Counterpath X-Lite Softphone

Introduction

Counterpath has been known for high quality SIP softphones for some time. Initially, Counterpath (then Xten) launched X-Pro and the free, but slightly limited version, X-Lite. Their 2nd generation softphone, eyeBeam, then replaced X-Pro in September 2004. The most significant change was the switch to the reSIProcate SIP stack from SIPfoundry. This enabled the phone to support all the advanced NAT traversal protocols and added a lot of functionality. The Counterpath eyeBeam softphone also supports video with consistently high quality.

? Unknown Attachment

Different versions of softphones from Counterpath:

1. X-Lite V3.0 for Windows (free)
2. X-Lite V2.0 for Linux (free)
3. X-Lite V3.0 for Max OS X (free)
4. X-Lite V4.0 beta for Windows (free)
5. X-Lite V4.0 beta for Mac (free)
6. eyebeam V1.5 for Windows and Mac OS X
7. Bria and Bria Professional (offers auto provisioning)

How to use the Xten eyeBeam Softphone

Settings using the sipX Configuration Server

The Xten family of softphones is not yet supported for plug & play management by the sipX Configuration Server. Adding support would not be too difficult.

All the Xten softphones can be manually configured to work with the sipX system. In order to use it you first have to create a new user in the sipX Configuration Server who will be using the phone. The SIP credentials (User ID and SIP password) then have to be transferred to the phone manually and the phone is configured using its own configuration menu.

sipXconfig

Home Help Logout Search

| Users | Devices | Features | System | Diagnostics | |

▶ **Identification**

▸ Phones
▸ Call Forwarding
▸ Schedules
▸ Speed Dial
▸ Group Supervisor
▸ Personal AutoAttendant
▸ Registrations
▸ Permissions
▸ Caller ID

User: 300

Show Advanced Settings

Existing Groups:
administrators, import

New Groups: You can create new groups simply by adding the new group name to the Groups form value.

User ID
`300`
The User ID can be a numeric extension like "123" or a name like "jsmith". The User ID is displayed by the phone and it is therefore recommended to use the internal extension or the name of the user. If using Direct Inward Dialing (DID), then it is recommended to define the DID number (or its DNIS portion) as an alias.

Last name
`John`

First name
`Donovan`

Active greeting
`default system greeting`
Voicemail prompt callers will hear before leaving a message.

E-mail address
`donovan@example.cor`
Used for sending notification about new voicemail left for this user. Leave empty to disable e-mail notification.

Attach voicemail ☑
If checked, the voicemail message will be attached to the notification e-mail. Otherwise, the e-mail will contain a link to retrieve voicemail message.

Additional E-mail address
Used for sending voicemail message notification to the additional e-mail address.

Attach voicemail ☐
If checked, the voicemail message will be attached to the notification email sent to the additional e-mail address.

PIN
`********`

Confirm PIN
`********`
The PIN is a password used to log in to voicemail or to the user portal. Numeric PINs are recommended, since only numbers can be dialed.

Groups
`import`
List all groups for this user. If a group does not exist, it will be created. When entering multiple groups, separate them with spaces.

Aliases
`donovan`
Aliases are additional names for the user. Like the user ID, an alias can be either a numeric extension or a name. When entering multiple aliases, separate them with spaces.

OK   Apply   Cancel

⚠ **Release 3.0** and higher of sipX automatically generate a SIP password when a new user is created. The SIP password has to be programmed into the phone and is required for the line to register. This password is **different from the PIN** entered on the "New User" page. In order to see or change the assigned SIP password, click on the "Show Advanced Settings" link.

ⓘ If you would like the softphone to have an entry under *Devices* in the sipX Configuration Server, you can create an *unmanaged phone*. The user line can then be assigned to that phone. Even though no profile is generated for this phone, the administrator can see what users and lines are assigned to which phones. Choose a random serial number (12 digits) as an identifier or alternatively you could enter the MAC address of the laptop or PC on which the softphone is installed.

Configuring the eyeBeam Phone

Install the eyeBeam softphone application following the instructions provided by Xten. Launch the eyeBeam application and right-click your mouse and select *Settings*. Go to *SIP Accounts* and create a new line as follows:

> ⚠️ The example provided above uses DNS SRV. If DNS SRV is not used please enter the fully qualified host name of the sipX server in the field labeled *Domain*. Add *User name* and *Authorization user name* that matches the *User ID* defined for the user created in sipX. The *Password* is the **SIP Password** auto-generated by the sipX Configuration Server and not the PIN. Check *Register with domain* and don't forget to enable the account (check box at the top).

