Reference Voice system.

For an ADSL system, the available interfaces can be Pvc 0, Pvc 1, etc. Any of the available PVCs can be chosen from the drop down list. (PVC refers to the Permanent Virtual Circuit typically available on an ADSL+Voice systems.)

**BLAM Server:** This section allows the user to start and stop the BLAM server. The fields are **RTrace fifo size**, **RDump fifo size**, and **Port** number.

To start the BLAM server, enter a valid set of values into the **RTrace fifo size**, **RDump fifo size**, and **Port** fields, then click **Submit Changes**.

To stop the BLAM server, set both the **RTrace fifo size** and **RDump fifo size** values to 0, then click **Submit Changes**.

- **RTrace fifo size:** Enter the Rtrace FIFO size in KB. A non-zero value is required for starting the BLAM server. The maximum value is 100. Enter 0 for stopping the BLAM server.
- **RDump fifo size:** Enter the RDump FIFO size in KB. A non-zero value is required for starting the BLAM server. The maximum value is 100. Enter 0 for stopping the BLAM server.
- **Port:** Enter the port number on which the BLAM server listens for connection from the BLAMski application. The range is 1 to 32767.
- **BLAM Current State:** This display-only field indicates the current state of the BLAM server. This status is refreshed upon clicking **Submit Changes**.

**NAT Traversal Parameters:** This section configures for the NAT Traversal technique support in ATA.

- **NAT Traversal Technique:** Select **USE STUN** to enable STUN (default) if the ATA is behind a NAT enabled router and the router has no ALG for SIP, or **NONE** to disable STUN (ATA is not to use STUN for NAT traversal). ATA also supports a proprietary implementation of NAT traversal where the Service

provider is expected to provide some relay support. If **NONE** is selected, then based on the responses received, the ATA will dynamically determine if the SIP Server supports the proprietary implementation.

*Note: Even when STUN is enabled, the ATA does an automatic detection of the presence of SIP ALG and disables the use of STUN. This is to avoid some media problems arising out of the behavior of some ALGs when STUN is used at the user end.*

- **STUN Server:** Enter the IP address or Domain Name of the STUN Server. The default is 66.7.238.210. This field is applicable only if USE STUN is selected as the NAT traversal technique.
- **STUN Port:** Enter the port number on which the STUN server listens for requests from the STUN Client on ATA. The range is 1 to 65535. The default is 3478. This field is applicable only if USE STUN is selected as the NAT traversal technique.
- **Force Keep Alive:** Only valid when STUN is not used. If STUN is not enabled, and keep alive is still expected to be sent then select **Yes** otherwise select **No**.
- **Keep Alive Period:** The keep alive interval in seconds to be used when STUN is not enabled.

**Default Dial Plan Parameters:** This field provides the default dial plan string that can be configured for the system (see Section 2.2.4 for more details on the default value for this field). Edit the dial plan as required.

When a new service provider is added, the initial dial plan string for the service provider is taken from this default dial plan string. If this field is left empty, the system attaches an empty dial plan string to the new service provider, in which case it applies the default rule upon save and reboot (see Section 2.2.4). Refer to Section 2.2 for more details on how to build the dial plan string.

**Submit Changes:** Click **Submit Changes** to save the settings on this page to system RAM.

*Note: After clicking **Submit Changes** to save page settings to system RAM, you must permanently save the configuration and reinitialize the system as follows.*

*Note: For **BLAM Server**, click **Submit Changes** to start or stop the BLAM server, based on the entered values.*

**Save Configuration:** Click this link to go to the **Save Settings / Reboot** page. On the **Save Settings / Reboot** page, click **Save & Reboot** to permanently save the settings to system flash memory and to reinitialize the system to the new settings.

### 2.1.3 SIP Service Provider Configuration Page

The **SIP Service Provider** configuration page sets the configuration related to the SIP service provider.

**SIP Service Provider**

Service Providers List:

New Service Provider:

Registration Interval (in secs):  Authentication Method:

Registrar Address:  Registrar Port:

Proxy Address:  Proxy Port:

OutboundProxy Address:  OutboundProxy Port:

Dial Plan String:

Service Provider Action:

Note: Save changes to flash to restore on power up.

[Save Configuration](#)

**Service Provider List:** Enter the name of the service provider to be configured. When a service provider is selected from this drop-down list, the respective parameters are automatically displayed.

A DEFAULT service provider is provided with a default set of parameters. This can be edited. New service providers can be manually defined and added. An existing service provider can be edited or deleted.

**New Service Provider:** Enter the name of the service provider to be added if the **Service Provider List** field is blank, or a new string to rename the service provider displayed in the **Service Provider List** field.

**Registration Interval (in secs):** Enter the re-registration interval in seconds. The range is 0 to 2147483347 seconds. The default is 3600 seconds.

**Authentication Method:** Select the authentication method. Only MD5 is supported.

- **AUTH\_NONE:** Disable any authentication method.
- **AUTH\_MD5:** Use MD5 authentication method.

**Registrar Address:** Enter the IP address or Domain Name of the registrar with which the ATA must register in order to receive or send calls.

**Registrar Port:** Enter the port number of the registrar on which it will listen for Register requests from the ATA. The range is 1 to 65535. The default port is 5060.

**Proxy Address:** Enter the IP address or Domain Name of the SIP proxy server.

**Proxy Port:** Enter the port number on which the SIP proxy server will listen for messages. The range is 1 to 65535. The default port is 5060.

**OutboundProxy Address:** Enter the IP address or Domain Name of the Outbound proxy server. This is useful in cases where the ATA is behind a NAT.

**OutboundProxy Port:** Enter the port number on which the Outbound proxy server listens for messages from the ATA. The range is 1 to 65535. The default port is 5060.

*Note: Refer to RFC 3261 [2] for more SIP definitions.*

**Dial Plan String:** This parameter provides the dial plan string as required by the service provider. Modifying this field while adding a new service provider **will not** take effect after **ADD** has been selected and **Submit Changes** has been clicked. While adding a new service provider, the dial plan string takes the value from the default dial plan string specified in the **VoIP General Parameters** web page only. To modify this field, complete adding the service provider, and then edit it, select **EDIT** in the drop-down box of **Service Provider Action** and click **Submit Changes**. *Please refer to Section 2.2 for more details on how to build the dial plan string.*

*Note: Dial Plan String of the current selected service provider does not require a Save & Reboot to take effect.*

**Service Provider Action:** Select the drop-down option (**DISPLAY SP RULES**, **ADD NEW SP**, **DELETE SEL SP**, or **EDIT SEL SP**) to display and manipulate the SIP and dial plan parameters for the service provider selected in the **Service Provider List**.

- **DISPLAY SP RULES:** Select **DISPLAY SP RULES** to enable display of the selected service provider parameters when a service provider is selected in the **Service Provider List** field. This is the default selection.
- **ADD NEW SP:** Select **ADD NEW SP** to add a new service provider after clicking **Submit Changes** according to the value that appears in the **New Service Provider** field. This field must not be empty.

*Note: The maximum number of service providers is 4.*

- **DELETE SEL SP:** Select **DELETE SEL SP** to delete the selected service provider from the **Service Provider List**.
- **EDIT SEL SP:** Select **EDIT SEL SP** to overwrite the selected service provider's (in the **Service Provider List** field) parameters with the current parameters displayed on the web page. The **New Service Provider** field is optional and needs to be filled only when the service provider name also has to be changed.

**Submit Changes:** Click **Submit Changes** to save the settings on this page to system RAM.

*Note:* After clicking **Submit Changes** to save page settings to system RAM, you must permanently save the configuration and reinitialize the system as follows.

**Save Configuration:** Click this link to go to the **Save Settings and Reboot** page. On the **Save Settings and Reboot** page, click **Save & Reboot** to permanently save the settings to system flash memory and to reinitialize the system to the new settings.

## 2.1.4 User Login Account Configuration Page

The **User Login Account** configuration page sets and configures login accounts for the service provider chosen in the index web page, i.e., for the currently selected service provider in the main web page.

**User Login Account**

Service Provider Name: **DEFAULT**

Login Account List: **DEFAULT**

New Account Name:

User ID:  Password:

Auth User ID:  Display Name:

Login Action: **DISPLAY** ☐ User

Note: Settings need to be saved to Flash and the system needs to be re-booted for the changes to take effect.

[Save Configuration](#)

**Service Provider Name:** This field displays the service provider for which the login information is being configured. The service provider is selected on the **ATA Configuration** page.

**Login Account List:** Select the login account to be configured. When a login account is selected from this drop-down list, the respective parameters are automatically displayed. A default set of parameters is provided for every new login account added. These parameters can be edited. New login accounts can be defined and added. An existing login account can also be edited or deleted.

**New Account Name:** Enter the name of the new login account to be added, or a new string to rename an existing login account. The login account edited will be the one selected from the **Login Account List** field.

**User ID:** Enter the registration ID of the user with the registrar.

**Password:** Enter the password used for authentication with the registrar.

**Auth User ID:** Enter the Authorization User ID for authentication with the registrar. If not specified explicitly by the service provider, this is same as the User ID.

**Display Name:** Enter the Display Name as it should appear on Caller ID.

*Note:* **Display Name** of the current selected login does not require a **Save & Reboot** to take effect.

**Login Action:** Select a drop-down option (**DISPLAY**, **ADD**, **EDIT**, or **DELETE**) to manipulate the various login parameters for the login account selected in the **Login Account List**.

- **DISPLAY:** Select **DISPLAY** for the selected login account details to be displayed after clicking **Submit Changes**. This is the default selection.
- **ADD:** Select **ADD** to add a new login account after clicking **Submit Changes** according to the value that appears in the **New Account Name** field. This field must not be empty.
- **EDIT:** Select **EDIT** to overwrite the selected login account (in the **Login Account List** field) parameters with the displayed parameters. The **New Account Name** field is optional and needs to be filled only when the login account name is to be changed.
- **DELETE:** Select **DELETE** to delete the selected login account from the **Login Account List**.

*Note:* The above parameters are specific to the service provider selected in the **ATA Configuration** page.

*Note:* Up to four login accounts can be added per service provider.

**Submit Changes:** Click **Submit Changes** to save the settings on this page to system RAM.

*Note:* After clicking **Submit Changes** to save page settings to system RAM, you must permanently save the configuration and reinitialize the system as follows.

**Save Configuration:** Click this link to go to the **Save Settings and Reboot** page. On the **Save Settings and Reboot** page, click **Save & Reboot** to permanently save the settings to system flash memory and to reinitialize the system to the new settings.



## 2.1.5 Timer Parameters Configuration Page

The **Timer Parameters** configuration page displays and configures timers used at the system level. This section explains the various timers available for configuration. These timers are not applicable for PSTN calls.

**Timer Parameters**

Selected EndPoint = 0

Predial Timeout (Secs):	<input type="text" value="16"/>	Call Progress Timeout (Secs):	<input type="text" value="60"/>
Alert Timeout (Secs):	<input type="text" value="60"/>	Call Waiting Timeout (Secs):	<input type="text" value="40"/>
Disconnect Timeout (Secs):	<input type="text" value="10"/>	Call Forward No Ans Timeout (Secs):	<input type="text" value="30"/>
RingBack Timeout (Secs):	<input type="text" value="60"/>		

Note: Settings need to be saved to Flash and the system needs to be rebooted for the changes to take effect.

