User's Guide for X-Lite and X-PRO

Xten SIP User Agents by Xten Networks, Inc.

Version 2.0 for
Windows: 98SE, NT 4.0, ME, 2000, XP, Pocket PC 2000/2002, and Mac OS X
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1.0 Introduction to Xten SIP User Agents

Welcome to X-Lite and X-PRO SIP User Agents from Xten Networks, Inc. Either of these products transforms your PC, Laptop or Pocket PC (including wireless) into a multi-function SIP softphone client. You can make calls using X-Lite and X-PRO within your enterprise LAN or between any two points which have Internet access. Both X-Lite and X-PRO can connect to any other SIP telephone including standards-compliant SIP softphones and hardphones, allowing quality voice communication over the Internet, for free.

1.1 What are X-Lite and X-PRO?

X-Lite and X-PRO, produced by Xten Networks, Inc. are both SIP User Agents - a software application that can transform your PC or Pocket PC into a phone, giving you the advantage of using the same system (for example your personal computer) to make and receive voice calls as you use to check your email and do other computing tasks. Session Initiation Protocol (SIP) is a signaling protocol used for establishing sessions between hosts over an IP network. SIP is an RFC standard and both X-Lite and X-PRO are compliant with RFC 3261 from the Internet Engineering Task Force (IETF).

This guide covers the various features of both X-Lite and X-PRO, which you can use to make and receive calls. With X-PRO (full-featured version) you can also provide full conferencing on a personal computer running Windows 98SE, NT 4.0, ME, 2000 and XP, a PDA running Pocket PC 2000/2002 or a Macintosh running Mac OS X version 10.2 or later. Xten User Agents deliver award-winning design and unsurpassed performance.

Quality of Service

These full-featured IP softphones were made for use with Windows PCs, Pocket PC handheld devices, and Macintosh machines. A full range of audio compression codecs means excellent quality of service on dial-up, broadband & wireless connections. Using broadband Internet connections and associated audio compression codecs, the quality of service is often equivalent to that of traditional telephone systems.

Compatibility and Scalability

Xten's SIP User Agents' underpinning is 100% compliant with SIP standards making interoperation with the world's Voice-over-IP networks a cinch. Here are some of the reasons why X-Lite and X-PRO are quickly becoming the SIP User Agents of choice for many IP Telephony Service Providers.

List of Features and Benefits

Features

- SIP Compliant (RFC-3261)
- Touch-tones [DTMF] (RFC 2833)
- 6 Lines (3 Lines for **X-Lite**)
- Line Hold
- Blind Call Transfer [IP & PSTN] (**X-PRO** Only)
- Supervised Call Transfer [IP & PSTN] (**X-PRO** Only)
- Do Not Disturb (**X-PRO** Only)
- Inbound Call 'Ignore' (**X-PRO** Only)
- Inbound Call 'Send to Voicemail' (**X-PRO** Only)
- Call Forwarding URI/URL (**X-PRO** Only)
- Voicemail URL (**X-PRO** Only)
- 7 Party Conferencing [IP & PSTN] (**X-PRO**)
- Auto-Conference (**X-PRO** Only)
- Multiple SIP Proxy Registration [10 Proxies]
- Dial/ Redial/Hang up
- Auto-Answer (**X-PRO** Only)
- Dynamic CODEC Selection
- Caller ID [SIP ID]
- Call Timer
- Silence Threshold
- Backspace/Clear/Delete
- Mute
- Microphone & Speakers Levels
- Microphone & Speakers Meters
- Push-to-Talk [PocketPC] (**X-PRO** Only)
- Speakerphone
- Last Caller-ID
- Last Call Duration
- Recent Calls Dialed
- Recent Calls Received
- Sound Device Selection [not for CE]
- Direct IP to IP Calling
- Easy-to-use Menu System
- Phonebook [Import/Export CSV]
- Speed Dial
- SIP REFER Method Supported (RFC-3515) (**X-PRO** Only)
- NAT/Firewall Traversal [Reverse UDP]
- STUN Firewall Discovery (**X-PRO** Only)
- X-NAT Firewall Discovery (**X-PRO** Only)
- X-Tunnels Firewall Solution (**X-PRO** Only)
- "Hot Keys" for Many Major Functions
- Tool tips over items on User Interface

