## **USER MANUAL**

## Introduction

This program has been designed to be used as a multiplatform videoconference aplication. It uses a session establishment protocol called SIP, and has been programmed for the Java Platform. Therefore it can be run under any operational system which has Java Virtual Machine installed, available at the following web site http://www.java.com. For audio and video to work correctly, Java Media Framework JMF neededs to be installed, it can be found at the web site http://java.sun.com/products/java-media/jmf.

To run the program just run the script umasip.bat, for Windows systems, or umasip.sh for Linux systems, running the graphical user interface in the Figure 1:



Figure 1. User Interface

## **Program Requirements**

For the program to work properly some hardware and software requirements are needed.

The software requirements are:

- Java Virtual Machine 1.4 or higher
- Java Media Framework 2.1.1e or higher

The minimun hardware requirements are:

- Processor, at lest, Pentium 166 MHz, PowerPC 160 MHz, or UltraSparc 166 MHz
- At lest 32 MB of RAM memory
- Sound Board
- Speakers
- Microphone (optional, just in case audio transmission is desired)
- Webcam (optional, just in case video transmission is desired)

## **Option Menu**

The graphical interface shows to the user an option menu, as shown in the Figure 2:



Figure 2. umasip option menu

These options are available through the menu and its functions are the followings in the Table 1:

Opción	Menú	Funcionalidad
New Call	Call	It lets make a new call. A window
New Call		asking for the remote address is shown.
Hang	Call	It finish the current call. It is available only when there is a call.
Quit	Call	Quit the program.
See SIP	Configuration	You can see the SIP protocol
Information		information in the main window.
Transmission Information	Configuration	It opens a window with the configuration about de multimedia preferences.
Autoresponse	Configuration	It lets answer afirmatively and automatically to incoming calls.
Preferences	Configuration	It opens a window that lets configure the preferences.
"Language"	Language	It lets choose the language to use.
Help Themes	Help	It shows the available themes of help. When choosing one, a window with the help is opened.
License	Help	It opens a window with the license information.
About	Help	It opens a window with the information about umasip.

Table 1. Option Menu

The options in the *Language* menu are the name of the different languages, available by the interface, written in their own languages. In the option *Help Themes* there is another submenu from where to choose each available help theme.

## New Call

This option from the Call menu will be used to make a new call. The name of the user to be called and its host will be asked. If the user to be called has not specified a user name, it

will take anonymous by default. To refer the host, apart from its name, it can also be used its IP address.

As an example, to call a user that has not change the default name, with the following IP address 192.168.1.7, the following locator will be used: anonymous@192.168.1.7, as shown in Figure 3.

ର	sip:usuario@l	nost
0	anonymous@192.168.1.7	
	Aceptar	Cancelar

Figure 3. New Call window

A called user will received a notice in which he will be asked if the comunication with the calling user is desired.

## **Incoming Call**

If while the program is in a waiting state a new call is received, a new window shows up noticing this call, and giving you the option of accepting or rejecting it, as shown in Figure 4. If the call is accepted the comunication is established, and if it is rejected a busy tone is sent to the user who is trying to establish the call.

	Aviso de Llamada 🛛 🔤 🗙
2	Llamada procedente de sip:anonymous@192.168.1.7. ¿Descolgar?
	<u>Sí</u> <u>N</u> o

Figure 4. Call Notice

If a new call arrives while there is already a call in course, the program answer automatically sending a busy tone to the user who is trying to establish the new call.

## Hang a Call

While an active call exists, there is a window in which the communication is shown, and in which there is a button where hanging and finishing the communication is available, see Figure 5. There is a possibility in the Call menu to hang with an option that is enabled in the moment a call is established.



Figure 5. Active call window

Anyone of the participants can finish a call, the caller or the receiver.

#### **SIP** Protocol Information

With the option See SIP Information in the Configuration menu, it is possible to see at the main window the debugging information about the SIP Protocol. Like this, an expert user who would like to see how this protocol for establishing sessions works, can see how all sessions in each call has been established, negotiated and finished.

The program functioning with this option active can be seen in Figure 6.