[Save Configuration](#)

**Left Sidebar Navigation:**

- Status**
  - Home
  - PPP
  - ADSL
- Configuration**
  - WAN
  - LAN
  - VoIP
  - PPP
  - NAT
  - Virtual Server
  - Bridge Filtering
  - DNS
  - Dynamic DNS
  - User Level Username/Password
  - Save Settings / Reboot
- Admin Privilege**
  - WAN Status
  - ATM Status
  - ADSL Configuration
  - Route Table
  - Learned MAC Table
  - RIP Configuration
  - Misc Configuration
  - TCP Status
  - Admin Level
  - Username/Password
  - Reset to Factory Default
  - Diagnostic Test
  - System Log
  - Local Code Image Update
  - Network Code Image Update
  - Firmware
  - Boot Code

**Selected Endpoint:** The endpoint for which these timer parameters are applicable. This corresponds to the current selected endpoint on the ATA Configuration web page.

**Predial Timeout (Sec):** Enter the length of time in seconds the dial tone will be generated once the phone has been lifted off hook. At the end of this period, if no digits have been pressed, the ATA will start playing the Fast-Busy tone.

**Call Progress Timeout (Sec):** Enter the length of time in seconds the ATA will wait for the initial response from the other end point once an outgoing call has been made.

**Alert Timeout (Sec):** Enter the length of time in seconds the ATA will play the Ring tone when an incoming call has arrived and the phone is on-hook. At the end of this period, the ATA will automatically stop the ring and reject the call.

**Call Waiting Timeout (Sec):** Enter the length of time in seconds the Call Waiting tone will be played when an incoming call arrives in the connected/held state. The Call

Waiting tone is played at an interval of 10 sec. for USA. It is configurable using the Call Waiting tone parameters.

**Disconnect Timeout (Sec):** Enter the length of time in seconds the Fast-Busy tone will be played once a call has been disconnected by the remote-end. At the end of this period, the Warble tone will be played until the user hangs up the phone.

**Call Forward No Ans Timeout (Sec):** Enter the length of time in seconds after which the call will be forwarded when it is not answered. This timer is applicable when **Call Forwarding on No Answer** is enabled from either the **Call Feature** configuration page Section 2.1.6) configuration page or the dial-pad.

**RingBack Timeout (Sec):** Enter the length of time in seconds ATA will wait while the RingBack tone is being played for the final response from the other end point once an outgoing call has been made and the initial response has been received.

*Note: All Timeout parameters do not require a Save & Reboot to take effect.*

**Submit Changes:** Click **Submit Changes** to save the settings on this page to system RAM.

*Note: After clicking **Submit Changes** to save page settings to system RAM, you must permanently save the configuration and reinitialize the system as follows.*

**Save Configuration:** Click this link to go to the **Save Settings and Reboot** page. On the **Save Settings and Reboot** page, click **Save & Reboot** to permanently save the settings to system flash memory and to reinitialize the system to the new settings.

## 2.1.6 Call Feature Configuration Page

The **Call Feature** configuration page displays and modifies call features. Enabling any of these options allows the user to apply the appropriate configuration as specified in the dial plan by using the dial-pad.

**Selected Endpoint:** The endpoint for which these call features parameters are applicable. This corresponds to the current selected endpoint on the ATA Configuration web page.

### Call Forwarding:

- **Enable Call Forwarding Unconditionally:** Click this check box to enable Unconditional Call Forwarding. When enabled, an incoming call will be forwarded to the entered Call Forwarding Number when using the dial-pad as described in Section 3.3.8.
- **Enable Call Forwarding On Busy:** Click this check box to enable Call Forwarding on Busy. When enabled and the telephone is connected, an incoming call will be forwarded to the entered Call Forwarding Number when using the dial-pad as described in Section 3.3.8.

- **Enable Call Forwarding On No Ans:** Click this check box to enable Call Forwarding on No Answer. When enabled and the incoming call is not answered within a time limit, the incoming call will be forwarded to the entered Call Forwarding Number when using the dial-pad as described in Section 3.3.8. The **Call Forward No Ans Timeout** timer is configured as described in Section 2.1.5.
- **Call Forwarding Number:** Enter the telephone number to be used when Call Forwarding is enabled as described above. The user can over-ride this number from the dial-pad if desired as described in Section 3.3.8.

**Call Waiting:** Click **Yes** to enable Call Waiting or **No** to disable Call Waiting.

The user can temporarily disable Call Waiting from the dial-pad (see Section [3.3.7](#)) only when this feature is enabled.

**Call Return:** Click **Yes** to enable Call Return or **No** to disable Call Return.

The user can do Call Return from the dial-pad (Section 3.3.9) only when this feature is enabled.

**Caller ID Transmit Block:** Click **Yes** to allow Caller ID Transmit Blocking or **No** to disable Caller ID Transmit Blocking.

The user can use Caller ID Transmit Block from the dial-pad (see Section 3.3.12) only when this feature is enabled.

**Privacy Display Name:** The name to display when using Caller ID Transmit Blocking.

**Conferencing:** Click **Yes** to enable 3-way conferencing or **No** to disable 3-way conferencing.

The user can temporarily disable 3-way conferencing from the dial-pad (see Section [3.3.6](#)) only when this feature is enabled. When disabled, the user can switch between the calls using hookflash.

**Submit Changes:** Click **Submit Changes** to save the settings on this page to system RAM.

*Note: All Call Feature parameters do not require a **Save & Reboot** to take effect.*

*Note: After clicking **Submit Changes** to save page settings to system RAM, you must permanently save the configuration and reinitialize the system as follows.*

**Save Configuration:** Click this link to go to the **Save Settings and Reboot** page. On the **Save Settings and Reboot** page, click **Save & Reboot** to permanently save the settings to system flash memory and to reinitialize the system to the new settings.

## 2.1.7 Address Book Configuration Page

The Address Book Configuration web page allows for configuration of address book entries which can be used for speed dial execution of calls.

**Address Book**

Selected EndPoint = 0

Address Book Index	Display Name	Number	IP Address/ Domain Name	Port Number
0	Tom	12345	fwd.pulver.com	5060

**Edit Address Book**  
Use this portion to add a new entry or delete or edit an existing entry

Display Name:

Number:

IP Address/Domain name:  Port Number:

Speed Dial Code:

Address Book Action:

Note: Settings will be saved to Flash directly and the system DOES NOT need to be rebooted for the changes to take effect.

The top half of the web page displays the current address book table for the selected Endpoint.

The bottom half of the web page can be used for editing the address book. This includes adding new entries, deleting existing entries, modifying existing entries and displaying the values corresponding to an entry index or speed dial index in the address book.

**Selected Endpoint:** The endpoint for which this address book is applicable. This corresponds to the current selected endpoint on the ATA Configuration web page.

**Address Book Table:** This table displays the current address book as configured for this endpoint.

**Display Name:** Enter the Display Name for this address book entry.

**Number:** The user phone number or name for this entry. This field is optional, if the IP Address is specified. A phone number that can be reached through the current configured proxy server can also be added as an entry in which case the IP address/Domain name and the Port number fields are not necessary.

**IP Address/Domain Name:** Enter the IP address or the domain name that corresponds to this address book entry. If this field is left empty, then the User number or name must be specified, in which case the current configured proxy server for this endpoint will be used as the domain name.

**Port Number:** Specify the SIP port number on which the remote end will receive our call. This is useful when you want to specify a non-phone number entry, where the call can be made directly without going through the configured proxy server. When this field is not specified, the default SIP port of 5060 will be assumed.

**Speed dial code:** This refers to the index in the address book as well as the speed dial entry code. This needs to be specified following \*78 (or the configured speed dial service code – see Section [2.2.2](#)) to dial out the number corresponding to this address book entry.

**Address Book Action:** Select a drop-down option (**DISPLAY**, **ADD**, **EDIT**, or **DELETE**) to manipulate the various address book parameters for the entry index selected from the speed dial code drop down box.

- **DISPLAY:** Select **DISPLAY** for the selected speed dial code details to be displayed after clicking **Submit Changes**. This is the default selection.
- **ADD:** Select **ADD** to add a new address book entry after clicking **Submit Changes**.
- **EDIT:** Select **EDIT** to overwrite the selected address book entry.
- **DELETE:** Select **DELETE** to delete the selected address book entry

*Note:* The above parameters are specific to the endpoint selected in the **ATA Configuration** page.

*Note:* Up to ten address book entries can be added per endpoint.

**Submit Changes:** Click **Submit Changes** to save the settings on this page to system RAM and Flash also.

*Note:* Address book entries addition/deletion/editing **do not** need a Save and Reboot. The changes are reflected immediately.

## 2.1.8 Advanced Telephony Settings Page

The **Advanced Telephony Settings** page allows configuration of the various endpoint level telephony parameters.

**Conexant - Hasbani - Microsoft Internet Explorer**

File Edit View Favorites Tools Help

**Advanced Telephony Settings**

Selected EndPoint = 0

VAD

- ☐ Disable VAD
- ☐ Enable VAD with COMFORT NOISE
- ☒ Enable VAD with STANDARD SID
- ☐ Enable VAD and SUPPRESS SID

---

Echo Cancellation

- ☐ Disable EchoCancellation
- ☐ Enable EchoCancellation
- ☐ Enable EchoCancellation with NLP
- ☒ Enable EchoCancellation with CNG\_NLP

Start Attenuation:

Max Attenuation:

Echo Cancellation Tail Length:  ms

---

DTMF Relay

- ☐ NONE (IN AUDIO)
- ☒ RFC 2833

DTMF Payload Number:

---

Hook Flash Time

HookFlash Max Time (ms):

---

Debounce Times

Debounce On-Off Time (ms):

Debounce Off-On Time (ms):

Done Internet

The screenshot shows the Conexant web interface in a Microsoft Internet Explorer browser window. The left sidebar contains a navigation menu with links for Status, Configuration (WAN, LAN, VoIP, PPP, NAT, Virtual Server, Bridge Filtering, DNS, Dynamic DNS, User Level Username/Password, Save Settings / Reboot), and Admin Privilege (WAN Status, ATM Status, ADSL Configuration, Route Table, Learned MAC Table, RIP Configuration, Misc Configuration, TCP Status, Admin Level, Username/Password, Reset to Factory Default, Diagnostic Test, System Log, Local Code Image Update, Network Code Image Update, Firmware, Boot Code). The main content area is titled 'Echo Cancellation' and contains the following settings:

- Echo Cancellation:**
  - ☐ Disable EchoCancellation
  - ☐ Enable EchoCancellation
  - ☐ Enable EchoCancellation with NLP
  - ☒ Enable EchoCancellation with CNG\_NLP
  - Start Attenuation:
  - Max Attenuation:
  - Echo Cancellation Tail Length:  ms
- DTMF Relay:**
  - ☐ NONE (IN AUDIO)
  - ☒ RFC 2833
  - DTMF Payload Number:
- Hook Flash Time:**
  - HookFlash Max Time (ms):
- Debounce Times:**
  - Debounce On-Off Time (ms):
  - Debounce Off-On Time (ms):
- Codec Preference:**
  - G711U:
  - G711A:
  - G729A:

At the bottom of the form is a 'Submit Changes' button. Below the button, a note states: 'Note: Settings need to be saved to Flash and the system needs to be rebooted for the changes to take effect.' A link for 'Save Configuration' is also present.