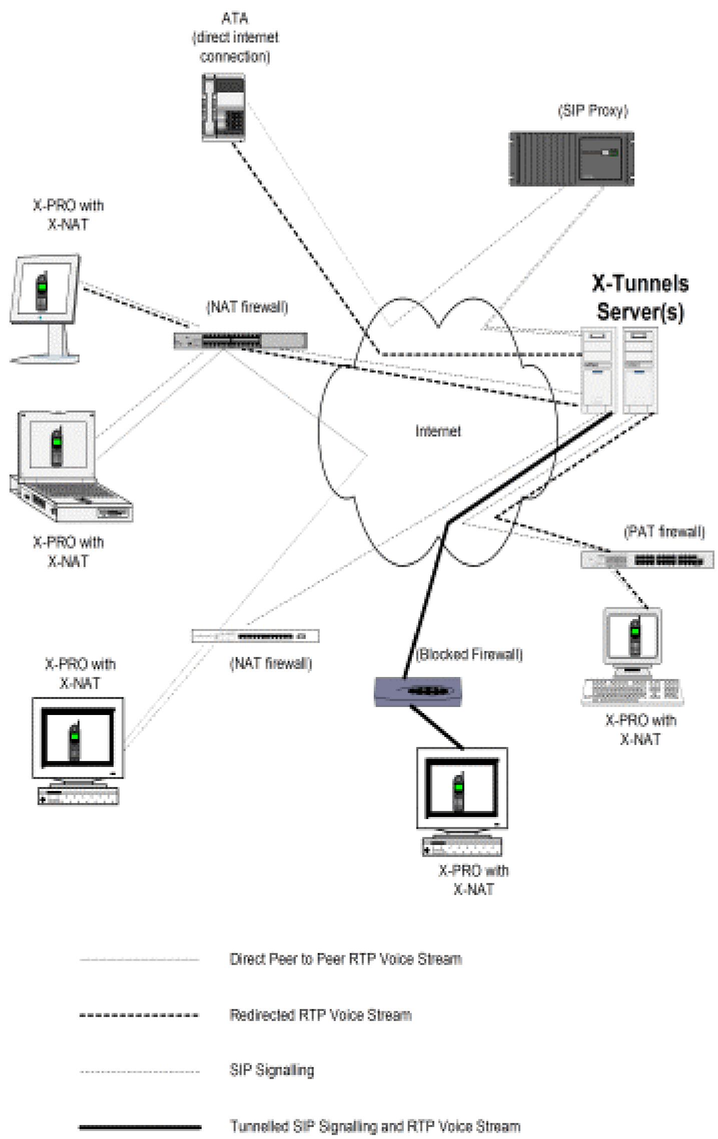
Benefits

- Easy to install
- Intuitive user interface and Menu. Easy to navigate and configure.
- G.711/G.723/SPX/iLBC/GSM codecs included for **X-PRO** [G729 licensed separately] *X-PRO for MAC OS X does not include G.723 or G.729*
- G.711/XPX/iLBC/GSM codecs for **X-Lite**
- Branding and Customization available (**X-PRO** Only)
- Standard PC and Pocket PC hardware
- NAT/Firewall support
- Specify NAT IP to be written in SIP messages
- Supports Windows 95/98/NT4/ME/2000/XP
- Supports Windows CE (**X-PRO** Only)
- Supports MAC OS X (version 10.2 or later)
- 7 party [PSTN + IP] Tele-Conferencing reduces costs for any business (**X-PRO** Only)
- RFC 3261, 2833 Compliant
- Multiple Network support reduces technical fuss and expense
- 6 Lines to either IP and/or PSTN allows for multi-tasking (3 Lines for **X-Lite**)

1.2 Available Xten Solutions

1.2.1 Firewall Tunneling for X-PRO Using X-Tunnels

Xten SIP User Agents are built to take advantage of reverse UDP mapping NAT firewalls. No firewall modifications are necessary. Additionally, XPRO along with X-Tunnels provides a full firewall solution for Xten SIP User Agents. X-Tunnels which employs X-NAT technology along with STUN firewall type discovery allows for traversal over various firewall configurations as illustrated below:



X-Tunnels, equipped with STUN server technology allows **X-PRO** to discover the following types of firewalls and to make adjustments when implementing the SIP and RTP protocols:

- ✓ External IP/No Firewall
- ✓ NAT Firewall
- ✓ PAT/Symmetric Firewall
- ✓ Blocked Firewall

X-Tunnels allows **X-PRO** to decide when to stream voice data through the **X-Tunnels** server or when peer-to-peer communication is appropriate. Additionally, X-NAT helps to prevent port conflicts on NAT-based firewalls to allow more than one **X-PRO** User Agent to function from behind the same firewall without having to choose a different set of ports for each client.

