



AVAZ SIP PHONE

User Manual

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1 Installation

Installing AVAZ SIP Phone on your computer consists of the following steps

1. Install DirectX 8.0
2. Download and Unzip AVAZ SIP Phone Setup ZIP File
3. Run the AVAZ SIP Phone Setup

1.1 Install DirectX 8.1

1. If you are using Windows XP® DirectX 8.1 is already there
2. If you are using Windows 2000®, download the DirectX 8.1 engine from:

<http://www.microsoft.com/windows/directx/downloads/drx81.asp>

1.2 Download and Unzip AVAZ SIP Phone Setup ZIP File

1. Close any applications on your computer.
2. Navigate to the AVAZ SIP Phone Download Page and scroll down to the Downloads section.
3. Click on Setup Program.
4. When prompted select 'Save to Disk'
5. Browse to the Desktop.
6. Save the AVAZ SIP Phone Setup ZIP File on the Desktop.
7. Unzip AVAZ SIP Setup ZIP File in Folder 'AVAZ SIP Setup'

1.3 Run the AVAZ SIP Phone Setup

1. Navigate to the Folder where you unzipped the Setup ZIP File.
2. Click on the 'Setup.exe' – Setup is launched
3. Choose the Destination Folder for AVAZ SIP Phone and click 'Next'
4. Choose Program Folder for AVAZ SIP Phone and Click 'Next'
5. Click 'Finish' to complete Setup.
6. AVAZ SIP Phone is installed.

2 Introduction

The Session Initiation Protocol (SIP) is an application-layer protocol for creating, modifying and terminating sessions with one or more participants. [RFC 2543]

The AVAZ SIP Phone is a SIP User Agent and has been built on AVAZ SIP signaling stack.

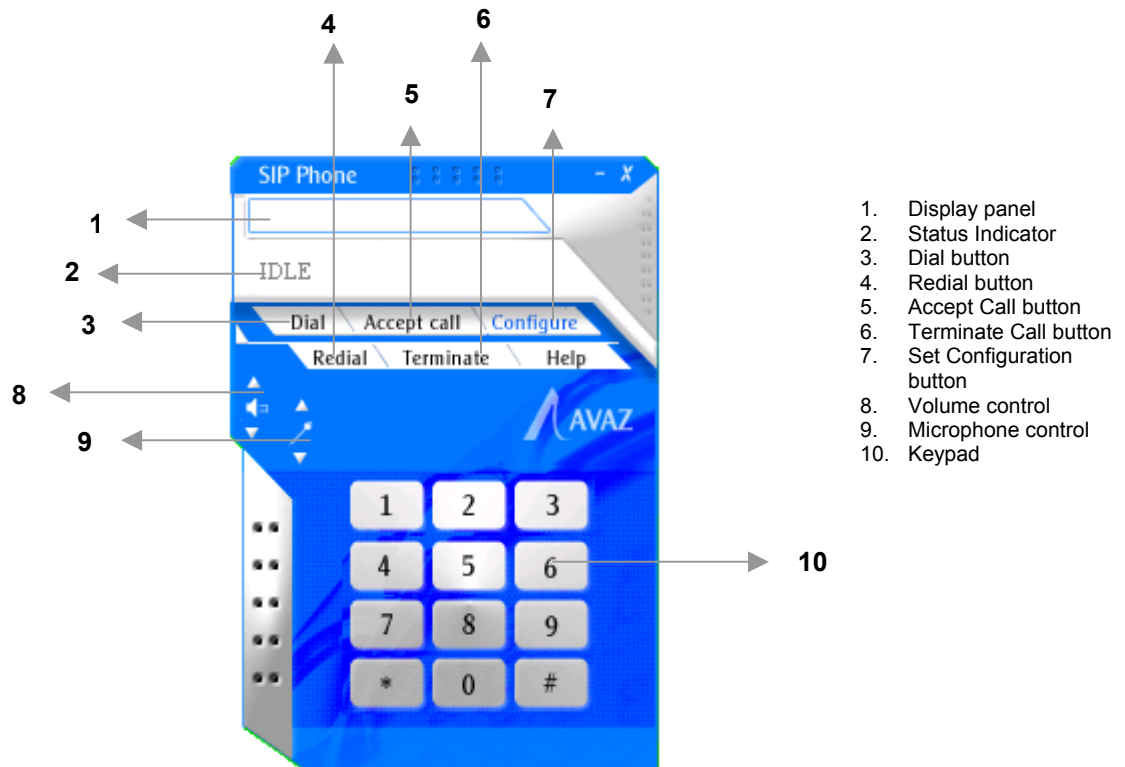
2.1 Features

AVAZ SIP Phone supports following features:

- RTP media
- UDP Transport
- G.711 64kbps audio
- Message Retransmission timers
- INVITE, ACK, BYE, CANCEL supported
- Protocol Message display window
- Protocol Message logging
- Ring-back / Ringing tone
- Status display

2.2 User Interface

Figure shows different controls of AVAZ SIP Phone.



