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InstaVoIP Modules – Embedded Web Pages

For Use with ControlSpace® Designer

User Guide

Software Version 4.4.0

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Getting Started

This document describes the embedded web pages used in the Bose® EX-1280C to configure the VoIP features.

Overview

The web pages are designed for modern browsers, and make heavy use of JavaScript, JQuery, and AJAX queries.

A moderate level of login security is provided by enabling HTTP Digest Authentication. Although not as secure as SSL/TLS, this feature is supported in nearly all web browsers and HTTP clients and libraries, and is a common solution for handling authentication in embedded software.

⚠ ***CAUTION:*** *Digest Authentication can interact badly with a browser's page caching. This may cause some AJAX requests to fail, resulting in a page with no settings displayed or a totally blank page If this should happen, refreshing the page (either with the refresh button, F5, Shift-F5, or Ctrl-R depending on the browser) usually helps rectify the problem.*

Before You Begin

Follow the instructions in the documentation for your InstaVoIP application/device to set up and run the application/device.

Logging In

The default username and password are "admin" and "admin", respectively. The can be changed on the **System** page.

⚠ ***CAUTION:*** *Digest Authentication can interact badly with a browser's page caching. This may cause some AJAX requests to fail, resulting in a page with no settings displayed or a totally blank page If this should happen, refreshing the page (either with the refresh button, F5, Shift-F5, or Ctrl-R depending on the browser) usually helps rectify the problem.*

Navigating the Web Interface
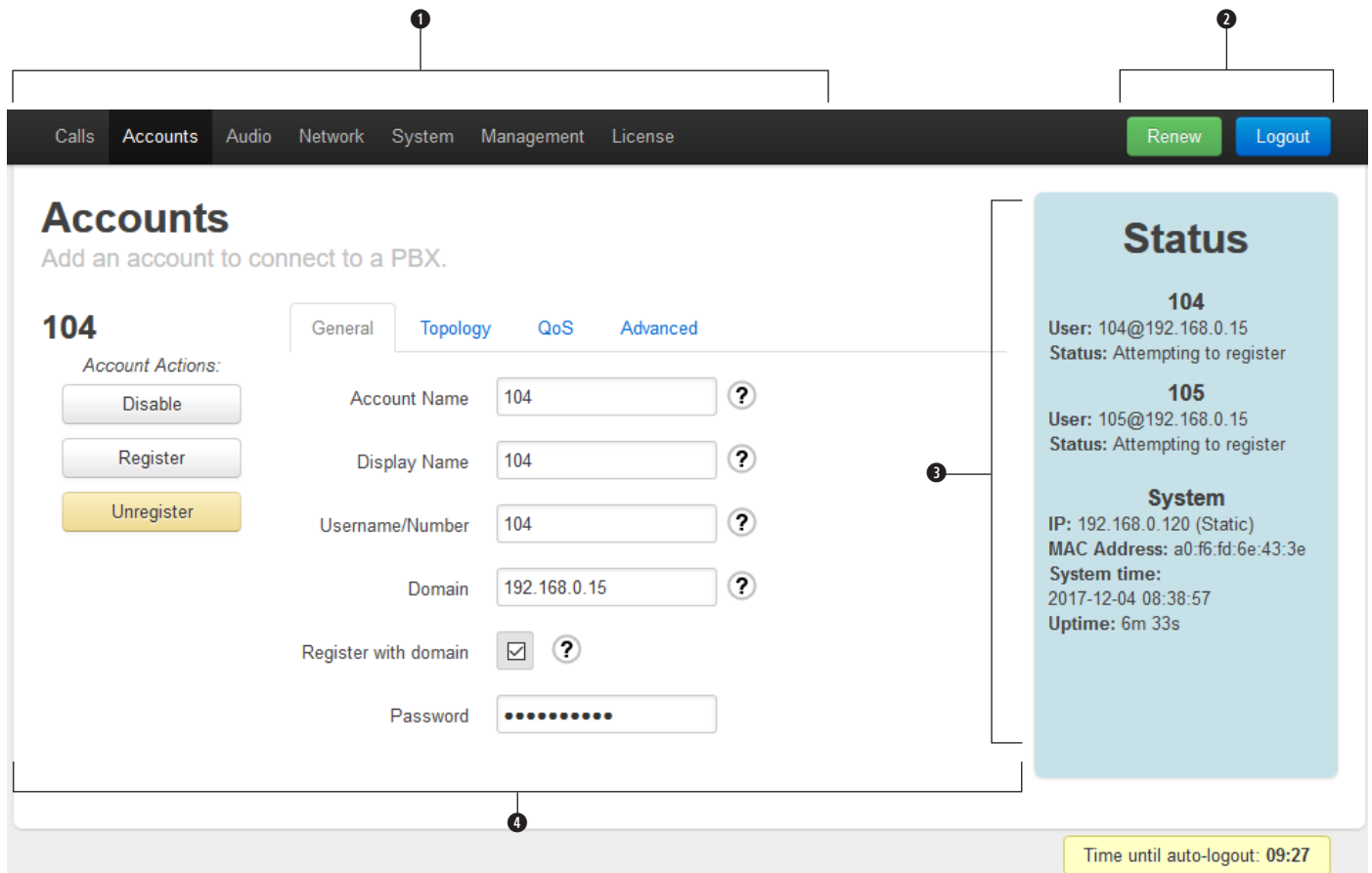
The Web Interface has the following elements, which are always present:

