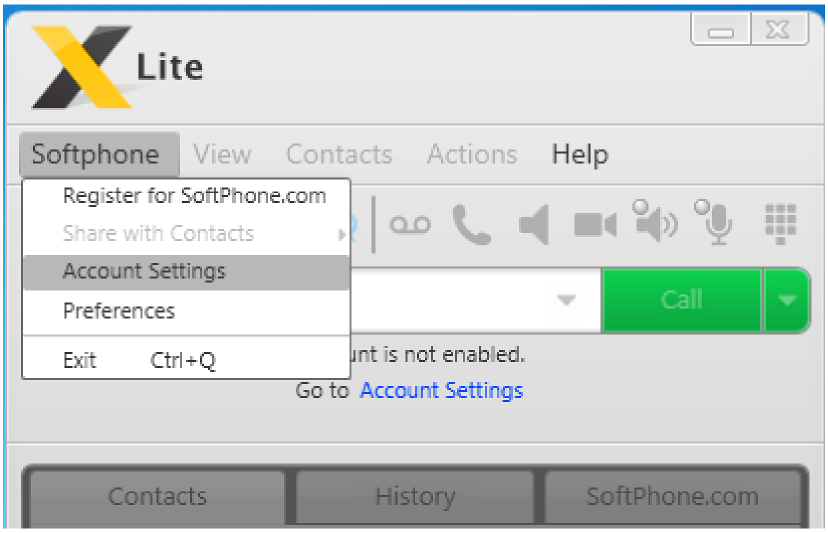
X-Lite setup

Account settings

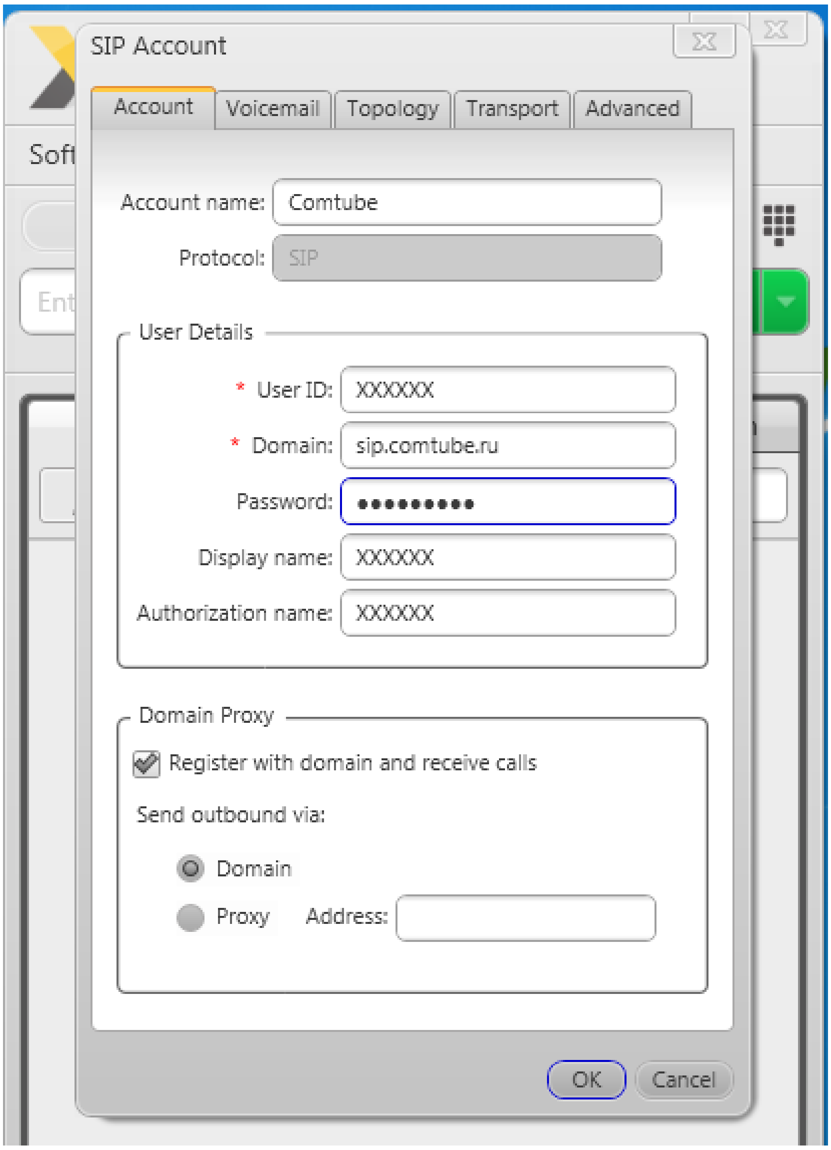
1. Open **«Account settings»** in tab **«Softphone»**.