Other features of the **X-Tunnels** solution include:

- ✓ AES and 3-DES Encryption from **X-PRO** to **X-Tunnels** server
- ✓ **X-PRO** supports settings from a primary and a secondary STUN Server and provides for fail-over detection
- ✓ **X-Tunnels** provides load-balancing among available X-Tunnels servers

X-Tunnels is only available to clients running the **X-PRO** User Agent. For more information on the **X-Tunnels** Complete Firewall Traversal solution, visit http://www.xten.com and follow the product link.

1.2.2 Turnkey Wireless Telephony

Our partnerships with some of the world's largest IP Telephony Service Providers means that Wireless Providers can now offer cellular-like services on their existing 802.11x networks without provisioning additional equipment.

1.2.3 Xten SIP User Agents for Pocket PC

This new wireless communications software was developed with mobility in mind making it a perfect fit for Wireless ISPs and remote enterprise work environments.

Xten SIP User Agents for Pocket PC are compatible with all 802.11x networks and have been tested on the majority of Pocket PC devices available today.

1.2 Requirements and Installation

1.2.1 System Requirements

This section covers the minimum hardware and software you will need to use with Xten SIP User Agents on a personal computer.

- ✓ **Processor**: Pentium II 300 or faster
- ✓ **Memory**: 128 MB RAM or better
- ✓ **Operating Systems**: Windows 98SE, NT 4.0, ME, 2000, XP,
- ✓ **Online Capabilit**y: Wired or wireless broadband Internet connection
- ✓ **Sound Card**: 16 bit sound card (SoundBlaster or equivalent)
- ✓ **Color Setting**: 16 bit (High Color)

For a PDA or Tablet PC running Pocket PC 2000/2002 3.0 or Pocket PC 2002 a full-duplex sound card with a headset is preferred, push to talk is otherwise recommended.

Any Macintosh running Mac OS X version 10.2 or later will be sufficient to run the Xten SIP User Agent.

Any kind of TCP/IP connection including wired and wireless Ethernet or wide area connections (WAN) like xDSL, Frame Relay, full or fractional E1/T1, etc. You will get a better sound quality with a broadband connection but a v.90 analog modem is enough if you are using voice compression (ex. G.723)

1.2.2 Before You Install

Important: You must sign up for an account with an SIP service provider before you can make a call with **X-Lite/X-PRO**. For a list of Xten Networks' providers, visit the following link http://www.xten.com/index.php?menu=ippstn. After signing up for an account, you will have a Username or SIP Number and a Password

Some User account information is required from your SIP Proxy administrator before you install **X-Lite/X-PRO** and run the software for the first time.

- ✓ **Phone number or SIP account number** is required to make and receive calls
- ✓ **SIP Proxy Server.** To process SIP calls, you must have the address of the SIP Proxy Server.
- ✓ **SIP Proxy Server Port.** The default SIP port is 5060 used by the SIP Proxy.

1.2.3 Installing X-Lite/X-PRO

This section outlines how to install, and how to uninstall **X-Lite/X-PRO**.

To install X-Lite/X-PRO

1. Make sure to exit any software applications that use sound, such as CD and media players.
2. Run the **X-Lite/X-PRO** install program.
   (X-Lite is available for free downloaded from http://xten.com/lite.php .)
3. Follow the instructions at each step, progressing through the InstallShield Wizard until the installation is complete. At the InstallShield Wizard Complete step, click **Finish**.
4. **X-Lite/X-PRO** displays a Checkbox **"Would you like to run X-lite/X-Pro now?"** It is recommended you click **Yes**, in order immediately configure your **X-Lite/X-PRO**.

Important: To make and receive calls, you must configure the application to connect to the service provider for which you have an account. This information registers you with the SIP Proxy Server; please note the Login Status line on the **X-Lite/X-PRO** display.

1.2.4 Uninstalling X-Lite/X-PRO

To uninstall **X-Lite/X-PRO** go to **Control Pane**l->**Add/Remove Program**s->**X-Lite/X-PRO**->**Install/Uninstall**. When a dialog box asking whether you want to restart your computer appears, click **Yes**.

2.0 X-Lite/X-PRO Control Panels and Dialogs

Dial Pad Graphical User Interface (GUI)

X-Lite/X-PRO provides a handy dial pad to quickly dial numbers for calls and also to dial special dial codes to display configuration and control dialog screens.