0	umasip	
Llamad	a Configuración Idioma	Ayuda
SIP Protoc	ol Information:	
TIME   Lo	:al User ==MESSAGE (Request/Response)== Remote User	
Client Sip	ready with user: anonymous@192.168.1.200	
Client Sip	ready with user: anonymous@192.168.1.200	
12:55:16	sip:anonymous@192.168.1.200 <<==== Request INVITE <<==== sip:anonymous@19	92.168.1.7
	Receiving AUDIO: Payload Type 5 by port 23322	
	Receiving VIDEO: Payload Type 26 by port 23326	novani seren andera - Er
	sip:anonymous@192.168.1.200 ====>> Response TRYING (100) ====>> sip:anonymous@192.168.1.200 ====>> sip:anonymous@192.1.200 =====>> sip:anonymous@192.1.200 =====>> sip:anonymous@192.1.200 ====>> sip:anonymous@192.1.200 =====>> sip:anonymous@192.1.200 =====>> sip:anonymous@192.1.200 ====>> sip:anonymous@192.1.200 =====>> sip:anonymous@192.1.200 =====>> sip:anonymous@192.1.200 =====>> sip:anonymous@192.1.200 ====>> sip:anonymous@	
	sip:anonymous@192.168.1.200 ====>> Response RINGING (180) ====>> sip:anony	
	sip:anonymous@192.168.1.200 ====>> Response DECLINE (603) ====>> sip:anony	
12:55:24	sip:anonymous@192.168.1.200 <<==== Request ACK <<==== sip:anonymous@192.	.168.1.7;tag=2960
12:57:04	sip:anonymous@192.168.1.200 ====>> Reguest INVITE ====>> anonymous@192.1	68.1.7
	Transmitting AUDIO: Payload Type 0 by port 23322	
	Transmitting VIDEO: Payload Type 34 by port 23326	
	sip:anonymous@192.168.1.200 <<==== Response TRYING (100) <<==== anonymou	
	sip:anonymous@192.168.1.200 <<==== Response RINGING (180) <<==== anonymo	
12:57:05	sip:anonymous@192.168.1.200 <<==== Response OK (200) <<==== anonymous@1	92.168.1.7
	Receiving AUDIO: Payload Type 5 by port 23332	
	Receiving VIDEO: Payload Type 26 by port 23336	
	sip:anonymous@192.168.1.200 ====>> Request ACK ====>> anonymous@192.168	
28 Text222 20 28 28	<pre>  sip:anonymous@192.168.1.200 &lt;&lt;==== Request BYE &lt;&lt;==== anonymous@192.168   sip:anonymous@192.168.1.200 ====&gt;&gt; Response OK (200) ====&gt;&gt; anonymous@1</pre>	
15.01.11	Top. anonymous at s2, too, t.200 ====>> Response on (200) ====>> anonymous at .	52.100.1.7

**Figure 6. SIP Protocol Information** 

#### **Transmission Information**

With the option Transmission Information in the Configuration menu, information about the audio and video formats, that are going to be used or that are being used during a call, can be known. Using this method, you can also know if during a call the information corresponding to audio is been transmitted or not, the format in which it is been transmitted and its characteristics. The same applies with the information about video.

An example of the information that can be shown would be the one in Figure 7:



**Figure 7. Transmission Information** 

#### **Autoresponse of calls**

There is an option in the Configuration menu that makes autoanswering calls available. If this options is activated, while the program is in standby mode, all incoming calls will be answered automatically. Therefore, if accepting any arriving call is desired, this option can be activated and no warning window will be shown asking about accepting or rejecting the call. In this case, the window with the call image will appear directly, and the multimedia information specified at the Preferences will start to be transmited automatically.

## Configuration

For the configuration of the program functioning characteristics will be used the option Preferences in the Configuraton menu. With this option it will be shown a window, like the one in Figure 8, where can be configured the user name, the host in which it is, the audio and video devices to be used, and the corresponding formats to transmit. In the field User Name can be specified the desired name to use. By default, in case that no name has been specified, the program uses the name anonymous.

In the field Host/IP can be specified the host in which the user can be found, that in this version of the program will have to be local host from which it s been running.

In the field Audio Device there are all the available audio devices in the computer, and is possible to choose the desired device for the audio information capture.

In the field Audio Format can be selected the transmission format in which it is desired to code the information.

In the field Video Device can be chosen among the different existing devices for the video information input.

In the field Video Format can be chosen the video format in which to code to transmit.

•	Preferencias	_0×	
Nombre del Usuario:	žanonymous	Por Defecto	
Máquina/IP:		Autodetectar	
Dispositivo de Audio:	JavaSound audio capture	<u></u>	
Formato de Audio:	ULAW/rtp, 8000.0 Hz, 8-bit, Mono, FrameSize=8 bits		
Dispositivo de Vídeo:	v4l:Labtec Webcamil:0		
Formato de Vídeo:	H263/RTP	_	
	Aceptar	Cancelar	

#### **Figure 8. Preferences**

At the bottom of the window, there are two buttons to accept or cancel the changes done. If accepted the program do the pertinent changes, and if rejected the previous preferences are restored.

