UBIQUITI NETWORKS

UniFi® VoIP

Quick Provisioning Guide for Third-Party PBX

User Guide

Table of Contents

Chapter 1: Overview

This User Guide provides intructions on how to configure a third-party Private Branch eXchange (PBX) for use with the UniFi® VoIP Phone (models UVP, UVP-Pro, and UVP-Executive). Throughout the rest of this document, *UniFi VoIP Phone* refers to any and all models of the UniFi VoIP Phone.

This configuration involves two steps:

- Configure the PBX with the extension of each phone.
- Configure each UniFi VoIP Phone's SIP settings so that it can connect to the PBX.

For detailed instructions, refer to the appropriate section for your PBX:

"Asterisk Configuration" on page 2

"FreeSwitch Configuration" on page 6

"3CX Configuration" on page 9

"Elastix Configuration" on page 13

"FreePBX Configuration" on page 17

"FusionPBX Configuration" on page 21

Chapter 2: Asterisk Configuration

This chapter provides detailed instructions on how to configure a UniFi VoIP Phone on the Asterisk® (**http://www.asterisk.org/**) open-source PBX. This process, which must be done once for each phone, involves modifying the SIP configuration file for the phone, editing the dial plan file to enable calls to the extension, and configuring the SIP settings on the phone.

Creating a Phone Extension on Asterisk

Each PBX comes with a default configuration that contains a dial plan, extensions, and all initial settings needed. By default, the file `/etc/asterisk/sip.conf` contains the extensions. To add extension 100, add the following text snippet to this file (bold italic text indicates user-specified values):

```
[sip-temp](!)
type=friend
host=dynamic
disallow=all
allow=ulaw
qualify=1000
canreinvite=no
nat=force_rport
dtmfmode=rfc2833
context=from-internal

[100](sip-temp)
username=100
callerid=Your Name <100>
secret=password
dial=SIP/100
```

To be able to call this extension, you need to hook it up to the corresponding dial plan (found in file `/etc/asterisk/extensions.conf` by default). To do so, create the context `from-internal` that is specified as the outbound context for the SIP extension. For extension 100, the lines to be added are:

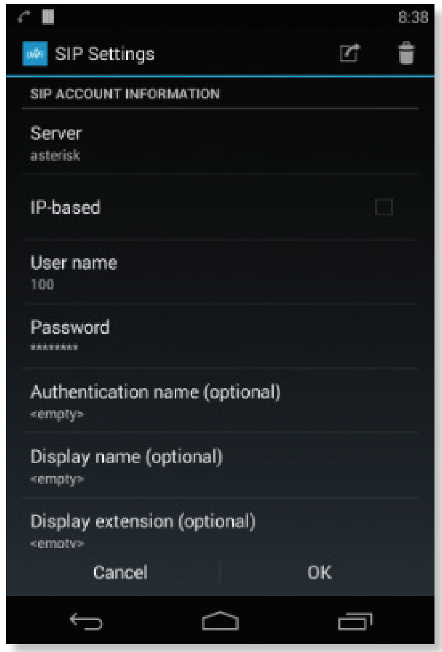
```
[from-internal]
exten => 100,1,Dial(SIP/${EXTEN}|40|Ttr)
```

Configuring the Phone's SIP Settings

Before you can configure the UniFi VoIP Phone's SIP settings, perform initial configuration on the phone by following the instructions in **"Initial Configuration" on page 26**. Then, configure the phone's SIP account by following these steps:

1. Press the **Settings** icon at the bottom left of the *Welcome* screen to display the *Phone Settings* page.
2. Press **SIP service**.
3. Press **SIP accounts**.

4. Press **Add account**. The *SIP Settings* page is displayed.



5. Complete the following SIP account information and press **OK**:

Field	Description
Server	SIP server IP address or hostname for registration
IP-based	Select this option if you are using IP-based authentication
User name	SIP user ID
Password	SIP password
Authentication name (optional)	The username used for authentication, if required
Display name (optional)	The name that will be displayed to other users as the Caller ID
Display extension (optional)	The extension that will be displayed to other users
Voicemail number	The extension where the voicemail service can be reached at your PBX
Outbound proxy (optional)	The SIP proxy used for outbound calls, if required

The minimum Asterisk SIP configuration requirements are:

- **Server** Set this to the IP or hostname of your Asterisk server.
- **User name** Set this to the SIP username.
- **Password** Set this to the SIP password.

Enter the SIP settings that you configured in Asterisk in **"Creating a Phone Extension on Asterisk" on page 2**. For the *Password* field, use the setting of the `secret` option.

6. The new account will be displayed on the *SIP accounts* page.

If the new account was properly configured, *Receiving calls* will be displayed.

If it was not properly configured, *Not connected to server (error: 408)* will be displayed.

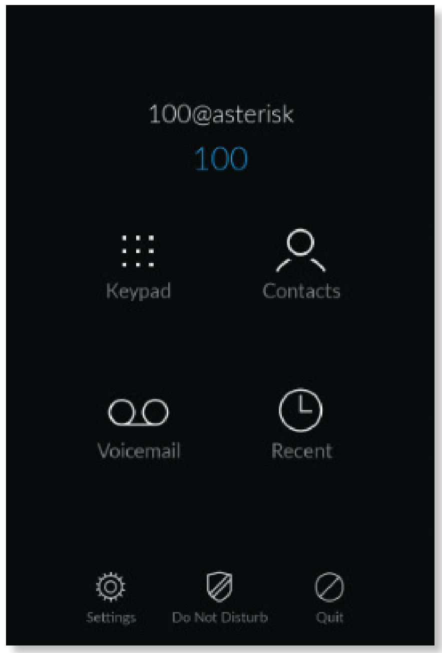
If configuration was successful:

- The Asterisk log will show the following to indicate that the UniFi VoIP Phone has registered:

```
    -- Registered SIP '100' at XX.XX.XX.XX:1024

[Jun  3 13:36:06] NOTICE[4093]: chan_sip.c:23522 handle_response_
peerpoke: Peer '100' is now Reachable. (139ms / 5000ms)
```

- The UniFi VoIP Phone's *Dialer* screen will show the phone as connected, and will allow you to make calls if your Asterisk server is set up for outbound calls (SIP, IAX, PRI, etc.).



Configuring Calls Between Phones

To enable calls between UniFi VoIP Phones (extensions 100 and 101 in this example), first add the following lines to the `sip.conf` file (bold italic text indicates user-specified values):

```
[sip-temp](!)
type=friend
host=dynamic
disallow=all
allow=ulaw
qualify=1000
canreinvite=no
nat=force_rport
dtmfmode=rfc2833
context=internal-calls
```

```
[100](sip-temp)
username=100
callerid=Your Name <100>
secret=Password

[101](sip-temp)
username=101
callerid=Your Name <101>
secret=Password
```

Then, add the following lines to the `/etc/asterisk/extensions.conf` file:

```
[100-in]
exten => s,1,Dial(SIP/100,30)
exten => s,2,Hangup()

[101-in]
exten => s,1,Dial(SIP/101,30)
exten => s,2,Hangup()

[internal-calls]
exten => 100,1,GoTo(100-in,s,1) ;Extension 100
exten => 101,1,GoTo(101-in,s,1) ;Extension 101
```

Chapter 3: FreeSwitch Configuration

This chapter provides detailed instructions on how to configure a UniFi VoIP Phone on the FreeSWITCH® (**http://www.freeswitch.org/**) open-source PBX. This process, which must be done once for each phone, and involves creating a unique configuration file for the phone, editing the dial plan file to enable calls to the extension, and configuring the SIP settings on the phone.

Creating a Phone Extension on FreeSWITCH

Each PBX comes with a default configuration that contains a dial plan, extensions and all initial settings needed. By default, the files containing the extensions in FreeSWITCH are in the `/etc/freeswitch/directory/default` directory. To add extension 100 to FreeSWITCH, create a file named `100.xml` with the following contents (bold italic text indicates user-specified values):

```
<include>
   <user id="100">
      <params>
         <param name="password" value="password"/>
         <param name="vm-enabled" value="true"/>
         <param name="vm-password" value="8761"/>
      </params>
      <variables>
         <variable name="user_context" value="default"/>
         <variable name="effective_caller_id_name" value="Your Name"/>
         <variable name="effective_caller_id_number" value="100"/>
      </variables>
   </user>
</include>
```

To be able to call this extension, you need to edit the FreeSWITCH dial plan (found in file `/etc/freeswitch/dial plan/default.xml` by default) and add the following contents:

```
<extension name="100">
  <condition field="destination_number" expression="^100$">
    <action application="bridge" data="user/100"/>
  </condition>
</extension>
```

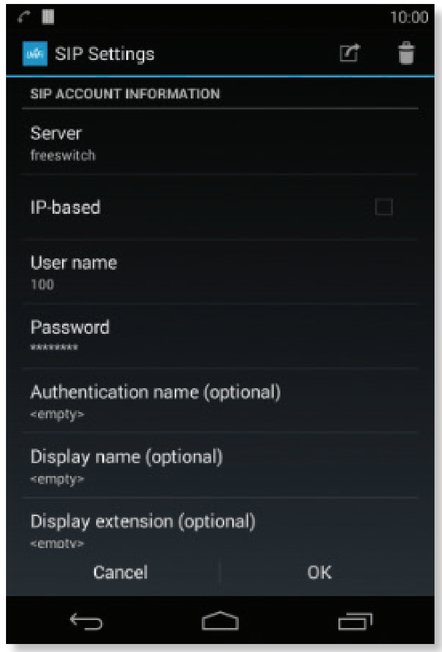
Once you have completed these steps and configured your UniFi VoIP Phone, you will be ready to make and receive calls with it.

