

Ekiga Manual 2.00

Damien Sandras

Christopher zanee Warner

Matthias Redlich

This documentation is for version 2.00 of Ekiga.

Copyright © 2003-2006 Damien Sandras

Copyright © 2003-2004 Matthias Redlich

Copyright © 2003-2004 Christopher Warner

Revision History

Revision Ekiga Manual 2.0 2006-01-22
Damien Sandras

User manual for the Ekiga Voice over IP, IP Telephony and Video-Conferencing application

1. Introduction

1.1. Ekiga

Ekiga is a free Voice over IP, IP Telephony and Video-Conferencing application for Linux and other Unices (e.g BSD, OpenSolaris or MacOSX). It was written by Damien Sandras and is licensed under the GNU/GPL.

Ekiga is able to use modern Voice over IP protocols like SIP, and H.323. It supports all major features defined by those protocols like *call hold*, *call transfer*, *call forwarding*, ... It also supports basic *instant messaging*, and has advanced support for *NAT traversal*. Ekiga supports the best *free* audio and video codecs, and has wideband support for a superior audio quality, together with echo cancellation.

1.2. SIP and H.323

The Session Initiation Protocol (SIP) is a protocol developed by the IETF MMUSIC Working Group and

proposed standard for initiating, modifying, and terminating an interactive user session that involves multimedia elements such as video, voice, instant messaging, online games, and virtual reality. In November 2000, SIP was accepted as a 3GPP signaling protocol and permanent element of the IMS architecture. It is one of the leading signalling protocols for Voice over IP.

H.323 was originally created to provide a mechanism for transporting multimedia applications over LANs but it has rapidly evolved to address the growing needs of VoIP networks. One strength of H.323 was the relatively early availability of a set of standards, not only defining the basic call model, but in addition the supplementary services, needed to address business communication expectations. H.323 was the first VoIP standard to adopt the IETF standard RTP to transport audio and video over IP networks. H.323 is based on the ISDN Q.931 protocol and is suited for interworking scenarios between IP and ISDN, respectively between IP and QSIG. A call model, similar to the ISDN call model, eases the introduction of IP Telephony into existing networks of ISDN based PBX systems.



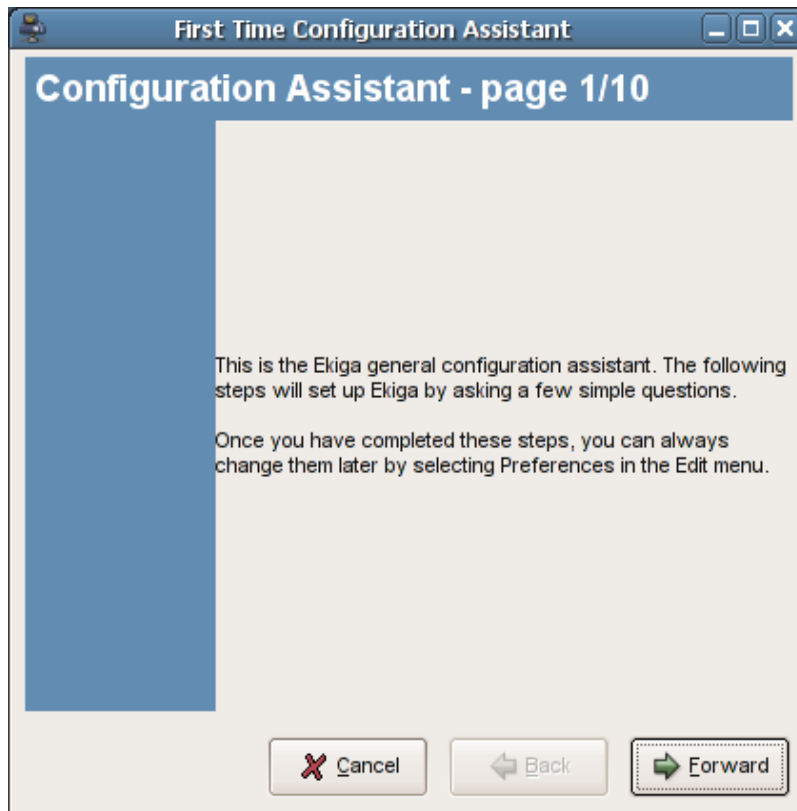
2. Getting Started

When starting Ekiga for the first time the configuration assistant will show automatically. The Configuration Assistant is a step-by-step questionnaire that will guide you through all the steps involved in creating the basic configuration you will need to operate Ekiga. You should go through all of these steps properly, otherwise the assistant will re-appear (when it has not been completed) or Ekiga will not function appropriately (if some of your answers have not been correct). You may run the Configuration Assistant at any time from the Edit menu.

Tip: All settings can be changed via the preferences window at anytime.

2.1. Configuration Assistant Introduction

Figure 1.



Throughout the entire configuration process navigation is available at the bottom of the window. You will be able to navigate through the questions using Back, Forward and Cancel. If you hit Cancel during the setup Ekiga will not be affected by your changes and all entered information will be discarded.

This page welcomes you to the Configuration Assistant. There is nothing to change or edit here. Press the 'Forward' button towards the bottom of the window to start the configuration.

2.2. Personal information

Figure 2.

The screenshot shows a window titled "First Time Configuration Assistant" with a standard Linux-style title bar (minimize, maximize, close buttons). The main content area has a blue header bar that reads "Personal Information - page 2/10". Below this, the text "Please enter your first name and your surname:" is followed by a text input field containing "Damien Sandras". A note below the field states: "Your first name and surname will be used when connecting to other VoIP and videoconferencing software." At the bottom of the window, there are three buttons: "Cancel" (with a red X icon), "Back" (with a green left arrow icon), and "Forward" (with a green right arrow icon).

The Personal Information window requires you to supply personal information to use Ekiga. This information is displayed when connecting to other audio/video applications.

2.3. ekiga.net Account

Figure 3.

The screenshot shows a window titled "First Time Configuration Assistant" with a sub-header "ekiga.net Account - page 3/10". The main content area contains the following text and controls:

Please enter your username:

Please enter your password:

The username and password are used to login to your existing account at the ekiga.net free SIP service. If you do not have an ekiga.net SIP address yet, you may first create an account below. This will provide a SIP address that allows people to call you.