Enable Presence Indication

The eyeBeam softphone is able to subscribe to and display presence of other users. We have tested this with eyeBeam revision *eyeBeam 1.1 3010n stamp 19039* and **Polycom phones** (SIP firmware 1.5.2 and 1.6.2).

Open the right sliding drawer window of the eyeBeam phone. A buddy list appears that at the top indicates your own presence, gives access to received call and dialed call logs, as well as allows the creation of a buddy list. Using the SIP presence capabilities of the eyeBeam softphone you can subscribe to the state of any line registered on a Polycom phone. The state of a line is either changed automatically by the Polycom phone (e.g. when you are on the phone), or you can manually change state using the softkey *MyStat* on the Polycom phone. The following states are supported:

- **Green:** A *green* icon indicates that the line is registered (i.e. the phone is plugged in and configured).
- **Red:** A red circle indicates that the person is either on the phone or manually set the state to *Busy*.
- **Clock:** A little clock icon indicates that the person manually set the state to *Away* or *Be right back*.
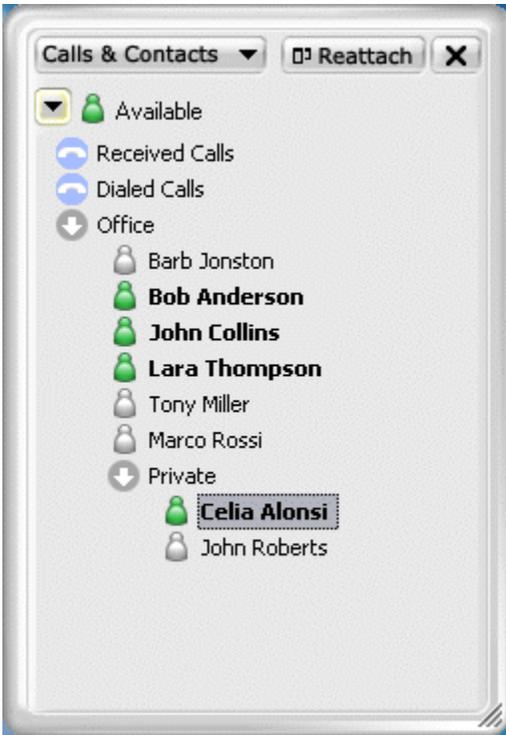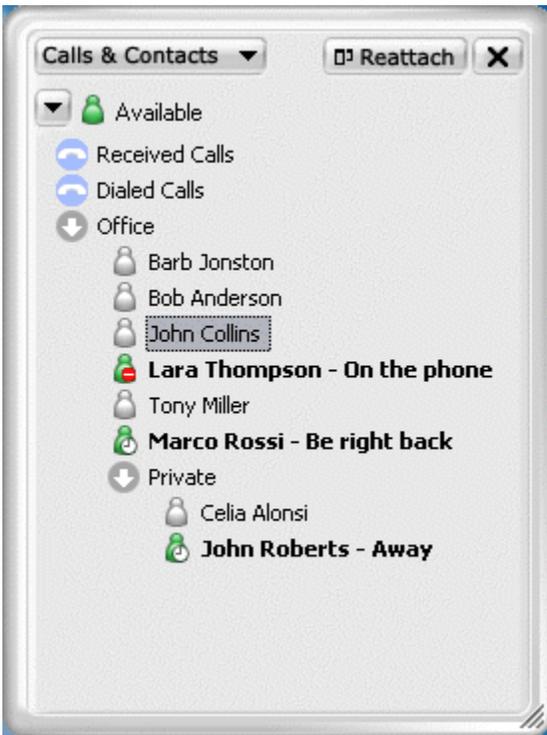
⚠ Contacts can be exported and imported. Therefore, it is easily possible to create a company-wide contact list that can be loaded into all the eyeBeam softphones.

ⓘ **Be Careful**

The eyeBeam softphone can be used as a simple attendant console where an assistant can check the state of the line for a group of people.

⚠ This feature should also work with Snom phones. However, we have not yet tested this fully.

⚠️

How to Configure Presence Indication

Nothing has to be configured on the Polycom phone. The default profile generated by the sipX Configuration Server provides all the necessary settings.

On the eyeBeam phone presence has to be enabled as follows:

⚠️ The presence setting is *peer-to-peer* because the eyeBeam phone subscribes to the state of every phone directly without using a presence server.

