



## **Application User Manual**

### **Configuring and Using SIP-RAVA Client V02.12D**

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# 1 Glossary

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## CGI

The Common Gateway Interface is a standardised method used by a Web server to interact with specific application (ex. a database). A CGI script processes the request from the Web server to a database, gets the output and returns it to the Web client.

## DHCP

Short for Dynamic Host Configuration Protocol, a protocol used to assign an IP address to a device connected to a Network.

## DTMF

Short for Dual Tone Multiple Frequency, a signaling method developed by Bell Labs for sending telephone dialing information over the same analog, voice-quality phones lines that carry voice.

## FDX

Short for Full Duplex. A full-duplex system allows communication between two parties to be done in both directions simultaneously. See [this](#) Wikipedia article for more information.

## HDX

Short for Half Duplex. In a half-duplex system simultaneous communication in both directions is not possible, i.e. if one of the parties is sending, the other have to wait it to finish before replying. See [this](#) Wikipedia article for more information.

## PBX

Private Branch Exchange; a telephone system that handles the internal and external calls of a building, company, etc.

## RTP

Real time transport protocol; a layer added to the internet protocol (IP) used for transporting media content in real-time to reduce latency issues.

## SIP

The Session Initiation Protocol (SIP) is a signalling protocol, widely used for controlling multimedia communication sessions such as voice and video calls over Internet Protocol (IP).

## UAC

SIP terminology, means “user agent client”, to indicate generally the device that start a transaction, sending a request to a server (UAS).

## UAS

SIP terminology, means “user agent server”, to indicate generally the device that start a transaction, receiving a request from a client (UAC).

## 2 Introduction

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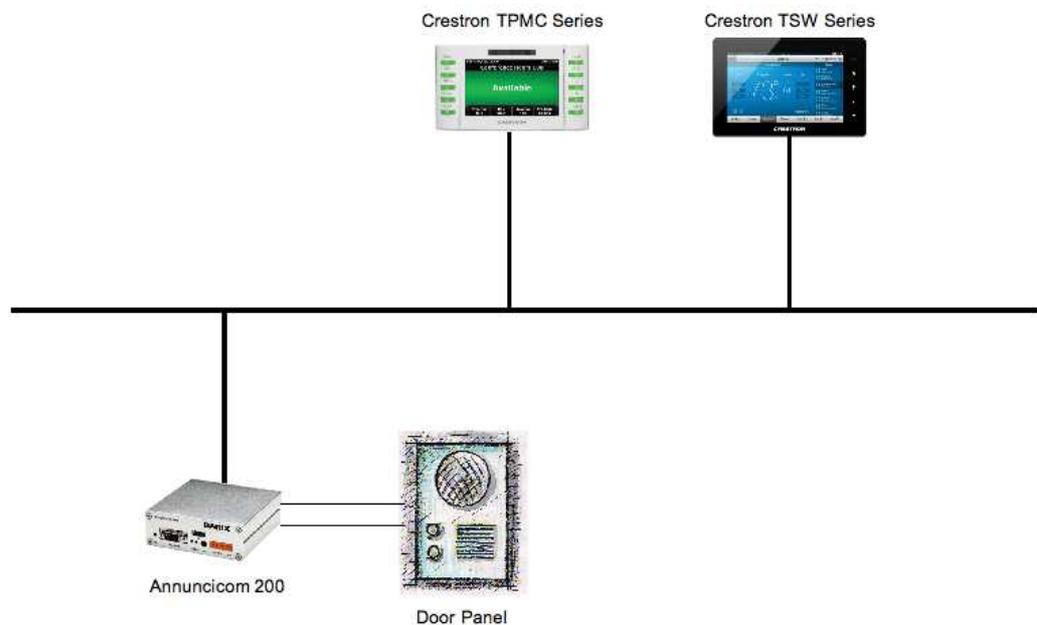
### 2.1 About the “SIP-RAVA Client”

The SIP-RAVA Client Application is a customised version of the standard Barix SIP Client application, optimised to work in a peer to peer environment using a standard “telephone fashion” voice over IP communication, using the widely used application-layer control protocol known as "SIP" (Session Initiation Protocol, RFC 3261), and the RAVA based SIP extension provided by CRESTRON.

A basic knowledge of the SIP protocol features and terminology is required in order to fully understand this document.

Although the main focus of the SIP-RAVA Client application is the Peer to peer use case, it can be also configured to work in a proxy based (PBX) connection mode.

An example SIP-RAVA system is shown below:



*Illustration 1: SIP-RAVA System Overview*

In this case the Barix SIP-RAVA client is being used as a door-station, while the Crestron panels can be installed in several rooms around the building. Pressing the door-station call button will ring those devices assigned to the device via its configuration. A SIP peer-to-peer call will be established with the first device to answer. The door-station relay can be remotely activated by the entry of a DTMF code.

### 2.2 General features

- Supports profiles for easy configuration
- Configurable destination number to call for each input contact
- Configurable call “pick up”/”hang up” timeout interval
- Configurable “call/close on level” feature

- Configurable “peer to peer” or “proxy based” connections
- Configurable SIP and audio RTP ports
- Support for G.711 and G.722 audio with Acoustic Echo Cancellation (AEC)
- DTMF door open key sequence
- Configurable relay activation at call answer/call ring
- Configurable use of beep tone on call answer
- Serial and CGI control interface
- Profile based WEB configuration UI
- 10/100 Mbit Ethernet connection supports automatic network configuration (BOOTP, DHCP, IPzator, and as well as manual static IP configuration)
- Syslog debugging messages.