You may skip this step if you use an alternative SIP service, or if you would prefer to specify the login details later.

[Get an ekiga.net SIP account](#)

☐ I do not want to sign up for the ekiga.net free service

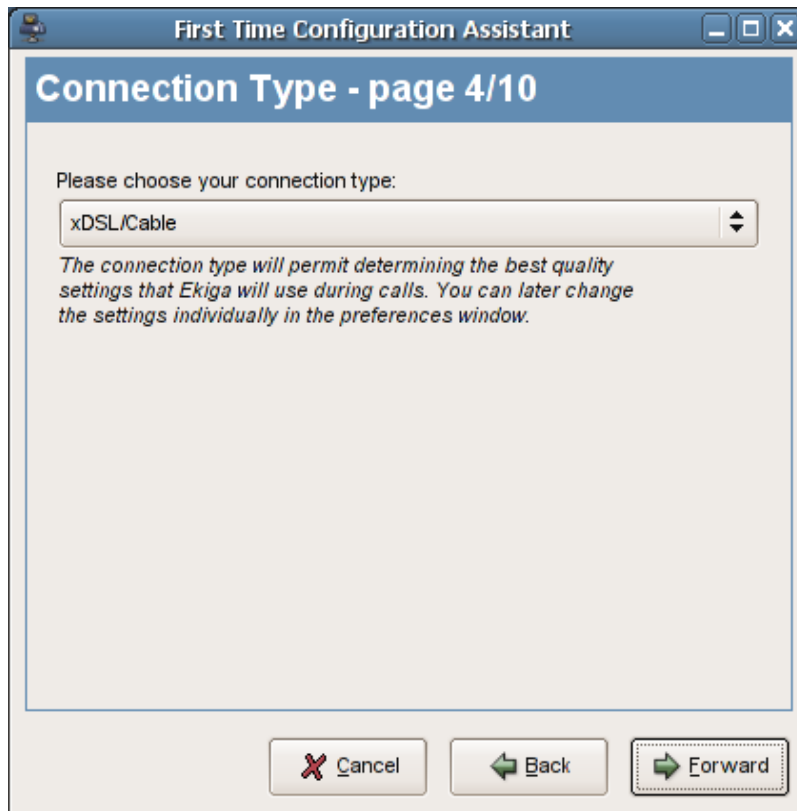
At the bottom, there are three buttons: "Cancel" (with a red X icon), "Back" (with a left arrow icon), and "Forward" (with a right arrow icon).

ekiga.net is a free SIP services platform provided to Ekiga users. If you want to call other users and to be callable, you need a SIP address. You can get one from <http://www.ekiga.net>. ekiga.net also offers additional services like conference rooms, voice mail or online white pages. Please see <http://www.ekiga.net> for more information.

Just follow the link given in the dialog to get an account if you do not have one, then fill in your username and password. Please press 'Forward' after having entered all required information to continue.

2.4. Connection Type

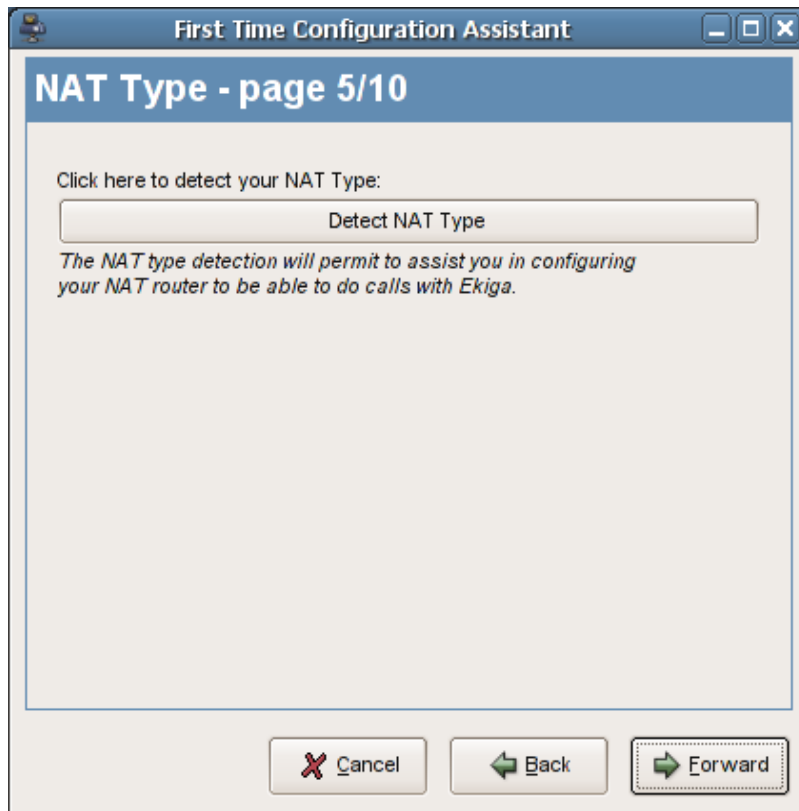
Figure 4.



Ekiga supports several audio and video codecs. It includes codecs with excellent quality as well as codecs with medium to good quality. The higher the quality of a codec, the more bandwidth it requires. Moreover, video codecs can adapt their quality to the available bandwidth. This option is necessary in the initial configuration of Ekiga so that it chooses the optimal codec suited to your network connection and so that it adjusts the video quality settings. If your connection type is not mentioned in the list you should select the one closest to your network connection and adjust Ekiga manually with the preferences window (codecs section) later on.

