LINPHONE ’S USER MANUAL

1. Introduction

Linphone is a simple web-phone. It allows you to make two party-calls using an IP network like the internet. What you need to run Linphone is :

- Linux (maybe it works also on BSD or other UNIX, but nobody test it under other envirronnements than linux.)
- Gnome 1.2 or more, installed but not necessary running.
- a sound card correctly configured.
- headphones or speakers.
- a microphone
- a connection to a network (the Internet for example), using a modem, an ethernet card or anything.

You should close any application that is using the audio device before running linphone.

Linphone is free, it is released under *GNU Public License*.

WARNING: This software is provided with NO WARRANTY see file COPYING for details. This means you MUST NOT use linphone for confidential conversations: there is NO encryption, so it is easy for any bad-intentionned person to catch audio streams. Note also that it is not recommended to run Linphone as root.

2. Running linphone

Linphone can be run as three different ways:

- as normal application: in the gnome menu, linphone should appear in network sub-menu. If you are not running gnome, you can execute linphone by typing linphone in a terminal, for example. When linphone is not running, you cannot receive calls.
- as a gnome applet: by clicking on the gnome panel with the right button, add the applet. Linphone appears in the network menu. By running silently in the panel, linphone is able to receive calls even if its window is not shown. If you want the main window to appear, click on the applet. When a call arrives, the main window is shown and you will hear the ring normally.
- as a silent daemon: this is for non-gnome users. For example in kde, you have a /home/user/.kde2/AutoStart/ directory where you can add links to the application you want to be started when logging. So go to the above directory with the file manager, and do 'add link' by right clicking. The command to run is "linphone –daemon". You will not show linphone's interface, but to make it appear, just runs linphone normally, for example in a terminal.

3. Making a call

3.1. Basic principles

Linphone uses the Session Initiation Protocol to establish the connection with a remote host. In this protocol each caller or callee is identified by a SIP url: sip:user_name@host_name. A sip url is very closed in syntax to an email address, excepted the “sip:“ prefix.

User_name is like a login name on an Unix machine, and host_name is the name or the IP address of the machine the user can be joined.

Note that Sip is a new telecommunication protocol designed to be simple, and it is not compatible with H323 at all.

3.2. Application: two users connected to the internet by a modem

Here is a "simple" way to proceed. The network requirements are :

- a 28k or more modem.

Scenario:

- Bob has to call Tom at 21:00. At 21:00 Tom connects to the internet using kppp, or gppp, or wvdial, (or other).
- Once it is connected to the internet, he can run linphone.By opening the statistics box of kppp or gppp, he can see its IP address (if not type /sbin/ifconfig to see it). The name of the modem interface should be ppp0.
- Then he sends an e-mail to Bob where it tells "My IP address is xxx.xxx.xxx.xxx.".
- Bob receives the e-mail, and then types in linphone window the name and ip address of the person he wants to contact: Tom @ xxx.xxx.xxx.xxx(Tom's IP address), and finaly press the call button.
- Then linphone rings at Tom's house while Bob can hear the ringback sound that informs him that linphone is ringing remotely. Then Tom has just to answer the call by clicking on the answer button, and then should talk each other.

If you encounter problems, see section 4-problems.

3.3. test trial: you have no friends to call at the moment (because it is too late for example), but would like to know if linphone really works

Since version 0.3.0, linphone comes with a test program called '*sipomatic*'. Sipomatic can answer automatically to calls from linphone. To do this:

- run sipomatic from a terminal. Dont’t be surprised, sipomatic does not have a graphical interface, but you don’t have to interact with it.
- From linphone, go to the property box, section network, and choose "lo" as the default interface. Apply changes by clicking OK.
- Then type the following sip url in the main window: sip:robot@127.0.0.1:5064 . 127.0.0.1 is a local address for your computer, and robot is the name to use for calling sipomatic. 5064 is the port where sipomatic can be joined. Normally you should always use 5060 (i.e the default port when no port is specified) to call somebody, but sipomatic is the exception: it runs on port 5064. The reason for this is that linphone already runs on 5060, and you cannot have two applications running on the same port, in the same time and on the same machine.
- Then press the call button. After one second, sipomatic should answer to your call and you should here a short annoncement.