## 2.3 SIP features

- SIP (RFC 3261) compliant architecture:
- Processed methods and status messages: REGISTER, INVITE, CANCEL, BYE, INFO (for DTMF), ACK
- Processed status messages:180, 200, 401, 403, 407, 503, 603
- Supports Peer to Peer mode which enables any two SIP user agents (UAS/UAC) to communicate directly with one another without the mediation of a SIP server
- Can register to a SIP Registrar (PBX)

## 2.4 RAVA related features

- Support for Crestron over Internet Protocol (CIP). The following CIP messages are supported:

<i>Message type</i>	<i>RX</i>	<i>RX Action</i>	<i>TX</i>	<i>TX Action</i>
0x14	Yes	Reply with 0x15	No	
0x15	No		Yes	Sent in response of 0x14
0x16	Yes	Reply with 0x16	No	
0x17	No		Yes	Sent in response (unicast) of 0x16, and broadcasted 3 times at device boot
0x18	No		No	
0x19	No		No	
0xA6	Yes	Check if its ID is requested, and reply with 0xA7	Yes	Broadcast this query to get the IP to start a SIP call
0xA7	Yes	If in reply of its own 0xA6 request, then start SIP call	Yes	Sent in response of received 0xA6 query
0xA8	Yes	Check if its own SIP group is requested, and reply with 0xA9	Yes	Broadcast this query to collect info about all locations it is trying to reach
0xA9	Yes	Build IP table (max 8) of all locations, and start a group INVITE to all	Yes	Sent in response of received 0xA8 query

- Supports CIP IP\_ID Tables of maximum 8 entries
- Receiving SIP Multicast paging
- SIP Group call (sending multiple INVITE SIP messages, one active call only) up to 8 peers

## 2.5 Supported hardware

The SIP-RAVA Client solution is designed to run on the Annunicom 200-RAVA device.

While the application may run on other Barix devices, its complete functionality is not guaranteed in such a case.

## 2.6 Additional documents

Technical specifications for the supported devices can be found in the corresponding product sheet which can be downloaded from the Barix site [www.barix.com](http://www.barix.com).

The SIP-RAVA Client Application has been developed in BCL and is available with its source code enabling users to further customise it. For detailed technical information about the BCL programming language please download the “Barix Control Language (BCL) Programmers Manual” from the Barix website.

## 2.7 ABCL SIP firmware

Barix provides the SIP-RAVA Client application as included in a pre-built ABCL SIP (abcl\_sip\_vXX.XX\_DATE) package. The SIP-RAVA Client is guaranteed to work only when it is run with its original ABCL package. Using the SIP-RAVA Client with a different version of the ABCL package may not work correctly.

## 2.8 About this manual

### Links to chapters

References to chapters (e.g. [X Chapter name](#)) are red and underlined and serve as direct links when viewed in Adobe Acrobat Viewer. Click on the link to jump to the referenced chapter, click on the left arrow icon to jump back to where you came from.

### Bookmarks pane in Adobe Acrobat

The complete “Table of Contents” is available in Adobe Acrobat Viewer. Click on the “Bookmarks” pane tab on the left side of Adobe Acrobat Viewer to open it. Click on any bookmark to directly jump to the corresponding part of the manual.

### Chapter overview

This manual is divided into the following chapters:

- [Chapter System requirements](#) describing some basic SIP environment and system configurations.
- [Chapter Error: Reference source not found](#) explaining how to quickly set up the device for use with the SIP client.
- [Chapter Error: Reference source not found](#) giving a detailed guide for the WEB UI SIP-RAVA Client configuration.
- [Chapter Extra features in the SIP-RAVA Client application](#) describing in details how to use some of the advanced features provided by the application.
- [Chapter Common issues](#) explaining how to solve the most common issues to have the application working.

- [Chapter Additional Information](#) explain how to update a firmware package through the WEB UI, how to setup the device network interface, and how to rescue a device via the serial port

## 3 System requirements

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### 3.1 SIP environment

The SIP-RAVA Client application can run in the following configurations:

- Peer to peer mode:
  - receiving and initiating RAVA based SIP calls;
  - receiving RAVA based SIP multicast paging
  - initiating RAVA based SIP group call
- standard SIP “user agent client/server” (UAC/UAS) in a PBX environment.

### 3.2 SIP server compatibility

The SIP-RAVA Client application has been validated with the following SIP Proxy/Registrar Servers:

- Asterisk 1.4.29
- OpenSER 1.3.2-notls (i386/linux)
- FreeSWITCH 1.0.head (git-00207ce 2010-09-30 02-37-57 -0400)
- Cisco CUCM 8.6

## 4 SIP-RAVA WEB UI configuration

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This chapter explains in detail all configuration options for the SIP-RAVA Client, basic use of the application is explained in the SIP-RAVA Quick Start Guide document available from the Barix website.