**Selected Endpoint:** The endpoint for which these call advanced telephony settings are applicable. This corresponds to the current selected endpoint on the ATA Configuration web page.

**VAD:** This section configures Voice Activity Detection.

- **Disable VAD:** Click this option button to disable VAD, i.e., voice activity will not be detected.
- **Enable VAD with COMFORT NOISE:** Click this option button to enable VAD with comfort noise generation.
- **Enable VAD with STANDARD SID:** Click this option button to enable VAD with SID according to G.729 and PT13 of G.711.
- **Enable VAD and SUPPRESS SID:** Click this option button to enable VAD with discontinuous transmission. When silence is detected, transmission of packets will be paused until voice is again detected.



**Echo Cancellation:** This section configures Echo Cancellation with NLP or with CNG\_NLP options.

- **Disable Echo Cancellation:** Click this button to disable Echo Cancellation.
- **Enable Echo Cancellation:** Click this button to enable Echo Cancellation with no other options.
- **Enable Echo Cancellation with NLP:** Click this button to enable Echo Cancellation using NLP (Non Linear Processor). With this option, the **Start Attenuation** and **Max Attenuation** can be set.
- **Enable Echo Cancellation with CNG\_NLP:** Click this button to enable Echo Cancellation using NLP and CNG (Comfort Noise Generation).
- **Start Attenuation:** Enter the minimum or start attenuation, in dB, for Echo Cancellation when NLP is enabled. The default value is 8192.
- **Max Attenuation:** Enter the maximum converged attenuation, in dB, for Echo Cancellation when NLP is enabled. The default value is 16384.
- **Echo Cancellation Tail Length:** Select the Echo Cancellation Tail Length, in ms. The default value is 24 ms.

**DTMF Relay:** This section selects how packets containing Mid-Call DTMF tones are sent.

- **NONE (IN AUDIO):** Click this button to send Mid-Call DTMF tones in RTP packets with the same payload as voice, i.e., dynamic payload negotiation for DTMF digits will not be done.
- **RFC 2833:** Click this button to send Mid-Call DTMF tones in RTP packets separately using RFC2833, i.e., dynamic negotiation of RTP payload for DTMF digits will be done.
  - **DTMF Payload Number:** This field is configurable when RFC 2833 is selected as the DTMF Relay mechanism. Specify the payload number that needs to be used for DTMF information negotiated in SDP during SIP signaling.

**HookFlash Time:** This section configures the HookFlash time.

- **HookFlash Max Time (ms):** Enter the maximum time, in ms, of Hook Flash. For a HookFlash to be detected, there should be an on-hook occurrence followed by an off-hook occurrence within the entered **HookFlash Max Time** period. If **HookFlash Max Time** expires before an off-hook, an on-hook event will be sent.

If debounce time is configured, OffOnDebounce must be less than HookFlash MaxTime.

**Debounce Times:** This section configures two debounce delays.

- **Debounce On-Off Time (ms):** Enter the delay, in ms, before informing on an off-hook event to ensure it is not a spike. If an on-hook event is received during this delay time, the two events will be ignored. If 0 is entered, no debounce will be done.
- **Debounce Off-On Time (ms):** Enter the delay, in ms, before informing on an on-hook event to ensure it is not a spike. If an off-hook event is received during this delay time, the two events will be ignored. If a **HookFlash Max Time** is configured, it must be more than OffOnDebounce. If 0 is entered, no debounce will be done.

**Codec Preference:** This section determines the order in which supported codecs will be placed in a call setup message sent to any destination line. It also helps determine the selected codec when a message indication for an incoming call is received from the remote end with codec preference information.

- **G711U:** Select priority 1, 2, or 3 to be assigned to the G711U codec, or NONE if the G711U codec is not to be used.
- **G711A:** Select priority 1, 2, or 3 to be assigned to the G711A codec, or NONE if the G711A codec is not to be used.
- **G729A:** Select priority 1, 2, or 3 to be assigned to the G729A codec, or NONE if the G729A codec is not to be used.

*Note: If two codec types are assigned the same priority, then the priority is assigned in the order as G711U > G711A > G729 in the decreasing order of priority. For example, if 1, 2, and 2 is selected for G711U, G711A, and G729, respectively, priority will be assigned as 1, 2, and 3 for G711U, G711A, and G729, respectively.*

**Submit Changes:** Click **Submit Changes** to save the settings on this page to system RAM.

*Note: All **Advanced Telephony setting** parameters **do not** require a **Save & Reboot** to take effect.*

*Note: After clicking **Submit Changes** to save page settings to system RAM, you must permanently save the configuration and reinitialize the system as follows.*

**Save Configuration:** Click this link to go to the **Save Settings and Reboot** page. On the **Save Settings and Reboot** page, click **Save & Reboot** to permanently save the settings to system flash memory and to reinitialize the system to the new settings.

## 2.1.9 Advanced System Telephony Parameters Page

The **Advanced System Level Telephony Parameters** page allows configuration of the system level telephony parameters for Caller ID and Jitter Buffer.

**Codec Preferences:** This section configures the frames per RTP packet for the selected codec.

- User can select 1, 2, or 3 in the “Frames per packet” text field.
- 1, 2, 3 corresponds to 20, 40, 60 ms voice size with respect to G711u, G711a and G729

**Jitter Buffer Configuration:** This section configures the jitter buffer information for the selected codec.

- **Max Reorder delay (ms):** When the Reorder buffer reaches **max reorder delay** value specified, it is rapidly emptied.
- **Max accept late seq num:** If the new packet sequence number is low by “**Max Accept late seq num**” from the reorder buffer tail sequence number, then a new sequence has started and the reorder buffer is initialized.
- **Initial delay (ms):** Sets the **initial delay** for the jitter buffer.

- **Excessive frame deletion mode (0/1):** Once the reorder buffer grows past `peak_transit_delay`, the ATA limits further growth in one of the two user selectable modes:

Soft (1): Every time a frame is read from the reorder buffer for decoding, a second frame is read, decoded and deleted. This is continued until the reorder buffer is below the deletion threshold.

Hard (0): Frames past the deletion threshold are immediately deleted, before decoding.

- **Max Transit delay (ms):** The maximum delay in the delay buffer, in milliseconds.
- **Peak Transit delay (ms):** The deletion threshold, in milliseconds. When the delay exceeds this limit, frames are deleted according to excessive frame deletion mode.
- **Delay Buffer Increment (ms):** This field specifies the value by which the size of delay buffer is increased when the delay buffer underruns.
- **Transit delay decrease threshold (ms):** The minimum interval of decreased jitter before the delay buffer adapts downwards.

**Caller ID Info:** This section configures Caller ID (CID).

**Modulation:** Select **CID\_BELL202** to use Bell 202 modulation or **CID\_V23** to use V.23 modulation.

- **Delay to CID SLIC State (ms):** Enter the time delay, in milliseconds, to change the SLIC state, i.e., the delay after the end of the ring and before sending the CID.
- **Delay to CID Transmission (ms):** Enter the time delay, in milliseconds, between the end of the first ring and the beginning of CID transmission.
- **FSK AMP 0:** Enter the deviation of logic 0 frequency, in Hz.
- **FSK AMP 1:** Enter the deviation of logic 1 frequency, in Hz.

**Submit Changes:** Click **Submit Changes** to save the settings on this page to system RAM.

*Note:* All **Advanced System Telephony** parameters **do not** require a **Save & Reboot** to take effect.

*Note:* After clicking **Submit Changes** to save page settings to system RAM, you must permanently save the configuration and reinitialize the system as follows.

**Save Configuration:** Click this link to go to the **Save Settings and Reboot** page. On the **Save Settings and Reboot** page, click **Save & Reboot** to permanently save the settings to system flash memory and to reinitialize the system to the new settings.

## 2.1.10 FXO Parameters

The **FXO Configuration** page allows configuration of system level FXO parameters. See Section 3.5 for more details on FXO support.

**FXO Configuration**

Attach FXO to FXS:

CID Enable: ☒ Yes ☐ No

PSTN Backup: ☐ Yes ☒ No

FXO Bridging Enable: ☐ Yes ☒ No

Tone Detection Wait Timer (Secs):

Settings need to be saved to Flash and the system needs to be rebooted for the changes to take effect.

[Save Configuration](#)

**Attach FXO to FXS:** This field determines which FXS is to be attached to the FXO line. Outgoing and incoming PSTN/FXO calls can be made and received on the attached FXS only.

**CID Enable:** Click **Yes** to enable PSTN Caller ID display or **No** to disable PSTN Caller ID display.

**PSTN Backup:** Click **Yes** to enable PSTN Backup or **No** to disable PSTN Backup (default).

*Note: This field is displayed only when the vophwcfg.ini is appropriately configured (see Appendix C.1).*

In case of WAN down, the system will switch to PSTN Relay only when this PSTN Backup is enabled and the system has a PSTN Relay. Section 3.4 of this document describes the PSTN call features of the ATA.

**FXO Bridging Enable:** Not supported in this release.

**Tone Detection Wait Timer (Secs):** Enter the length of time in seconds the ATA will wait for the PSTN dial tone to be detected (to make sure CO supports call on hold) after the user has pressed hook flash in a connected PSTN call. In case of tone not being detected or a timeout, a fast busy tone will be played. The user can press hook flash again to come back to the first PSTN call.

Tone Detection Wait Timer not supported in this release

**Submit Changes:** Click **Submit Changes** to save the settings on this page to system RAM.

*Note: After clicking **Submit Changes** to save page settings to system RAM, you must permanently save the configuration and reinitialize the system as follows.*

**Save Configuration:** Click this link to go to the **Save Settings and Reboot** page. On the **Save Settings and Reboot** page, click **Save & Reboot** to permanently save the settings to system flash memory and to reinitialize the system to the new settings.

## 2.2 Building Dial Plan String

### 2.2.1 Syntactic Format for Dial Plan Strings

The dial plan string consists of a set of dial plan rules specified according to the following syntax:

1. DTMF digits 0–9, A–F, \*, and #.
2. Range and sub-range of digits (“[ ]”). For example, [135] specifies digits 1, 3, or 5, [5–9] specifies digits 5 to 9, and [125–8] specifies digits 1, 2, and 5 to 8.
3. Zero or more repetitions of the preceding event (“.”). For example, X.7 specifies 7 digits where each digit is 0–9, N.5 specifies 5 digits where each digit is 2–9, and [1–5].5 specifies 5 digits where each digit is 1–5.
4. Inter-digit timeout (IDT) in seconds (“t”) followed by a character that specifies the timeout interval in seconds for the preceding event. For example, X.5t5 specifies 5 digits where each digit is 0–9 with an IDT of 5 seconds.
5. Dial plan rule separator (“|”).
6. A suffix character (“>”) that defines the end of the required dialing digits; optional dialing digits can follow.
7. “X” or “x” signifies digits 0–9.
8. “N” or “n” signifies digits 2–9.
9. Star service serial code separator (“:”) indicating the end of the serial code.
10. Star service dial-pad code separator (“;”) indicating the end of the dial-pad code.
11. Digit not allowed in a given position (“!”).