2.2.1 Display panel

Display panel is used to enter callee sip address. Callee address can be entered using either keyboard or keypad.



2.2.2 Status Indicator

Status Indicator shows current status of the phone. For example, when phone is not in use it shows "IDLE" status, similarly, it shows "DIALING", "RINGING" etc.



2.2.3 Dial Button

This button is used for dialing a given address.



2.2.4 Redial Button

This button is used to redial the recent address.



2.2.5 Accept Call Button

User can accept call by pressing this button.



2.2.6 Terminate Call Button

This button is used for terminating ongoing call session. It can also be used to cancel the call negotiation.



2.2.7 Configuration Button

This button is used to configure the phone.



2.2.8 Volume Level Control

This control is used to adjust volume levels.



2.2.9 Microphone Level Control

This control is used to adjust microphone level.




2.2.10 Keypad

Key Pad is used for dialing the desired phone number. Number entered through the pad will appear on display box.



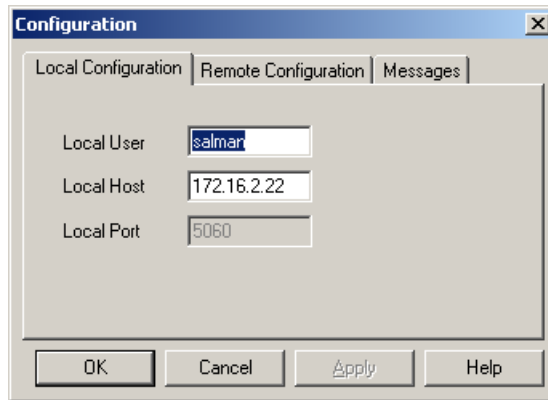
3 Operating AVAZ SIP Phone

3.1 Configuration

Configuration is required before using AVAZ SIP Phone. User clicks on  button a configuration dialog pops up.

3.1.1 Local Configuration

To set the local configuration, user selects the “Local Configuration” tab in the dialog box.



The image shows a 'Configuration' dialog box with three tabs: 'Local Configuration', 'Remote Configuration', and 'Messages'. The 'Local Configuration' tab is selected. It contains three input fields: 'Local User' with the value 'salmar', 'Local Host' with the value '172.16.2.22', and 'Local Port' with the value '5060'. At the bottom are four buttons: 'OK', 'Cancel', 'Apply', and 'Help'.



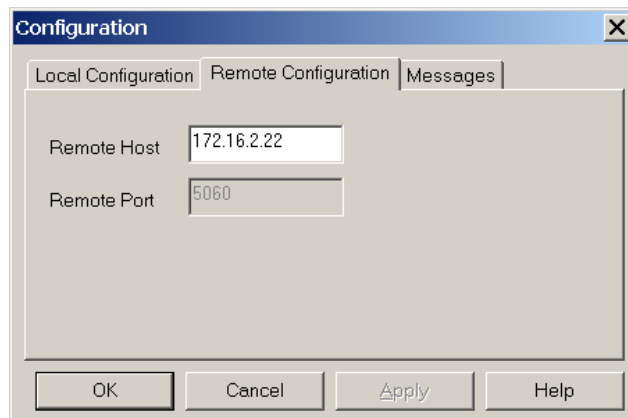
The image shows two input fields. The first is labeled 'Local User' and contains the text 'salmar'. The second is labeled 'Local Host' and contains the text '172.16.2.22'.

Set local user's id. e.g. john, 97372001

Set local IP address.

3.1.2 Remote Configuration

To change the configuration of outbound proxy user selects the “Remote configuration” tab in the configuration dialog box.



The image shows the same 'Configuration' dialog box, but with the 'Remote Configuration' tab selected. It contains two input fields: 'Remote Host' with the value '172.16.2.22' and 'Remote Port' with the value '5060'. The 'Local Configuration' tab is now disabled. The buttons at the bottom remain the same: 'OK', 'Cancel', 'Apply', and 'Help'.