❶ **PAGE BAR** (or **TAB BAR**) – Used to navigate to each of the various configuration pages available in this application.

❷ **LOGIN CONTROLS** – Consisting of the "**Renew**" button (immediately extends the auto-logout countdown to the full duration of 10 minutes) and the "**Logout**" button (pops up a dialog box asking if the user is sure, and if so immediately logs out the user).

❸ **STATUS BAR** – Displays basic status about VoIP accounts (such as registration status), the system or application (such as uptime), and other application-specific details. May also display notification boxes when settings have been modified and must be saved, or when settings have been saved and require a restart of a service or the whole InstavoIP application/system.

❹ **PAGE CONTENTS** – Displays the information and configuration options for the page selected in the Page Bar.

To access different pages, click the page's name in the Page Bar.

The auto-logout function defaults to auto-logout after 10 minutes. The countdown display indicates how much time is remaining. To remain logged in for longer than 10 minutes, click the "Renew" button before the time has expired. To immediately log out, click the "Logout" button; you will be prompted whether or not you really wish to log out. When the auto-logout countdown reaches 0 seconds, you will be automatically logged out without prompting, and any unsaved configuration changes you have made will be lost.

To change settings, edit their values as normal (depending on whether the value is a checkbox, text field, etc.). After making changes, a popup in the Status Bar will indicate that there are unsaved changes. Click the "Save Changes" button to immediately save changes, or the "Revert" button to return all settings on the web page to their previous saved values.

After saving settings, most settings will take effect immediately. However, in order for some settings to take effect, it may be necessary to restart VoIP accounts (causing them to de-register with their previous servers and register anew with their newly-configured servers) or reboot the system. In these cases, a red popup in the Status Bar will indicate the type of restart that is required. Clicking the button will apply that restart. (There is no way to hide or cancel the popup, but you can choose to ignore it and continue using the InstaVoIP application with the old settings. Changing the settings again to have their previous values will not cancel the need for a restart.)

Using the Web Interface for Configuration

In the demo applications, the web interface is the primary and most user-friendly method for configuring settings in InstaVoIP.

Only general InstaVoIP configuration items, common to all InstaVoIP applications, are described here. Configuration items that are specific to a single application are described in that product's or application's documentation.

Accounts Page

The Accounts page is used for configuring VoIP accounts.

Currently only SIP accounts are supported, although InstaVoIP has a modular design that can support any type of VoIP account, such as IAX2 (Inter-Asterisk eXchange, a protocol used by Asterisk PBXs).