Configuring the Phone’s SIP Settings

Before you can configure the UniFi VoIP Phone’s SIP settings, perform initial configuration on the phone by following the instructions in **“Initial Configuration” on page 26**. Then, configure the phone’s SIP account by following these steps:

1. Press the **Settings** icon at the bottom left of the *Welcome* screen to display the *Phone Settings* page.
2. Press **SIP service**.

3. Press **SIP accounts**.
4. Press **Add account**. The *SIP Settings* page is displayed.



5. Complete the following SIP account information and press **OK**:

Field	Description
Server	SIP server IP address or hostname for registration
IP-based	Select this option if you are using IP-based authentication
User name	SIP user ID (ask your PBX administrator)
Password	SIP password (ask your PBX administrator)
Authentication name (optional)	The username used for authentication, if required
Display name (optional)	The name that will be displayed to other users as the CallerID
Display extension (optional)	The extension that will be displayed to other users
Voicemail number	The extension where the voicemail service can be reached at your PBX
Outbound proxy (optional)	The SIP proxy used for outbound calls, if required

The minimum FreeSWITCH SIP configuration requirements are:

- **Server** Set this to the IP or hostname of your FreeSWITCH server.
- **User name** Set this to the SIP Username.
- **Password** Set this to the SIP Password (secret above).

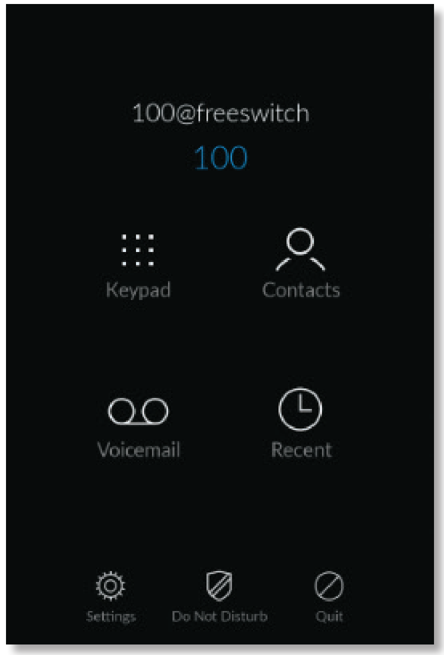
Enter the SIP settings that you configured in FreeSWITCH in **"Creating a Phone Extension on FreeSWITCH" on page 6**.

6. The new account will be displayed on the *SIP accounts* page.

If the new account was properly configured, *Receiving calls* will be displayed.

If it was not properly configured, *Not connected to server (error: 408)* will be displayed.

If configuration was successful, the UniFi VoIP Phone's *Dialer* screen will show the phone as connected, and will allow you to make calls if your FreeSWITCH server is set up for outbound calls (SIP, IAX, PRI, etc.).



Chapter 4: 3CX Configuration

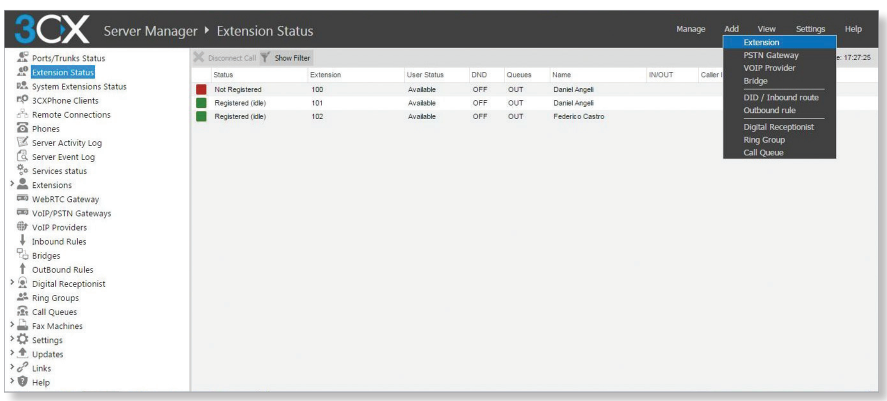
This chapter provides detailed instructions on how to configure a UniFi VoIP Phone on the 3CX® (**http://www.3cx.com**) PBX. This process, which must be done once for each phone, involves creating an extension for the phone using the 3CX Management Console, and configuring the SIP settings on the phone.

Creating a Phone Extension on 3CX

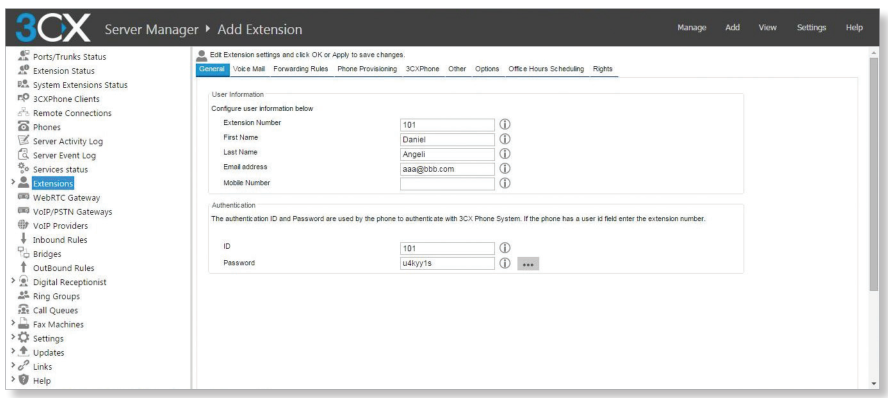
To configure phone extensions on your 3CX installation, follow these steps:

1. Run the *3CX Management Console* and log in to your 3CX system:

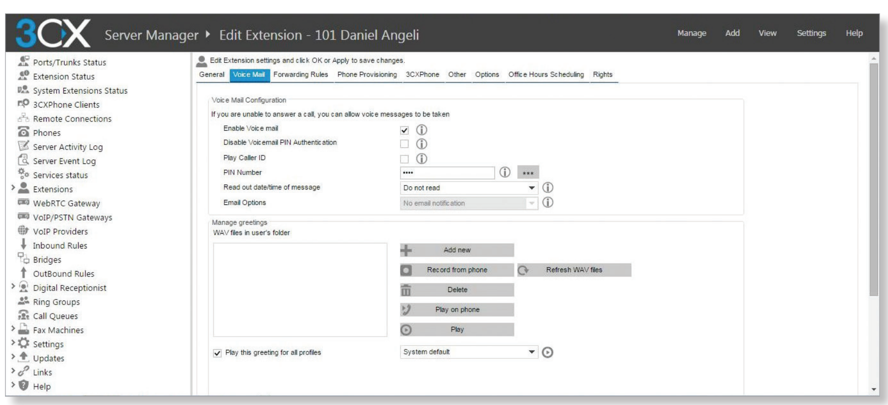
2. After you successfully log in, the *Extension Status* page is displayed. To create new extensions, click **Add**, and then click **Extension**.