### 4.1 Home page

Using an Internet browser, connect to the device by writing the IP address in the browser URL bar, the application home page will be displayed, showing the most important configuration and status information grouped in the following sections:

HOME	PROFILES	CONFIGURATION	STATUS	DEFAULTS	UPDATE	REBOOT	Annuaicom 200 MAC: 00:08:E4:02:9F:A7 FW V01.11
------	----------	---------------	--------	----------	--------	--------	--

# SIP-RAVA™ CLIENT

V2.12D(08 Apr 2013)

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**SIP Door Station**

**APPLICATION STATUS**

Application Mode: Peer to Peer

Call State: Idle

P2P Remote Party:

---

**AUDIO STATUS**

Current Set Volume: 0 %

Left Output Peak Level: 0 dBFS

Right Output Peak Level: 0 dBFS

Left Input Peak Level: 0 dBFS

Right Input Peak Level: 0 dBFS

---

**DEVICE & XB I/O STATUS**

I/O Contacts

	7	6	5	4	3	2	1	0
Inputs	⊗	⊗	⊗	⊗	⊗	⊗	⊗	⊗
Relays	⊗	⊗	⊗	⊗	⊗	⊗	⊗	⊗

**Help**

**Home page**  
Gives an overview of the most important settings of the unit.

**APPLICATION STATUS**

**Application Mode**  
Shows the current mode of the application, and may take the following values:

- *Device is still booting ...*  
The Boot process has not finished yet.
- *SIP mode*  
The device is in SIP mode. The SIP server name, and the SIP ID are also shown in this case
- *Peer to peer mode*  
The device is in P2P mode, and configured to call to only one remote peer. Incoming calls will be accepted only from this peer.

**Time till next Registration**  
Shows the remaining time till the next registration attempt. The current registration status is shown with different colours of the text:

Illustration 2: SIP-RAVA Home Page

## APPLICATION STATUS

### Application mode

Shows the current mode of the application, and may take the following values:

- **Device is still booting:** The Boot process has not finished yet
- **SIP mode:** The device is in SIP mode. The SIP server name, and the SIP ID are also shown in this case. Direct IP (P2P) calls are ignored
- **Peer to peer mode:** The device is in P2P mode, and configured to accept and make direct IP calls

### Time till next Registration

If the device is configured to use a SIP server it will show the remaining time till the next registration attempt. The current registration status is shown with different colours of the text:

- **Device not registered**
- **Registration in progress**
- **Device Registered**

### Call Status

Shows the current call state, and may take one of the following values:

- **Idle:** No audio stream is being received, and the SIP client is able to accept calls;
- **Getting incoming call:** The SIP client has received an INVITE message and is ringing. The ID of the remote party is also displayed;
- **Outgoing call:** The SIP client has initiated an outgoing call, and the remote party ID is displayed as well;
- **In active call:** The SIP client is in a call session with the displayed remote party.

## DEVICE I/O STATUS

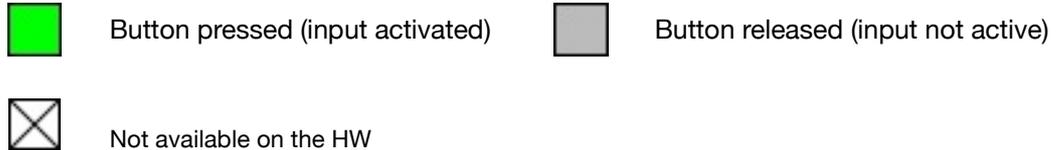
Displays the available I/Os of the device.

### **Device I/O**

Device I/O enumeration; bit "0" corresponds to the first device input ( for example IN0 for Annunicom 200-RAVA).

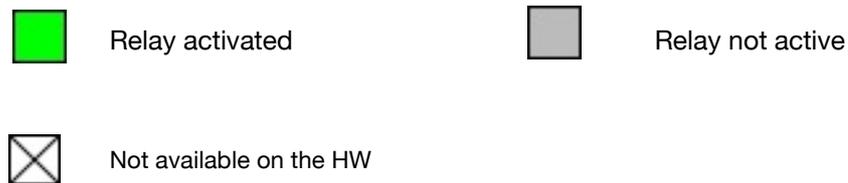
### **Inputs**

The following icons are used to indicate the inputs state:



### **Relays**

Monitors the device relays, indicated with the following states:



## AUDIO STATUS

Shows the status of the following audio parameters:

- **Volume:** Displays the current volume in %
- **Left output peak level:** Displays the current left output peak level in dBFS (dB full scale)
- **Right output peak level:** Displays the current right output peak level in dBFS (dB full scale)
- **Left input peak level:** Displays the current left input peak level in dBFS (dB full scale)
- **Right input peak level:** Displays the current right input peak level in dBFS (dB full scale)

## 4.2 Profiles Page

The Profiles page lets you easily set the application for a given purpose. When a profile is selected, some of the non relevant settings are set to default and hidden, and some are left unchanged, so use this feature with caution. After applying the desired profile, the SIP client should be further customised only for those fields described in this section.

All profiles allow access to the complete set of settings from the following sections:

- SIP Protocol Settings
- Door Settings

- Control Interfaces Settings

In addition, each profile provides extra customisation settings.



Illustration 3: SIP Client Profiles page

The following profiles are available:

### SIP Phone

This is the typical use of the SIP client. In this profile you can:

- Establish a full-duplex phone communication;
- Call up to 2 extensions
- Select the audio input

The following additional options are enabled in this profile:

- Close on timeout
- Close on Input 0 and Input 1
- Input audio buffer level
- Audio source input type
- Encoding
- Volume
- Mic and A/D gains

### SIP Door Station

The device is to be used as a door intercom station. Pushing the button causes the device to ring. If the call is not answered the device will close the call automatically after a preconfigured timeout. In this mode the SIP client can be used in half duplex mode with door panels. The phone pickup mode is preset to “*auto answer*” and cannot be changed.