The Accounts page displays settings for all VoIP accounts in the system. Often there are only 1 or 2 accounts, although the InstaVoIP SDK can support unlimited accounts subject only to resource constraints (memory, CPU, and bandwidth).

Each SIP account's section has the following elements:

❶ **ACCOUNT NAME** – The name of this account (based on either the Account Name setting or Username and Domain settings, depending on which are configured)

❷ **ACCOUNT ACTIONS** – Controls for this account, which let you selectively disable/enable this account only, and register/deregister this account only.

❸ **ACCOUNT SETTINGS TAB BAR** – Because SIP accounts have many settings, most of which are not needed for ordinary configuration, the settings are divided into multiple tabs.

❹ **TAB CONTENTS** – Displays the configuration items for the tab selected in the Account Settings Tab Bar.

The most common settings for SIP accounts are on the General tab. Most users should not have to modify settings on the Topology, QoS, or Advanced tabs.

Typical SIP Account Settings

For the vast majority of users, only a few settings need to be configured for a SIP account:

- Enter your username, extension, or number on the General tab under "Username/Number".
- Enter the SIP server's name (DNS hostname or IP address) on the General tab under "Domain"
- Almost all users should leave "Register with domain" on the General tab **checked**

Often, these three settings are all that is needed to correctly configure a SIP account. However, some users may need to enter a few additional settings.

- Most SIP servers require authentication to prevent unauthorized access.
  - If your SIP server requires a password to register, enter this on the General tab under "Password".
  - Rarely, the authentication username is different from the registration username. If this is the case, enter the authentication username on the Advanced tab under "Auth Username".
- A few additional settings may also be required in rare cases:
  - If your InstaVoIP application/device is behind a NAT and requires a STUN server to communicate with external SIP servers or VoIP endpoints, enable the use of STUN for this account on the Topology tab by checking "STUN".
  - If your SIP server operates on a port different from the standard port (5060 for normal SIP, 5061 for Secure SIP), enter this on the Advanced tab under "Proxy port".
  - If your SIP server operates through a SIP proxy, on the Advanced tab enter the proxy's hostname or IP address under "Proxy server". If the proxy operates on the standard port (5060 for normal SIP, 5061 for Secure SIP), you can leave "Proxy port" set to 0; otherwise, enter the port under "Proxy port".

⚠ **CAUTION:** *Settings other than the ones mentioned above should only be changed if you know they are required. Using inappropriate values for some settings may prevent InstaVoIP from being able to communicate with the SIP server or VoIP endpoints!*

General Tab

Account Name – The name used to identify this account. It will only be shown on this page and in system log messages.

Display Name – Equivalent to "Caller ID"; the name to report to the SIP server and to other callers as your personal name.

Depending on the SIP server's configuration, it may ignore this setting and apply a display name from its own database, or ignore the display name entirely.

Username/Number – Equivalent to an "extension" or "phone number"; the number, extension, or address this phone is configured with on the SIP server.

Domain – The address of the SIP server. This can be a DNS name (like "sip.company.com") or numeric IP address (like 12.34.56.78).

Register with domain – Determines whether InstaVoIP will register with a SIP server, or operate in peer-to-peer (P2P) mode.

In most cases, a centralized SIP server coordinates VoIP calls among endpoints (phones) that register directly with it, and sometimes with external services such as a connection to PSTN (public switched telephone network) through a
"termination service".

This is the normal method of using SIP. Most users should therefore leave the Register with domain setting **checked**.

In some cases, a centralized SIP server is not used; this is usually called **peer-to-peer mode (P2P)**. In this case, dialing a SIP URI requires knowing both the username/extension and the address of the peer, which are entered like
<username@server.address.com>. Calls cannot be dialed by entering only a username, since there is no server to route the call.

Password – The authentication password to use when connecting to the SIP server.