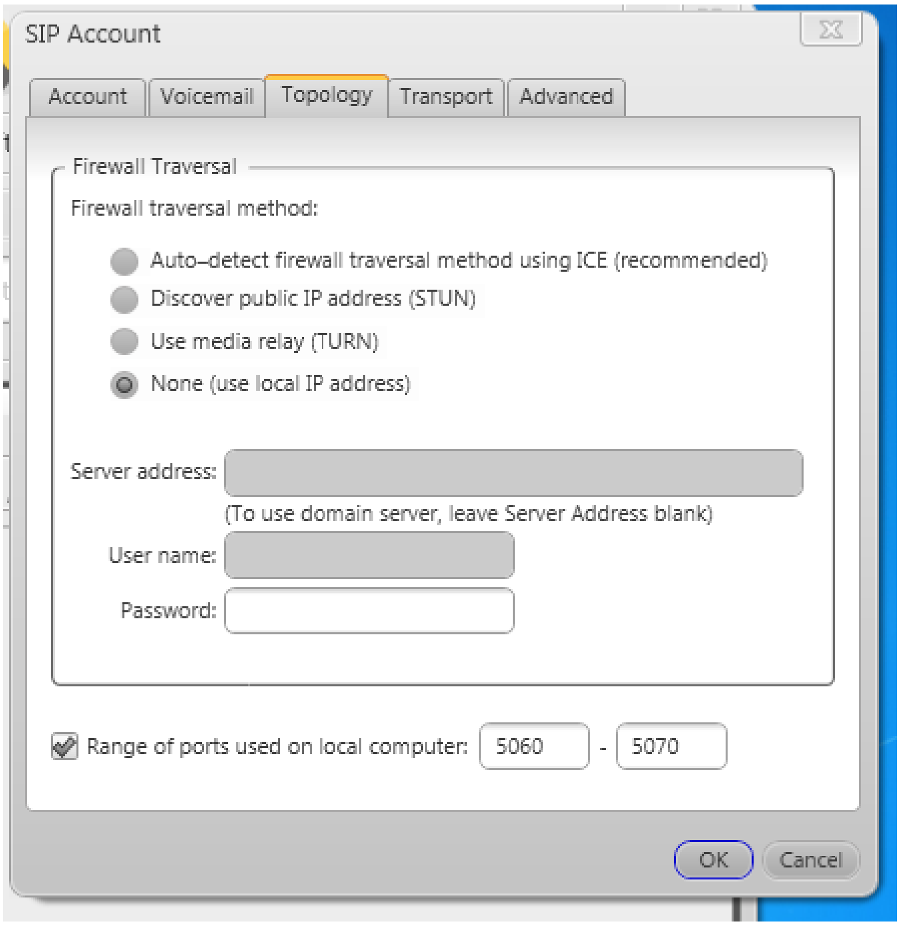
2. Specify the following settings in «**SIP Account**»:

- **UserID:** your 6-digit SIP-number
- **Domain:** sip.comtube.com
- **Password:** your password on comtube.com
- **Display name:** your 6-digit SIP-number
- **User name:** your 6-digit SIP-number

If you want to receive incoming calls, set «**Register with domain and receive incoming calls**» option. Otherwise X-Lite will not register on our server and will not be able to receive incoming calls.