The following additional options are enabled in this mode:

- Close on timeout
- Close on Input 0 and Input 1
- Input audio buffer level

- Audio input type
- Encoding
- Volume
- Mic and A/D gains
- Talk mode (HDX or FDX)
- Output trigger level and trigger level timeout for the Voice Activity Detection (VAD)

## 4.3 Configuration Page

The configuration page consists of several areas - for the menu navigation, settings, and help.

The menu navigation section contains two parts - Basic and Advanced. While the Basic menu shows only the most essential settings needed to initially configure the application, the Advanced menu gives access to all settings available for the selected application profile.

When the required changes have been made, click the "Apply" button to save the changes. The device will reboot

**NOTE:** The changed settings are not preserved if you switch between the *Basic* and *Advanced* menus. Make sure you apply the changes you have already done before switching to the other submenu!

The screenshot shows the 'SIP-RAVA™ CLIENT' interface. At the top, there is a navigation bar with tabs: HOME, PROFILES, CONFIGURATION, STATUS, DEFAULTS, UPDATE, REBOOT. The current page is 'SIP Door Station' under 'CONFIGURATION'. The main content area is titled 'SIP PROTOCOL SETTINGS' and includes the following settings:

- Peer to Peer:** Radio buttons for No (selected) and Yes.
- SIP ID (username):** An empty text input field.
- Listen SIP Port:** Input field with '0'.
- Listen RTP Port:** Input field with '0'.
- Default REGISTER Time:** Input field with '1800'.
- Send NAT-keepalives:** Radio buttons for No (selected) and Yes.
- Periodically Renew DNS:** Radio buttons for No (selected) and Yes.
- Debug Mode:** Radio buttons for Off (selected) and On.
- CIP PROTOCOL SETTINGS:**
  - CIP Port Number:** Input field with '4794'.
  - SIP Groups:** Input field with 'ALL'.
  - SIP Display ID:** An empty text input field.

On the right side, there is a 'Help' panel titled 'SIP PROTOCOL SETTINGS' with detailed instructions for each setting, such as 'Peer to Peer: Choose whether peer to peer calls should be allowed.' and 'SIP ID: Enter the SIP ID (username) that has been created for this device.'

Illustration 4: SIP Client Advanced Configuration page

Below is the full list of configuration options available in the Advanced menu. Please be aware that not all of them may be visible, depending on your selected profile.

## NETWORK SETTINGS

### Use SonicIP

If set to "yes", the device will announce its IP address over the audio output during device startup.

Default: "yes"

### **IP Address**

Enter the 4 values of the desired device IP address e.g.:

"0.0.0.0" for automatic discovery (DHCP/BOOTP, IPzator, AutoIP)

"192.168.0.12" for an internal LAN

Default: "0.0.0.0"

### **Netmask**

Enter the 4 values of the desired Static IP e.g.:

"0.0.0.0" for a default Netmask depending on the used IP Address.

"255.255.255.0" for a C class network

Default: "255.255.255.0"

### **Gateway IP Address**

Enter the 4 values of the desired Gateway IP address e.g.:

"0.0.0.0" for no Gateway

"192.168.0.1" for a Gateway in a LAN

Note: The Gateway has to be set only when connecting to other devices over the WAN (through a router).

Default: "0.0.0.0"

### **Primary DNS**

In this field you can give the desired primary and alternative DNS IP address to be able to connect to URLs (e.g. www.radio.com).

Example: "195.186.1.111"

Default: "0.0.0.0"

### **Alternative DNS**

In this field you can give the desired alternative DNS IP address in case the primary DNS is not reachable.

Example: "195.186.4.111"

Default: "0.0.0.0"

### **Syslog Address**

Destination address for syslog messages sent by the BCL program via the SYSLOG command. Set this to your syslog logging machine, if your syslog messages are

recorded centrally.

If set to 0.0.0.0, syslog messages are broadcast.

Default: "0.0.0.0"

### **DHCP Host Name**

Name of the device to be sent in the DHCP requests. If left empty, the device will use internally an automatically generated name based on its MAC address (this one will not be displayed to the user). Enter up to 15 characters.

### **Web Server Port**

Defines the port where the web server of the device can be reached. If set to "0" the default HTTP port (80) is used.

### **SNMP System Name**

SNMP MIB entry for system name (system.sysName.0)

### **SNMP System Location**

SNMP MIB entry for system location (system.sysLocation.0)

### **SNMP System Contact**

SNMP MIB entry for system contact (system.sysContact.0)

This parameter can be queried using any SNMP browser but can not be updated.

## **SIP PROTOCOL SETTINGS**

### **Peer to Peer**

Choose whether peer to peer calls should be allowed.

### **SIP Server (PBX) / Remote Peer**

Enter either the hostname/IP address of a SIP server, or of the remote peer.

### **SIP ID**

Enter the SIP ID (username) that has been created for this device.

### **SIP Password**

Leave this field empty if the PBX doesn't require authentication.

### **Listen SIP Port**

Listening port for the SIP protocol messages. A value of 0 means a default value of 5060.