Topology Tab

SIP Transport Mode – The method used to communicate with the SIP server. Automatic allows InstaVoIP to select an appropriate mode, which will usually result in UDP (which is the normal method for SIP servers) but may attempt TCP if UDP fails. TCP forces the use of a TCP connection.

Local Port – The UDP or TCP port that data will be sent from when communicating with a SIP server. A setting of 0 means to choose a local port randomly, which is the normal behavior.

STUN – Enables use of a STUN server to determine the InstaVoIP application/device's public IP address when it is behind a NAT. This only enables the use of a STUN server; to configure the STUN server's address and port, use the settings on the **Network page**.

QoS Tab

These configure Quality-of-Service settings, which lets some routers prioritize VoIP data above normal data when bandwidth is constrained.

CAUTION: The default values are set to standard values recognized by all compliant routers. Modifying these values could prevent the InstaVoIP application/device from being able to communicate with other SIP servers or VoIP endpoints!

Audio RTP DSCP – The DSCP value for RTP audio data.

Call Control DSCP – The DSCP value for SIP data.

Advanced Tab

General	Topology	QoS	Advanced

Auth Username [   ]

Proxy [   ]

Proxy Port [0   ]

Registration Lifetime [3600   ]   seconds

Keep-Alive [☐]

Silence Suppression [☐]

Auth Username – The username to use for authenticating with the SIP server. If this is left blank, the "Username/Number" from the General tab will be used as the authentication username, which is the most common scenario.

Proxy – If a SIP proxy is being used, this is the hostname or IP address of the SIP proxy. When no SIP proxy is in use, leave this setting blank.

Using the Web Interface for Configuration

Proxy Port – The port of the SIP server or SIP proxy. A setting of 0 means to use the standard port (5060 for normal SIP servers, 5061 when Secure SIP is being used).

Follow these instructions to set the SIP server and SIP proxy correctly:

- SIP server at "sip.server.com" on the standard port
    - Domain – sip.server.com
    - Proxy – (leave blank, which is the default)
    - Proxy Port – 0 (default) or 5060
- SIP server at "sip.server.com" port 1234 (nonstandard)
    - Domain – sip.server.com
    - Proxy – (leave blank, which is the default)
    - Proxy Port – 1234
- SIP proxy at "proxy.server.com" on the standard port, SIP server at "sip.server.com" on the standard port
    - Domain – sip.server.com
    - Proxy – proxy.server.com
    - Proxy Port – 0 (default) or 5060
- SIP proxy at "proxy.server.com" on the standard port, SIP server at "sip.server.com" port 1234 (nonstandard)
    - Domain – sip.server.com:1234
    - Proxy – proxy.server.com
    - Proxy Port – 0 (default) or 5060
- SIP proxy at "proxy.server.com" port 9876 (nonstandard) , SIP server at "sip.server.com" on the standard port
    - Domain – sip.server.com
    - Proxy – proxy.server.com
    - Proxy Port – 9876
- SIP proxy at "proxy.server.com" port 9876 (nonstandard) , SIP server at "sip.server.com" port 1234 (nonstandard)
    - Domain – sip.server.com:1234
    - Proxy – proxy.server.com
    - Proxy Port – 9876

Registration Lifetime – How long to request registration with the SIP server to last. This value is only a request; the SIP server may impose a minimum or maximum beyond this value. Whatever values the SIP server permits, registration renewal is attempted several times before the end of the duration, usually starting halfway through the duration.

Keep-Alive – Enables sending periodic keep-alive data to the SIP server, to ensure that the communication channel remains open. (When running behind a NAT, the NAT may assume the channel is no longer valid and close it prematurely; enabling keep-alive prevents this from happening.)

Silence Suppression –Enables detecting when the outgoing audio on calls using this account is nearly silent, and ceasing transmission of RTP audio until voice is detected again. This conserves bandwidth, but requires additional CPU on the InstaVoIP device and is not well supported by all VoIP endpoints. (Also called **VAD (Voice Activity Detection)**, although VAD is technically a distinct but necessary component of silence suppression.)