2.5. NAT Type

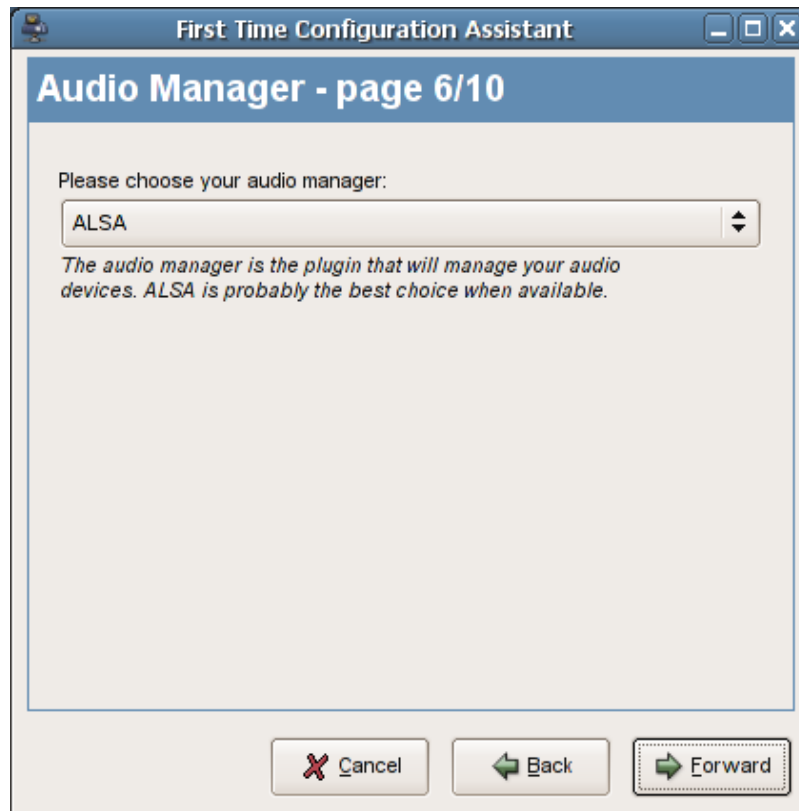
Figure 5.



Ekiga has extended support for NAT. The NAT Type detection page will allow you to detect which type of NAT you are using (if any) and help configuring Ekiga appropriately. Clicking on the detection button will bring a popup indicating which type of NAT was detected and automatically configure Ekiga to transparently cross your router. In most of the cases, it will be totally transparent. Please refer to the Ekiga FAQ (<http://www.ekiga.org>) for more information.

2.6. Audio Manager

Figure 6.



The Audio manager manages everything audio. It is dependant on the operating system on which Ekiga is running, and some operating systems offer different alternatives.

2.7. Audio Devices

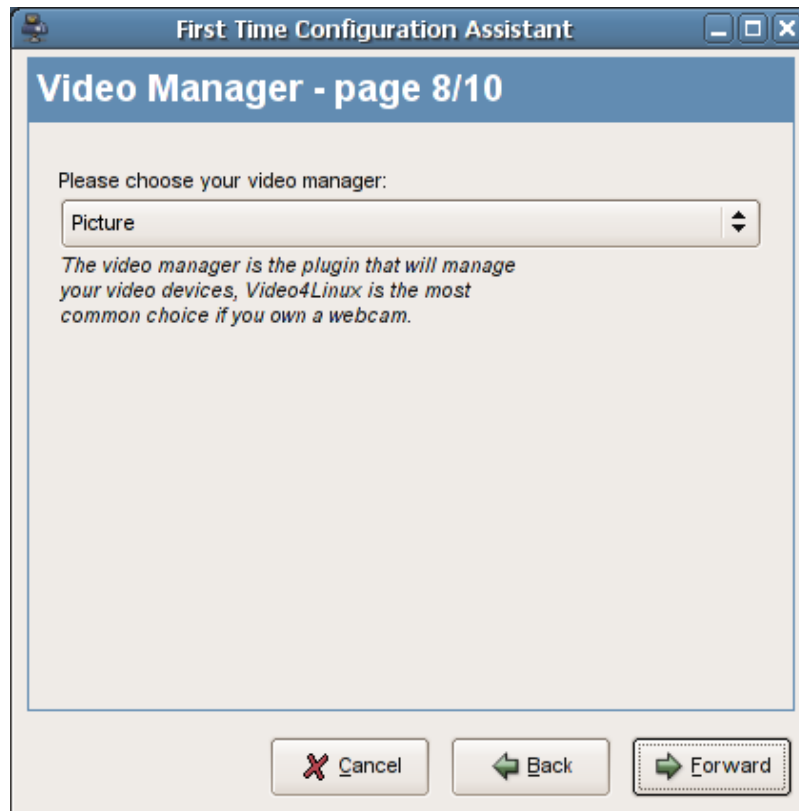
Figure 7.



Ekiga requires audio devices to play and record sound. The audio output device outputs the incoming sound stream during a call. Please select the device that your headset or speakers are connected to. The audio input device is where your microphone is connected to. These settings might be the same as the settings for the audio player if you have only one soundcard. But please note that it is also possible to record sound via another device (e.g. internal microphone in a webcam) too. It is generally recommended that you test your settings after having selected all the appropriate devices. Please press the 'Test Settings' button on the right. If this test was successful you can continue on to the next page in the Configuration Assistant. Otherwise you should change your devices and test your configuration again until you have a setup that works for you.

2.8. Video Manager

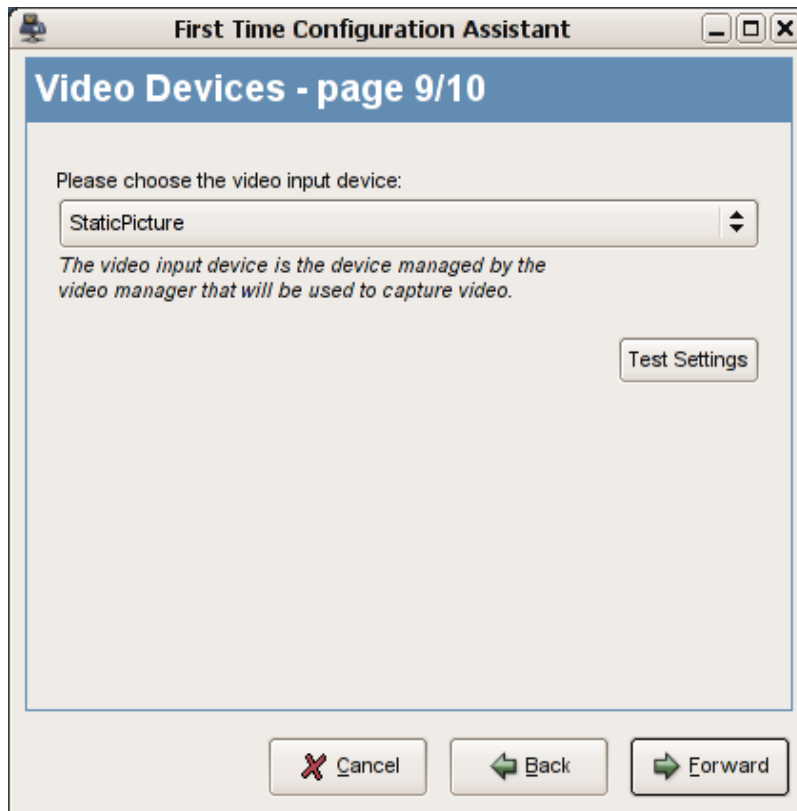
Figure 8.