### **Listen RTP Port**

Listening port for the RTP audio blocks. A value of 0 means a default value of 5004.

### **Default REGISTER Time**

The value that the SIP client suggests to the SIP server when sending the REGISTER request. If this value is accepted by the SIP server, the SIP client has to register after this amount of time. Allowed values in the range of 60-3600 seconds.

**NOTE:** The SIP server may overwrite this value in its reply to the REGISTER request.

### **Send NAT-Keepalives**

Enable this function if the device resides behind a NAT.

### **Periodically Renew DNS**

If enabled, it will force the SIP client to renew the DNS of the SIP server every time the registration to the server fails. Use this feature if you have a backup SIP server with the same DNS name and you wish to enable the SIP client to switch to it if the main server fails.

### **Registration Fail Timeout**

The SIP client will automatically resolve the DNS of the SIP server if the REGISTER request fails. However, if the SIP server is down, or the REGISTER message gets lost, the SIP client will not get a reply from the SIP server. Use this field to set a timeout after which the registration will be assumed to have failed if no reply has been received from the SIP server.

**NOTE:** Use this feature with caution. Setting this value too low may result in registration malfunctioning. If unsure, leave it to the default value.

### **Debug Mode**

Sends received or sent SIP messages via Syslog.

## **CIP PROTOCOL SETTINGS**

### **CIP Port Number**

Set here the port to send and receive CIP control messages.

Default: 41794

## SIP Groups

Set here a comma delimited list of groups in which this device is a member of (eg. ALL,LAB\_DOOR).

**NOTE:** All devices must be members of the group "ALL", so do not delete it from here.

## SIP Display ID

The name set here will be displayed to users called by this device.

**NOTE:** This field can be used only with CIP capable SIP clients.

## OUTBOUND CALL SETTINGS

### Auto Hangup Time

If enabled, the device will automatically hang-up after the configured amount of time if no one answers the call. This option is suitable for door station panels that have only one ring button, and no button to cancel the initiated call.

Values: 5 to 30 seconds.

Default: 15 seconds

### Beep On Call Answer

If enabled, the SIP client will play a short beep sound when the remote party answers the call.

Default: Off

### Call on Device Inputs

Extensions to be called when the corresponding digital input is closed. Maximum two inputs are available.

In "Peer to Peer" mode this field is used to assign the IP of the remote peer to be called. This field can contain just remote peer ID, or the IP preceded with its call ID.

For example "9999@192.168.0.1", or just "192.168.0.1"

## INBOUND CALL SETTINGS

### Input Buffer Level

Maximum delay of the input audio buffer in milliseconds. Decrease this value to minimize delay, increase this value to prevent audio dropouts.

Default setting is "300" ms.

### Phone Pickup Mode

Choose how calls should be answered:

- **auto-answer:** the call is auto-answered immediately.
- **auto-pick-up after timeout:** in this mode either input button can be used to answer

and terminate the call. If the call is not answered inside the "Pick/Hang up After" time period, the device will auto answer the incoming call.

- **auto-hang-up after timeout: (default)** in this mode the input button is used to answer and terminate the call. If the call is not answered inside "Pick/hang up after" time period, the device will decline the call.
- **not callable:** in this mode incoming calls will be declined.

### **Pick/Hang up After**

This field sets the timeout delay used to auto-pick-up, or auto-hang-up, depending upon the Phone Pickup mode.

### **Stream Timeout**

In some scenarios the remote party may go offline without explicitly closing the active call. In this case the SIP-RAVA client may stay in an active call state for an unlimited amount of time. Set here the time in minutes after which the SIP client will close the active call if there is no audio stream received. You can change between 0 and 600 minutes.

### **Beep On Call Answer**

If enabled, the SIP client will play a short beep sound when the incoming call is answered.

Default: Off

### **Door Open Code**

Set the secret code to open the door by entering this DTMF sequence while communicating with the device. When the sequence is detected from the remote peer, the first relay contact is closed. Leave it empty to disable this functionality.

### **Open Door Relay for**

Set the contact closure duration. After this period, the relay contact is switched back to open.

### **Enable Relay**

Select here when you would like to have the relay enabled.

Options are: "on door open command" (default), "on call answer", "on incoming ring"

## AUDIO SETTINGS

### Input Source

Choose between "Mic" and "Line in" as desired input source.

Default: "Mic".

### Encoding

Choose between "uLaw", "aLaw" or G.722.

**Note:** The G.722 selection option is visible only if supported by the HW.

Default: "uLaw".

### Volume

Choose between "0%" and "100%" in 5% steps.

Default: "50%".

### Microphone Gain

Microphone gain dB, increase if your microphone is too faint, decrease if it's too loud or over-driven.

Default: "21dB"

### A/D Amplifier Gain

A/D converter pre-amplification for Line-in audio input in dB. Increase if the audio signal too faint, decrease if it's too loud or over-driven.

Default: "0dB"

### Talk Mode

Choose between "full-duplex" and "half-duplex" talk mode. This option is available only in the SIP Door Station profile.

Default: "FDX" at factory defaults, and "HDX" when the SIP Door Station Profile is applied.

### Output Trigger Level

Defines the audio level from an incoming audio RTP stream that will switch the SIP client into "listen" mode. This setting is ignored if the door station is configured in FDX mode. Used to detect if the remote party is talking.