TIP:

- ✓ Right Click on the Call Status Area with your mouse to invoke a SIP Proxy Selector for your next outbound call.
- ✓ Scroll your mouse down to select which Proxy you want to use for your call and Left Click.
- ✓ Enter the SIP URL or Phone Number you wish to call and click on the Green Phone to make your call.

NOTE: The Check marks indicate that you are registered to the Proxy. If you do not see a check mark next to the Proxy you want to use your call will not go through nor will you receive calls. Your software must be registered with the appropriate SIP Proxy in order to function. Please check with your network administrator for additional assistance.

Help, Minimize, Close

Call Status
Displays the current call status

Call Data
Dialed number, Call Timer
SIP Caller ID

Do Not Disturb [DND]
Auto-Answer [AA]
Auto Conference [AC]

Hang Up [Red Phone]

Microphone Fader Control
Click on Microphone Icon to enable/disable LEDs. Click again to enable Silence Threshold Control.

CODEC Selection Control
Black type is active.

Lines 1 – 6
Click to Select Line, 2nd Click to place on hold.

Dial [Green Phone]

1) Phonebook Control
2) Clear / Backspace
3) Menu Control

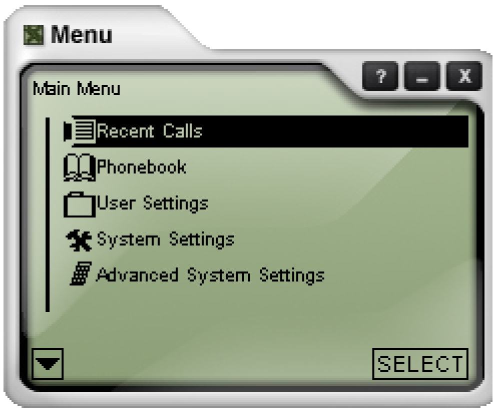
Speaker Fader Control

MUTE, Call Transfer, Conference

2.1 Main Menu

The menu system of X-Lite/X-PRO can be accessed by clicking on the Menu Control found on the Dial Pad GUI as shown in the previous diagram.

The XPRO Menu System

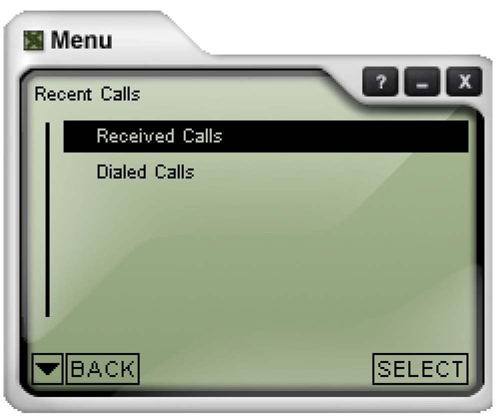


The following items are currently available from the Main Menu:

- **Recent Calls:** Dialed and Received
- **Phonebook:** Phonebook & Speed Dial List
- **User Settings:** Typical User Settings
- **System Settings:** Network Configuration, etc.
- **Advanced System Settings:** Advanced Audio, SIP Protocol, and Display settings among others. If unsure do not alter default settings.

2.2 Recent Calls

Viewing Recent Calls



Select **Recent Calls**.

- ✓ To select a menu item, double-click on the item or highlight the item and click on the SELECT button at the bottom right of the menu panel.
- ✓ A menu item can also be selected by pressing the ENTER key when the item is highlighted.
- ✓ Clicking on either of the two arrows at the bottom left of the menu panel will move focus from one item to the next.
- ✓ The UP and DOWN arrows on the keyboard or scroll-wheel on your mouse can also be used to move between items.

A menu item can also be highlighted by a single click of the mouse over the item.

The **Recent Calls** menu allows the user to review both received and dialed calls and stores up to a maximum of 50 for each category.

Note: To return to a parent menu click on the BACK button at the bottom left of the panel or press the ESC key on your keyboard.

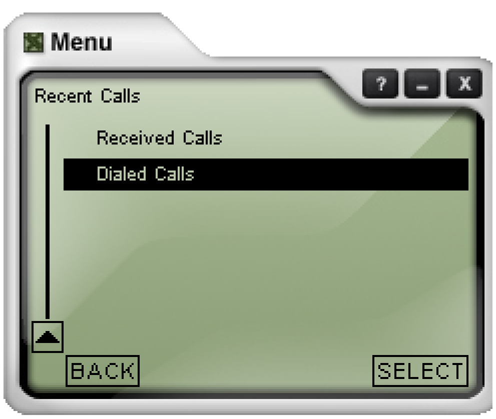
Selecting a Received Call

- ✓ To view the details of a given call, highlight and then select the call entry.