Please select the Video Manager from the list. It can be Video4Linux to manage webcams, or AVC / DC for Firewire cameras, or any other choice depending on the operating system on which Ekiga is running.

2.9. Video Devices

Figure 9.



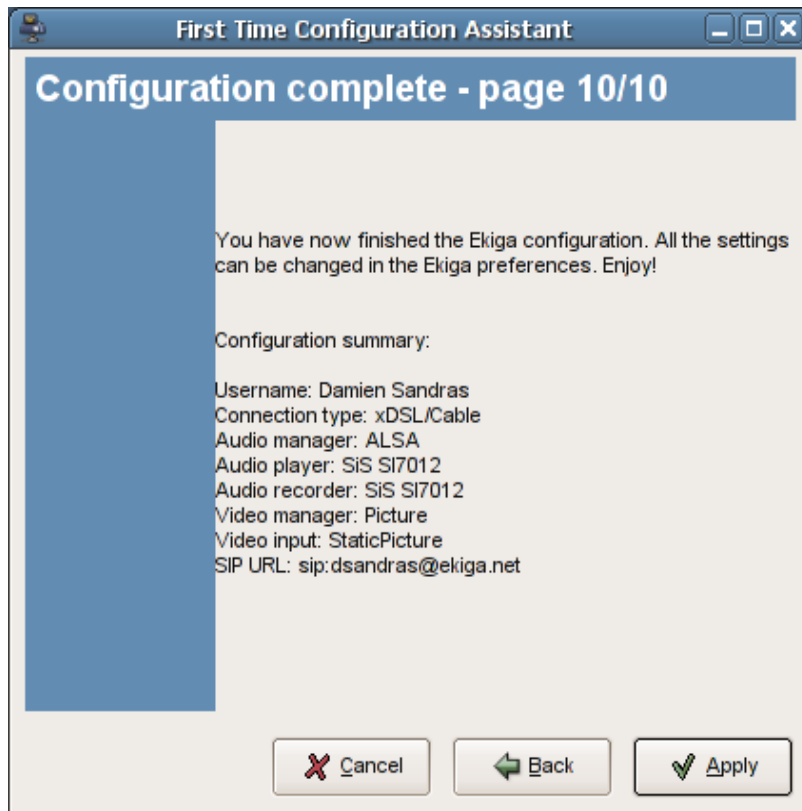
This step is optional and concerns users with video devices (e.g. webcams) only. If you do not have any video devices you may skip this page.

If you have a webcam or video device in the list you may select it here.

Please hit the "Test Settings" button to ensure that your device works with Ekiga, if so, continue on with the Configuration.

2.10. Configuration Complete

Figure 10.



The configuration of Ekiga is now completed. The last window only shows a short configuration summary of the settings you have chosen. Please verify that all these settings are correct. If something is incorrect you may use the 'Back' button in the lower right hand corner of the window to move to any page of the assistant and correct the mistake.

If everything is correct please press the 'Apply' button to save the configuration. The assistant will be closed and the main Window of Ekiga will now appear. Remember, all settings can be changed via the preferences window at anytime.

3. Basic Usage

3.1. Calling and being called



3.1.1. From computer to computer (PC-To-PC)

If you want to call other users and to be callable, you need a SIP address. You can get a SIP address from <http://www.ekiga.net> as described above.

The SIP address can be used by other users to call you. Similarly, you can use the SIP address of your friends and family to call them. You can for example use *sip:dsandras@ekiga.net* to call the author of Ekiga.

You can use the online address book of Ekiga to find the SIP addresses of other Ekiga users. It is of course possible to call users who are using another provider than ekiga.net. You can actually call any user using SIP software or hardware, and registered to any public SIP provider

If you know the URL address of the party that you wish to call, you may enter that URL into the sip: input box at the top of the screen and press the Connect button; eg: *sip:foo@ekiga.net* and pressing the Connect button would call the user at that address. With the default setup, you can simply type *sip:foo* to call user *foo@ekiga.net*.

Tip: Ekiga also supports H.323 and as such can call any H.323 software or hardware. Please refer to the section related to URLs to learn more about the various types of URLs that can be used to call remote H.323 and SIP users.

3.1.2. From computer to real phones (PC-To-Phone)

Ekiga can be used with several Internet Telephony Service Providers. Those providers will allow calling real phones from your computer using Ekiga at interesting rates. We are recommending you to use the default Ekiga provider.

If you want to create an account and use it to call your friends and family using regular phones at interesting rates, simply go in the Tools menu, and select the "PC-To-Phone Account" menu item. A dialog will appear allowing you to create an account using the "Get an Ekiga PC-to-Phone account". Once the account has been created, you will receive a login and a password by e-mail. Simply enter them in the dialog, enable "Use PC-To-Phone service", and you are ready to call regular phones using Ekiga

With the default setup, you can simply use sip:003210444555 to call the real phone number 003210444555, 00 is the international dialing code, 32 is the country code, 10444555 is the number to call.

3.1.3. From real phone to computer (Phone-To-PC)

Ekiga can be used to receive incoming calls from regular phones. To allow this, you can simply login to your PC-To-Phone account using the Tools menu as described above, and buy a phone number in the country of your choice. Ekiga will ring when people will call that phone number.

Tip: You can actually use any H.323 or SIP ITSP provider, including your own PBX at work. However we recommend using the integrated provider.

3.2. Sending instant messages



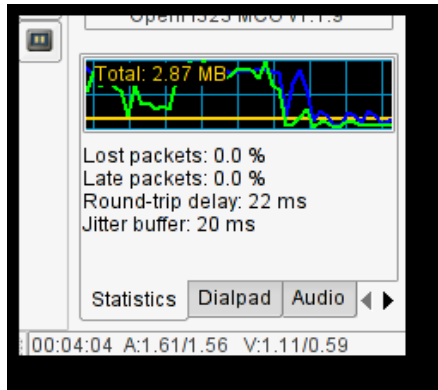
Ekiga allows you to send instant messages to remote users provided that you know their URL. You can by opening the chat window by selecting Tools -> Chat Window. To send a text message to an user, simply enter his SIP address in the URL field, enter your text message, and click on Send. You can later decide to call that user by clicking on Call User.