## 2.2.2 Dial Plan String and Serial Codes

This data defines the dial plan string as entered or configured by the user or the service provider.

The user or service provider enters the dial string consisting of supported call-related dial patterns and the star service dial patterns.

The call patterns can generally be entered in the order preferred by the user or service provider. The normal call patterns do not have any serial codes associated with them.

The serial codes used for specifying the dial-pad code related to each star service are fixed and the user must follow them.

- Notes:*
1. The dial plan interpreter analyzes the collected digits from the user on a first-come first-served basis. If, at the end of the collection and analysis, the interpreter determines that there are two or more possible candidates for a match, it chooses the first available match in the order in which the rules appear in the dial plan string.
  2. The maximum storage space allocated for the dial plan string is 512 characters.

The serial codes with associated star service features and default dial-pad codes are defined in Table 2-1.

*Table 2-1. Serial Codes, Star Service Features, and Default Dial-Pad Codes*

Serial Code	Star Service Feature	Default Dial-Pad Code
1	Enable Unconditional Call Forwarding	*72
2	Disable Unconditional Call Forwarding	*73
3	Enable Call Forwarding on Busy	*74
4	Enable Call Forwarding on No Answer	*75
11	Temporary Disable Call Waiting	*70
12	Call Return	*69
16	Blind Transfer	*90
18	IP Dialing	*47
19	Speed Dialing from Address Book	*78
20	PSTN Call	#
22	Temporary Disable 3-Way Conferencing Call	*83
23	Disable Call Forwarding on Busy	*76
24	Disable Call Forwarding on No Answer	*77
25	Emergency Call	N11
26	Temporary Enable Caller ID Transmit Block	*67
<b>Note:</b> Missing number codes (5-10, 17, 21, and 27-31) are not supported.		



### 2.2.3 Framing Dial Plan Rules

Dial plan interpretation and parsing is based on the incoming character and a set of expected characters as follows:

<b>0–9, A–D, #, or *</b>	A DTMF digit is recognized as valid if it is one of the following: between 0–9 or A or B or C or D or # or *.
<b>X</b>	Recognized as any valid DTMF digit in the range 0–9. The expected characters following this are “.” or “t” or any other valid syntactic character.
<b>N</b>	Recognized as any valid DTMF digit in the range 2–9. The expected characters following this are “.” or “t” or any other valid syntactic character.
<b>t</b>	The associated inter-digit timer (IDT) value in seconds. The expected character following this <b>must</b> be between 0–9 or a–z to represent the timer value. For a–z, “a” corresponds to 10 seconds, “b” to 11 seconds, and so on. The maximum timer value is “z” which corresponds to 35 seconds.
<b>.</b>	This character represents zero or more repetitions of the previous character (which <b>must</b> be a valid DTMF digit). This <b>must</b> be followed by a value between 0–9 and a–f to specify the number of repetitions. Letters a–f correspond to numbers 10–15, respectively.
<b>&gt;</b>	<p>This character is used for two purposes:</p> <ol style="list-style-type: none"><li>To specify the suffix associated with the dial plan rule. A single valid DTMF digit <b>must</b> follow this character.</li><li>To specify the minimum number of digits required to be dialed for the corresponding dial plan rule. The &gt; character must be included at the minimum position required for meaningful match.</li></ol> <p>For the example pattern, 0&gt;#t411t8x.etfxt2, only digit 0 is required for a meaningful match.</p> <p>For the example pattern, 0t411x&gt;#t8x.etfxt2, a 4-digit dial string of 011X is required for a meaningful match.</p>
<b>!</b>	This character, in association with a following DTMF digit, represents what a digit in the specified position <b>should not</b> be. A DTMF digit representation/range/sub-range <b>must</b> follow this character.
<b>[X-X]</b>	Sub-range starting from first value to next value. For example, [2-8] means a digit in the range 2 through 8.
<b>[XX]</b>	Specifies an array of digits allowed in the position mentioned. For example, [128] means digit 1, 2, or 8.

## 2.2.4 Default Dial Plan String

The default dial plan string corresponds to the North American Numbering Plan (NANP). This default dial plan string is (where each rule is separated by the separator “[”]):

```
0>#t4|Nx.5t8xt2>#|011x>#x.et8xt2|1Nx.2Nx.5t8xt2>#|1:*72;>#x.etfxt2|
2:*73;>#t4|3:*74;>#x.etfxt2|4:*75;>#x.etfxt2|11:*70;>#t4|12:*69;>#t4|
16:*90;x>#x.dtfxt2| 18:*47;x>#[0-9*].f[0-9*].ft8
[0-9*].ft4|19:*78;x>#t4|20:#;x.3>#x.atfxt2|22:*83;x>#x.dtfxt2|
23:*76;>#t4|24:*77;>#t4|25:N1t41;># |26:*67;>#t4| [0-9*]>#[0-9*].e[0-9*].ft4
```

The individual default dial plan rules are summarized in Table 2-2.

*Table 2-2. Default Dial Plan Rules*

Default Dial Plan Rule	Dial Plan Rule Function	Reference
0>#t4	Dial local operator	2.2.4.1
Nx.5t8xt2>#	Dial a local call	2.2.4.2
011x>#x.et8xt2	Dial a long distance call	2.2.4.4
1Nx.2Nx.5t8xt2>#	Dial an international long distance call	2.2.4.3
1:*72;>#x.etfxt2	Star service for Enable Unconditional Call Forwarding	2.2.4.5
2:*73;>#t4	Star service for Disable Unconditional Call Forwarding	2.2.4.8
3:*74;>#x.etfxt2	Star service for Enable Unconditional Call Forwarding on Busy	2.2.4.6
4:*75;>#x.etfxt2	Star service for Enable Unconditional Call Forwarding on No Answer	2.2.4.7
11:*70;>#t4	Star service for Temporary Disable Call Waiting	2.2.4.9
12:*69;>#t4	Star service for Call Return	2.2.4.10
16:*90;x>#x.dtfxt2	Star service for Blind Transfer	2.2.4.11
18:*47;x>#[0-9*].f[0-9*].ft8[0-9*].ft4	Star service for IP Dialing call	2.2.4.12
19:*78;x>#t4	Star service for Speed Dialing from Address Book	2.2.4.13
20:#;x.3>#x.atfxt2	Dial PSTN call	2.2.4.14
22:*83;x>#x.dtfxt2	Star service for Temporary Disable 3-Way Conferencing call	2.2.4.15
23:*76;>#t4	Star service for disabling Call Forwarding on Busy	2.2.4.16
24:*77;>#t4	Star service for disabling Call Forwarding on No Answer	2.2.4.17
25:N1t41;>#	Dial an emergency call	2.2.4.18
26:*67;>#t4	Star Service for Temporary blocking of Caller ID Transmission	2.2.4.19
[0-9*]>#[0-9*].e[0-9*].ft4	Default number call	2.2.4.20

There are three major components of any star service rule:

1. The star service rule must begin with the serial code identified in Table 2-1. This serial code identifies the feature/dial plan action to be executed. This serial code is followed by the separator “.”.
2. Following this is the dial-pad code (digits the user must dial first) to execute the feature. The separator “;” follows this dial-pad code.
3. Then the rule for the digits or number needed to execute the star service is specified.

*Note: If the IDT elapses during the minimum number of digits to be dialed, the dialed string does not match the plan, in which case a match to another plan is attempted. If the IDT elapses during the optional digits to be dialed, the dialed number is sent.*

#### **2.2.4.1 Default Dial Plan Rule 0>#t4**

This is the dial plan rule for dialing the local operator.

Digit 0 must be dialed, optionally followed by pressing #.

After the 0 digit is dialed, press # to send the digit immediately, otherwise there is a 4-second delay before the digit is sent.

<b>0</b>	One digit 0
<b>&gt;#</b>	Position of >specifies 1 digit minimum; suffix is #
<b>t4</b>	IDT of 4 seconds

#### **2.2.4.2 Default Dial Plan Rule Nx.5t8xt2>#**

This is the dial plan rule for dialing a local call within the geographical region covered by the NANP.

Seven digits must be dialed, each digit within 8 seconds of the preceding digit, optionally followed by pressing #.

After the seven digits are dialed, press # to send the digits immediately, otherwise there is a 2-second delay before the digits are sent.

<b>N</b>	One digit in range 2–9
<b>x.5</b>	Five digits in range 0–9
<b>t8</b>	IDT of 8 seconds
<b>x</b>	One digit in range 0–9
<b>t2</b>	IDT of 2 seconds
<b>&gt;#</b>	Position of >specifies 7 digits minimum; suffix is #

#### **2.2.4.3 Default Dial Plan Rule 1Nx.2Nx.5tftx2>#**

This is the dial plan rule for dialing a long distance call.

Eleven digits must be dialed starting with 1, each digit within 15 seconds of the preceding digit, optionally followed by pressing #.

After the 11 digits are dialed, press # to send the digits immediately, otherwise there is a 2-second delay before the digits are sent.

<b>1</b>	One digit 1
<b>N</b>	One digit in range 2–9
<b>x.2</b>	Two digits in range 0–9
<b>N</b>	One digit in range 2–9
<b>x.5</b>	Five digits in range 0–9
<b>tf</b>	IDT of 15 seconds
<b>x</b>	One digit in range 0–9
<b>t2</b>	IDT of 2 seconds
<b>&gt;#</b>	Position of >specifies 11 digits minimum; suffix is #

#### 2.2.4.4 Default Dial Plan Rule 011x>#x.et8xt2

This is the dial plan rule for dialing an international long distance call.

Four digits must be dialed starting with 011, followed by 15 optional digits and optionally followed by pressing #. Each digit must be dialed within 8 seconds of the preceding digit.

After the 011x digits are dialed, press # to send the digits immediately, otherwise the digits will be sent 8 seconds after the last dialed digit for digits 4-18 is dialed, or 2 seconds after digit 19 is dialed.

<b>011</b>	Three digits with 011 pattern
<b>x</b>	One digit in range 0–9
<b>&gt;#</b>	Position of >specifies 4 digits minimum; suffix is #
<b>x.e</b>	Fourteen digits in range 0–9
<b>t8</b>	IDT of 8 seconds
<b>x</b>	One digit in range 0–9
<b>t2</b>	IDT of 2 seconds

#### 2.2.4.5 Default Dial Plan Rule 1:\*72;>#x.etfxt2

This is the dial plan rule for Enable Unconditional Call Forwarding star service.