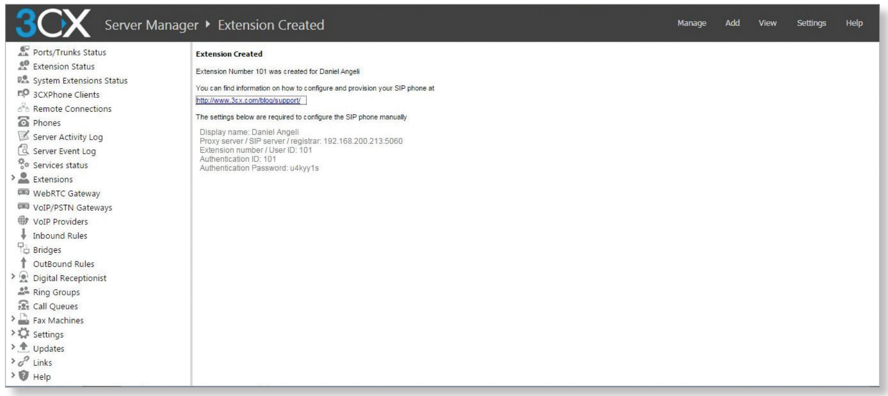
3. The *Add Extension* page displays a form with fields to be completed.
   a. Select the *General* tab. In the *User Information* section, complete the following fields:
      - **Extension Number** This is the extension number to dial to reach this user.
      - **First Name** This becomes the first name in the Caller ID. It is also displayed in the extension status.
      - **Last Name** This becomes the last name in the Caller ID. It is also displayed in the extension status.
      - **Email Address** This is the email address used for voicemail notifications for this extension. It is disabled if left blank.
   b. In the *Authentication* section, fill in the following fields:
      - **ID** This is the registration ID. It must be the same as the *User Name* setting on the phone.
      - **Password** This is the password used for the extension; it is automatically generated when you create a new extension.



4. Select the *VoiceMail* tab, and set these values:
   - **Enable Voice mail** Set this to **Enabled**.
   - **PIN Number** This is the password used to access the voicemail system. This password can only contain numbers (it is automatically generated). A user can change the password you enter here after logging into the voicemail system (999) with a phone.



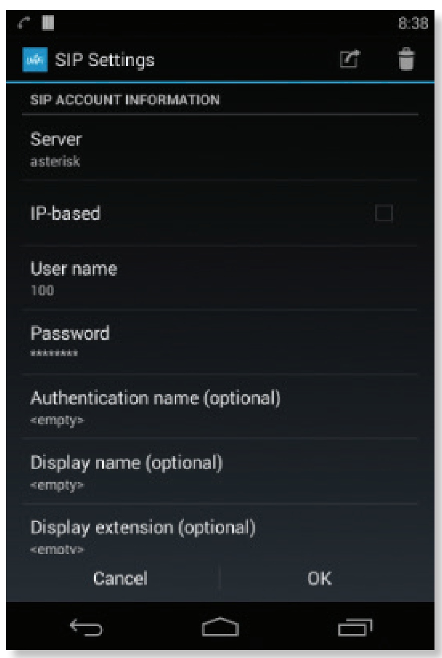
5. When you are finished configuring the settings, click **Apply**. You will be redirected to the *Extension Created* page that shows all the information related to the extension you have just created.



Configuring the Phone's SIP Settings

Before you can configure the UniFi VoIP Phone's SIP settings, perform initial configuration on the phone by following the instructions in **"Initial Configuration" on page 26**. Then, configure the phone's SIP account by following these steps:

1. Press the **Settings** icon at the bottom left of the *Welcome* screen to display the *Phone Settings* page.
2. Press **SIP service**.
3. Press **SIP accounts**.
4. Press **Add account**. The *SIP Settings* page is displayed.



5. To complete the phone configuration, fill in the following fields:
    - **Server** Set this to the IP or hostname of your 3CX PBX.
    - **User name** Set this to the value of the *Extension Number* field (in the *User Information* section of the *Add Extension* page in step **3** above).
    - **Password** Set this to the value of the *Password* field (in the *Authentication* section of the *Add Extension* page in step **3** above).
    - **Voicemail number** Set this to **999**.
6. The new account will be displayed on the *SIP accounts* page.

    If the new account was properly configured, *Receiving calls* will be displayed.

    If it was not properly configured, *Not connected to server (error: 408)* will be displayed.

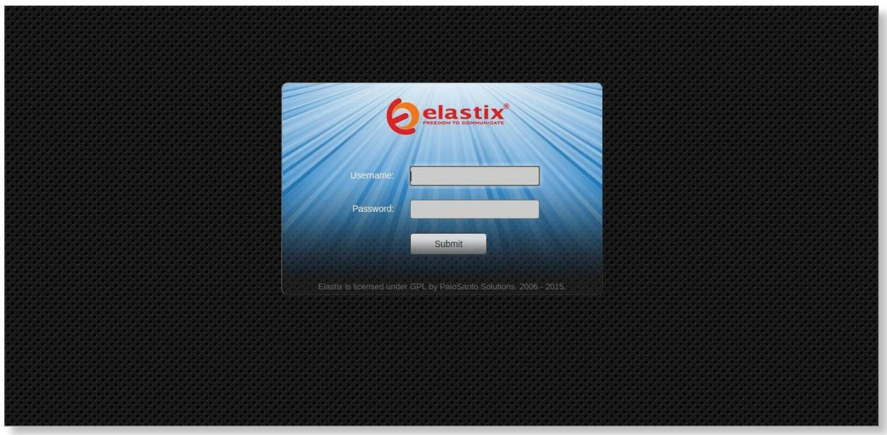
Chapter 5: Elastix Configuration

This chapter provides detailed instructions on how to configure a UniFi VoIP Phone on the Elastix® (**http://www.elastix.org**) PBX. This process, which must be done once for each phone, involves creating an extension for the phone using the Elastix administration portal, and configuring the SIP settings on the phone.

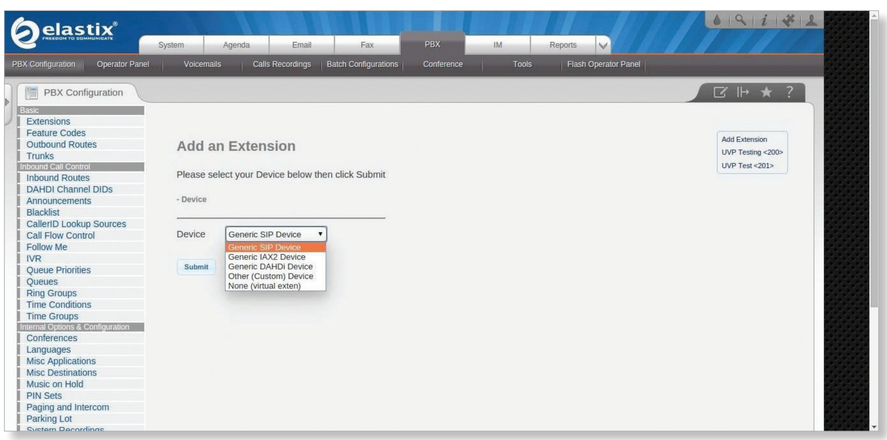
Creating a Phone Extension on Elastix

To configure phone extensions on your Elastix installation, follow these steps:

1. Log in to the administration portal:



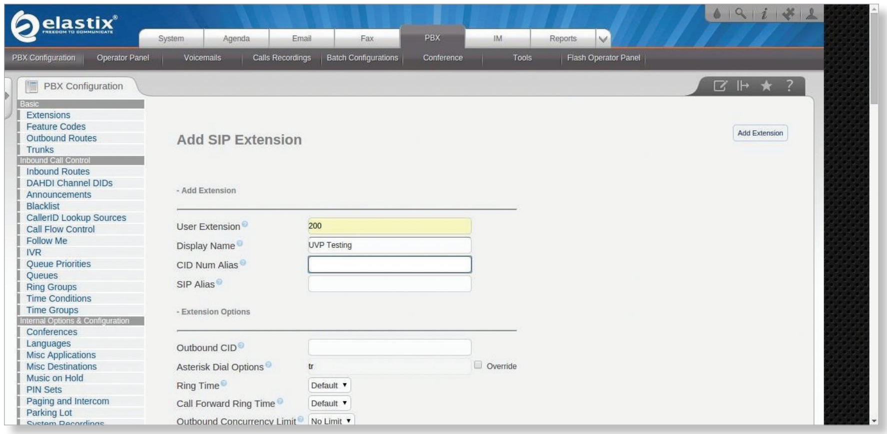
2. Click the **PBX** tab and then click **PBX Configuration** at the top left to display the *Add an Extension* page, where you can choose the type of device that you want to add. From the *Device* drop-down menu, select **Generic SIP Device** for the phone.



3. The *Add SIP Extensions* page displays a form with fields to be filled in.

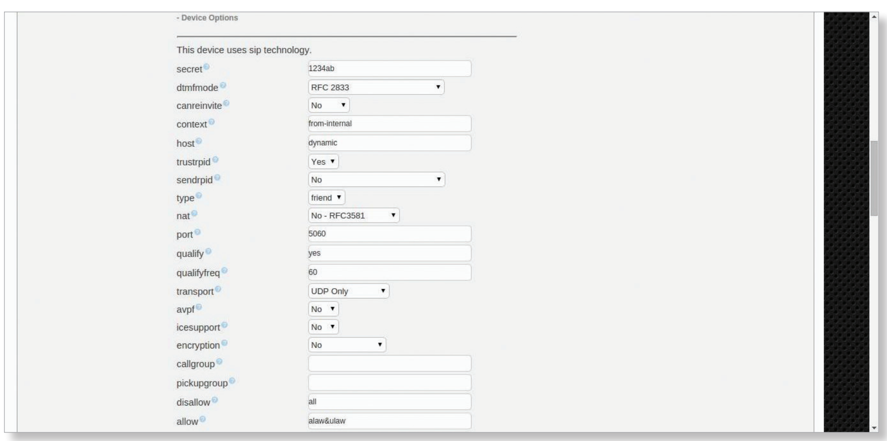
   a. In the *Add Extension* section, fill in the following fields:

   - **User Extension** This is the extension number to dial to reach this user.
   - **Display Name** This is the Caller ID name for outgoing calls from this extension. Enter the name only, not the number.