You can also use the white pages described later to send instant messages to online users. To do this, simply highlight an user, and select Contact -> Send Message. The chat window will appear and allow you to do a conversation with the selected remote user.

Tip: You can also exchanges text messages with H.323 Ekiga users, but only while being in a call. To do this, simply click on the new tab icon, and a new tab will automatically be created allowing a conversation with the user you are in a call with.

3.3. Managing Calls

3.3.1. Understanding the statistics



To view the statistics, please select the Statistics tab in the control panel.

The statistic visualizes the network traffic caused by Ekiga. It draws a graph for each RTP stream. This means that - if audio and video are enabled in Ekiga and the client of the remote party - you will see four different graphs. (incoming audio stream, incoming video stream, outgoing audio stream, outgoing video stream)

- **Lost packets:** The percentage of lost packets, ie of packets from the remote user that you did not receive. A too high packets loss during the reception can result in voice and/or video distortion and is usually caused by a bad network provider or by settings requiring much bandwidth.
- **Late packets:** The percentage of late packets, ie of packets from the remote user that you received but too late to be taken into account, Ekiga being sending and receiving real-time video and audio.
- **Round-trip delay:** The required time for a packet to arrive at its destination and come back. You can see the Round-Trip delay during a call as a connection quality indicator together with the Lost and Late packets statistics.
- **Jitter buffer:** The Jitter buffer is the buffer where received sound packets are accumulated. When the buffer is full, then the sound is played. If your network is of bad quality, then you need a big jitter buffer, ie a big delay before sound is played back, because you need more time before being able to play audio back.

3.3.2. Adjusting the audio and video settings

Your audio and video settings can be adjusted through the control panel while you are in a call. If you want to change the audio input or output devices during a call, simply select the Audio tab in the panel. The brightness, whiteness, color and contrast of your video input device are changed via the Video tab.

3.3.3. Controlling the call

Ekiga supports several actions which can be performed when in a call. These actions enable you to control active sessions.

- Ending a call: The communication to the remote user can be ended by selecting Call->Disconnect.
- Holding a call: You can hold a remote party call by selecting Call->Hold. This effectively pauses Video and Audio transmission, to continue transmission again you select Call->Retrieve Call and Video and Audio Transmission will begin again.
- Mute Audio: This effectively prevents all Audio communication to your respective party.
- Suspend Video: This effectively prevents all Video transmission to your respective party.
- Transferring the remote party: You can transfer the remote user to another H.323 or CALLTO URL by using the appropriate menu entry in the Call menu or by double-clicking on an user in your address book, or in the calls history.

Tip: All URLs supported by Ekiga (SIP, H.323, CALLTO and Speed Dials) can be used for call transfer.

3.3.4. Taking a snapshot

While in a call you can take a snapshot of the remote party via Call -> Save Current Picture. A PNG-file will be saved in the current directory. The filename consists of three parts: the save_prefix, date and current time. (e.g. Ekiga-snap-2003_06_19-024316.png).

3.3.5. Watching calls execution using the history windows

History windows in Ekiga are comparable to logfiles. They keep chronological track of actions performed by Ekiga and provide additional information to the user.

3.3.6. General History

The General History window keeps track of many operations which are mainly performed in the background. It displays information about audio and video devices, calls, codecs and other details. The latest operations can be found at the bottom, older entries are shown on the top. You can access this information by opening Tools->Generic History.

3.3.7. Calls History

The Calls History window stores information (date, duration, URL, Software, Remote user) about all outgoing and incoming calls. They are divided into three groups - Received calls, Placed calls and Unanswered calls.

- Received calls contains all incoming calls which were accepted by Ekiga
- Placed calls keeps track of all attempts - succesful or not - to call another user.
- Unanswered calls shows incoming calls which timed out or were rejected (if Do Not Disturb is enabled, for instance) by Ekiga.

Tip: Double-clicking on a row in the Calls History will call back the selected user or transfer any active call to that user. Notice that you can also drag and drop entries from the Calls History into the Address Book to store contact information.

This information can be accessed by opening Tools->Calls History and by switching between the three tabs.

3.4. Managing Contacts

3.4.1. Managing my contacts with the Address Book

The Address Book is a feature which allows you to find users to call and/or to save locally your list of persons that you call on a regular basis. It respectively loads the list of users from the LDAP directory and will store locally their addresses and associated speed dials (if any).

3.4.2. Basics of the Address Book

To open the Address Book, select Tools -> Address Book and the Ekiga Addressbook window should appear. To your left there will be a list dialog showing the Servers you have added to the list as well as a list of local Address Books. The defaults are the Ekiga white pages, the contacts near you, and the personal address book from Novell Evolution (<http://www.novell.com/products/evolution>).

Ekiga is able to use several types of address books, allowing to search for remote contacts, and bookmark local contacts. The most common address book type is the LDAP directory where you can find information about registered users. Ekiga is able to browse any LDAP directory and use a specific attribute as calling URL. For example, you could have an LDAP directory in your company, with a specific attribute containing the local extensions of all your colleagues. Ekiga is able to use such an LDAP directory. Simply select in File -> New Address Book, and choose remote LDAP as type.

Ekiga is also able to detect other Ekiga users on the LAN using the Bonjour technology popularized by Apple (tm). That supposes you have a local mDNSResponder daemon running on your computer. Finally, Ekiga is able to bookmark contacts in the local address book, shared with the Novell Evolution (<http://www.novell.com/products/evolution>) suite.

To refresh the list of users for a specific address book, simply click the Find button. It will search for all users in that address book. You can contact people by double clicking on their highlighted field. You can also Drag-and-Drop to call a specific party by selecting the highlighted field and dragging it into the Main Window.

In certain cases you will want to search specifically for a person name, his or her call URL, or his location in the Ekiga white pages. The address book window allows you to apply filters when searching for contacts.

Tip: The Ekiga white pages will allow you to look for users in your region. It returns a limited number of results corresponding to your search. If the user is associated to a red icon, it means that he is online. If he is associated to a greyed out icon, it means he is offline. You can then add him to your personal address book to call him later.