Values: 0 to 32767.

Default: "1000"

### Trigger Level Timeout

Defines the timeout in milliseconds, after which if no audio is detected from the incoming RTP stream, the SIP Client should switch back to "talk" mode. This setting is ignored if the door station is configured in FDX mode.

Default: "200 ms"

### **Acoustic Echo Cancellation**

Enable or disable AEC here.

Default: "Off"

## **CONTROL INTERFACES**

### **Use Serial Port For**

- **No use:** In this mode the serial port is not used.
- **Serial Control Interface:** In this mode the serial port is used to send commands to the SIP-RAVA client. For more information how to use it please refer to the SIP User Manual.

Default: "No use"

### **Serial Port 1 Settings**

These settings adjust the first serial interface properties.

#### **Baud Rate**

Select the serial transmission speed ("300" to "230400" Baud).

Default: "9600"

#### **Data Bits**

Select "7" or "8" data bits.

Default: "8"

#### **Parity**

Select "no", "even" or "odd" parity.

Default: "no"

#### **Stop Bits**

Select "1" or "2" stop bits.

Default: "1"

#### **Flow Control**

Select the type of flow control:

RTS/CTS signals not used: "none";

RS232: "Software flow control(XON/XOFF)" or "Hardware flow control (RTS/CTS)";

RS485: "RS485 direction control";

Default: "none".

## SECURITY SETTINGS

### Reset Function

Enable or disable the "Reset" function on the Reset button and on the WEB UI. When enabled restart the device by pressing the Reset button once.

Default: "enabled"

### Factory Defaults

Enable or disable the "Factory Defaults" function on the Reset button. In order to revert all settings to factory defaults keep the Reset button pressed until the red LED starts blinking (approx. 10 seconds).

Default: "enabled"

### Update Function

Enable or disable the WEB Update function of the device. If the Update function is disabled, the only way to update the firmware is to use the serial rescue method.

Default: "enabled"

### Set Password

This is visible as long as no password is set.

Enter a password (up to 25 characters) and hit the "Apply" button. After the restart you should close the browser window and open a new browser window. You will be asked to supply user name and password. The user name can be omitted but the password has to be supplied in order to see the web configuration.

**Note:** After applying the changes for the password, please restart your browser and reload the home page of the device. If you do not do this, the browser may enter into endless loop asking for the password because of the non protected configuration page still stored in its cache.

### Old Password / New Password

These fields are visible as long as a password is set.

To allow free access (clearing the password) enter the old password and leave the field "New Password" empty, then hit the "Apply" button.

After the restart you will not be asked for user name and password anymore.

To change the password enter the old password and enter the new password in the field

"New Password", then hit the "Apply" button.

After the restart you will be asked for user name and password. The user name can be omitted but the new password has to be supplied in order to see the web configuration.

## 4.4 Status Page

This page shows the device status and configuration options, and can be used for diagnostic purpose. It consists of three sections, split with line separators:

**Device Information section**, showing all information about the device HW:

The screenshot shows the 'STATUS' page of the SIP-RAVA Client. At the top, there is a navigation bar with tabs: HOME, PROFILES, CONFIGURATION, STATUS (selected), DEFAULTS, and UPDATE. Below the navigation bar, the title 'SIP-RAVA™ CLIENT V2.12C (06 Mar 2013)' is displayed. The page is divided into several sections:

- System status:**
  - Ticks: 326262 ms
  - Uptime: 326 s
- Device information:**
  - Hardware type: Annunicom 100 (ID 7)
  - IPAM type: IPAM 102 (ID 2)
  - IP address: 192.168.1.24
  - MAC address: 00:08:E1:02:9F:7A
- FW version:** B1.11 (03/06/2013)
- WEB version:** 02.02
- Bootloader version:** 99.26
- Setup version:** 01.01
- Sg version:** 09.26 (Mar 6 2013)
- Fs version:** 09.26 (03/06/2013)

**BCL application:**

- Name: sip
- Version: V2.12C (06 Mar 2013)

**USB device info:**

- Device detected: no
- Vendor ID: 0x0000 Product ID: 0x0000
- Device class: 0x0 subclass: 0x0
- Interface class: 0x0 subclass: 0x0
- Maximum power: 0 mA
- Device capacity: 0 kB

**Filesystem Info:**

- Filesystem attached: yes
- Filesystem type: unknown
- Serial number: 00000000

Illustration 5: Device status page

**Application Status section**, showing a snapshot of the SIP application status:

The screenshot shows the 'SIP Status' section of the application status page. It contains a table of parameters:

SIP Status	
Active Profile	SIP Door Station
Call State	Idle
Remote Party	
Registration Status	
Current Set Volume	50 %
Left Output Peak Level	-99 dBFS
Right Output Peak Level	-99 dBFS
Left Input Peak Level	-99 dBFS
Right Input Peak Level	-99 dBFS

Illustration 6: SIP-RAVA Client application status page

**Application Configuration section**, showing all EEPROM configuration parameters for SIP, even the ones not being shown in the current profile configuration menu:

# SIP-RAVA™ CLIENT V2.12D(08 Apr 2013)

**System status:**  
 Ticks: 6244451 ms  
 Uptime: 6244 s

**Device information:**  
 Hardware type: Annunicom 200 (ID 19)  
 IPAM type: IPAM 102 (ID 2)  
 IP address: 192.168.11.176  
 MAC address: 00:08:E1:02:9F:A7

