

X-Lite setup HowTo

Create account

1. Open **«SIP account settings»** menu and press "Add" button.



Enabled	Acct #	Domain	Username	Display Name	<u>A</u> dd
	1				
					Remove
					Properties.
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- 2. Specify the following settings:
 - **Display name:** your 6-digit SIP-number
 - User name: your 6-digit SIP-number



- **Password:** your comtube.com password
- Authorization user name: your 6-digit SIP-number
- **Domain:** sip.comtube.com

Properties of Account 1		
Account Voicemail Topology	Presence Advanced	
-User Details		
Display Name	XXXXXX	
User name	XXXXXX	
Password	•••••	
Authorization user name	XXXXXX	
Domain	sip.comtube.com	
Domain Proxy Register with domain and receive incoming calls Send outbound via: Image: The second se		
Dialing plan	1	
	ОК Отмена Применить	

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 here:



roperties of Account 1	
Account Voicemail Topology	Presence Advanced
Firewall Traversal IP address Discover global addre Use local IP address STUN server Discover server Use specified server:	55
Port used on local computer	
_XTunnels	
Use XTunnels	Never
Server address	
Use SIP user name and p	assword
Username	
Password	
	ОК Отмена Применить

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 **«Presence»** page. Here you may set availability option, to make other users see your current status (Online, Do not disturb, etc). Set **«Mode»** parameter to **«Presence Agent»**.



Properties of Account 1		
Account Voicemail Topology	Presence Advanced	
Mode:	Presence Agent	
Poll time	300	seconds
Refresh interval	3600	seconds
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Note, that you can see other users statuses only if they are registered and if they use the same program (not all SIP phones support such a functionality).

When filling an address book in X-Lite with your comtube fellows, choose **«Softphone»** field in **«Contact methods»** list, and specify 6-digit SIPnumber. For other users, who use other SIP networks, specify full SIP number (number@domain).

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



Properties of Account 1		
Account Voicemail Topology	Presence Advanced	
Register Settings		
Reregister every	3600	seconds
Min. time	20	seconds
Max. time	180	seconds
Advanced Options		
Send SIP keep-alives	🔽 Use rport	
Enable session timers	Default session time 60	seconds
Session Timer Preference	None	~
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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 changins settings, press "**OK**". Right after this X-Lite will attempt to register on our server. You should 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 at 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 proxy to access to the internet (SIP protocol does not allow proxies). Also, you may encounter this problem in case of technical problems on our side. If you don't use proxy and encounter this error, please contact technical support.

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

To setup codecs choose **«Options…»** menu and go to **«Advanced»** tab.

Options	
General	Disabled codecs: Enabled codecs:
Advanced	BroadVoice-32 BroadVoice-32 FEC DVI4 DVI4 Wideband G711 uLaw GSM iLBC L16 PCM Wideband Speex Speex FEC Speex Wideband Speex Wideband FEC
8	Codec Properties
Quality of Service	Description:
Diagnostics	Bitrate range (bps): 0 - 0 Fidelity:
	0.0 4.5
	Apply Revert OK Cancel

On the right you can see active (enabled) codecs, on the left – any other codecs that are available.

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 codesc, because video calls are not supported.



Outgoing calls

To make an outgoing call dial a number in international (E.164) 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, 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 project Tel/Fax: **+7 (495) 961-00-08** E-mail: <u>support@comtube.com</u> www: <u>www.comtube.com</u>