Digits \*72 must be dialed, followed by 15 optional digits and the optional # suffix. Each digit must be dialed within 15 seconds of the preceding digit.

If only the \*72# digits are dialed or the IDT elapses before dialing any digits after \*72, the default number configured on the **Call Feature** configuration page (Section 2.1.6) is sent.

After the \*72 digits are dialed, press # to send the digits immediately, otherwise the digits will be sent 15 seconds after the last dialed digit for digits 4-17 is sent, or 2 seconds after digit 18 is dialed.

<b>1:</b>	Serial code for Enable Unconditional Call Forwarding followed by “:” separator
<b>*72;</b>	Dial-pad code for Enable Unconditional Call Forwarding followed by “;” separator
<b>&gt;#</b>	Position of >specifies 3 digits minimum; suffix is #
<b>x.e</b>	Fourteen digits in range 0–9
<b>tf</b>	IDT of 15 seconds
<b>x</b>	One digit in range 0–9
<b>t2</b>	IDT of 2 seconds

*Note: The default above rule states the number specified can have digits in the range 0–9 and can be a maximum length of 15 digits. This is flexible; for example, a Call Forwarding rule could be specified as 1:\*72;>#[78]x.6t4. In this case, Call Forwarding is done only to a number beginning with either 7 or 8.*

#### 2.2.4.6 Default Dial Plan Rule 3:\*74;>#x.etfxt2

This is the dial plan rule for Enable Unconditional Call Forwarding on Busy star service. Digits \*74 must be dialed, followed by 15 optional digits and the optional # suffix. Each digit must be dialed within 15 seconds of the preceding digit.

If only the \*74# digits are dialed or the IDT elapses before dialing any digits after \*74, the default number configured on the **Call Feature** configuration page (Section 2.1.6) is sent.

After the \*74 digits are dialed, press # to send the digits immediately, otherwise the digits will be sent 15 seconds after the last dialed digit for digits 4-17 is sent, or 2 seconds after digit 18 is dialed.

<b>3:</b>	Serial code for Enable Unconditional Call Forwarding on Busy followed by “:” separator
<b>*74;</b>	Dial-pad code for Enable Unconditional Call Forwarding on Busy followed by “;” separator
<b>&gt;#</b>	Position of >specifies 3 digits minimum; suffix is #
<b>x.e</b>	Fourteen digits in range 0–9
<b>tf</b>	IDT of 15 seconds
<b>x</b>	One digit in range 0–9
<b>t2</b>	IDT of 2 seconds

#### 2.2.4.7 Default Dial Plan Rule 4:\*75;>#x.etfxt2

This is the dial plan rule for Enable Unconditional Call Forwarding on No Answer star service.

Digits \*75 must be dialed, followed by 15 optional digits and the optional # suffix. Each digit must be dialed within 15 seconds of the preceding digit.

If only the \*75# digits are dialed or the IDT elapses before dialing any digits after \*75, the default number configured on the **Call Feature** configuration page (Section 2.1.6) is sent.

After the \*75 digits are dialed, press # to send the digits immediately, otherwise the digits will be sent 15 seconds after the last dialed digit for digits 4-17 is sent, or 2 seconds after digit 18 is dialed.

<b>4:</b>	Serial code for Enable Unconditional Call Forwarding on No Answer followed by “:” separator
<b>*75;</b>	Dial-pad code for Enable Unconditional Call Forwarding on No Answer followed by “;” separator
<b>&gt;#</b>	Position of >specifies 3 digits minimum; suffix is #
<b>x.e</b>	Fourteen digits in range 0–9
<b>tf</b>	IDT of 15 seconds
<b>x</b>	One digit in range 0–9
<b>t2</b>	IDT of 2 seconds

#### 2.2.4.8 Default Dial Plan Rule 2:\*73;>#t4

This is the dial plan rule for Disable Unconditional Call Forwarding star service.

Digits \*73 must be dialed, each digit within 4 seconds of the preceding digit, optionally followed by pressing #.

*Note: The \*73 code will disable all enabled Call Forwarding star services previously enabled by the \*72, \*74, or \*75 codes.*

After the \*73 digits are dialed, press # to send the digits immediately, otherwise there is a 4-second delay before the digits are sent.

<b>2:</b>	Serial code for Enable Unconditional Call Forwarding followed by “:” separator
<b>*73;</b>	Dial-pad code for Enable Unconditional Call Forwarding followed by “;” separator
<b>&gt;#</b>	Position of >specifies 3 digits minimum; suffix is #
<b>t4</b>	IDT of 4 seconds

#### 2.2.4.9 Default Dial Plan Rule 11:\*70;>#t4

This is the dial plan rule for Temporary Disable Call Waiting star service.

Digits \*70 must be dialed, each digit within 4 seconds of the preceding digit, optionally followed by pressing #.

After the \*70 digits are dialed, press # to send the digits immediately, otherwise there is a 4-second delay before the digits are sent.

<b>11:</b>	Serial code for Temporary Disable Call Waiting followed by “:” separator
<b>*70;</b>	Dial-pad code for Temporary Disable Call Waiting followed by “;” separator
<b>&gt;#</b>	Position of >specifies 3 digits minimum; suffix is #
<b>t4</b>	IDT of 4 seconds

#### 2.2.4.10 Default Dial Plan Rule 12:\*69;>#t4

This is the dial plan rule for Call Return star service.

The digits \*69 must be dialed, each digit within 4 seconds of the preceding digit, optionally followed by pressing #.

After the \*69 digits are dialed, press # to send the digits immediately, otherwise there is a 4-second delay before the digits are sent.

<b>12:</b>	Serial code for Call Return followed by “:” separator
<b>*69;</b>	Dial-pad code for Call Return followed by “;” separator
<b>&gt;#</b>	Position of >specifies 3 digits minimum; suffix is #
<b>t4</b>	IDT of 4 seconds

#### 2.2.4.11 Default Dial Plan Rule 16:\*90;x>#x.dtfxt2

This is the dial plan rule for Blind Transfer star service.

Digits \*90 followed by a single-digit number must be dialed, each digit within 15 seconds of the preceding digit, followed by 14 optional digits and the optional # suffix.

After the \*90x digits are dialed, press # to send the digits immediately, otherwise the digits will be sent 15 seconds after the last digit for digits 4-17 is dialed, or 2 seconds after digit 18 is dialed.

<b>16:</b>	Serial code for Call Return followed by “:” separator
<b>*90;</b>	Dial-pad code for Call Return followed by “;” separator
<b>x</b>	One digit in range 0–9
<b>&gt;#</b>	Position of >specifies 4 digits minimum; suffix is #
<b>x.d</b>	Thirteen digits in range 0–9
<b>tf</b>	IDT of 15 seconds
<b>x</b>	One digit in range 0–9
<b>t2</b>	IDT of 2 seconds

#### 2.2.4.12 Default Dial Plan Rule 18:\*47;x>#[0-9\*].f[0-9\*].ft8[0-9\*].ft4

This is the dial plan rule for IP Dialing star service.

Digits \*47 followed by at least a single digit must be dialed, each digit within 8 seconds of the preceding digit (up to the first 31 digits), optionally followed by pressing #.

After the \*47<User URL> digits are dialed, press # to send the digits immediately, otherwise there is a 4-second delay before the digits are sent. See [Section 3.3.10](#) for more clarity on how to enter user numbers, IP address and Port using dial pad.

<b>18:</b>	Serial code for IP Dialing followed by “:” separator
<b>*47;</b>	Dial-pad code for IP Dialing followed by “;” separator
<b>x</b>	One digit in range 0–9
<b>&gt;#</b>	Position of >specifies 1 digit minimum; suffix is #
<b>[0-9*].f</b>	Fifteen digits in range 0–9 or *
<b>[0-9*].f</b>	Fifteen digits in range 0–9 or *
<b>t8</b>	IDT of 8 seconds
<b>[0-9*].f</b>	Fifteen digits in range 0–9 or *
<b>t4</b>	IDT of 4 seconds

### 2.2.4.13 Default Dial Plan Rule 19:\*78;x>#t4

This is the dial plan rule for speed dialing an entry in the address book.

Digits \*78 followed by a single digit must be dialed, each digit within 4 seconds of the preceding digit, optionally followed by pressing #.

*Note: The single digit is an index into the address book maintained by the system. The required address book entries must be configured according to Section 2.1.7.*

After the \*78x digits are dialed, press # to send the digits immediately, otherwise there is a 4-second delay before the digits are sent.

<b>19:</b>	Serial code for Speed dial followed by “:” separator
<b>*78;</b>	Dial-pad code for Speed Dialing followed by “;” separator
<b>x</b>	One digit in range 0–9
<b>&gt;#</b>	Position of >specifies 3 digits minimum; suffix is #
<b>t4</b>	IDT of 4 seconds

### 2.2.4.14 Default Dial Plan Rule 20:##;x.3tf>#x.atfxt2

This is the rule for dialing a PSTN call.

Digit # followed by three single-digit numbers, each digit within 15 seconds of the preceding digit, optionally followed up to 11 optional digits, and optionally followed by pressing #.

After the four required digits are dialed, press # to send the digits immediately, otherwise the digits will be sent 15 seconds after the last digit for digits 4-13 is dialed, or 2 seconds after digit 14 is dialed.

<b>20:</b>	Serial code for PSTN call followed by “:” separator
<b>##;</b>	Dial-pad code for PSTN call followed by “;” separator
<b>x.3</b>	Three digits in range 0–9
<b>tf</b>	IDT of 15 seconds
<b>&gt;#</b>	Position of >specifies 4 digits minimum; suffix is #
<b>x.a</b>	Ten digits in range 0–9
<b>tf</b>	IDT of 15 seconds
<b>x</b>	One digit in range 0–9
<b>t2</b>	IDT of 2 seconds



**2.2.4.15 Default Dial Plan Rule 22:\*83;>#t4**

This is the dial plan rule for Disabling 3-Way Conferencing temporarily.

Digits \*83 must be dialed, each digit within 4 seconds of the preceding digit, optionally followed by pressing #. No number needs to be dialed.

After the \*83 is dialed, press # to send the rule immediately, otherwise there is a 4-second delay before the rule is executed.

<b>22:</b>	Serial code for Temporary Disable of 3-Way Conferencing followed by “:” separator
<b>*83;</b>	Dial-pad code for Disabling 3-Way Conferencing followed by “;” separator
<b>&gt;#</b>	Position of >specifies 4 digits minimum; suffix is #
<b>t4</b>	IDT of 4 seconds

**2.2.4.16 Default Dial Plan Rule 23:\*76;>#t4**

This is the dial plan rule for Disable Call Forwarding on Busy star service.

Digits \*76 must be dialed, each digit within 4 seconds of the preceding digit, optionally followed by pressing #. No number needs to be dialed.

After the \*76 is dialed, press # to send the rule immediately, otherwise there is a 4-second delay before the rule is executed.