Settings Not Found in the Accounts Page

A few settings typically associated with SIP accounts are actually global settings and are not found on the Accounts page.

STUN server – Although the use of a STUN server is enabled on a per-account basis, only a single STUN server needs to be configured for the whole InstaVoIP application. This is configured on the **Network page**.

RTP ports – The ports used to receive RTP audio are configured globally for the whole InstaVoIP applications. This is configured on the **Network page**.

Audio Page

The Audio page has settings for digital audio codecs.

Codec Selection

This selection displays a list of available and preferred digital audio codecs.

Digital audio codecs are ways of encoding audio for transmission between computers. Depending on the algorithms used by a codec, different codecs all have varying levels of sound quality, bandwidth usage, and CPU usage.

Not all VoIP implementations support the same set of codecs. Therefore, VoIP devices always indicate to other systems their preferences for codecs to use.

Available – Lists all codecs that this InstaVoIP application supports.

Codecs are identified by name, such as "G.711 uLaw" or "DVI4".

Some basic codecs such as G.711 only support **narrowband audio** (8 kHz sampling rate, allowing audio to be heard up to 4 kHz), sometimes called "NB", "voiceband", or "standard definition". Other codecs support higher sampling rates, such as **wideband audio** (16 kHz sampling rate, allowing audio to be hard up to 8 kHz, sometimes called "WB", "HD", or "high definition") or **ultra-wideband audio** (32 kHz sampling rate, allowing audio to be heard up to 16 kHz, sometimes called "UWB", "UHD", or "ultra-high definition"). InstaVoIP also supports a 48 kHz sampling rate (allowing audio to be heard up to 24 kHz), which it refers to as "CD".[1]

InstaVoIP applications have a compile-time setting for the maximum sampling rate they permit, which is often either 8 kHz (narrowband) or 16 kHz (wideband). **Codecs above the InstaVoIP application's permitted sampling rate will not be displayed in the list**.

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[1] InstaVoIP does not currently support the true sampling rate used on Compact Discs, 44.1 kHz. When support for this added, the designation for 48 kHz codecs will be modified to reflect this.

Preferred – Lists codecs that may be used during a call, in order from most preferred (top) to least preferred (bottom). A codec that is in the Preferred list is **enabled**, and a codec which is not in the Preferred list is **disabled**.

To add a codec to the Preferred list, double-click the desired codec in the Available list. You can also click the codec in the Available list to highlight it and click the "Enable" button.

To remove a codec from the Preferred list, double-click the desired codec in the Preferred list. You can also click the codec in the Preferred list to highlight it and click the "Disable" button.

To change the order of codecs in the Preferred list, click the codec you wish to move to highlight it. Then click the "Move Up" button to move it higher in the list (making it more preferable), or click the "Move Down" button to move it lower in the list (making it less preferable).

Codec Last Used – Shows the most recent digital audio codecs used by the system, both Rx and Tx.

Recommended audio codecs

As per Internet Standard RFC 7007, all VoIP devices SHOULD include G.711 uLaw as one of their preferred codecs. This is the standard audio codec for PSTN telephones in North America and Japan. It requires very little CPU to process, but it is comparatively bandwidth-intensive and limited to only narrowband audio. We suggest always enabling this codec, but making it the least-preferred codec (or second-least, if G.711 aLaw is enabled) in the Preferred list.

G.711 aLaw is similar to G.711 uLaw, with only minor differences in the range of signals it can encode. It is the standard audio codec for PSTN telephones in Europe, and by convention is used for international calls if at least one country uses it. We suggest enabling this codec, but making it the least-preferred codec in the Preferred list.

Description of available codecs

G.711 uLaw and **G.711 aLaw** (sometimes called "PCMU" and "PCMA", respectively) are the standard codecs for PSTN telephony. As such, they offer identical quality to land-line telephones. Their CPU usage is very low, but bandwidth is high at 64 kbit/s.