4. Call parameters

4.1. Network

- List of network interfaces: you must choose a network interface to use with linphone. If you want to contact somebody on the internet, you should choose the network interface that connects your computer to the internet. For example, if you are using a 56k modem, it should be interface ’ppp0’. If you are not connected to any network, only the local interface called "lo" will appear in the list. The only thing you can do in this case is to call sipomatic.
- Connection type: select here the way you are connected to the network you want to use (in most case the internet). This help linphone in configuring itself according to the bandwidth of your connection type.

4.2. RTP

RTP (Real Time Protocol) is a protocol used to send media streams over networks.

- RTP port: linphone uses default port 7000 to send and receive audio streams. If you think port 7000 is used by another application, change it as you want.
- Jitter compensation: This number represents the number of audio packets linphone is waiting for before starting to play them. If sometimes some audio packets are late, they have more chance for being played. Increase this parameter if you hear 'cutted voice' to improve the quality of the transmission, but it will increase the delay (you will hear the voice of the remote user a few second later). On the other hand, if you are using a perfect network, and if you have good audio drivers, you can set this parameters down to three packets, and so you will have a short delay.

4.3. SIP

SIP (Session Initiation Protocol) is a protocol to establish media sessions over a network. In simpler words, this is the thing that makes the ring at the remote user, starts the call and terminates it when one of the two parties hangs up.

- SIP port: linphone uses default port 5060 to send and receive SIP packets. This is higly recommended by SIP 's rfc to use port 5060. So don't change this unless you really know what you are doing.
- Use registrar: toggle this button if you want to suscribe services to a remote sip server. Those services can be: redirection, proxy or outbound proxy. See section "Registering on a remote server" for details about this.

4.4. Codecs

Codecs are algorithms especially designed to compress voice data. For example, digitalised voice in 16bit / 8000 Hz represents a data flow of 128 kbits/second. Using the GSM vocoder, this flow is reduced to 13 kbits/second, without significant loss in quality.

- Codec choice: linphone can use several codecs. Use buttons on the bottom of the codec list to put them in an order of preference. Note, that according to your network connection type (given in the network section), some codecs are not usable. They appear in red and they are not selectable. You can decide to use or not a usable codec (in blue) by changing its status with the enable/disable buttons on the bottom of the list.

4.5. Audio parameters

In this section you will find parameters related to your sound equipment.

- Drivers choice: in linux, you have two different kinds of soundcard drivers: OSS (called also kernel drivers) and ALSA. A program can use ALSA driver the same way as OSS ones, but ALSA drivers have better performance when used passing trough the ALSA library. So if you have alsa drivers (names begin by snd_*), select the ALSA mode. If you don't know, choose OSS.
- Source choice: in this combo box you can choose the recording source for your voice. In most case it will be the microphone (mic).
- Auto-kill option: by toggling this option, linphone will try to stop sound daemons (esd and artsd), that may lock permanently your audio device, and so cause linphone to fail in open the audio device when it needs it. It is recommended to have this option on.

5. Address book

You can store and recall sip address using the address book (in the connection menu).

A sip address is in the form: sip:user@domain_name. You can also add a display name, that is used only for your information. To recall a sip address, select it in the address list, and click on the "select" button. Then you will see the address you have selected in the main window. Just press the call button to call the person.

6. Registering on a SIP server

You can suscribe for services on remote SIP servers. These services can be:

- redirection: linphone will ask the server to create a sip account <sip:your_user_name@the_sip_server>. This account can be used by your friends in order to call you. For example, you are a simple internet user with a 56k modem, and your IP address cannot be static, and you don't have a host name known by name servers. It is impossible for your friends to contact you, unless you send them an email to inform them of your IP address. What you can do easily is to register on the a sip registrar <sip:example_registrar.com> for example. Your username is "bob". When registering to the server, linphone sends your ip address to the server and an account "sip:bob@example_registrar.com" is created on the server. Now your friends Jim can call you using the address <sip:bob@example_registrar.com>. Of course "example_registrar.com" is supposed to be a well-known domain name. Then Jim's linphone will receive a redirection message to tell the exact location of Bob. And so linphone will ring at Bob's house.
- proxy: this is exactly the same principle, excepted that the server will not send a redirection message to jim's phone. Instead of that it will redirect directly the call to Bob's linphone.

The registrar you've registered on can also be used as an outbound proxy. This is usefull when using linphone behind a firewall. The outbound proxy takes the same role as a http proxy in private subnets. All outcoming calls from linphone will be addressed to the proxy, that will be in charge to redirect the call to person which is supposed to be in the external network.

