

Quick Provisioning Guide for Third-Party PBX

User Guide

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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:



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 <u>"Initial Configuration" on page 26</u>. Then, configure the phone's SIP account by following these steps:

- 1. Press the **Settings** ⁽²⁾ 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.

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🍻 SIP Settings	Ľ	Û
SIP ACCOUNT INFORMATION		
Server asterisk		
IP-based		
User name 100		
Password		
Authentication name (optional) <empty></empty>		
Display name (optional) <empty></empty>		
Display extension (optional) <emotv></emotv>		
Cancel	ОК	
ţ (1

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 <u>"Creating a Phone Extension on</u> <u>Asterisk" on page 2</u>. For the *Password* field, use the setting of the <u>secret</u> 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: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>

</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 <u>"Initial Configuration" on page 26</u>. Then, configure the phone's SIP account by following these steps:

- 1. Press the **Settings** ⁽²⁾ 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 <u>"Creating a Phone</u> 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:

SCA FIDIle System
Language
English
User Name
Password
Login Maintenance Not Available
3CX Phone System Management Console v12.5.39117.982

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

3CX Server Mai	nager 🕨	Extension St							anage A	dd View	Settings Help
Ports/Trunks Status	×	Disconnect Call 🌱 Sho	w Filter							PSTN Gatewa	IV e: 17:27:2
📫 Extension Status		Status	Extension	User Status	DND	Queues	Name	IN/OUT	Caller I	VOIP Provide	r
R System Extensions Status		Not Registered	100	Available	OFF	OUT	Daniel Angel			bridge	
C 3CXPhone Clients		Registered (idle)	101	Available	OFF	OUT	Daniel Angeli			DID / Inboun	d route
and Remote Connections		Registered (idle)	102	Available	OFF	OUT	Federico Castro			Outbound ruk	0
Phones										Digital Recept	tionist
Server Activity Log										Ring Group	
🕄 Server Event Log										Call Queue	
So Services status											
> Extensions											
WebRTC Gateway											
VoIP/PSTN Gateways											
I VOIP Providers											
Inbound Bules											
P- Bridges											
OutBound Bules											
Digital Recentionist											
På Ding Creuns											
Call Guarant											
S Par Call Quedes											
Tax machines											
settings											
Updates											
C Links											
> 🕼 Help											

- 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.

3CX Server Ma	nager 🕨 Add Extension		ka ka shekara nga sa	Manage	Add V	ew Settings	Help
Ports/Trunks Status Ports/Trunks Status Portsension Status Portsension Status Portsension Status Portsension Status Portsension Portsensi	tet Einnison settings and ciki OK or Apply to diment Vote Mail Porwarding Rules Phone User Information Configue user Information below Etamsion Number First Name Last Name Email address	save changes. Provisioning SCXPhane Other Options 101 Daniel Angeli aa@bbb.com	Office Hours Scheduling Rights				
Metanticat WebAtC Gateway WebAtC Gateway WoldPSTN Gateways VolPPonViders thobund Rules Oral Bruides A Cutbound Rules Oral Bruides A Cutbound Rules A Cut	Model Number Aufberts (Mon D) and Passend are used ID Passand	y the phone to authentic ate with 3CX Phone Sy 101 (u44)yy1s (U assem. If the phone has a user id field enter the D • • • •	extension number.			
>							

- 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.

Ports/Trunks Status	Edit Extension settings and click OK or Apply to sar	ve changes.			
Extension Status	General Voice Mail Forwarding Rules Phone Pro	ovisioning 3CXPhone Other Options Office Hours Scheduling Rights			
System Extensions Status					
3CXPhone Clients	When we upable to accurate a call you can allow w	the measures to be taken			
Remote Connections	Enable Visice mail				
D Phones	Disable Voicemail PIN Authentication				
Server Activity Log	Play Caller ID				
Consistent Log	PIN Number				
Extensions	Read out date/time of message	Do not read 🔻 🧊			
WebRTC Gateway	Email Options	No email notification			
VoIP/PSTN Gateways	Manage grantings				
VoIP Providers	WAV files in user's folder				
Inbound Rules		Add new			
b Bridges		Bernard form reference			
T OutBound Rules					
Digital Receptionist		Delete Delete			
Ring Groups		2 Play on phone			
Ex Machines		Play Play			
C Settings		0			
•	 Play this greeting for all profiles 	System default 🔻 🕞			

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.