3. Go to «**Topology**» page. Here you can setup STUN server if needed. We recommend the following parameters:



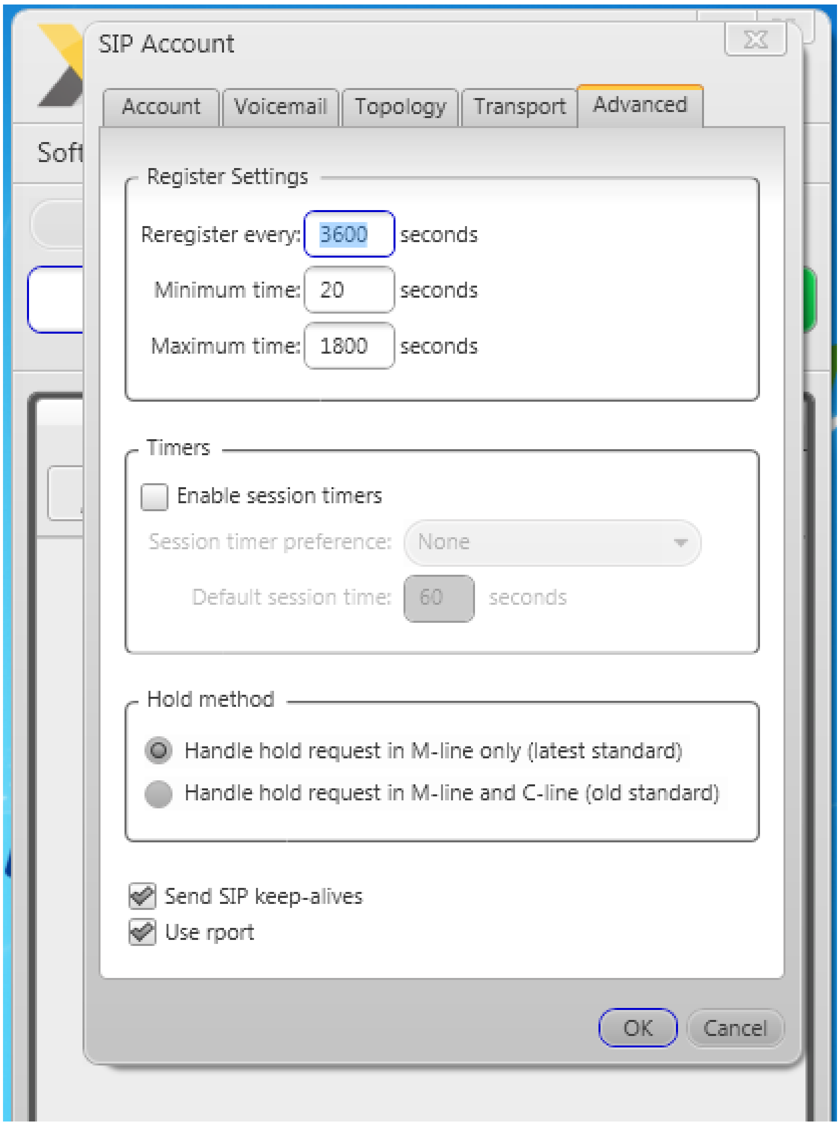
In this case STUN server is not used. If you need one, set the following parameters:

- Discover global address
- Use specified server: **stun.xten.com** (or **sip.comtube.com**):

NOTE!

We recommend to use STUN server ONLY in case of problems with incoming or outgoing calls.

4. Go to **«Advanced»** page. Here you may change re-register timeout.



We recommend settings shown on the screenshot. Do not set low re-register timeout. Typical values are 3600 (1 hour) or 600 (10 minutes). After all settings press "**OK**". After that X-Lite will attempt to register on our server. You may see the following messages (in order of appearance):

- Discovering network
- Registering
- Ready

Since "Ready" you should be able to make and receive calls.

Possible problems

Here are typical possible problems and ways to solve them.

401 – Unauthorized

Check if login and/or password are specified correctly (login is 6-digit SIP number, not your login on comtube.com). Check authorization data (see p.1), fix it (if needed) and retry. Also check “Domain” – probably you specified different server or made a typo.