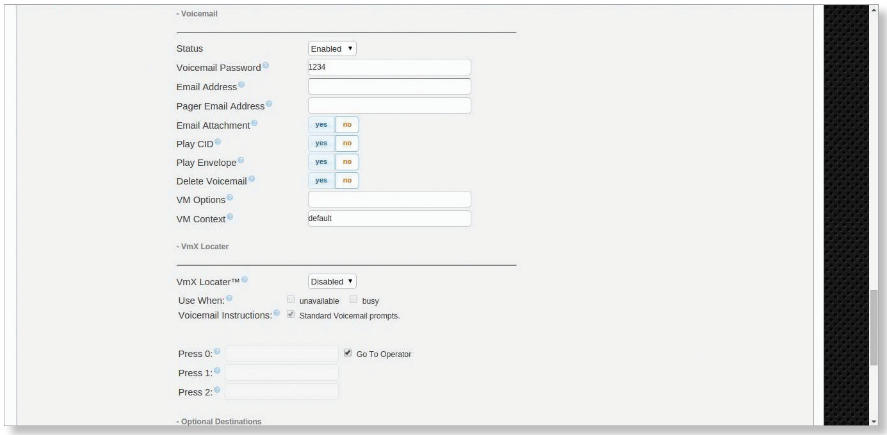
   b. In the *Device Options* section, fill in the following fields:

   - **secret** This is the secret password configured for the device's extension. It should be alphanumeric with at least two letters and numbers to make it secure.
   - **disallow** This is a list of disabled codecs. Set this to **all** to remove all codecs defined in the general settings.
   - **allow** This is a list of enabled codecs, with codecs separated by the "&" character, in order of precedence. For the UniFi VoIP Phone, set this field to **ulaw&alaw&g722**.

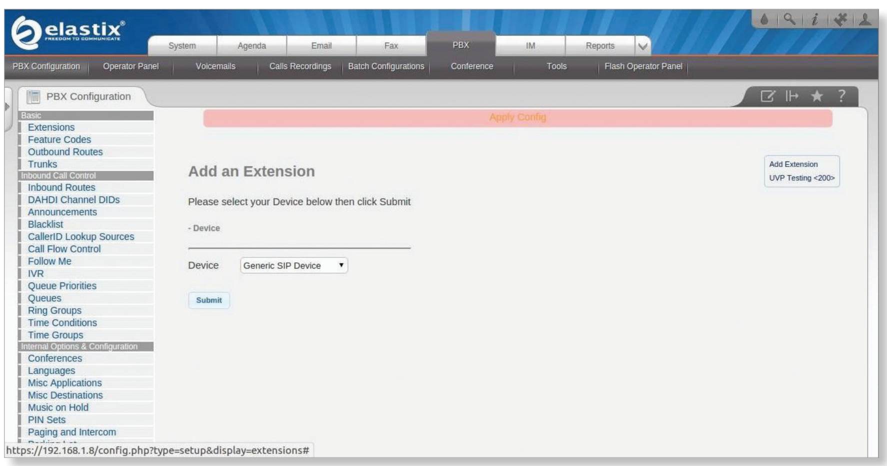


c. In the *Voicemail* section, fill in the following fields:

- **Status** Set this to **Enabled**.
- **Voicemail Password** This is the password used to access the voicemail system. This password can only contain numbers. A user can change the password you enter here after logging into the voicemail system (*98) with a phone.



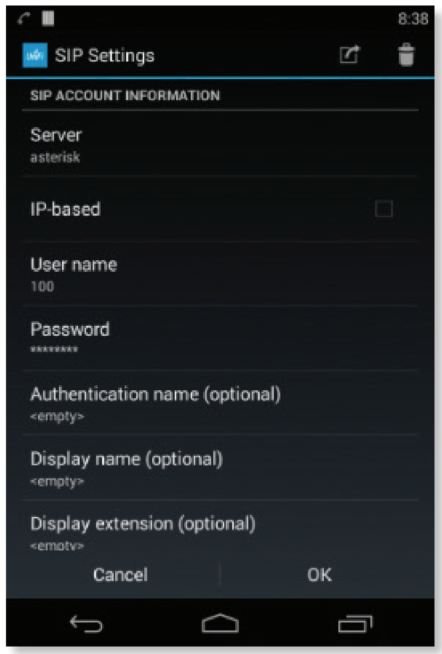
4. When you are finished configuring the settings, click **Submit**, and on the next page, click **Apply Config**.



Configuring the Phone's SIP Settings

Before you can configure the UniFi VoIP Phone's SIP settings, perform initial configuration on the phone by following the instructions in **"Initial Configuration" on page 26**. Then, configure the phone's SIP account by following these steps:

1. Press the **Settings** icon at the bottom left of the *Welcome* screen to display the *Phone Settings* page.
2. Press **SIP service**.
3. Press **SIP accounts**.
4. Press **Add account**. The *SIP Settings* page is displayed.



5. To complete the phone configuration, fill in the following fields:
   - **Server** Set this to the IP or hostname of your Elastix PBX.
   - **User name** Set this to the value of the *Extension Number* field (in the *Add Extension* section of the *Add SIP Extensions* page in step **3** above).
   - **Password** Set this to the value of the *secret* field in (the *Device Options* section of the *Add SIP Extensions* page in step **3** above)
   - **Voicemail number** Set this to ***98**.
6. The new account will be displayed on the *SIP accounts* page.

   If the new account was properly configured, *Receiving calls* will be displayed.

   If it was not properly configured, *Not connected to server (error: 408)* will be displayed.

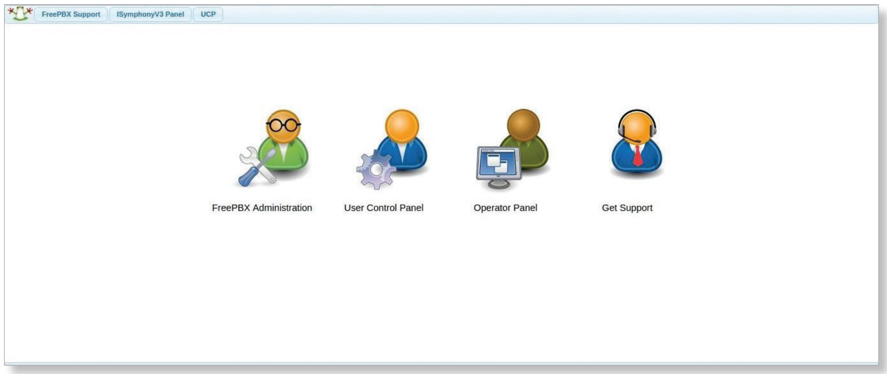
Chapter 6: FreePBX Configuration

This chapter provides detailed instructions on how to configure a UniFi VoIP Phone on the FreePBX® (http://www.freepbx.org/) PBX. This process, which must be done once for each phone, involves creating an extension for the phone using the FreePBX administration tool, and configuring the SIP settings on the phone.