Viewing the details of a call

- ✓ The following details are available for the call as shown to the left
- ✓ Click BACK to return to the Recent Calls menu.

Viewing Dialed Calls

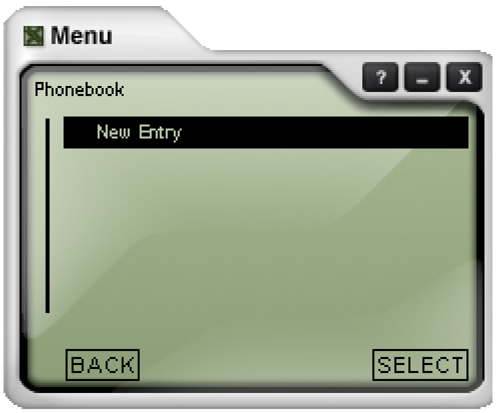


- ✓ To view the **Dialed Calls**, Highlight and SELECT.
- ✓ After entering the Dialed Calls menu, select the call from the list to view its details.
- ✓ The same details are available that were available for "Received Calls".
- ✓ A tick mark beside a call indicates that the call dialed was answered.

2.3 Phonebook

Much like a cellular telephone menu you can ADD, DELETE & EDIT your Phone book entries.

Adding a new entry to the Phonebook

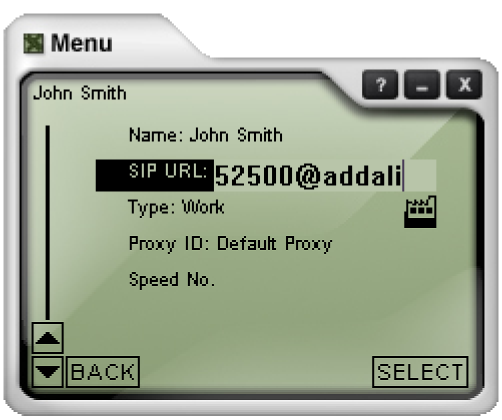


- ✓ You can reach the Phonebook by selecting it from the Main Menu.
- ✓ To add a new entry to the Phonebook, highlight and select the New Entry item from the list.

Filling in the fields of a new Phonebook entry

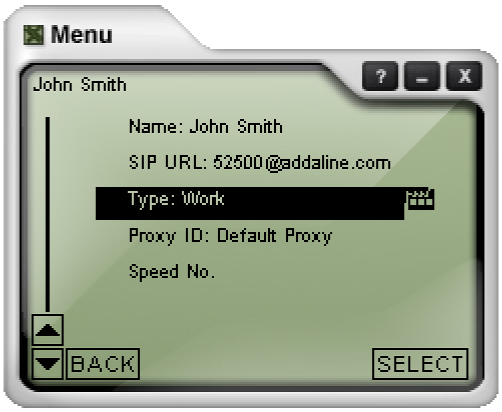
- ✓ Highlight and select "New Entry" to add a new entry to the Phonebook.
- ✓ Enter the Name of the person that you would like to be displayed in the Phonebook.

Entering the SIP URL of the contact



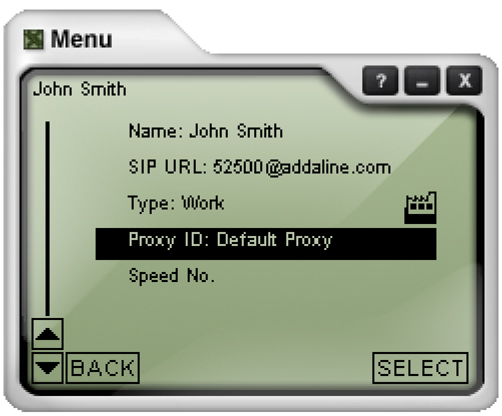
- ✓ Move down to the **SIP URL** field and enter the SIP URL of this contact.

Specifying the Type of contact

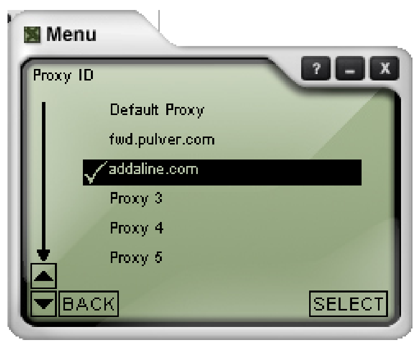


- ✓ To specify the **Type** field of this contact, select this item to enter a list of contact types.
- ✓ Highlight and select the type of contact from the list and click on BACK to return to the Phonebook entry.