Configuring the buddy list:

Again, this setting is for an installation with DNS SRV enabled. Enter the fully qualified host name of the sipX server if DNS SRV is not used.

The eyeBeam phone allows to setup groups.

How to Configure Message Waiting Indication (MWI)

The eyeBeam softphone release *1.1 3010n stamp 19039* (November 2005) does not yet properly support Message Waiting Indication. Xten has promised to fix this in the near future. The phone needs to send a subscribe message to the sipX server in order to receive a *notify* about message state changes.

eyeBeam has been corrected as of version 1.5.

This is apparently still an issue with X-LITE Windows version 3.0 build 47546.

Configure X-Lite to register with sipX

X-Lite does not distinguish realm from domain so you must edit configure DNS SRV so you domain resolves to the machine you have the PBX installed for SIP traffic OR change you realm to be your fully qualified domain name. See this issue for more

http://thread.gmane.org/gmane.comp.telephony.pbx.sipfoundry.general/3547

Add a USER who will be using the X-Lite softphone

The Xten family of softphones is not yet supported by ConfigServer. Adding support would not be too difficult. In order to use it you first have to create a new user in the sipX ConfigServer who will be using the phone. The SIP credentials (User ID and SIP password) have to be transferred to the phone manually and the phone is configured using its configuration menu.

sipXconfig

| Users | Devices | Features | System | Diagnostics |

▸ Identification
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First name : `Donovan`

Active greeting : `default system greeting`

Voicemail prompt callers will hear before leaving a message.

E-mail address : `donovan@example.cor`

Used for sending notification about new voicemail left for this user. Leave empty to disable e-mail notification.

Attach voicemail : ☑

If checked, the voicemail message will be attached to the notification e-mail. Otherwise, the e-mail will contain a link to retrieve voicemail message.

Additional E-mail address :

Used for sending voicemail message notification to the additional e-mail address.

Attach voicemail : ☐

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PIN : `********`

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Aliases : `donovan`

Aliases are additional names for the user. Like the user ID, an alias can be either a numeric extension or a name. When entering multiple aliases, separate them with spaces.

[OK] [Apply] [Cancel]

⚠ **Release 3.0** and higher of sipX automatically generates a SIP password when a new user is created. The SIP password is programmed in the phone and required for the line to register. This password is **different from the PIN** entered on the "New User" page. In order to see or change the assigned password, click on the "Show Advanced Settings" link.

Configure the X-Lite 2.0 phone on Linux

1. Select MENU (the button on the left side of the green telephone button).
2. Double-click on SYSTEM SETTINGS.
3. Double-click on SIP PROXY.
4. Double-click [DEFAULT] to set up your sipX user configs for registration here.

```
ENABLED - yes
DISPLAY NAME - any string
USER NAME - sipX USER ID only
AUHTORIZATION USER - sipX USER ID only
PASSWORD - the PIN you assigned to the user
DOMAIN/REALM - if using DNS SRV records, then enter the domain.com only
(this will also fill in the [DEFAULT] menu field on the previus config page)
SIP PROXY - FQDN of the sipX server
```

5. Click BACK to see the [DEFAULT] menu
   You should see the entry you made in the DOMAIN/REALM field here.
6. Go back to SYSTEM SETTINGS and then to the NETWORK menu

```
AUTOSELECT IP - yes
OUT BOUND SIP PROXY - IP of your sipX server
PRIMARY DNS SERVER - IP of your primary DNS server
```

7. Click BACK
8. Restart the X-Lite phone

Configure the X-Lite 3.0 phone on Windows

1. Right click on Display Area.
2. Select SIP ACCOUNT SETTINGS from Menu.
3. Click on the ADD button.

```
DISPLAY NAME - any string
USER NAME - sipX USER ID only
PASSWORD - the PIN you assigned to the user
AUHTORIZATION USER - sipX USER ID only
DOMAIN - if using DNS SRV records, then enter the domain.com only
(this will also fill in the [DEFAULT] menu field on the previus config page)
```

4. Click on the VOICEMAIL tab at the top of the window

```
NUMBER TO DIAL FOR CHECKING VOICEMAIL - set it to 101
NUMBER FOR SENDING CALLS TO VOICEMAIL - 8 and extension (extension 200 would set this to 8200)
```

5. Click the OK button and then click the CLOSE button on the next window.

6. Phone should register

Force X-Lite to use UDP Transport

1. Use ***7469
2. Filter for "transport"
3. change "auto" to "udp" for "proxies:proxy0:transport" and "system:network:transport".
4. Close the window and save changes.
5. Restart.