FW version: B1.11 (04/08/2013)  
 WEB version: 02.02  
 Bootloader version: 99.26  
 Setup version: 01.01  
 Sg version: 09.26 (Apr 8 2013)  
 Fs version: 09.26 (04/08/2013)

**BCL application:**  
 Name: sip  
 Version: V2.12D(08 Apr 2013)

**USB device info:**  
 Device detected: no  
 Vendor ID: 0x0000 Product ID: 0x0000  
 Device class: 0x0 subclass: 0x0  
 Interface class: 0x0 subclass: 0x0  
 Maximum power: 0 mA  
 Device capacity: 0 kB

**Filesystem info:**  
 Filesystem attached: yes  
 Filesystem type: unknown  
 Serial number: 00000000

Illustration 7: SIP Client EEPROM settings status page

## 5 Extra features in the SIP-RAVA Client application

### 5.1 Inputs, outputs and relays

The SIP-RAVA Client WEB UI shows the device input, output and relay states.

**NOTE:** The support for X8 has been disabled in SIP-RAVA client

DEVICE & X8 I/O STATUS							
I/O Contacts							
7	6	5	4	3	2	1	0
Inputs	<input checked="" type="checkbox"/>	<input type="checkbox"/>					
Relays	<input checked="" type="checkbox"/>	<input type="checkbox"/>					

Graphically, inputs, relays and outputs are always numbered from 0 to 7. First input, output, or relay (number 0) corresponds to the first Barix device input, output or relay.

### 5.2 DTMF commands

After a call is successfully established, the SIP-RAVA Client application can accept DTMF commands.

**NOTE:** There must be no longer that 5 seconds between each key entry, otherwise the sequence will be cleared, and the command must be re-entered.

The supported commands are listed in the table below:

Command	Description	Examples
NNNN	Web UI configured DTMF door open code.	1234 if the sequence match the preconfigured Web UI door open code, the relay is closed for the preconfigured time period.
#2*NNN*	Set relays state, NNN is the decimal value (0 to 255), representing an 8 bit bit-mask, to be applied.	#2*255* 255 mean binary "11111111" so the command set all relays to "1" (all closed).  #2*2* 2 mean binary "00000010" so the command set relay "1" to 1(closed) and all other to 0 (open).

### Remarks:

Using the DTMF command 2 may interfere with the configured relay behaviour, so use it with caution:

- If the relay is configured to be used for door open, and the DTMF command is used to set the relay to ON, it will be automatically set to OFF if the door open code is typed, and the door lock timeout expires;
- If the relay is configured to switch ON at call answer, it will be automatically switched OFF at call close independently of the whatever state has been set with DFMF 2 command;
- If the relay state is changed with a DTMF command 2, and the relay is set to switch ON at call ring, it will be set to ON the next time the SIP client rings, and set to OFF when the call is answered, thus overriding the relay state set with the DTMF command 2.

## 5.3 Control interface

A special control interface allows for remote control of the device from the network, or from a serial cable connection.

A set of commands is supplied for this purpose. Commands can be sent using one of the following methods:

- through an http cgi command, using the syntax:  
`http://IP_DEST/basic.cgi?command`
- through an RS-232 command

The available commands are shown in the table below.

Command	Description	Examples
D[H][S]XXXX<CR>	Dial command, [H] means high priority, XXXX is the SIP extension to call. [S] means stop the active call. In this case all the remaining characters (if any) from the command string are ignored	DH1001<CR> the command dial the 1001 extension, with an high priority [H] level. DS<CR> will stop the active call.
Q<CR>	Query device status, returns "UID,Sc,ii,oo,rr<CR>" with UID being the UserID (phone number), "Sc" - the connection state (0 = idle, 1 = calling, 2 = incoming call, 3 = connected, 4 = getting notification message), ii,oo,rr being respectively inputs , outputs and relays states as 2 digit hex	

Command	Description	Examples
Xhhh<CR>	<p>numbers.</p> <p>Send data to serial port 1, as hex string (2 digits per byte).</p> <p>If this command is received over UDP or TCP, the data received at the serial port will be forwarded as Xhhhhh&lt;CR&gt; to the same interface.</p> <p><b>NOTE:</b> no streamback for serial and CGI is available. This command works only if the selected serial port is not already used by accessories such as X8, and if it is not issued from the SERIAL control channel.</p>	<p>X148454c4c4f&lt;CR&gt;</p> <p>Send an "HELLO" ascii string as hexadecimal values through serial port 1.</p>
XS	Stops the serial bridge, started by a previous X command.	

## 6 Common issues

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### 6.1 SIP-RAVA client cannot initiate CIP enabled calls

The SIP-RAVA client does not support limited broadcast (sending to address 255.255.255.255), but only local broadcast. Make sure that the Barix SIP-RAVA client and the remote RAVA capable SIP client reside on the same subnet.