3CX Server Manage	er 🕨 Extension Created	Manage	Add	View	Settings	Help
Image: Status Image: Statu	Extension Created Extension Transfer 10 was created tor Cancil Argolia Visus in for al demonstration on how to configure and providion your SP phone at <u>Hardynour AccanobiologueanCit</u> The extension transmot Cancil Argolia Cancil					

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 <u>"Initial Configuration" on page 26</u>. 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 <u>3</u> 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.

aelastix"						691×12
	System Agenda En	ai Fax	РВХ ІМ	Re	ports	
3X Configuration Operator Panel	Voicemails Calls Recordin	gs Batch Configurations		Tools	Flash Operator Panel	
PBX Configuration						
Basic Extensions Feature Codes Outbound Routes Trunks Inbound Gall control Inbound Routes	Add SIP Extension					Add Extension
DAHDI Channel DIDs Announcements Blacklist	- Add Extension					
CallerID Lookup Sources Call Flow Control	User Extension	200				
Follow Me IVR	Display Name	UVP Testing				
Queue Priorities	CID Num Alias					
Queues Ring Groups Time Conditions	SIP Alias®					
Time Groups Internal Options & Configuration	- Extension Options					
Languages Misc Applications	Outbound CID®					
Misc Destinations	Asterisk Dial Options	tr	Ove	rride		
Music on Hold	Ring Time [®]	Default *				
Paging and Intercom	Call Forward Ring Time	Default •				
Parking Lot System Decordings	Outbound Concurrency Limit	No Limit *				

- 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.

This device uses sip to	chnology.	
secret®	1234ab	
dtmfmode 🔍	RFC 2833	
canreinvite®	No *	
context®	from-internal	
host®	dynamic	
trustrpid [©]	Yes •	
sendrpid [©]	No	
type®	friend •	
nat®	No - RFC3581 •	
port®	5060	
qualify	yes	
qualifyfreq	60	
transport®	UDP Only	
avpf®	No •	
icesupport [®]	No 🔻	
encryption [®]	No	
callgroup®		
pickupgroup		
disallow®	al	
allow	alaw&utaw	

- 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.

- Voicemail		
Status	Enabled •	
Voicemail Password	1234	
Email Address		
Pager Email Address		
Email Attachment®	yes no	
Play CID®	yes no	
Play Envelope®	yes no	
Delete Voicemail	yes no	
VM Options		
VM Context®	default	
- VmX Locater		
VmX Locater™ [©]	Disabled •	
Use When:	unavailable 🗍 busy	
Voicemail Instructions:	6 Standard Voicemail prompts.	
Press 0:0	Go To Operator	
Press 1:0		
Press 2:0		
- Optional Destinations		

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

PRESDOM TO COMMUNICATE		
System	i Agenda Email Fax BBX IM Reports 🗸	
PBX Configuration Operator Panel	Voicemails Calls Recordings Batch Configurations Conference Tools Flash Operator Panel	
PBX Configuration	Agely Config Agely Config Hease select your Device below then click Submit Device evice Generic SIP Device • Selent	Add Extension UVP Testing -2005

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 <u>"Initial Configuration" on page 26</u>. 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.

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🍻 SIP Settings		$\square_{\mathbf{t}}$	Î
SIP ACCOUNT INFORMATION	4		
Server			
asterisk			
IP-based			
User name			
100			
Password			
Authentication name (<empty></empty>	optional)		
Display name (optional <empty></empty>			
Display extension (opti <emotv></emotv>	onal)		
Cancel		ок	
\leftarrow		ā	

- 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.

Admin • Applications •	Connectivity Reports Settings UCP	Logout: a
Add an Extension		Add Extension
Add an Extension		
Please select your Device	elow then click Submit	
- Device		
Device	Generic CHAN SIP Device *	
Submit		

- 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.

Admin - Applications - Connectivity	Reports Settings UCP	Logout: admin
Add SIP Extension	Itension Are Eleman 0 200 > UAVP Testing 0	
- Add Extension		
User Extension [©]	200	
Display Name 🕫	UVP Testing	
CID Num Alias 8		
SIP Alias		
- Extension Options		
Queue State Detection ®	Use State	
Outbound CID		
Asterisk Dial Options	Ttr Override	
Ring Time 9	Default *	
Call Forward Ring Time	Default *	
Outbound Concurrency Limit	3 •	
Call Waiting	Enable *	

- 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.