G.722 is a wideband audio codec with the same bandwidth requirements as G.711.

G.726 has only slightly reduced quality compared to G.711, but uses around half the bandwidth (depending on the bitrate chosen). "Fixed payload" is the most common bitrate (32 kbit/s), but it can also be run at 16, 24, or 40 kbit/s.

DVI4 (occasionally called "ADPCM DVI" or "IMA ADPCM") was formerly recommended as a standard codec for all VoIP devices to support. It also supports wideband and ultra-wideband audio, providing superior audio quality over G.711 using roughly half the bandwidth and without much additional CPU. However, it is not widely used today, and few VoIP devices support it.

Linear PCM codecs perform no compression. As a result, they have the highest audio quality and least CPU usage, but also consume the most network bandwidth. They are rarely used in VoIP, and we do not suggest enabling them for ordinary usage.

iLBC (Internet Low Bitrate Codec) is designed to offer acceptable audio quality with low bandwidth (15.2 kbit/s for iLBC-20 and 13.33 kbit/s for iLBC-30), and good robustness against lost data packets. In bandwidth-constrained scenarios, it may be useful. However, as its audio quality is comparatively low and it is CPU-intensive, it has limited value for average users.

Speex is optimized for voice communication over Internet networks, as opposed to many other codecs that were designed for PSTN or mobile networks. It is commonly supported in open-source applications. Its developers, Xiph.Org, consider it obsolete; its successor is Opus, which offers better performance in every aspect.

SILK was originally developed for Skype, but has since been made freely available. It offers high audio quality for voice communication and low latency, but is very CPU-intensive. It has been superseded by Opus, which incorporates SILK's algorithms but adds support for high-fidelity music.

Network Page

The Network page is used to configure network settings, such as IP addresses, STUN server, and RTP ports.

Network
Network settings for the device.

WAN
Outgoing network settings

General

Host	
Domain	
Connection Type	○ Dynamic IP
	◉ Static IP

Static IP Address

Address	192.168.0.120
Mask	255.255.255.0
Default Router	192.168.1.1
DNS Primary	0.0.0.0
DNS Secondary	0.0.0.0
DNS Tertiary	0.0.0.0

VLAN

Enabled	☐
ID	0

STUN
Global STUN server settings

Server	
Port	3478

RTP
Configure Port Range

Port Range Start	23456 to 23556

WAN

The WAN section configures IP address settings and other features of the WAN interface.[2]

Host – This is the system hostname that is reported to applications and services that require one, such as DHCP or DNS. This setting is generally not needed for most InstaVoIP users and can be left as the default setting.

Domain – This is the DNS domain that InstaVoIP is part of. This value may be ignored if DHCP is used to obtain an IP address automatically. This setting is generally not needed for most InstaVoIP users and can be left as the default setting.

Connection Type – **Dynamic IP** enables DHCP, which attempts to contact a DHCP server to automatically obtain an IP address and DNS servers. **Static IP** enables manual configuration of IP address and DNS servers.

Dynamic IP Address

| Connection Type | ⦿ Dynamic IP |
| | ◯ Static IP |

There are no configuration settings for Dynamic IP.

Static IP Address

Static IP Address	
Address	192.168.0.120
Mask	255.255.255.0
Default Router	192.168.1.1
DNS Primary	0.0.0.0
DNS Secondary	0.0.0.0
DNS Tertiary	0.0.0.0
VLAN	
Enabled	☐
ID	0

Address – The IP address of this system.

Mask – The subnet mask for this IP address, such as "255.255.0.0".

Default Router – The default gateway, or first hop, for this IP address.

DNS Primary/Secondary/Tertiary – The IP addresses of one or more DNS servers to use for name resolution.

VLAN Enabled – Select to enable VLAN, if used.

VLAN ID – ID of VLAN, if used.