3.4.3. Managing remote and local contacts

To add an address book, select File -> New Address Book. A dialog will appear. You then select the type of address book you want to add. The type can be Local, or remote LDAP or remote ILS. Enter the server name. Enter the name, the various parameters and select 'OK' and the new address book should now appear in the address books list. If you do not know what parameters to use for a remote LDAP address book, please ask them to your administrator. The address book parameters can be changed at any time by selecting File -> Properties when the address book is highlighted. It can also be deleted by selecting File -> Delete.

To add a contact to one of your local address books, simply select the address book you wish to add the contact and select Contact -> New Contact. The option of adding a New Contact will appear and you may now enter his name and VoIP URL as well as other settings. After complete select 'OK' and now your contact has been added. You can only add contacts to local address books. The contact parameters can be changed at any time by selecting File -> Properties when the contact is highlighted. He can also be deleted by selecting File -> Delete.

You can also add a contact from the white pages (or any other local or remote address book) by selecting the highlighted contact and dragging him to the specific local address book you wish to add him to or by selecting Contact -> Add Contact to Address Book when selecting that contact.

Finally, you can edit the groups your users belong to using the User Properties dialog from the main menu or from the right-click menu, or using drag-and-drop between groups.

3.5. Managing Incoming Calls

3.5.1. Managing incoming calls

Ekiga supports different policies for incoming calls. Per default it displays a popup window which allows you to decide whether you want to refuse or accept the request for an incoming call. Furthermore Ekiga offers three additional behaviors: Busy mode, Free for Chat and Forward.

3.5.1.1. *Busy mode*

If this mode is enabled Ekiga refuses all incoming requests and only allows outgoing calls. You are not able to receive any call and do not notice if another user tries to contact you except when looking at the Calls History.

This mode can be enabled by selecting Call -> Busy in the main window.

3.5.1.2. *Free for Chat mode*

If this behavior is activated Ekiga accepts all incoming calls. It does not display a popup window but tries to establish the connection to the remote party immediately.

This mode can be enabled by selecting Call -> Free For Chat in the main window menu.

3.5.1.3. *Forward*

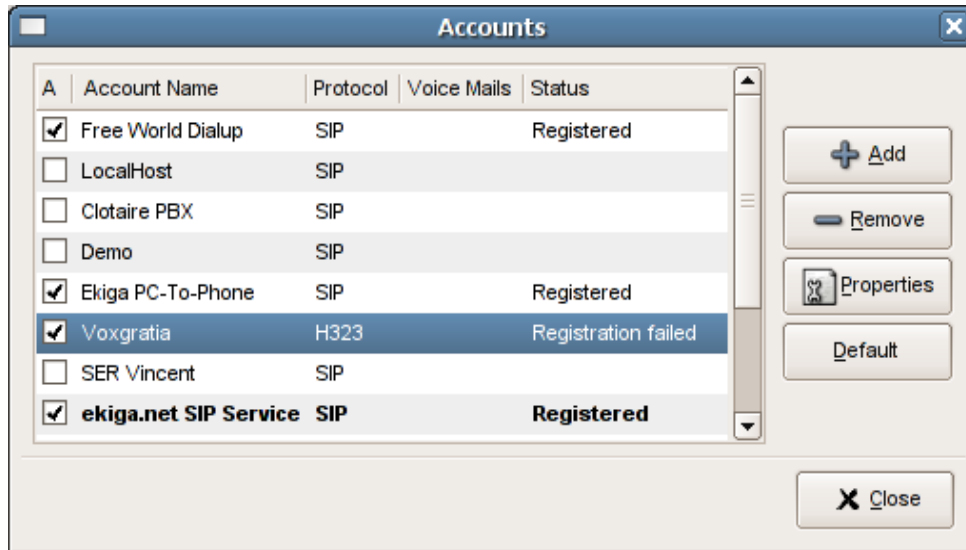
Ekiga has the ability to forward calls to another host. Which allows you to configure Ekiga to forward all incoming calls to a specified URL. Furthermore it is able to forward calls interactively when you do not answer the call after a configurable amount of time or when you are busy.

Call Forwarding can be configured by selecting Call -> Forward in the main menu or through the preferences window. Notice that you need to specify an URL where to forward calls in the preferences to be able to activate tht option. Open the preferences window by choosing Edit -> Preferences in the main window and select Call Forwarding on the left. You will now see the appropriate section. It contains three checkboxes for the three cases described above and one textfield for the IP address/hostname of the host the calls shall be forwarded to.

4. Advanced Usage

4.1. Registering Additional Accounts

4.1.1. The accounts window



You can open the accounts window by selecting Edit -> Accounts. This will open the Accounts Window. The Accounts Window will allow you to add SIP and H.323 accounts and to register to them. An account describes the user login and password parameters to register to SIP and H.323 services. Those *services* can be an Internet Telephony Service provider (like ekiga.net), or an IPBX (like CISCO, Nortel, or Asterisk).

4.1.2. Adding a SIP account

To add a SIP account, simply click on the Add button. A dialog will appear and allow you to enter several parameters:

- *Account Name*: You can enter the account name.
- *Protocol*: You can choose SIP.
- *Registrar*: The registrar to which you want to register. This is usually an IP address or an host name that will be given to you by your Internet Telephony Service Provider, or by your administrator if you are trying to register to a SIP IPBX.
- *User*: You can enter your login.
- *Password*: You can enter your password

Tip: Ekiga will do a best guess concerning the identity that will be used when calling out. Sometimes, you will need to force that identity. You can do this by specifying the identity in the user field. e.g.: dsandras@ekiga.net to force dsandras@ekiga.net to be used as outgoing identity for that account.

You can also control some advanced parameters. Like the Registrar, User and Password, they will be given to you by the ITSP you are using or by your administrator. Those parameters are:

- *Authentication Login*: If it is different from the user parameter you provided above. In that case, the user field will be used to control the outgoing identity for the account you are adding, while the login will be used during the authentication phase.
- *Realm/Domain*: It is globally unique and dependant on the ITSP or the IPBX. It is generally identical to the registrar domain.

- *Registration Timeout*: The timeout after which the registration should be updated.