<b>23:</b>	Serial code for Disable Call Forwarding on Busy followed by “:” separator
<b>*76;</b>	Dial-pad code for Disable Call Forwarding on Busy followed by “;” separator
<b>&gt;#</b>	Position of >specifies 3 digits minimum; suffix is #
<b>t4</b>	IDT of 4 seconds

**2.2.4.17 Default Dial Plan Rule 24:\*77;>#t4**

This is the dial plan rule for Disable Call Forwarding on No Answer star service.

Digits \*77 must be dialed, each digit within 4 seconds of the preceding digit, optionally followed by pressing #. No number needs to be dialed.

After the \*77 is dialed, press # to send the rule immediately, otherwise there is a 4-second delay before the rule is executed.

<b>24:</b>	Serial code for Disable Call Forwarding on No Answer followed by “:” separator
<b>*77;</b>	Dial-pad code for Disable Call Forwarding on No Answer followed by “;” separator
<b>&gt;#</b>	Position of >specifies 3 digits minimum; suffix is #
<b>t4</b>	IDT of 4 seconds

#### 2.2.4.18 Default Dial Plan Rule 25:N1t41;>#

This is the dial plan rule for Emergency Call. However, this rule also handles other N11 services.

One digit in range 2–9 followed by the digit 1 and the digit 1 must be dialed, each digit within 4 seconds of the preceding digit.

After digits N11 are dialed, the digits are immediately sent.

<b>25:</b>	Serial code for Emergency Call followed by “:” separator
<b>N1t41;</b>	Dial-pad code for Emergency Call followed by “;” separator
<b>N</b>	One digit in range 2–9
<b>1</b>	One digit 1
<b>t4</b>	IDT of 4 seconds
<b>1</b>	One digit 1
<b>&gt;#</b>	Position of >specifies 3 digits minimum; suffix is # (# is included because a character must be provided after >, however, pressing # is not an option because the dialed digits are sent immediately)

- Notes:
1. Digits N11 match the [0-9\*]>#[0-9\*].e[0-9\*].ft4 dial plan, except the 25:N1t41;># dial plan rule allows a 911 call will be sent without delay.
  2. This emergency rule is not part of the default service provider’s dial plan string as Pulver does not support 911 calls (see Appendix B).
  3. Any number of such emergency rules can be specified in the dial plan string. In such emergency call match cases, the dial plan interpreter will not wait for any timeout and will send out the matched digits immediately.

#### 2.2.4.19 Default Dial Plan Rule 26:\*67;>#t4

This is the dial plan rule for Temporary Enable Caller ID Transmission Block.

Digits \*67 must be dialed, each digit within 4 seconds of the preceding digit, optionally followed by pressing #.

After the\*67 digits are dialed, press # to send the digits immediately, otherwise there is a 4-second delay before the digits are sent.

<b>26:</b>	Serial code for Temporary Enable Caller ID Transmission Block followed by “:” separator
<b>*67;</b>	Dial-pad code for Temporary Enable Caller ID Transmission Block followed by “;” separator
<b>&gt;#</b>	Position of >specifies 3 digits minimum; suffix is #
<b>t4</b>	IDT of 4 seconds

#### 2.2.4.20 Default Dial Plan Rule [0-9\*]>#[0-9\*].e[0-9\*].ft4

This is the dial plan rule for Default Number Call, which applies if no previous rule matches.

This rule takes a minimum of 1 digit and a maximum of 30 digits. All dialed digits will be sent. The service provider decides what to do with the dialed digits. All digits dialed can be between 0 through 9, or \*.

After the first digit is dialed, press # to send the digits immediately, otherwise the digits will be sent 4 seconds after the last dialed digit.

[0-9*]	One digit in range 0–9 or *
>#	Position of >specifies 1 digit minimum; suffix is #
[0-9*].e	Fourteen digits in range 0–9 or *
[0-9*].f	Fifteen digits in range 0–9 or *
t4	IDT of 4 seconds

- Notes:*
1. If this default call rule is removed from the dial plan string, then any sequence of digits which **does not** match any of the other rules will be blocked by the ATA i.e., the dialed digits **will not** be sent to the service provider.
  2. The default call rule is a superset of all the other dial plan rules. The dial plan interpreter works on a first-come first-serve basis. Ensure that this rule is included and is always specified at the end of the dial plan string.

#### 2.2.5 Empty Dial Plan String

If a default dial plan string is not required, the **Default Dial Plan String** field on the **General** configuration page (Section 2.1.2) can be left empty, in which case the default dial pattern to accept all dialed digits will be incorporated.

The default dial pattern, [0-9\*]>#[0-9\*].e[0-9\*].ft4, is transparent to the user and will not be displayed on the **General** configuration page (see Section 2.1.2).

Please note that when the dial plan string is left empty, there will no star services support available locally at ATA. Since the default dial plan rule sends all collected digits to the service provider, it is expected that the service provider handles the star services using his dial plan.

#### 2.2.6 Incorrect Dial Plan Rule

If the user enters a syntactically incorrect dial plan rule, the rule is rejected while parsing but the rule is not removed from the dial plan string. In other words, syntactic analysis is not performed during the parsing stage **after** the **Save and Reboot**, but after parsing the dial plan string, the invalid dial plan rule is skipped.

If all the dial plan rules are invalid, the default dial pattern is used. This default dial pattern, [0-9\*]>#[0-9\*].e[0-9\*].ft4, is transparent to the user and will not be displayed on the **General** configuration page (see Section 2.1.2).

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## 3. Using Conexant ATA

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### 3.1 Setting up ATA for VoIP Calls

This section describes the setup and configuration needed before the ATA can be used.

1. Ensure the ATA software has been flashed onto the VoIP reference board.
2. When power is applied to the board, the system initializes with the default configuration which is **not** initially set up to make VoIP calls.
3. The configured default service provider is Pulver. You can either use Pulver or configure a new service provider. See Section 2.1.3 for adding or selecting a new service provider.
4. If you choose to use Pulver, then a free login account must be obtained for each line from the website <http://www.fwd.pulver.com>.
5. After the login accounts have been obtained, configure the login account information for the service provider you intend to use. See Section 2.1.4 for adding and selecting the login account to use.
6. The other default parameters are sufficient enough to start making and receiving calls.
7. After you have saved your changes, **Registration** status on the **ATA Configuration** page (see Section 2.1) will be displayed as **REGISTERED** if you have chosen to go through a service provider.

*Note: If your system has PSTN backup hardware, you must have configured vophwcfg.ini for PSTN backup (see Appendix C.2).*

### 3.2 Making Basic Calls

Follow these steps to make an outgoing phone call:

1. Pick up the handset.
2. When you hear a dial tone, dial the number you want to reach.
3. Press # to send the number immediately, or wait for the IDT to expire and the number will be sent automatically.

*Note: The ATA supports Mid-Call DTMF in order to call toll-free numbers and other interactive voice response (IVR) systems. ATA is configured automatically during call establishment to support RFC 2833-based dynamic payload negotiation and to send DTMF tones to the other end point.*

## 3.3 Advanced Call Features

The supplementary services described in this section, and their configuration and implementation, depend on the system of the country in which the service is activated.

This section describes the following topics:

- Caller ID
- Call-Waiting Caller ID
- Consultation Hold
- Blind Transfer
- Attended Transfer
- 3-Way Conferencing
- Call Waiting
- Call Forwarding
- Call Return
- IP Dialing
- Speed Dialing
- Caller ID Transmit Blocking

*Notes:*

1. *Web configuration overrides any dial-pad based configuration, and will be reflected only upon **Save and Reboot**, expecting a few parameters which have been noted in the above sections.*
2. *The digit sequences to be dialed for different actions to be executed are based on the default dial plan rules specified in this document.*
3. *Some features require that a hook flash be performed. To perform a hook flash, quickly press and release the switchhook (push button) on the phone cradle or quickly press and release the flash button or other dedicated key/button on the telephone base unit or handset, depending on the telephone.*  
  
*Be careful not to hold the switchhook or flash button down too long, e.g., for three seconds, which will disconnect your call. In this case, when you release the switchhook or flash button, you will hear a regular dial tone.*

### 3.3.1 Caller ID

When the telephone rings, the ATA sends a Caller ID signal to the telephone between the first and second ring (with name, telephone number, time, and date information, if these are available).

### 3.3.2 Call-Waiting Caller ID

The Conexant ATA plays the first Call Waiting tone, and then sends the name, telephone number, time, and date information, if these are available, before the second Call Waiting tone is played.

### 3.3.3 Consultation Hold

This feature allows a user to put the existing call on hold and call another number.

**How to Use:**

1. During an existing call, perform a hook flash to put the other party on hold and to get a dial tone.
2. When you hear the dial tone, dial another number.
3. When you are finished with the second call, perform a hook flash to revert to the first party. You can also alternate between calls by performing a hook flash.

### 3.3.4 Blind Transfer

This feature allows a user (transferor) to transfer an existing call to another telephone number (transfer target) without connecting to the transfer target number.

**How to Use:**

1. During an existing call, perform a hook flash to put the other party on hold and get a dial tone.
2. When you hear the dial tone, press \*90 on your telephone dial-pad.
3. When you hear the alert tone indicating that the ATA is expecting a number, dial the phone number to which you want to transfer the other party, then press # (optional).
4. When you hear the Fast Busy Tone, hang up your phone.

*Note: This dial-pad code can be changed as part of the dial plan rules which can be modified for the service provider on the **SIP Service Provider** configuration page as described in Section 2.1.3.*

### 3.3.5 Attended Transfer

This feature allows a user to transfer an existing call to another telephone number after first consulting with the dialed party (transfer target) before hanging up.

**How to Use:**

1. During an existing call, perform a hook flash to put the other party on hold and get a dial tone.
2. When you hear the dial tone, dial the telephone number to which the existing party is to be transferred.
3. When the target transfer answers the phone, you may consult with the target transfer, and then hang up your phone to transfer the call to the target transfer.

### 3.3.6 3-Way Conferencing

#### How To Use:

1. Dial the first number.
2. During connection to the first party, perform a hook flash to put the first party on hold and get a dial tone.
3. When you hear the recall dial tone, dial another number and talk with the second person.
4. To conference with both callers at the same time, perform a hook flash.
5. To drop the second call, perform a hook flash after entering into conferencing mode.

*Note: If you hang up during conferencing, both the other parties will be disconnected from the call.*

#### How To Temporarily Disable 3-way conferencing:

In some cases, the user may not want to engage in conference after calling the second callee. In such cases, he can choose to switch between calls (using hookflash) instead of entering into a 3-way conferencing mode. 3-way conferencing can be disabled in the following manner:

1. After going off-hook, if the ATA is configured to use 3-way Call Conferencing (see Section 2.1.3), press \*83 (optionally followed by #) on your telephone dial-pad to disable Call Conferencing for the duration of the following call.
2. You will hear a confirm tone followed by the dial tone. If you hang up with or without making the call, then the ATA enables Call-Conference again.
3. If the Conexant ATA is not configured to use Call Conference (see Section 2.1.3), pressing \*83 (optionally followed by #) results in fast-busy tone.