Admin * Applications * Connectivity	Reports Settings UCP Apply Config	Logout: ad
Device Options		
This device uses CHAN_SIP technolo	gy listening on :5060	
Change To CHAN_PJSIP Driver [©]	Changing SIP Driver unavailable	
Secret 0	1234ab	
DTMF Signaling ⁰	RFC 2833 *	
Can Reinvite 🕫	No T	
Context [®]	from-internal	
Host ⁹	dynamic	
Trust RPID 8	Yes •	
Send RPID [©]	Send P-Asserted-Identity header *	
Connection Type	friend •	
NAT Mode	Yes - (force_rport.comedia)	
Port	5060	
Qualify ⁶	yes	
Qualify Frequency	60	
Transport [©]	All - UDP Primary *	
Enable AVPF ⁹	No T	
Force AVP	No •	
Enable ICE Support	No T	
Enable Encryption	No	
Call Groups [©]		
Pickup Groups		
Disallowed Codecs		
Allowed Codecs		
Dial 0	SIP/200	
Account Code 🕫		
Mailbox 🛛	200@device	
Voicemail Extension 🛛		
Deny 9	0.0.0.00.0.0.0	
Permit 0	0.0.000.0.0	

- 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.

Admin • Applications • Connectivity	Reports Settings UCP Apply Config	Logout: adr
Voicemail		
itatus	Enabled •	
oicemail Password [©]	200	
equire From Same Extension 🛛	yes no	
mail Address 🔮		
ager Email Address 🛛		
mail Attachment 🛛	yes no	
Play CID 🕫	yes no	
Nay Envelope 🤨	yes no	
elete Voicemail 🕫	yes no	
M Options		
M Context 😌	default	
iSymphony Settings		
dd to iSymphony	yes no	
uto Answer 🕫	yes no	
DTLS		

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

Admin + Applications + Connectivity	Reports + UCP Apply Config	Logout
Extension: 200		Add Extension UVP Test <200>
S Delete Extension 200		
Sedit Follow Me Settings		
- Edit Extension		
Display Name	UVP Test	
CID Num Alias		
SIP Alias		
- Extension Options		
Queue State Detection	Use State *	
Outbound CID 9		
Asterisk Dial Options	Ttr 🔲 Override	
Ring Time ⁹	Default *	
Call Forward Ring Time 9	Default •	
Outbound Concurrency Limit	3 •	
•	Coable +	

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 <u>"Initial Configuration" on page 26</u>. 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.

		8:38
SIP Settings	<u>r</u>	
SIP ACCOUNT INFORMATION		
Server		
asterisk		
IP-based		
User name		
Password		
Authentication name (optiona <empty></empty>	I)	
Display name (optional) <empty></empty>		
Display extension (optional) <emotv></emotv>		
Cancel	ок	
t A	-	

- 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 <u>4</u> 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[®] (<u>http://www.fusionpbx.com/</u>) 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**.

System Acc	counts Dialplan	Apps	Status	Advanced			
User Dashbo Quickly access in User Informat	is ited to your account. ager						
Username:	fusionpbx						
Voicemail:	View Messages						
Extension			Tools		Description		
Ring Group Extension				Tools		Description:	

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

System	Accounts Dialplan	Apps	Status	Advanced		
Extensions Use this to configure y	our SIP extensions.					SEARCH
Extension	Call Group		Context	Enabled	Description	8
200			default	True		× 🗙
						63

- 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*).

	ш
System Accounts Dialplan	Apps Status Advanced
Extension Add	BACK SAVE
EARTISION.	201 Enter the alphanumeric extension. The default configuration allows 2 - 7 digit extensions.
Number Alias:	Rusionpbx If the extension is numeric then number alias is optional.
Range:	Letter be mutation of extensions to oreade. Forcements each extension by 1.
Voicemail Password:	Enter the numeric voicemail bassword here.
Account Code:	1921581122 Effer the account code here.
Effective Caller ID Name:	Daniel Angell Enter the internal caller ID name here.
Effective Caller ID Number:	201 Enter the kitemal caller ID number here.
Outbound Caller ID Name:	UBNT Enter the external (public) caller ID name here.
Outbound Caller ID Number:	12011 Enter the external (public) caller ID number here.
Emergency Caller ID Name:	Enter the emergency caller ID name here.
Emergency Caller ID Number:	Enter the emergency caller ID number here.
Directory Full Name:	Enter the first mame followed by the last name.
Directory Visible:	Тие •
Directory Extension Visible:	Select whether so hade the name norm the arectory.
Limit May:	Select whether announce the extension when calling the directory.
Little (1984)	5 Enter the max number of outgoing calls for this user.
Limit Destination:	Enter the destination to send the calls when the max number of outgoing calls has been reached.
Voicemail Enabled:	True Enabledisable voicemail for this extension.
Voicemail Mail To:	Enter the email address to send voicemail to (optional).
Voicemail Attach File:	True Choose whether to attach the volcemail to the email as an audio file.
VM Keep Local After Email:	True Keep local file alter sending the email.
Toll Allow:	International Enter the toil allow value here. (Examples: domestic.international.local)
Call Timeout:	B0 Enter the call timeout
Call Group:	Enter the uner call group here. Groups available by default: sales, support, billing.
Record:	Disabled Choose whether to record local, inbound, outbound, or at.
Hold Music:	Default Select the MOH Category here.
Context:	default Ener the user context here.
	ADVANCED
Enabled:	True Set the status of the extension.
Description:	

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.