4.1.3. Adding an H.323 account

The screenshot shows a standard Windows-style dialog box titled "Edit the Account Information". It contains the following fields and controls:

- Account Name:** A text box containing "My gatekeeper".
- Protocol:** A dropdown menu currently set to "H323".
- Gatekeeper:** A text box containing "gk.mycompany.com".
- User:** A text box containing "dsandras".
- Password:** A text box containing masked characters "*****".
- More Options:** A link with a right-pointing triangle icon.
- Buttons:** "Cancel" and "OK" buttons at the bottom right.

To add an H.323 account, simply click on the Add button. A dialog will appear and allow you to enter several parameters:

- *Account Name*: You can enter the account name.
- *Protocol*: You can choose H.323.
- *Gatekeeper*: The gatekeeper to which you want to register. This is usually an IP address or an host name that will be given to you by your Internet Telephony Service Provider, or by your administrator if you are trying to register to an H.323 IPBX.
- *User*: You can enter your login.
- *Password*: You can enter your password

You can also control some advanced parameters. Those parameters are:

- *Gatekeeper ID*: The gatekeeper ID, if any.
- *Registration Timeout*: The timeout after which the registration should be updated.

4.2. Understanding URLs

4.2.1. SIP URL's

SIP URL's are formatted as such "sip:user@[host[:port]]"

This permits you to call the given user or extension on the specified SIP proxy: *sip:jonita@ekiga.net*

4.2.2. H.323 URL's

H.323 URL's are formatted as such "h323:[user@][host[:port]]"

This permits you to:

- Call a given host on a port different from the default port which is 1720: *h323:seconix.com:1740*
- Call a given user using their respective alias if registered to a gatekeeper: *h323:jonita*
- Call a given phone number if you are registered to a gatekeeper for a PC-To-Phone provider, or if that user has an ENUM record associated to an H.323 URL: *h323:003210111222*
- Call a given user using their alias through a specific gateway or proxy:
h323:jonita@gateway.seconix.com
- Call an MCU and join a specific room: *h323:myfriendsroom@mcu.seconix.com*

4.2.3. CALLTO URL's

Callto URL's are formatted as such "callto:[user@][host[:port]]"

Callto URL's and H.323 URL's are formatted exactly the same except however callto urls also support ILS lookups through callto URLs of the type: *callto:ils_server/user_mail*.

For example, calling *callto:ils.seconix.com/joe.user@somedomain.com* will look for the user with the *joe.user@somedomain.com* email address on the ILS server *ils.seconix.com* and proceed to initiate a call.

4.3. Speed dials

Ekiga is able to associate speed dials with URLs using the address book. You can thus for example associate the speed dial *1* to the URL *sip:600000@ekiga.net*. That speed dial can then be used as URL.

For example, calling *sip:1#* will call *sip:600000@ekiga.net* provided that both are associated together in the address book.

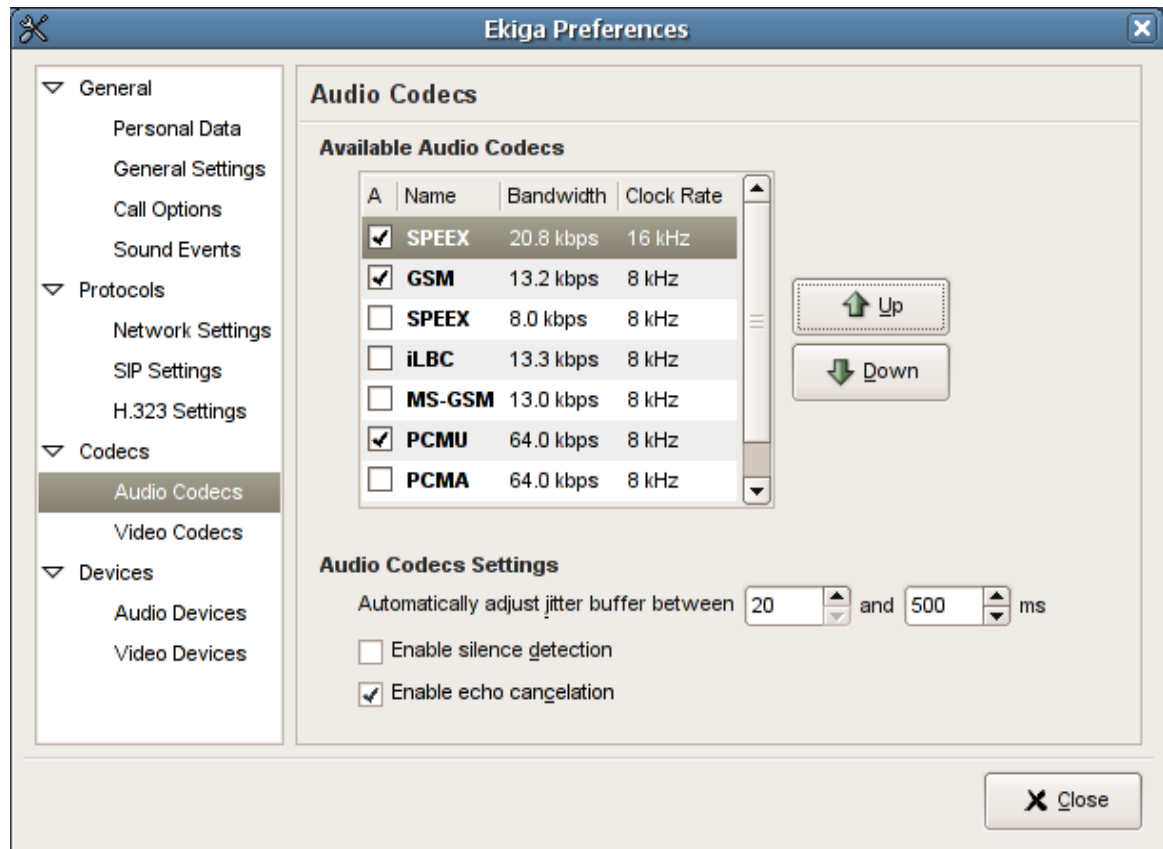
4.4. Controlling the Video Bandwidth

Ekiga is using a best-effort algorithm to maintain a low bandwidth when transmitting video. You can adjust the video quality settings following you prefer to have a good frame rate, or a good picture quality. It will permit Ekiga to dynamically adjust the video bandwidth and the number of transmitted images per second during a call while trying to respect the requested video bandwidth.