To use a registrar server with linphone, go to the property box, section SIP, and then toggle the button “use registrar”. Give the address of the registrar in the appropriate field, and then choose the type of services you request from this registrar: redirect or proxy, by toggling the corresponding buttons. You can also toggle the “Use registrar as an outbound proxy” if you need the registrar server to proxy your calls to an external network.

Finnaly click the OK button of the property box. Linphone will close it and immediately talk to the registrar server to inform it of your exact location. When linphone shutdowns, it will take a few seconds to unregister your location from the registrar.

A list of public registrar servers can be found at http://www.cs.columbia.edu/~hgs/sip/servers.html.

Unfortunately, many of these servers don’t work anymore, maybe caused by the recent crisis in telecommunications. Some other require authentification methods that are currently not supported by linphone.

In order for you not to lost your time, a list of working public sip servers usable with linphone is availlable on the old linphone ’s web site at http://simon.morlat.free.fr/english/servers.html.

7. Behind a firewall

Linphone is now able to work behind firewalls using a SIP proxy server. A SIP proxy server is responsible to forward calls from the private network to the external network, and vice versa. A sip proxy based on oSIP library is being developed at http://osipproxy.sourceforge.net.

You must specify the sip proxy to be used in the property box, section SIP, in the “Use SIP registrar” frame. Give the address of the SIP proxy and toggle the button “Use this registrar as an outbound proxy”.

There is one case where a sip proxy is not needed: when you are in a network where your machine has a public address and the firewall is just here to filter incoming and outgoing packets from the external network. In this case, all you have to do is to open the SIP port and RTP port on the firewall machine. The SIP port is given in the property box, section SIP, and the RTP port is given in the property box, section RTP. Both can be changed, but it es strongly recommended you to let the SIP port unchanged (5060).

8. Problems

8.1. Connection problems

> I try to phone to my friend <sip:toto@example.com>, but nothing happens, no ring, nothing at all.

You must verify that linphone uses the network interface that connects you to the internet (or to the network where calls should go). Use the property box, section Network, to select the correct network interface.

In other case, the person you are contacting may be not reachable at the moment...

8.2. Audio problems

> Linphone seems to connect to the remote sip url, it rings, but when the callee answers, nothing happens and we can't hear each other.

- Most people get problems because they don't choose the correct network interface in the property box, section network. For a dialup connection, it should be "ppp0". Note also that the "lo" interface SHOULD ONLY be used for testing with sipomatic. In other cases, it will fail.
- First rise up playback and recording level.
- If the voice is sometines cutted, you can modify parameter RTP->jitter compensation in the property box to greater values to avoid this. But it increases the

delay transmission.

- If linphone cannot open the audio device, check if it has the permission to open /dev/dsp, close all programs able to use audio device (xmms, kaiman...).
- Use alsa drivers (see http://www.alsa-project.org). Most distributions still use the old oss kernel-official drivers, that have big latency problems and are often buggy. ALSA drivers are much better. Note that you don't need to recompile linphone at all after installing alsa drivers, you even don't have to change for ALSA mode in the prpperty box, section Audio.

9. Bugs reporting and suggestions

First go to linphone's home page at http://www.linphone.org to check if you have the latest version if linphone.

If linphone crashes, send me directly a report at bugs@linphone.org. If linphone does not work, but does not crash, please ensure you have read all this manual before sending me a bug report at the above address. If you want to request something, don't hesitate to send me an email to help@linphone.org. Note that video support, and conferencing are planned features. If someone is interested in doing translations for linphone, send me a xx.po file based on the po/linphone.pot file of the distribution. You can also translate this user manual in other languages. In any case, please contact me if you want more details.

10. Authors

Simon MORLAT (simon.morlat@free.fr) wrotes:

- main program (src)
- RTP library (lprtplib)

- osipua : the user agent api built on top of osip stack.audio interface for oss and alsa drivers.
- (audio)wrappers for lpc10-1.5, gsm and g711 codecs

Aymeric Moizard (jack@atosc.org) wrotes the osip stack that is used by linphone.

The GSM library was written by : Jutta Degener and Carsten Bormann,Technische Universitaet Berlin.

The LPC10-1.5 library was written by: Andy Fingerhut Applied Research Laboratory <– this line is optional if Washington University, Campus Box 1045/Bryan 509 you have limited space One Brookings Drive Saint Louis, MO 63130-4899 jaf@arl.wustl.edu http://www.arl.wustl.edu/<732>jaf/ See text files in gsmlib and lpc10-1.5 directories for further information.

Icons by Pablo Marcelo Moia.

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