408 – Timeout

X-Lite is unable to communicate with our server – your firewall/router blocks incoming/outgoing UDP traffic. Check firewall/router settings and retry. Also this error may occur if you use proxy to access to the internet (SIP protocol does not allow proxies).

503 – Internal Server Error

This error occurs if you use HTTP/FTP proxy to access to the internet.

No voice

Probably your firewall/router blocks incoming/outgoing UDP traffic. Check firewall/router settings and retry.
Also it may be a codec problem. Check X-Lite’s codec setup (see below) and try to leave only one codec (for example, G.711).

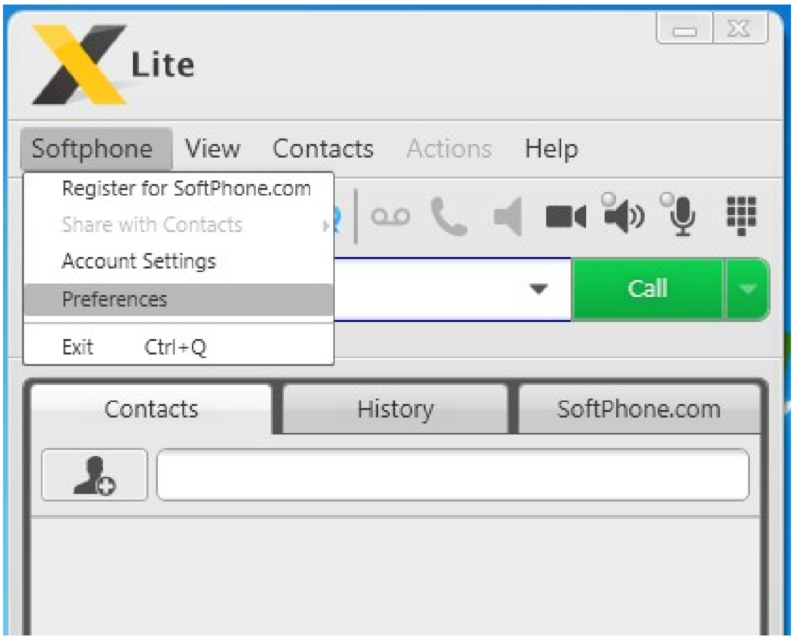
NOTE!
Even if there’s no voice, billing system works and makes write-offs. So if you encounter this problem, try to hang up in 6 seconds after a call is established.