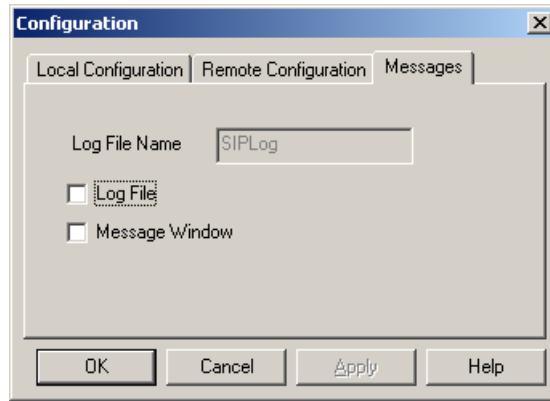


The image shows an input field labeled 'Remote Host' containing the text '172.16.2.22'.

Set the IP address of outbound SIP proxy or other User Agent.

3.1.3 Message logging

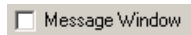
To set the message-logging configuration, user selects “Log file Info” tab from the configuration dialog box.



This field is used to enter the name of log file in which to store log information. User enters the log file name in box.

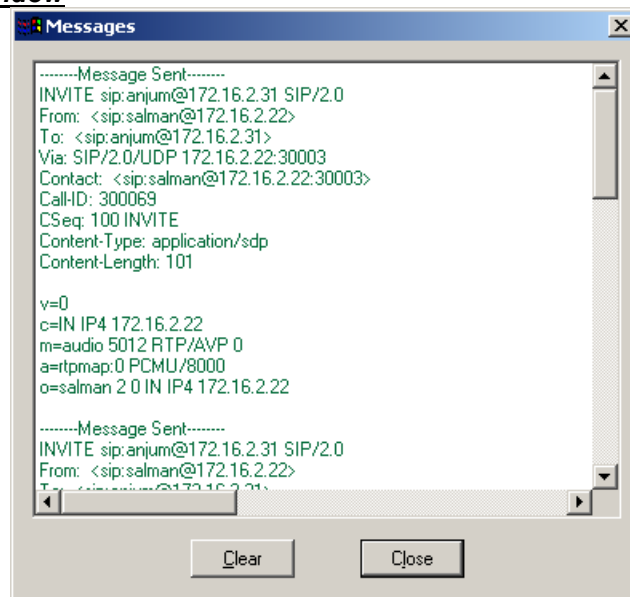


Enable/Disable message logging.



Enable/Disable messages view window.

Message window



3.2 Making a call

Dialing

User can dial to other SIP User Agent by:

1. Dialing other SIP user agent by key pad (by mouse click)

2. Typing phone number in Display panel

23768655


3. Typing SIP address in [user@IP](#) format e.g. [john@172.16.2.22](#)

Press  button after entering the number or press Enter.

Status indicator will indicate DIALING

Ring back tone will play if remote user is Ringing and Status indicator will indicate RING_BACK.

Redial Number

User can redial number by pressing  button.

3.3 Receiving a call


Ringing

On receiving an incoming call message from other user, Ring tone is played and the AVAZ SIP Phone's icon blinks in system tray.

Accept Call

User can accept an incoming call by pressing  button.

3.4 Call Termination

User can terminate an established call or cancel call negotiation by pressing  button.


3.5 Volume and Microphone level adjustment

Volume level

User can adjust volume level with  control.


Adjustment

 Increase

 Decrease

Microphone level adjustment

User can adjust microphone level with  control.

 Increase

 Decrease

4 Status Messages

In successful call establishment following messages are shown in the Status Panel.

Message	Meaning
IDLE	Termination is IDLE and in Off Hook Position.
Collecting Digits	User has started dialing Numbers.
DIALING	User has Pressed a valid Dial number. Signaling in progress.
RING_BACK	Remote Termination is Ringing and Ring Back is being played at Local Termination.
RINGING	An incoming call
TALKING	Call established

5 Error Messages

In case of any error following messages are shown in the Status Panel.

Message	Meaning
Sound card not initialized	Sound card is already in use
Remote user is Busy	Remote user is busy
Request time out	No response for SIP Request
Codec Mismatched	Codec mismatch

6 Known bugs/problems

Bugs	Meaning
Distorted GUI	GUI may appear distorted on some Display controllers
Volume adjustment	Volume and Microphone adjustment controls may not work properly