An alternate way to temporarily disable 3-way conferencing is as follows:

1. During connection to the first party, perform a hook flash to put the first party on hold and get a dial tone.
2. If the ATA is configured to use 3-way Call Conferencing (see Section 2.1.3), press \*83 (optionally followed by #) on your telephone dial-pad to disable Call Conferencing for the duration of the following call.
3. You will hear a confirm tone followed by the dial tone. If you hang up with or without making the call, then the ATA enables Call-Conference again.
4. If the Conexant ATA is not configured to use Call Conference (see Section 2.1.3), pressing \*83 (optionally followed by #) results in fast-busy tone.

*Note: When the sequence of digits \*83 is entered, the sequence could match the 3-way Call conference rule or the Default Number Call rule (Section 2.2.4.20). So, to execute the command immediately, press #. Otherwise, the action will execute after 4 seconds.*



### 3.3.7 Call Waiting in USA

If someone calls you while you are speaking on the telephone, the ATA indicates this by playing a Call Waiting Tone. Call Waiting can be disabled in the USA as follows.

**How to Use:**

1. If the ATA is configured to use Call Waiting (see Section 2.1.6), press \*70 (optionally followed by #) on your telephone dial-pad to disable Call Waiting for the duration of the following call.
2. You will hear a confirm tone followed by the dial tone. If you hang up with or without making the call, then the ATA enables Call-Waiting Tone again.
3. If the Conexant ATA is not configured to use Call Waiting (see Section 2.1.6), pressing \*70 (optionally followed by #) results in fast-busy tone.

*Note: When the sequence of digits \*70 is entered, the sequence could match the Call Waiting rule or the Default Number Call rule (Section 2.2.4.20). So, to execute the command immediately, press #. Otherwise, the action will execute after 4 seconds.*

### 3.3.8 Call Forwarding in USA

The ATA can control Call Forwarding at the end-point level.

The ATA supports three types of Call Forwarding:

- **Call Forwarding Unconditionally**—Forwards every incoming call.
- **Call Forwarding On Busy**—Forwards incoming calls only when the line is busy.
- **Call Forwarding on No Answer**—Forwards incoming calls only when the telephone is not answered after the configured period.

#### 3.3.8.1 Call Forwarding Unconditionally

**How to Use:**

1. Press \*72 on your telephone dial-pad.
2. After you hear a very brief alert tone (default is 100 milliseconds), enter the number (optional) you want to forward the call to; then press # (optional).
3. If you entered a number to forward the call to, the ATA will attempt to call that number. Once you disconnect the attempted call, in subsequent call attempts you will hear the stutter dial tone indicating that Unconditional Call Forwarding is enabled.

If you do not enter a number, then the **Call Forwarding Number** configured on the **Call Feature** configuration page (see Section 2.1.6) is used. But, in this case the ATA will not attempt to call the default number and you will hear a confirm tone, followed by the stutter dial tone. The duration of the stutter is configurable on the **Country Specific Ring & Tones** configuration page (see Section 2.1.1) by editing the last parameter of the Stutter Dial Tone configuration.

### 3.3.8.2 Call Forwarding On Busy

#### How to Use:

1. Press \*74 on your telephone dial-pad.
2. After you will hear a very brief alert tone (default is 100 milliseconds), enter the number (optional) you want to forward call to; then press # (optional).
3. If you entered a number to forward the call to, the ATA will attempt to call that number. Once you disconnect the attempted call, in subsequent call attempts you will hear a stutter dial tone indicating that Unconditional Call Forwarding is enabled.

If you do not enter a number, then the **Call Forwarding Number** configured on the **Call Feature** configuration page (see Section 2.1.6) is used. But, in this case the ATA will not attempt to call the default web configured number and you will hear a very brief confirm tone (default is 300 milliseconds only), followed by the normal dial tone.

### 3.3.8.3 Call Forwarding On No Answer

#### How to Use:

1. Press \*75 on your telephone dial-pad.
2. After you hear a very brief alert tone (default is 100 milliseconds), enter the number (optional) you want to forward the call to; then press # (optional).
3. If you entered a number to forward the call to, the ATA will attempt to call that number. Once you disconnect the attempted call, in subsequent call attempts you will hear normal dial tone only, but Call Forward on No Answer will be enabled.

If you do not enter a number, then the **Call Forwarding Number** configured on the **Call Feature** configuration page (see Section 2.1.6) is used. But, in this case the ATA will not attempt to call the default web configured number and you will hear a very brief confirm tone (default is 300 milliseconds), followed by the normal dial tone. You can set the **Call Forward No Answer** time on the **Timeouts** configuration page (see Section 2.1.5), which indicates the time after which Call Forwarding will take place. This timer is applicable for this feature only.

### 3.3.8.4 Canceling Call Forwarding

#### How to Use:

To cancel Unconditional Call Forwarding, press \*73 (optionally followed by #) on your telephone dial-pad.

To cancel Call Forwarding on Busy, press \*76 (optionally followed by #) on your telephone dial-pad.

To cancel Call Forwarding on No Answer, press \*77 (optionally followed by #) on your telephone dial-pad.

*Note: When the sequence of digits \*73, \*76 or \*77 is entered, the sequence could match the Cancel Unconditional Call Forwarding, Cancel Call Forwarding on Busy, Cancel Call Forwarding on no Ans, or the Default Number Call rule (Section 2.2.4.20). So, to execute the command immediately, press #. Otherwise, the action will execute after 4 seconds.*

*Note: In order for these codes to work, the corresponding features must be enabled as described in Section 2.1.6.*

### 3.3.9 Call Return in USA

The ATA provides the facility to call back the last incoming call that may have been missed. This is especially useful when the phone does not have Caller ID facility or does not support Call-Waiting Caller ID.

**How to Use:**

Press \*69 (optionally followed by #) on your telephone dial-pad to return the last incoming call.

- Notes:*
1. When the sequence of digits \*69 is entered, the sequence could match the Call Return in the USA rule or the Default Number Call rule (Section 2.2.4.20). So, to execute the command immediately, press #. Otherwise, the action will execute after 4 seconds.
  2. In order for these codes to work the corresponding features must be enabled on the **Call Feature** configuration page (see Section 2.1.6).

### 3.3.10 IP Dialing

The ATA provides the facility to do IP Dialing from the dialpad. This is especially useful for advanced users, who know the IP address of the other end and want to call it directly without going through any SIP proxy server.

**How to Use:**

This **How To** is based on the default dial plan rule configured for IP Dialing from [Section 2.2.4.12](#).

Press \*47 followed by the URL of the remote party, followed by a # sign to send out the call immediately.

The remote party's URL has the following syntax.

*<user number> \*\*<IP segment1> \*<IP segment2> \*<IP segment3> \*<IP segment4> \*<SIP Port>*

The user number and SIP Port fields are optional. In case the SIP port is not mentioned, the default SIP port of 5060 is used.

Some valid IP dial call examples are:

*\*4712345\*\*172\*25\*20\*178\*5060#*

*\*47172\*25\*20\*179#*

- Note:* When the sequence of digits \*47 is entered, the sequence could match the IP Dialing rule or the Default Number Call rule (Section 2.2.4.20). So, to execute the command immediately, press #. Otherwise, the action will execute after 4 seconds.

### 3.3.11 Speed Dialing for Address Book Entries

The ATA provides the facility of speed dialing for entries configured in the address book. See Section [2.1.7](#) for configuring the address book.

**How to Use:**

Press \*78 followed by the speed dial or address book entry index (0 – 9) (optionally followed by #) on your telephone dial-pad to dial the number corresponding to the entry specified in the address book. If the address book entry has just a phone number, the call will be made through the configured proxy server. If the entire URL is present in the address book, the call will be treated like an IP Dialing call.

*Note: When the sequence of digits \*78x is entered, the sequence could match the Speed dial rule or the Default Number Call rule (Section 2.2.4.20). So, to execute the command immediately, press #. Otherwise, the action will execute after 4 seconds.*

### 3.3.12 Caller ID Transmission Block

ATA provides the facility to block Caller ID transmission when you make a call. Caller ID transmission block can be enabled as follows.

**How to Use:**

1. If the ATA is configured to use Caller ID Transmission Block (see Section 2.1.6), press \*67 (optionally followed by #) on your telephone dial-pad to Enable Caller ID blocking for the duration of the following call.
2. You will hear a confirm tone followed by the dial tone. If you hang up with or without making the call, then the ATA disables Caller ID Transmission Block again. When this is done, the Caller ID Display Name sent when a call is made is taken from the Privacy Display Name (see Section 2.1.6).
3. If the Conexant ATA is not configured to use Caller ID Transmit Blocking (see Section 2.1.6), pressing \*67 (optionally followed by #) results in fast-busy tone.

*Note: When the sequence of digits \*67 is entered, the sequence could match the Call Waiting rule or the Default Number Call rule (Section 2.2.4.20). So, to execute the command immediately, press #. Otherwise, the action will execute after 4 seconds.*

## 3.4 PSTN Backup (Failsafe Relay Mode)

In case of power failure, the WAN IP is down, or SIP registration with the SIP server is lost, the ATA will automatically switch to Failsafe Relay (Hardware Relay) mode. In this case, all calls will be directed to the PSTN CO and handled by the CO. The CO will provide normal PSTN connectivity, with ATA not providing any call control. This automatic switch over provides the flexibility of allowing the user to make and receive PSTN and emergency calls in the event of power failure or IP down.

*Note: In systems where PSTN Backup is available, the user can make only VoIP calls when the system has registered with the VoIP service provider successfully.*

## 3.5 FXO Support

FXO support includes all VoIP features as well as additional PSTN related features:

- Initiate PSTN call using the VoIP dial plan.
- Receive an incoming PSTN call on the phone attached to the VoIP system.
- PSTN CID information display which can be enabled or disabled.
- Have a PSTN or VoIP call as a call waiting while making a PSTN or VoIP call.
- Place the first call on hold and make a second PSTN or VoIP call.
- Conferencing between VOIP and PSTN call.
- Switch between the two calls.

All the FXO actions can be done only from the FXS that is set through the FXO Parameters web page. (See Section 2.1.10.)

*Note:* See Appendix C on how to enable FXO support in `vophwcfg.ini`.

The call scenarios supported are:

- Incoming PSTN call
- Initiate PSTN call
- Call Waiting - Receiving a second call:

### 3.5.1 Incoming PSTN Call

There are two cases of handling a new PSTN call:

- CID Rx enabled: The ATA checks the FXO line state. If available, start ringing or start playing the Call Waiting tone to that Endpoint. The ATA then send the CID information to the phone if obtained.
- CID Rx disabled: The ATA checks the FXO line state. If available, start ringing or start playing the Call Waiting tone on that Endpoint.