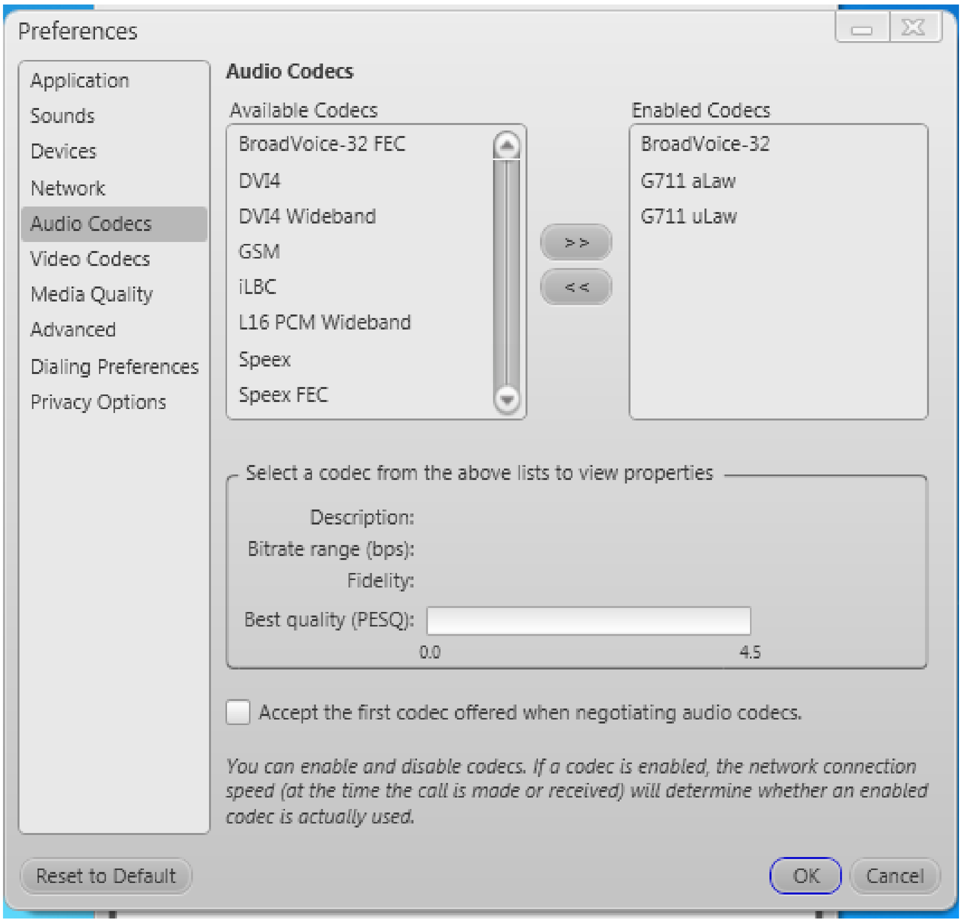
Incoming calls do not work

First of all check you call forward schedule on comtube.com - see «SIP Phone - Settings». For example, that call forwarding works from 10:00 till 23:00, but someone called you before 10:00 or after 23:00. This is also possible if “From” and “Till” values coincide – do not set “From 00:00 till 00:00”.
If schedule is correct, probably X-Lite has lost it’s registration. Restart the program and try to specify lower re-register timeout (see p.5).

Codec setup

In order to set codec you need to choose**«Preferences»** **in** menu **«Softphone»** and choose **«Audio Codecs»**.


Lite
Softphone
View
Contacts
Actions
Help
Register for SoftPhone.com
Share with Contacts
Account Settings
Preferences
Exit Ctrl+Q
Call
Contacts
History
SoftPhone.com


Preferences
Application
Sounds
Devices
Network
Audio Codecs
Video Codecs
Media Quality
Advanced
Dialing Preferences
Privacy Options
Audio Codecs
Available Codecs
BroadVoice-32 FEC
DVI4
DVI4 Wideband
GSM
iLBC
L16 PCM Wideband
Speex
Speex FEC
Enabled Codecs
BroadVoice-32
G711 aLaw
G711 uLaw
Select a codec from the above lists to view properties
Description:
Bitrate range (bps):
Fidelity:
Best quality (PESQ):
0.0
4.5
Accept the first codec offered when negotiating audio codecs.
You can enable and disable codecs. If a codec is enabled, the network connection speed (at the time the call is made or received) will determine whether an enabled codec is actually used.
Reset to Default
OK
Cancel

On the right list there are codecs used by program and on the left list there are other codecs which the program maintains.

Our platform supports only **G711 aLaw, G711 uLaw, iLBC**. We recommend to enable only codecs supported by our platform (or one of them).

We recommend to turn off video codecs, because video calls are not supported.

Outgoing calls

To make an outgoing call a number should be in the following format:

«country code» «city/operator code» «phone number»

Example: 74959610008

NOTE!

Do not use local prefixes for long distance and international calls, like 8 and 810 in Russia – only international format is allowed. Also, do not specify leading “+” sign.

Your account’s balance MUST BE POSIVIVE for outgoing calls!

Incoming calls

To connect to comtube’s SIP number from PSTN, dial access number:

+7 (495) 956-88-50

When you hear the answer, dial 6-digit SIP number. For example, 104705.

If someone calls you when you are offline or you did not set call forwarding, voicemail will answer the call and attempt to record a message (or receive fax). You can see new messages on «**Voicemail – Incoming» page**.

NOTE!

Your account’s balance MUST BE POSIVIVE for incoming calls!

Internal calls

To make a call to other comtube user, dial 6-digit number, for example 104706.

Comtube
Tel/Fax: **+7 (495) 961-00-08**
E-mail: support@comtube.ru
www: www.comtube.ru