### 6.2 SIP-RAVA client can receive SIP paging for group "ALL", but cannot initiate multiple INVITEs to that group

Check that the group name "ALL" has not been removed from the SIP Groups field in the CIP PROTOCOL SETTINGS section. If this is the case add it to the SIP groups list

### 6.3 SIP-RAVA client sends CIP query messages even when configured in SIP Server Mode

The SIP client sends a CIP SIP QUERY, or CIP SIP GROUP QUERY messages before initiating the SIP call to find the IP addresses corresponding to the SIP IDs it is trying to reach. This is a must for the SIP-RAVA client to work correctly in RAVA compliant P2P mode, but not needed in standard SIP server mode. To prevent the SIP-RAVA client from sending CIP query messages in this case set the CIP Port from the CIP PROTOCOL SETTINGS to 0

### 6.4 SIP call starts 2-3 seconds after the call button is pressed

The SIP-RAVA Client performs a CIP discovery of all RAVA compatible devices before sending a SIP INVITE to the discovered devices. This procedure may take 2-3 seconds which delays the actual call start.

# 7 Additional Information

This section contains details about operational and maintenance features:

## 7.1 Factory Defaults Page

This page helps you to revert to factory default if needed. To do so please click on "**Revert to defaults**" link to restore all factory settings except "Network configuration".

While restarting the device a screen appears showing a number counting down. On device start up a screen appears stating the successful reverting to factory defaults.

If you need to revert also the "Network configuration" settings to factory defaults please press and hold the Reset button for about 10 seconds while the device is powered.

HOME | PROFILES | CONFIGURATION | STATUS | DEFAULTS | UPDATE | REBOOT | Annunciom 200 MAC: 00:08:E1:02:9F:69 FW V31.11

### SIP-RAVA™ CLIENT V2.12C (06 Mar 2013)

**FACTORY DEFAULTS**

[Revert to defaults](#)

Reverts all settings except "Network configuration" to factory defaults.

**Help**

**Factory defaults**  
Click on "Factory defaults" to revert all settings except "Network configuration" to factory defaults.

While restarting the device a screen appears showing a number counting down.

Upon start up a screen appears stating the successful reverting to factory defaults.

**Hard default settings**  
To revert all settings (including the "Network configuration") to factory defaults the Reset button has to be pressed for about 10 seconds while the device is powered.

Illustration 8: Factory Defaults page

## 7.2 Update Page

This page enables you to do a full, or partial FW update of your device. For more details how to do it please refer to the Error: Reference source not found chapter.

HOME | PROFILES | CONFIGURATION | STATUS | DEFAULTS | UPDATE | REBOOT | Annunciom 200 MAC: 00:08:E1:02:9F:69 FW V31.11

### SIP-RAVA™ CLIENT V2.12C (06 Mar 2013)

**UPDATE**

Please read the instructions before applying the update.

Please click [here](#) to start the update

**Currently Loaded Version**

- Firmware VB1.11 (03/06/2013)
- Web UI V02.02
- Bootloader V99.26
- Setup V01.01
- Song V09.26 (Mar 3 2013)
- Filesystem V09.26 (33/06/2013)

**Help**

Barix constantly enhances the capabilities of their products. Therefore we recommend to keep the software on the device up to date. To download the latest firmware version please visit [www.barix.com](http://www.barix.com).

Download the firmware update package and unpack to a local drive.

**Updating files**  
Click on "Please click here to continue" to launch the update process. Before you click the link note that the update process can only be canceled by power cycling the device.

The device will restart in a special mode called Bootloader and a screen appears showing a number counting down. Please note that in this mode the standard http port 80 is always used.

Illustration 9: SIP-RAVA Client FW Update Page

## Updating files

Click on "**Please click here to continue**" to launch the update process. Please have in mind that the update process can **ONLY** be cancelled by power cycling the device.

The device will display a countdown, and then restart in a special mode called "Bootloader".

**Note:** In this mode the standard http port 80 is always used.

To upload a resource click on "Browse..." to locate the file you want to update. Once selected, click on "Upload". This upload process can take a few seconds.

After a successful upload the following text appears:

```
compound.bin successfully loaded.
```

```
Click on update before updating the next component or  
unplug the power supply to reboot the device.
```

If you choose "update" you may upload another resource or click the "reboot" button.

If you choose to reboot the following text appears:

```
rebooting...
```

```
Click here to reload the main page.
```

The device takes a few seconds to reboot. If you have a fixed IP address or DHCP resolves to the IP address used before, the main page will appear.

## 7.3 Updating a device using the RS-232 serial port

Sometimes when the device is not accessible via the LAN, or the image in the flash memory is corrupted for some reason, then a serial rescue may be needed to reset the device to the factory defaults.

The steps to do a serial rescue:

- Disconnect the power from the device
- Connect the COM port of the device to the COM port of your computer via a null modem serial cable
- Open a terminal in the update\_rescue directory in your unzipped copy of the ABCL SIP firmware. If using Linux or Mac, switch to update\_rescue/linux or update\_rescue/mac directory
- Start the rescue process by executing ipamresX, where X is the number (1-4) of the serial port on the computer
- If running on Linux /Mac execute the ipamresd.sh script, giving the device name of your COM port as a parameter. For example:

```
./seriald.sh /dev/tty.UC-232AC
```

- Power on the device, and wait for the update to finish. The device will reboot automatically.

**NOTE:** All other settings except the network settings will be lost !

## 8 Document details

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<b>Document Date:</b>	09. Jul. 2013
<b>Document Version:</b>	1.02
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05/11/2012	1.00	ASI	Created the first draft
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04/07/13	1.02	ASI	Added SIP-RAVA System diagram and fixed all references to Annunicom 100 to Annunicom 200

## 9 Legal Information

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