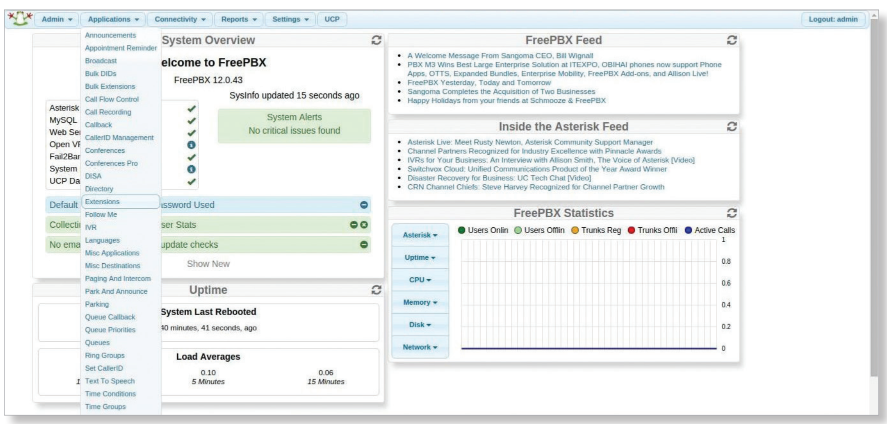
Creating a Phone Extension on FreePBX

To configure phone extensions on your FreePBX installation, follow these steps:

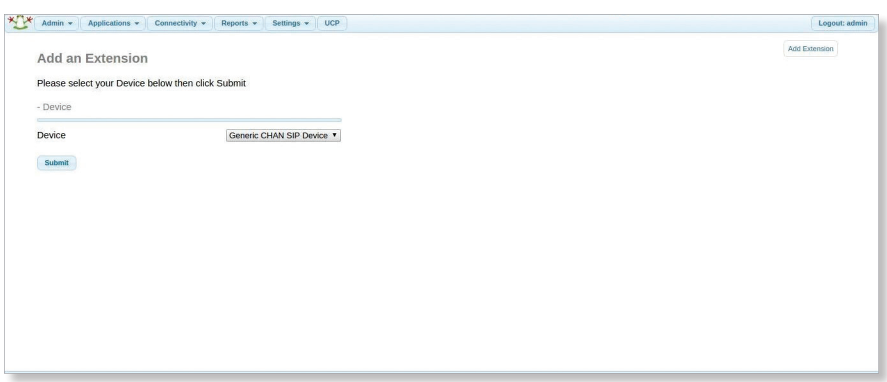
1. Select **FreePBX Administration** on your FreePBX home page:



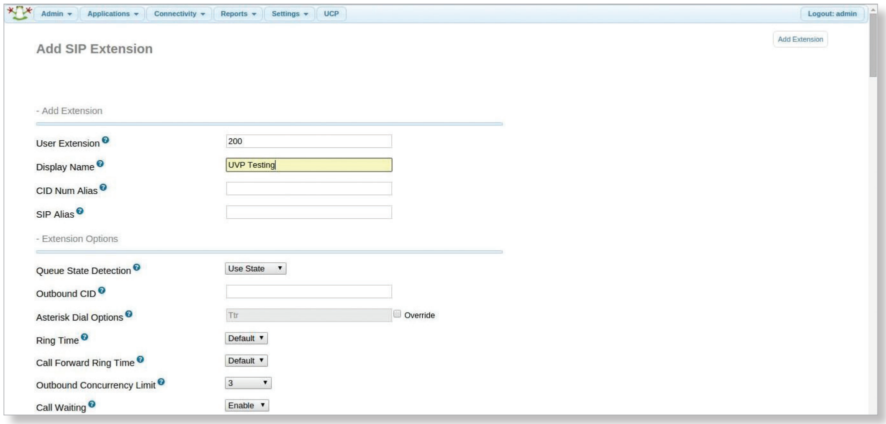
2. Click the **Applications** tab and then click **Extensions**.



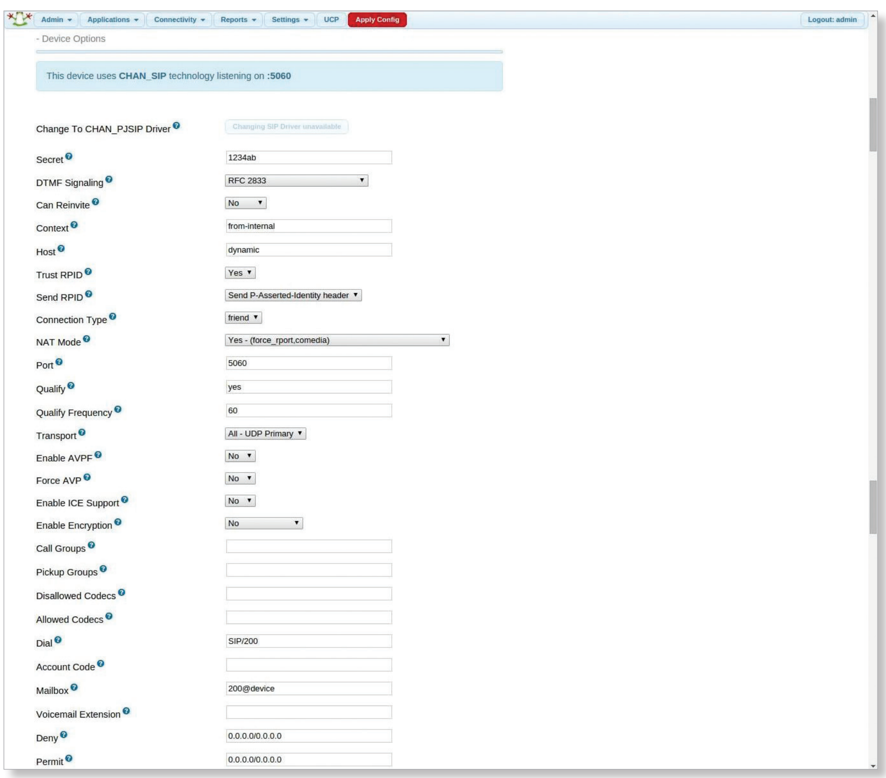
3. The *Add an Extension* page is displayed, where you can choose the type of device to be added. From the *Device* drop-down menu, select **Generic SIP Device** for the phone.



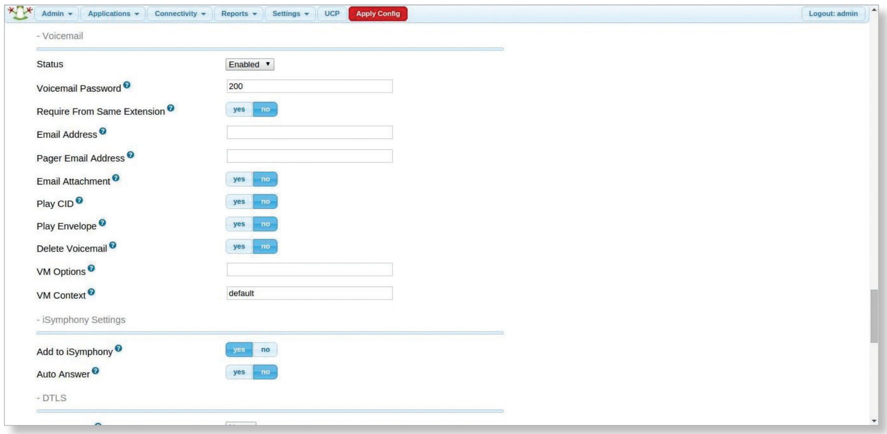
4. The *Add SIP Extensions* page displays a form with fields to be filled in.
   a. In the *Add Extension* section, fill in the following fields:
      - **User Extension** This is the extension number to dial to reach this user.
      - **Display Name** This is the Caller ID name for outgoing calls from this extension. Enter the name only, not the number.



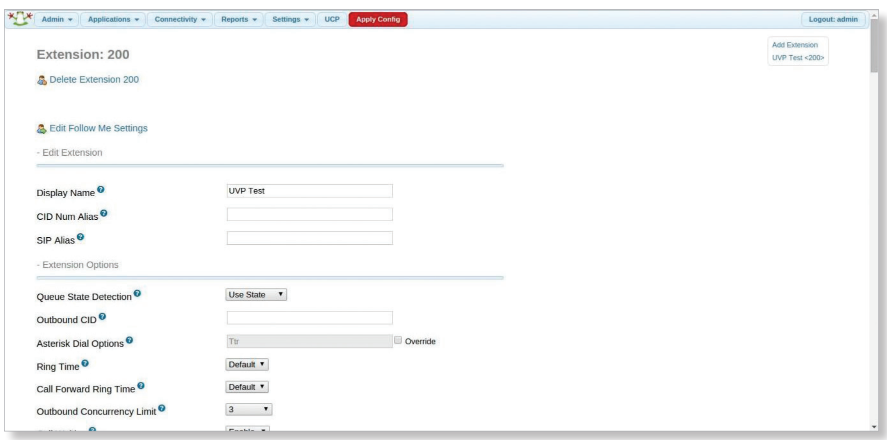
   b. In the *Device Options* section, fill in the following fields:
      - **Secret** This is the secret password configured for the device's extension. It should be alphanumeric with at least two letters and numbers to make it secure.
      - **Disallowed Codecs** This is a list of disabled codecs. Set this to **all** to remove all codecs defined in the general settings. The following codecs *must* be disallowed to enable outbound calling: **iLBC&speex&GSM&PCMA&PCMU**.
      - **Allowed Codecs** This is a list of enabled codecs, with codecs separated by the "&" character, in order of precedence. For the UniFi VoIP Phone, set this field to **ulaw&alaw&g722**.



c. In the *Voicemail* section, fill in the following fields:
   - **Status** Set this to **Enabled**.
   - **Voicemail Password** This is the password used to access the voicemail system. This password can only contain numbers. A user can change the password you enter here after logging into the voicemail system (*98) with a phone.