### 3.5.2 Initiate PSTN Call

1. To initiate a PSTN call the user picks up the handset and Dials #<PSTN number>#.
2. After the second '#' the line will be switched to PSTN and the <PSTN number> will be dialed.
3. While a PSTN call is active:
  - To finish the PSTN call, hang up.
  - Making a second call will cause the ATA to:
    - To play a dial tone and collect the digits.
    - The digits will go through the dial plan to make a decision whether it is a VoIP number, a PSTN number or a Peer2Peer number.
    - To act according to the dial plan (PSTN or VOIP or P2P Call).
    - If three way conferencing is disabled while doing a "flash", the following will happen to the calls:

1st Call	HookFlash Action	2nd Call	HookFlash action
PSTN	DSP dialtone	PSTN	PSTN hook flash -Conference the 2 calls
PSTN	DSP dialtone	VoIP	Switch between the 2 calls
VoIP	DSP dialtone	PSTN	Switch between the 2 calls
VoIP	DSP dialtone	VoIP	Switch between the 2 calls

*Note: Three way conferencing is not supported with FXO.*

### 3.5.3 Call Waiting - Receiving a Second Call

1st Call	2nd Call	Flash Action
VoIP	VoIP	Switch between the 2 VoIP calls.
VoIP	PSTN	Switch between the 2 calls.
PSTN	VoIP	Switch between the 2 calls.
PSTN	PSTN	Switch between the 2 calls.

In this case, no third calls accepted (PSTN or VoIP call).

## Appendix A      Glossary

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This glossary defines acronyms and keywords used in this document.

### A.1      Acronyms

<b>ATA</b>	Analog Telephony Adaptor
<b>BLAM</b>	Background Logging Application Mechanism
<b>FoIP</b>	Fax over Internet Protocol
<b>FXO</b>	Foreign Exchange Office
<b>FXS</b>	Foreign Exchange Station
<b>PSTN</b>	Public Switched Telephone Network
<b>PTM</b>	Packet Telephony Module
<b>RTP</b>	Real-time Transport Protocol
<b>SIP</b>	Session Initiation Protocol
<b>STUN</b>	Simple Transversal of UDP through NAT
<b>UA</b>	User Agent
<b>VoIP</b>	Voice over Internet Protocol

### A.2      Keyword and Definitions

<b>Caller</b>	Call Originating End is called the Caller
<b>Callee</b>	The Call Terminating End is called the Callee
<b>Transferor</b>	The End transferring the call
<b>Transferee</b>	The End being transferred
<b>Transfer Target</b>	The End to whom the transferee is being transferred

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## Appendix B Dial Plan for Pulver Service Provider

### B.1 Basic Dial Plan

An example dial plan string as would be used with Pulver (a globally available VoIP service provider - <http://www.fwd.pulver.com>) is given below. The dial plan string represents only the basic dialing call and service rules, which can be used with the Conexant VoIP solution.

*Note: The emergency call dial plan rule is not part of Pulver's dial plan string. Pulver does not support USA emergency call dialing. Please check their website for any updates on dialing support and add or change the dial plan rules accordingly.*

#### B.1.1 Pulver - USA Dial Plan

```
[1-9]x.2t8>#x.6t4|*18x.8t8xt2>#|**484x.7t4>#|1:*72;>#x.etfxt2|2:*73;>#t4|
3:*74;>#x.etfxt2|4:*75;>#x.etfxt2|11:*70;>#t4|12:*69;>#t4|16:*90;x>#x.dtfxt2|
18:*47;t4xt2>#|20:#;x.3>#x.atfxt2|22:*83;>#t4|23:*76;>#t4|24:*77;>#t4|
[0-9*]>#[0-9*].e[0-9*].ft4
```

#### B.1.2 Explanation of the Rules

[1-9]x.2t8>#x.6t4	Call any Pulver Number up to 9 digits, but minimum 3 digits
*18x.8t8xt2>#	Call any Toll Free Number through Pulver Account
**484x.7t4>#	Call any Global Village Number. This rule demands a minimum and a maximum of 7 digits following **484.
1:*72;>#x.etfxt2	Enable Unconditional Call Forwarding
2:*73;>#t4	Disable Unconditional Call Forwarding
3:*74;>#x.etfxt2	Enable On Busy Call Forwarding
4:*75;>#x.etfxt2	Enable No Answer Call Forwarding
11:*70;>#t4	Temporary Disable Call Waiting
12:*69;>#t4	Execute Call Return
16:*90;x>#x.dtfxt2	Execute Blind Transfer
18:*47;t4xt2>#	Execute IP Dialing Call
20:#;x.3>#x.atfxt2	PSTN Number Call
22:*83;>#t4	Temporary Disable 3-Way Conference Call
23:*76;>#t4	Disable On Busy Call Forwarding
24:*77;>#t4	Disable No Answer Call Forwarding
[0-9*]>#[0-9*].e[0-9*].ft4	Default Call if no rule above matches

*Note: If this rule is removed from the dial plan string, then any sequence of digits which does not match any of the above specified rules will be blocked at the ATA end itself, i.e., the collected digits will not be sent to the service provider.*

## B.2 Calling Other Service Provider Numbers through Pulver

Pulver also provides the facility to call other service provider numbers. The following is an example list of those service providers and the corresponding recommended dial plan rules that need to be configured for each of them.

Please check their website for the complete list of supported service providers:

[\(http://www.fwd.pulver.com/content/view/full/333/\)](http://www.fwd.pulver.com/content/view/full/333/)

Service Provider Partner	Recommended Dial Plan Rule
NIC.at -	**011x>#x.et8xt2
Vonage	**2431x>#x.et8xt2
CallUK..	**285x>#x.et8xt2
iConnectHere	**333x>#x.et8xt2
iConnectHere account	**334x>#x.et8xt2
Earthlink	**356x>#x.et8xt2
Intertex	**468x>#x.et8xt2
IPTel	**478x>#x.et8xt2
InterViVo..	**488x>#x.et8xt2
Livedoor.com	**555x>#x.et8xt2
XS4all 3991...	**666x>#x.et8xt2
MyPhones 393...	**697x>#x.et8xt2
SIPPhone, Inc	**747x>#x.et8xt2
sipgate.de	**777x>#x.et8xt2
Squillo.it	**778x>#x.et8xt2
Telphin	**835x>#x.et8xt2
Voz Telecom	**868x>#x.et8xt2
Voiptalk 71...	**878x>#x.et8xt2
Packet8	**8981x>#x.et8xt2
ZGWireless.net	**949x>#x.et8xt2
Inter-Asterisk Exchange (IAXTEL)	**1700x>#x.et8xt2

## Appendix C      Configuring \*.ini Files

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### C.1      vophwcfg.ini - Configuring PSTN Backup Support

This section explains how to configure the vophwcfg.ini for enabling and disabling PSTN support for a CX82310-based system and the 1-Line reference board.

#### C.1.1      Configuration for CX82310 Based System

The relevant fields in the vophwcfg.ini are:

HwPstnRelayCtrlGpio = 2	Indicates the GPIO number that controls the PSTN backup (PSTN hardware relay) (-1 if hardware support is not present).
HwPstnRelayPstnVal = 0	Indicates the value that should be written to HwPstnRelayCtrlGpio in order to switch to PSTN.
HwPstnHookDetGpio = -1	Indicates the GPIO number for the PSTN hook detection (-1 if not supported).
HwPstnOffHookVal = 1	Indicates the value for PSTN OffHook.
FXO_Support = 0	0 = FXO disabled, 1 = FXO enabled. (See note.)

#### C.1.2      Configuration for 1-Line Reference Hardware

The relevant fields in the vophwcfg.ini are:

HwPstnRelayCtrlGpio = 20	Indicates the GPIO number that controls the PSTN backup (PSTN hardware relay) (-1 if hardware support is not present).
HwPstnRelayPstnVal = 1	Indicates the value that should be written to HwPstnRelayCtrlGpio in order to switch to PSTN.
HwPstnHookDetGpio = 18	Indicates the GPIO number for the PSTN hook detection (-1 if not supported).
HwPstnOffHookVal = 1	Indicates the value for PSTN OffHook.
FXO_Support = 0	0 = FXO disabled, 1 = FXO enabled. (See Note.)

### C.1.3 Configuration for CX82100 (2FXS-1FXO) Based Reference Hardware

The relevant fields in the vophwcfg.ini are:

HwPstnRelayCtrlGpio = 18	Indicates the GPIO number that controls the PSTN backup (PSTN hardware relay) (-1 if hardware support is not present).
HwPstnRelayPstnVal = 1	Indicates the value that should be written to HwPstnRelayCtrlGpio in order to switch to PSTN.
HwPstnHookDetGpio = -1	Indicates the GPIO number for the PSTN hook detection (-1 if not supported).
HwPstnOffHookVal = 1	Indicates the value for PSTN OffHook.
FXO_Support = 2	0 = FXO disabled, 2 = FXO enabled. (See Note.)

*Note:*    *FXO\_Support*

*0 = Disable FXO*

*1 = Enable FXO on 1LGPIO Hardware*

*2 = Enable FXO on VoicePort based Hardware*

## Appendix D      Ring Tone Descriptions

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This appendix briefly describes each ring tone supported by the Conexant ATA.

**PSTN Dial Tone:** This dial tone indicates there is PSTN/FXO support and the ATA has not been initialized for network and SIP signaling. It is also played when the ATA has lost network connectivity. No VoIP calls can be made when this dial tone is present.

**Peer2Peer (IP Dialing) Dial Tone:** This dial tone indicates registration is disabled voluntarily for an endpoint or registration has failed. Only Peer-To-Peer calls can be made when this dial tone is present.

**Normal Dial Tone:** This is the regular dial tone played when the user goes off-hook to make a call when registration has been successfully performed.

**Busy Tone:** When the remote end responds with a busy message for the user call request, this tone is played to indicate the called party is busy.

**Ring Back Tone:** The tone played to the calling party while the called party phone is ringing.

**Call Waiting Tone:** If a third party calls when the user is talking to the first party on the telephone, the ATA plays a Call Waiting Tone.

**Alerting Tone:** The alerting tone is played during dial pad configuration if there is a star service code which allows the user an option to enter a number. Examples of these services are blind transfer and call forwarding enable.

**Congestion Tone:** The ATA plays this tone when the SIP proxy server responds with a timeout indicating there was no progress in the attempted call.

**Fast Busy Tone:** The ATA plays this tone during a call when the remote end hangs up the phone.

**Confirm Tone:** This high frequency tone is played for a few milliseconds to indicate that a configuration change was successfully made through dial pad.

**Warble Tone:** After the remote end disconnects during a call, the fast busy tone is played. The high-pitched warble tone is subsequently played if the user fails to hang up the phone within the disconnect timeout.

**Unobtainable Tone:** The ATA plays this tone if the called party cannot be reached by the SIP proxy server or if the called party is not registered with the SIP proxy server.

**Recall Tone:** When the user puts the first call on hold to dial another number, this dial tone, which is like a normal dial tone with a few stutters at the beginning, is played to indicate that a call on hold exists.

**Stutter Dial Tone:** When unconditional call forwarding has been enabled, the ATA plays the stutter dial tone instead of the normal dial tone. Applicable only when the ATA is registered to a SIP proxy server.

**VMI Dial Tone:** When the user has Voice Mail Waiting at the SIP proxy server, the ATA will play the VMI dial tone instead of the normal dial tone.

## NOTES

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