Notice that the algorithm is a best-effort algorithm, which means that if you specify too low video bandwidth settings, it can be impossible to respect them. However, if the video bandwidth permits to transmit with a better quality, or faster than the requested values, then Ekiga will dynamically increase them so that the quality and the framerate are always the best possible.

Choosing a higher framerate and a lower quality will have the same result in terms of video bandwidth than choosing a higher quality with a lower framerate. It depends if you prefer using your bandwidth to transmit more lower quality images or fewer big quality images.

4.5. Managing Codecs



4.5.1. Audio Codecs

The Ekiga audio codecs table in the preferences permits you to change the codecs order as well as disabling the codecs you don't want to use. Each codec has strong and weak points. For example, G.711 will give the best voice quality but will use the most bandwidth while SPEEX will give an average voice quality but requiring a very low bandwidth usage. Notice that there are two versions of SPEEX, one of them is SPEEX WideBand. You can see that to the 16 kHz clock rate.

4.5.2. Reordering the codecs

When you reorder the codecs, you are reordering the local capabilities table, ie the codecs you will use for sending. You will always transmit audio using the first codec in the table that is in common with the remote user. The remote user will transmit audio using the first codec in his table that is common with you.

4.5.3. Forcing the use of a specific codec

You can force the use of a specific codec by disabling all other codecs, but it will result in failed calls if the remote user doesn't allow that specific codec. The best is to put your preferred codecs at the top of the list so that you always transmit with them if the remote user allows it and to disable the codecs that you don't want to use for transmission and reception.

4.5.4. Adjusting the jitter buffer

You can adjust the delay to wait before playing the sound buffers that you have received using the jitter buffer adjustment. If there is too much packets loss, the delay required to have received all packets could be so important that it will exceed the jitter buffer. In such a case, the sound you are receiving will be of bad quality. A solution to that problem would be to increase the maximum limit of the jitter buffer to a few seconds, resulting in a big delay but in an improved voice quality. Notice that the jitter buffer will readapt itself to the lowest delay allowing for optimum transmission, and that a bad voice quality in reception is not due to a too low jitter buffer value, but to bad internet connection quality.

4.6. Changing Ports

4.6.1. The listen ports

The main port listening for incoming connections in Ekiga for SIP is port 5060 (UDP), while 1720 (TCP) is used by H.323. To change those ports you need to load "gconf-editor". Open gconf-editor, select apps from the left hand side menu and then select Ekiga. Then select "sip" or "h323", it should give you a list in the corresponding window to your right. Select listen_port and change it to your desired value. You can also change the UDP/RTP port ranges.

4.6.2. Explanation of the port ranges

1. The "listen_port" value is the port Ekiga will listen for incoming connections on. It is different for SIP and H.323.
2. The "rtp_port_range" value is the range of UDP ports that Ekiga will use for RTP (audio and video communication channels). Ekiga needs to be restarted for the new values to take effect.
3. The "udp_port_range" value is the range of UDP ports that Ekiga will use for SIP signalling or when registering to H.323 gatekeepers.
4. The "tcp_port_range" value is the range of TCP ports beside the listen_port that Ekiga will use for the H.245 channel with the H.323 protocol. That port range is not used by SIP. It is not used either when

H.245 Tunneling is enabled, which is in general always the case, except when calling old H.323 implementations like Netmeeting.

4.7. Controlling the SIP Settings

4.7.1. Misc Settings

Outbound Proxy

The outbound proxy is the SIP proxy that will relay your calls. The behavior of a SIP proxy is similar to the behavior of an HTTP proxy, ie some entity that issues the requests on your behalve and proxies the streams.

Forward URL

The URL to which SIP incoming calls should be forwarded if configured in the preferences.

4.7.2. Controlling the H.323 Settings

4.7.2.1. Misc Settings

Default gateway

The default gateway is the H.323 gateway to use when doing calls. For example, if you are calling *h323:123443* with a default gateway set to *foo*, gateway foo will dial 123443 on your behalve. Usually, you will be registered to a gatekeeper, and gateway is not used.

Forward URL

The URL to which H.323 incoming calls should be forwarded if configured in the preferences.

4.7.2.1.1. Advanced Settings

Ekiga permits a fine control of the H.323 settings in the Advanced H.323 Settings section of the preferences. You can enable H.245 Tunneling, Early H.245 and Fast Start.

H.245 Tunneling

H.245 Tunneling is the encapsulation of H.245 messages within H.225/Q.931 messages (H.245 Tunneling). If you have a firewall and enable H.245 Tunneling, there is one less TCP port that you need to allow for incoming connections.

Early H.245

This enables H.245 early in the setup and permits to achieve faster call initiation.

Fast Start

Fast Connect is a new method of call setup that bypasses some usual steps in order to make it faster. In addition to the speed improvement, Fast Connect allows the media channels to be operational before the CONNECT message is sent, which is a requirement for certain billing procedures. It was introduced in H.323 version 2.

5. About Ekiga

Ekiga is written by Damien Sandras (<dsandras@seconix.com>). To find more information about Ekiga, please visit the Ekiga Home Page (<http://www.ekiga.org>).

To report a bug or make a suggestion regarding this application or this manual, follow the directions in this document (ghelp:gnome-feedback).

This program is distributed under the terms of the GNU General Public license as published by the Free Software Foundation; either version 2 of the License, or (at your option) any later version. A copy of this license can be found at this link (ghelp:gpl), or in the file COPYING included with the source code of this program.

6. Appendix

6.1. Related Software

IPBX

- Asterisk PBX: <http://asterisk.org>

SIP

- SIP Express Router: <http://www.iptel.org/ser>

H.323

- OpenH323 Gatekeeper: <http://www.openh323.org>
- GNU Gatekeeper: <http://www.gnugk.org>
- OpenH323 Proxy: <http://openh323.sourceforge.net>
- H323 - ISDN Gateway: <http://www.telos.de/linux/H323/>

Conferencing/VoIP Software

- OpenMCU: <http://www.openh323.org>

Similar Clients

- XTen: <http://www.xten.com>
- XTen: <http://www.xten.com>
- SJPhone: <http://www.sjlabs.com/>
- OpenPhone: <http://www.openh323.org>
- Netmeeting: <http://www.microsoft.com>