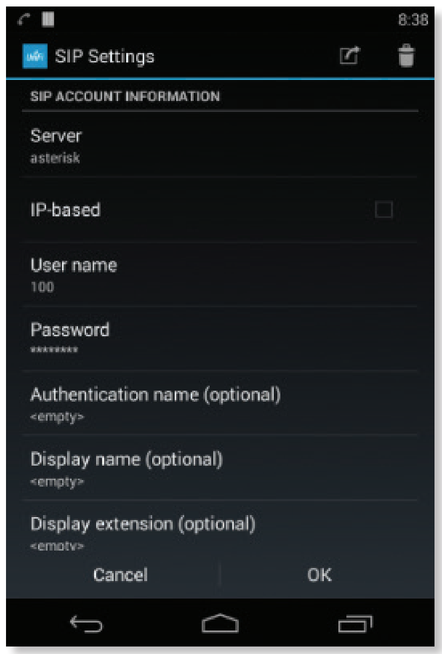
5. When you are finished configuring the settings, click **Submit**, and on the next page, click **Apply Config**.



Configuring the Phone's SIP Settings

Before you can configure the UniFi VoIP Phone's SIP settings, perform initial configuration on the phone by following the instructions in **"Initial Configuration" on page 26**. Then, configure the phone's SIP account by following these steps:

1. Press the **Settings** icon at the bottom left of the *Welcome* screen to display the *Phone Settings* page.
2. Press **SIP service**.
3. Press **SIP accounts**.
4. Press **Add account**. The *SIP Settings* page is displayed.



5. To complete the phone configuration, fill in the following fields:
   - **Server** Set this to the IP or hostname of your FreePBX PBX.
   - **User name** Set this to the value of the *User Extension* field (in the *Add Extension* section of the *Add SIP Extension* page in step **4** above).
   - **Password** Set this to the value of the *Secret* field (in the *Device Options* section of the *Add SIP Extension* page in step **4** above).
   - **Voicemail number** Set this to ***98**.
6. The new account will be displayed on the *SIP accounts* page.

   If the new account was properly configured, *Receiving calls* will be displayed.

   If it was not properly configured, *Not connected to server (error: 408)* will be displayed.

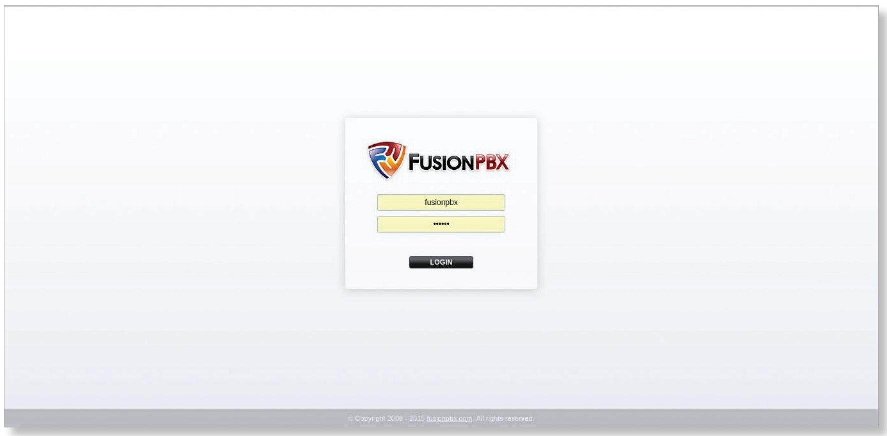
Chapter 7: FusionPBX Configuration

This chapter provides detailed instructions on how to configure a UniFi VoIP Phone on the FusionPBX® (**http://www.fusionpbx.com/**) PBX. This process, which must be done once for each phone, involves creating an extension for the phone using the FusionPBX administration tool, and configuring the SIP settings on the phone.

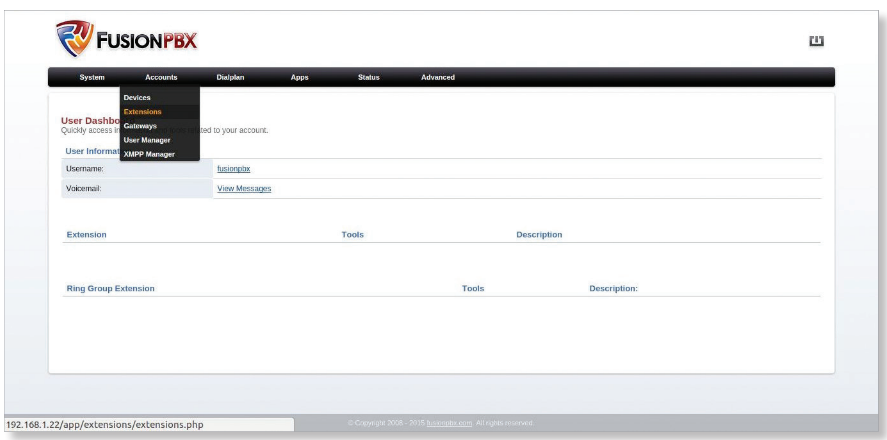
Creating a Phone Extension on FusionPBX

To configure phone extensions on your FusionPBX installation, follow these steps:

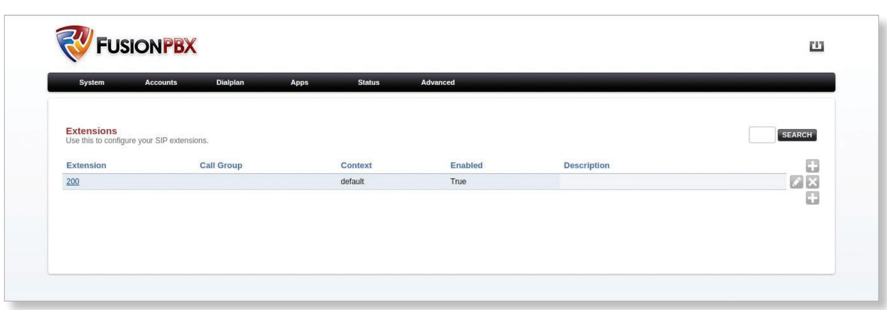
1. Log in to the FusionPBX administration portal:



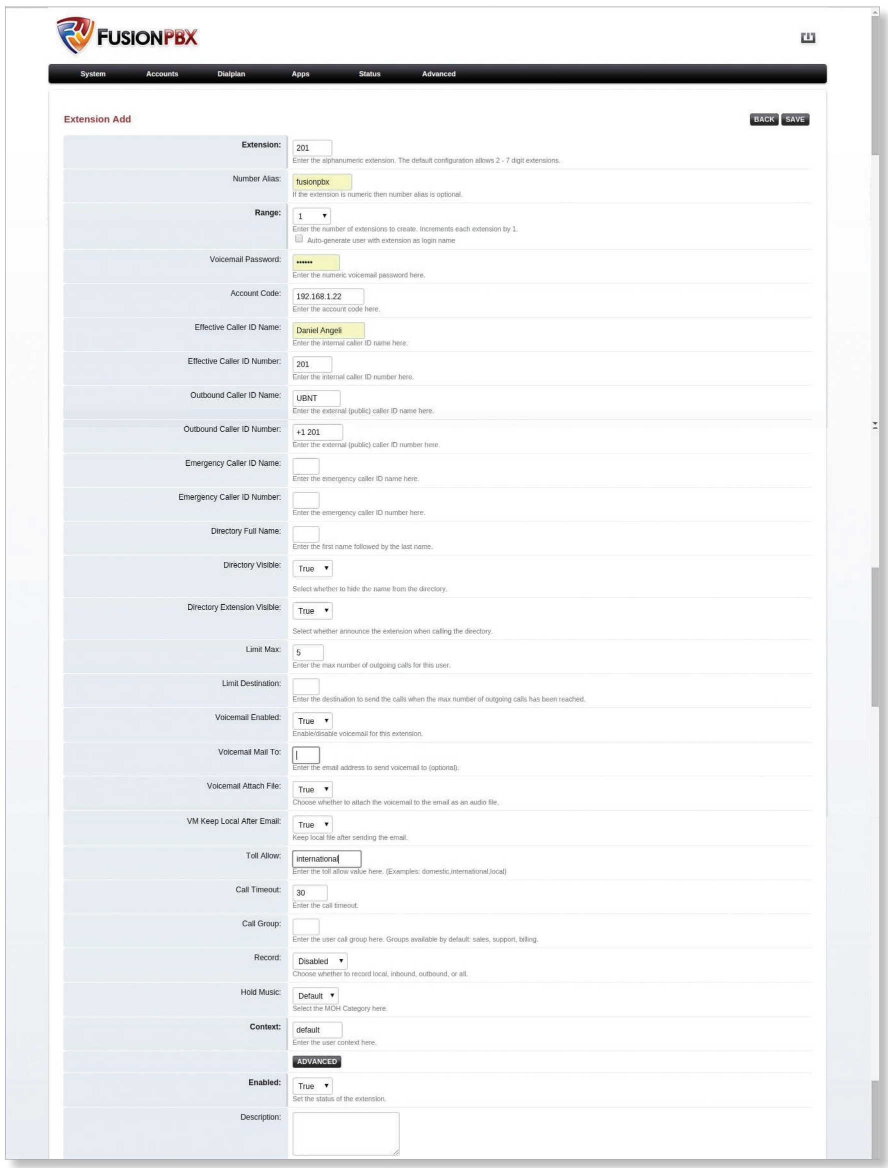
2. Click the **Account** tab and then click **Extensions**.