⚠ **CAUTION:** *Setting an MTU Size other than 1500 may render the InstaVoIP devices incapable of communicating with any other devices on your network.*

[2] The use of "WAN" (rather than "LAN") as the name of the primary network interface is due to InstaVoIP's support for acting as a gateway router. InstaVoIP's "WAN" interface is always the external interface which has access to the Internet, or at least to a wide but private network. InstaVoIP's "LAN" interface, in applications that have one, is the internal private network.

STUN

A STUN server is used to allow a host to discover its public IP address if it is located behind a NAT. This may be required for SIP or RTP to work properly.

Each VoIP account can be individually configured whether or not to use a STUN server; this is done from that account's settings on the Accounts page. However, the system only requires a single STUN server to be configured, which is done here.

STUN

Global STUN server settings

Server [   ]

Port [3478]

Server – The hostname or IP address of the STUN server.

Port – The port of the STUN server. The standard port for STUN is 3478.

RTP

InstaVoIP uses a global port range for all incoming RTP streams. It is not possible to configure VoIP accounts to have their own distinct RTP port ranges.

Outgoing data streams must necessarily be sent to the port specified by the peer. There is no way to configure or control this locally; only the peer can configure its incoming ports.

Although Internet Standard RFC 3551 specifies the use of port 5004 for RTP, there are multiple reasons why InstaVoIP supports configuring a range of ports:

- Supporting multiple independent calls simultaneously requires the use of multiple ports, since there is no signaling in SIP to indicate which RTP stream (identified by its SSRC) belongs to a particular call.
- In the process of traversing a NAT, the NAT must assign a different port number if port 5004 is already in used by another host behind the NAT.

In summary, the port range chosen for incoming RTP is largely arbitrary, and most users should not need to change the default settings.

RTP

Configure Port Range

Port Range Start [23456] to [23556]

Port Range – The start and end port numbers to use for incoming RTP streams. These should be even port numbers.

System Page

The System page has general administrative settings.

System
Various system specific settings

Authentication
Administrative user account

Username admin (?)

Change Password

Syslog
System logging
configuration

Report to server ☐

Server [   ]

Port 514

Date & Time
System date and time
configuration (Time Server /
NTP)

Enabled ☑

Server 0.debian.pool.ntp.org

Daylight savings ☐

Time Zone

(GMT-05:00) Eastern Time (US & Canada)

Set Date & Time

Authentication

This configures the authentication username and password, which are used to access the InstaVoIP web interface and command-line shell (via Telnet or UART).

Syslog

This configures Insta VoIP to send all of its log messages to a syslog server, which can display or archive the messages.

Report to server – When checked, enables the configuration options for sending log messages to a syslong server.

Server – The hostname or IP address of the syslog server.

Port – The port of the syslog server. The standard port for syslog is 514.

Date and Time

Because InstaVoIP hardware devices do not have battery-backed clocks, they must determine the current date and time anew every time they are restarted.

Enabled – When checked, enables using an NTP server to determine the current date and time. Disabling this option will leave the InstaVoIP device without a method for knowing the current date and time, which could result in it being unable to interoperate with other devices or network services.

Server – The NTP server to use for establishing the current date and time.

Daylight savings – Whether daylight saving time (also called "summer time") is currently in effect. This setting must be changed twice annually (once when entering daylight saving time, and once when returning to standard time).

Time zone – The local time zone of this InstaVoIP device, which specifies its offset from UTC and its additional offset during daylight saving time, if any.

Management Page

The System page has administrator functions for managing the entire InstaVoIP configuration and for upgrading a hardware InstaVoIP device's firmware.

Configuration

The section has controls for backing up (downloading) the complete configuration currently saved in the InstaVoIP application or device, and for restoring a previously backed up configuration.

Backup/Download – Click "Download configuration" to download complete configuration currently saved in the InstaVoIP application or device.

Restore/Upload – Use the uploader to select a previously backed up configuration file and restore it to the InstaVoIP application or device. After performing the upload, the settings will be saved, but will not take effect until the InstaVoIP application or device is restarted. **This will not be indicated on the web interface.**

Version

The version of VoIP application and library currently running.

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