## **Change of Language**

You do not need to restart the program to change the language, just select one among the available languages in the Language menu. Automatically, and while still running, the program will take the selected language, being able to change the language as many times as desired. In Figure 9 you can see the program functioning in several languages.

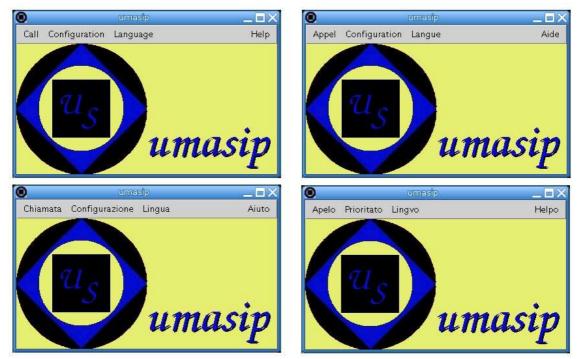


Figure 9. umasip in several languages

## **Help Themes**

In the Help menu you can find this manual, in case of help needed. There is a submenu *Help Themes* where there are different themes from wich to obtain help. When an option is clicked, a window with the information required is shown. For example, if *Introduction* is chosen, the wiondow in Figure 10 is opened:

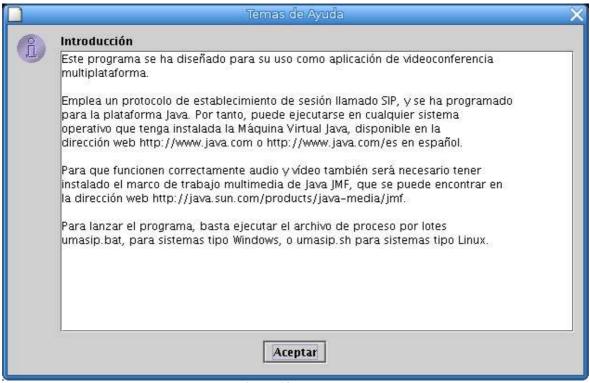
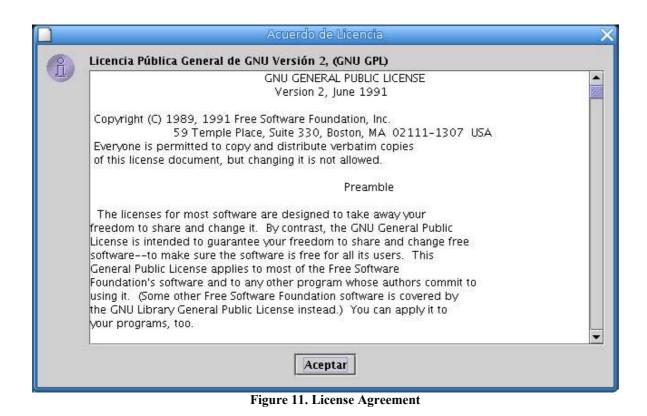


Figure 10. Help Themes

#### License

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The GPL license is a free license, also known as copyleft, and allows to use the software freely, to distribute copies, to change the software and to redistribute the software with changes. Anyone who distributes copies of the software, modified or not, must give the receivers all the same rights, so the GPL license will have to be present and without modifications. For further details, there is a copy of the license that can be seen with the option *License* from the *Help* menu as seen in Figure 11.



## **About umasip**

The information about umasip is shown in Figure 12:

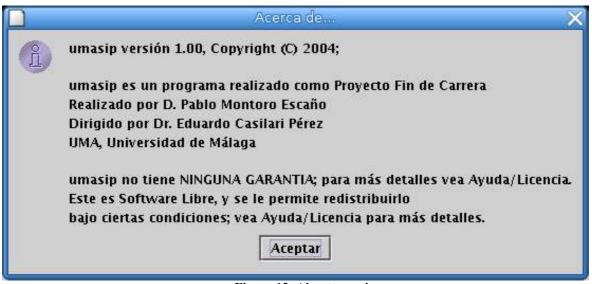


Figure 12. About umasip

This information can be seen through the option About... in the Help menu.

# Quit the program

To quit the program, just close the main window or select the *Quit* option from the *Call* menu.