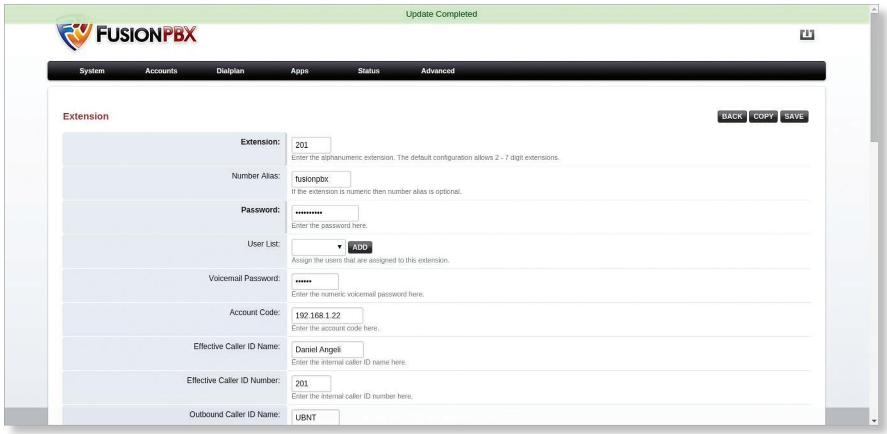
3. The *Extensions* page is displayed, where you can add, edit, or delete extensions. Click **+** on the right to add a new extension.



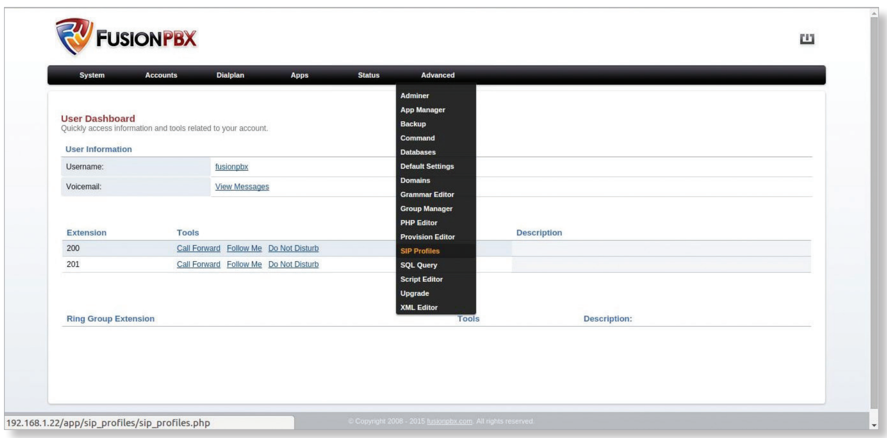
4. The *Extension Add* page displays a form. Fill in the following fields:
   - **Extension** This is the extension number to dial to reach this user.
   - **Password** This is the password configured for the device's extension. It is automatically created, but you can change it.
   - **Effective Caller ID Name** This is the Caller ID name for outgoing calls from this extension. Enter the name only; do not enter the number.
   - **Effective Caller ID Number** This is the Caller ID number for outgoing calls from this extension. Enter only enter the number; do not enter the name.
   - **Voicemail Password** This is the password used to access the voicemail system. This password can only contain numbers. A user can change the password you enter here after logging into the voicemail system (*98 by default) with a phone.
   - **Toll Allow** Enter the toll allow value (e.g. *domestic*; *international*; *local*).



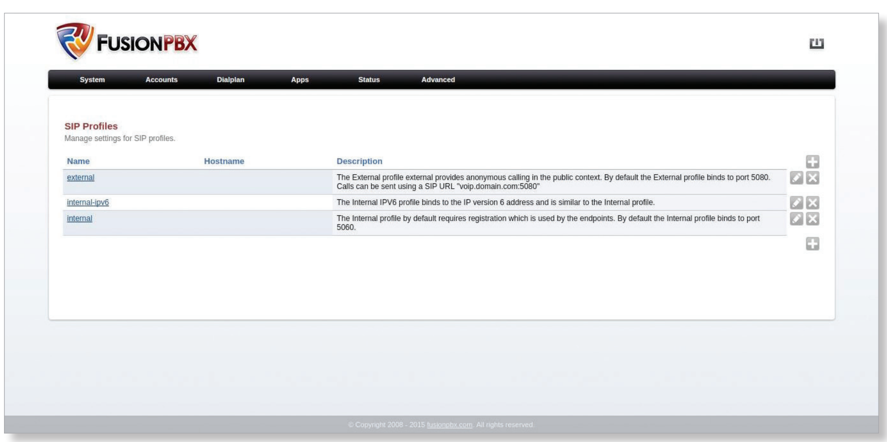
5. When you are finished configuring the settings, click **Save**. When the page is refreshed, *Update Completed* will be displayed at the top of the page.



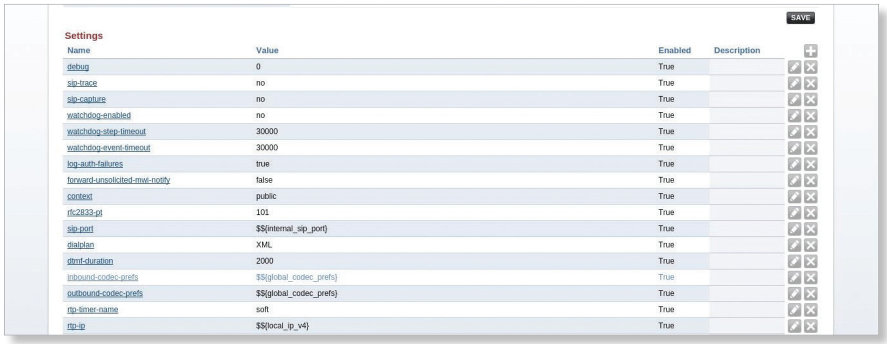
6. By default the codec priority is **PCMU**, **PCMA**, **GSM**. Select the **Advanced** tab to add the G722 codec as first priority if you want to use HD audio.



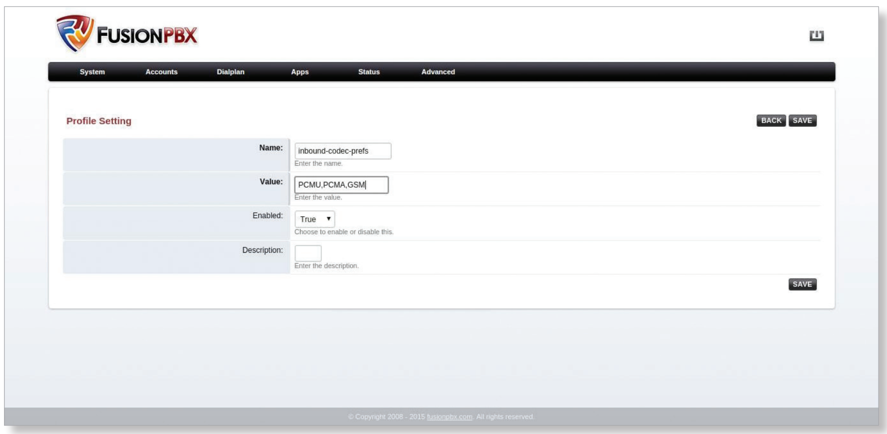
7. Select the profile where you want to change the codec priority (or add/delete a codec). In this example, we are updating the *internal* profile.



8. Inside the profile settings, select the inbound (**inbound-codec-prefs**) or outbound (**outbound-codec-prefs**) codec preferences you want to change:



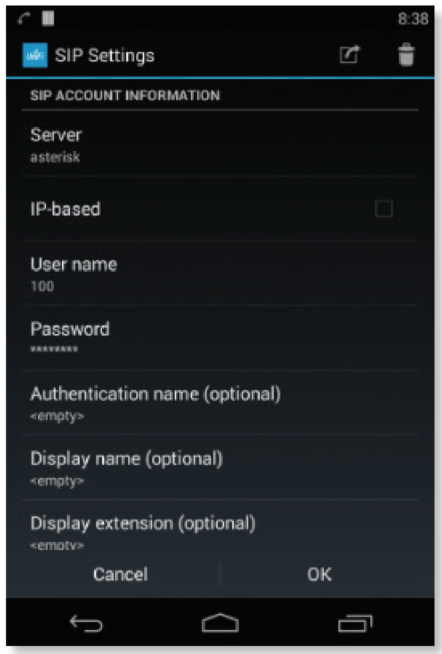
9. Finally, complete the *Value* field with the codecs of your preference in order of precedence, separated by commas. Then, click **Save** to save the form.