					Update Cor	npleted					
🤯 FUS	SIONPBX									ш	
System	Accounts	Dialplan	Apps	Status	Advance	al.					
Extension									BACK COP	Y SAVE	
		Extension:	201 Enter the alph	nanumeric extension.	The default confi	guration allows 2 - 7	digit extensions.				
		Number Allas:	fusionpbx If the extensio	on is numeric then nu	mber alias is optic	onal.					
		Password:	Enter the pas	sword here.							
		User List:	Assign the us	ADD ers that are assigned	I to this extension.						
		Voicemail Password:	Enter the num	neric voicemail passu	vord here.						
		Account Code:	192.168.1	22 ount code here.							
	Effe	ctive Caller ID Name:	Daniel Ang Enter the inte	rell mai caller ID name h	ere.						
	Effecti	ive Caller ID Number:	201 Enter the inte	mai caller ID number	here.						
	Outbr	ound Caller ID Name:	UBNT								

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.

System	Accounts	Dialplan	Apps	Status	Advanced		
					Adminer		
ser Dashboard					App Manager		
uickly access informat	tion and tools relate	d to your account.			Command		
User Information					Databases		
Username:		fusionpbx			Default Settings		
(ninemal)		Manu Massanan			Domains		
voiceman.		view messages			Grammar Editor		
					Group Manager		
Extension	Tools				PHP Editor	Description	
200	Call Eon	eard Eollow Ma	o Not Disturb		Provision Editor	boothphon	
201	Call Ford	rard Eollow Ma	o Not Disturb		SIP Profiles		
	<u>Start on</u>	the Longitude a	of their personality		Script Editor		
					Upgrade		
					XML Editor		
Ring Group Extens	lon				Tools	Description:	

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.

	Accounts Only	olan Apps	Status	Advanced		
SIP Profiles	for CID profiles					
Name	Hostna	me	Description			
external			The External profile of Calls can be sent usi	ternal provides anonymous calling in the public g a SIP URL "voip.domain.com:5080"	context. By default the External profile binds to port 5080.	
internal-ipv6			The Internal IPV6 pro	le binds to the IP version 6 address and is simil	ar to the Internal profile.	XX
internal			The Internal profile b 5060.	default requires registration which is used by th	e endpoints. By default the Internal profile binds to port	XX

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

Settings			
Name	Value	Enabled	Description
debug	0	True	🖉 🗙
sip-trace	no	True	🖉 🗙
sip-capture	no	True	🖉 🗙
watchdog-enabled	no	True	🖉 🗙
watchdog-step-timeout	30000	True	🖉 🗙
watchdog-event-timeout	30000	True	🖉 🗙
log-auth-failures	true	True	🖉 🗙
forward-unsolicited-mwi-notify	false	True	🖉 🗙
context	public	True	🖉 🗙
rtc2833-pt	101	True	🖉 🗙
sip-port	\$\${internal_sip_port}	True	✓ ×
dialplan	XML	True	🖉 🗙
dtmf-duration	2000	True	🖉 🗙
inbound-codec-prefs	\$\${global_codec_prefs}	True	🖉 🗙
outbound-codec-prefs	\$\${global_codec_prefs}	True	✓ ×
rto-timer-name	soft	True	<i>I</i> ×

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.

Profile Setting					1	BACK SAVE
	Name:	inbound-codec-prefs				
	Value:	PCMU,PCMA,GSM				
	Enabled:	True Choose to enable or disa	le this.			
	Description:	Enter the description.				
						SAVE

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 <u>"Initial Configuration" on page 26</u>. 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.

с 🔳		8:38
🐱 SIP Settings	Ľ	Ĵ
SIP ACCOUNT INFORMATION		
Server asterisk		
IP-based		1
User name		
Password		
Authentication name (optional) <empty></empty>		
Display name (optional) <empty></empty>		
Display extension (optional) <emotv></emotv>		
Cancel	ок	
Ĵ		

- 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 <u>4</u> 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:

• <u>"Initial Configuration" on page 26</u>

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

- <u>"Audio Codecs and DTMF Configuration (Optional)" on page 27</u>
- <u>"Incoming Call Action" on page 27</u>
- "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.



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 b 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** for twice to return to the *Phone Settings* screen. Then press **Call**.
- 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.
- 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* i 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: <u>support.ubnt.com</u> Community: <u>community.ubnt.com</u> Downloads: <u>downloads.ubnt.com</u>



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