Configuring the Phone's SIP Settings

Before you can configure the UniFi VoIP Phone's SIP settings, perform initial configuration on the phone by following the instructions in **"Initial Configuration" on page 26**. Then, configure the phone's SIP account by following these steps:

1. Press the **Settings** icon at the bottom left of the *Welcome* screen to display the *Phone Settings* page.
2. Press **SIP service**.
3. Press **SIP accounts**.
4. Press **Add account**. The *SIP Settings* page is displayed.



5. To complete the phone's configuration, fill in the following fields:
   - **Server** Set this to the IP or hostname of your FusionPBX PBX.
   - **User name** Set this to the value of the *Extension Number* field (on the *Extension Add* page in step 4 above).
   - **Password** Set this to the value of the *Password* field on the *Extensions* page (to display the page, click **Accounts** > **Extensions**).
   - **Voicemail number** Set this to ***98**.
6. The new account will be displayed on the *SIP accounts* page.

   If the new account was properly configured, *Receiving calls* will be displayed.

   If it was not properly configured, *Not connected to server (error: 408)* will be displayed.

Chapter 8: General Phone Configuration

This chapter explains how to perform initial configuration on a UniFi VoIP Phone for use with a third-party PBX. This initial configuration, which must be done for each phone in your system, is described in the following section:

- **"Initial Configuration" on page 26**

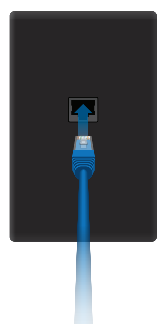
This chapter also explains how to perform the following additional configuration on a UniFi VoIP Phone:

- **"Audio Codecs and DTMF Configuration (Optional)" on page 27**
- **"Incoming Call Action" on page 27**
- **"Removing an Account (Optional)" on page 28**

Initial Configuration

For each UniFi VoIP Phone, perform initial configuration as follows:

1. Power on the phone:
   a. Connect one end of an Ethernet cable to the wall jack that is connected to your 48V, 802.3at-compliant switch.



   b. Connect the other end of the Ethernet cable to the port labeled *PoE/LAN* on the phone.

UVP, UVP-Pro

UVP-Executive

WARNING: Ensure that you refer to the correct illustration for your model (the *PoE/LAN* port is not in the same position on all models). Do NOT connect a powered Ethernet connection to the port labeled *PC*.

2. When the *Select Language* screen appears, select your language and press ▶ to continue.
3. The *Got Google?* screen appears. If you do not have a Google® account, create one as follows:
    a. Select **No**, and then select **Get an account**.
    b. Enter the user information for your account:
        i. The *Your name* screen displays a keyboard.
        ii. Enter your first name and press **Next**.
        iii. Enter your last name and press **Done**.
        iv. Press the down arrow below the keyboard.
        v. The *Your name* screen displays the *First* and *Last* names you entered. Click ▶ to continue or click ◀ to return to the previous step.
        vi. The *Choose username* screen displays a keyboard. Enter a username (6-30 characters in length) for your Google account. Click ▶ to continue or click ◀ to return to the previous step.
4. When the *Sign In* screen appears, enter your account's *Email* address and *Password*. Then press **Next**.
5. The *Google services* screen appears. Select or unselect the services you want, and then press ▶ to continue or click ◀ to return to the previous step.
6. The *Set up payment info* screen appears. To skip this step, press **SKIP**. To provide payment information, make the desired selection and press **Continue**.
7. The *Date & time* screen displays your time zone. To change time zones, press the time zone, scroll through the list, and select the desired time zone. Then press ▶ to continue or click ◀ to return to the previous step.

The *Welcome* screen is displayed. Proceed to the *Configuring the Phone's SIP Settings* section for your PBX.

Audio Codecs and DTMF Configuration (Optional)

If you want to select which audio codecs and DTMF type to use, follow these steps:

1. On the *Phone Settings* screen, press **SIP service**.
2. Press **Audio codecs**.
3. Select the codecs you want to use (by default all codecs are selected).
4. Press the **Back** ↩ icon twice to return to the *Phone Settings* screen. Then press **Call**.
5. Scroll down, press **DTMF type**, and select the DTMF type to be used (the preferred option is **RFC-2833**).

Incoming Call Action

1. On the *Phone Settings* screen, press **Call**.
2. Press **Incoming call action**, then select what the phone will do when an incoming call is received. The available options are:

Action	Description
Ring	The incoming call terminates on the UniFi VoIP Phone.
Do not disturb	The UniFi VoIP Phone will not accept calls and incoming calls will be sent to voicemail.
Forward	All incoming calls are redirected to a different destination.
Auto-answer	Incoming calls are answered automatically.

Removing an Account (Optional)

1. On the *Phone Settings* screen, press **SIP service**.
2. Press **SIP accounts**.
3. Drag the account you want to remove to the *Trash* icon in the top-right corner of the screen.

Note: The contacts you have added on Android will be available to the UniFi VoIP Phone but you won't be able to add contacts manually from within the UniFi VoIP Phone application.

Appendix A: Important Warning Regarding Emergency Calls

NOTE THAT THE LIMITATIONS SET FORTH BELOW ARE APPLICABLE TO ALL EMERGENCY CALLS, FUNCTIONS AND SERVICES, INCLUDING 911, ENHANCED 911 AND 112 CALLS (COLLECTIVELY, "EMERGENCY SERVICES").

1. You should be aware that:
   a. Emergency Services may not connect to the Public Service Answering Point ("PSAP"), or may ring to the administrative line of the PSAP, which may not be staffed after hours, or by trained emergency operators.
   b. VoIP customers may need to provide location or other information to their VoIP providers, and update this information if they change locations, for their Emergency Services to function properly.
   c. VoIP service may not work during a power outage, or when the Internet connection fails or becomes overloaded. Consider installing a backup power supply, maintaining a traditional phone line or having a wireless phone as a backup.
   d. VoIP service will not function if the telephone equipment or other equipment necessary to place calls is not correctly configured.
   e. Emergency Services may correctly connect to the PSAP, but may not transmit the user's phone number and/or location information.
   f. VoIP calls may not be capable of being received and/or processed by an emergency call center due to the center's technical limitations.
   g. VoIP calls may be affected by other factors or force majeure events, such as the quality of the broadband connection and network congestion.
2. You should:
   a. Provide your accurate physical address to your interconnected VoIP service provider to ensure that emergency services can quickly be dispatched to your location.
   b. Be familiar with your VoIP service provider's procedures for updating your address, and promptly update address information in the event of a change.
   c. Have a clear understanding of any limitations of your Emergency Services.
   d. Inform children, babysitters and visitors about your VoIP service and its Emergency Services limitations.
3. By installing the phone, you are affirmatively acknowledging that (i) you have read and understood this Warning, (ii) you understand that you may not be able to contact emergency services by dialing or using any Emergency Service, and (iii) you understand that you must inform users of the phone that they may not be able to contact emergency services by dialing or using any Emergency Service.
4. **IN NO EVENT SHALL UBIQUITI NETWORKS, ITS AFFILIATES, OFFICERS, DIRECTORS, EMPLOYEES, REPRESENTATIVES, AGENTS OR ANY OTHER THIRD-PARTY PROVIDER OR VENDOR WHO MAY FURNISH SERVICES OR PRODUCTS TO YOU IN CONNECTION WITH THE VOIP SERVICES OR THE EQUIPMENT BE HELD LIABLE FOR ANY CLAIM, DAMAGE, OR LOSS WHATSOEVER ARISING FROM OR RELATING TO EMERGENCY SERVICES, AND YOU HEREBY WAIVE ANY AND ALL SUCH CLAIMS OR CAUSES OF ACTION ARISING THEREFROM OR RELATING THERETO.**

Appendix B: Contact Information

Ubiquiti Networks Support

Ubiquiti Support Engineers are located around the world and are dedicated to helping customers resolve software, hardware compatibility, or field issues as quickly as possible. We strive to respond to support inquiries within a 24-hour period.

Ubiquiti Networks, Inc.
2580 Orchard Parkway
San Jose, CA 95131
www.ubnt.com

Online Resources

Support: **support.ubnt.com**

Community: **community.ubnt.com**

Downloads: **downloads.ubnt.com**

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