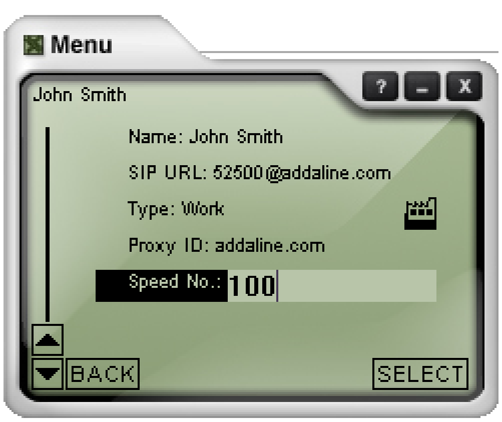
Specifying the Proxy ID from list of available proxies



- ✓ Select **Proxy ID** to bring up a list of available proxies to be used to reach this contact.
- ✓ Select the appropriate proxy and return to the Phonebook entry.
- ✓ Refer to **Section 2.5.1 System Settings – SIP Proxy** on how to configure different SIP Proxies to be used by your SIP client.

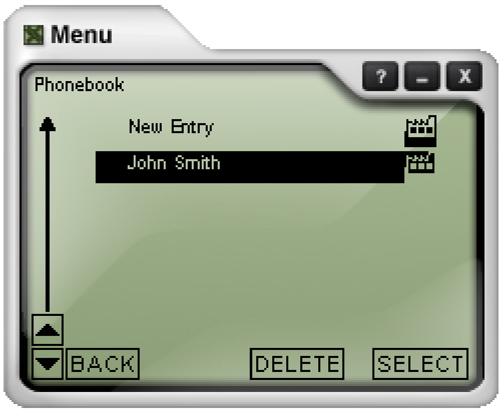


Assigning a Speed Dial number for your contact



- ✓ Finally, assign a **Speed Dial** number for this contact. This number can be used directly from the main interface to make a call to this user.
- ✓ Click BACK to add another entry or Click the X or Close button on the Menu to exit.

New Entry is now available

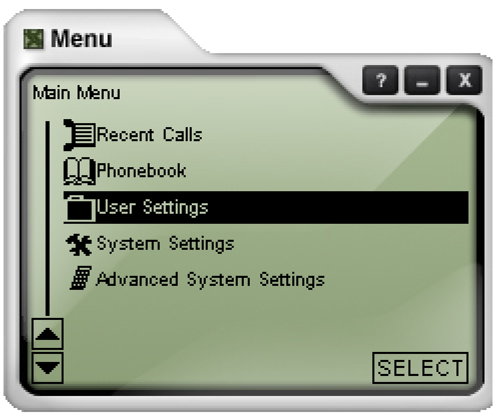


You new entry is ready to use.

- ✓ Click on the Dial control on the main interface to call this entry while it is highlighted. For more information on placing calls, refer to **Secion 3.1 How to Place a Call**.
- ✓ Use the DELETE key to clear the entire entry. All settings for this user will be lost.

2.4 User Settings

Selecting User Settings from the Main Menu



- ✓ Highlight and select **User Settings** from the Main Menu.

Viewing Phone Status

- ✓ This includes the user name you have registered for your default SIP Proxy (more information on this in **Section 2.5.1 System Settings – SIP Proxy**) as well as information relating to the Software Build Number.

Viewing Features

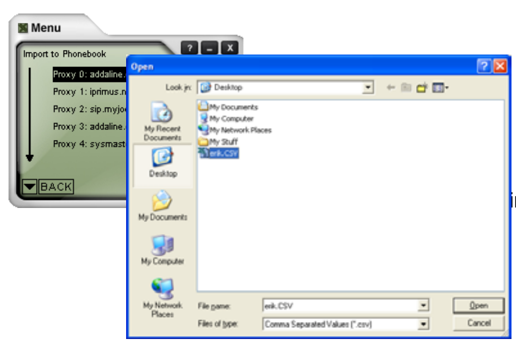
Call Features

- ✓ Enable DND (Do Not Disturb) - Yes/No
- ✓ Auto-Answer – Yes/No
- ✓ Auto-Answer Wait Time (ms) – specify value in milliseconds
- ✓ Enable Auto-Conference – Yes/No

For more information on these features, refer to **Section 4 Advanced Features**.

Import/Export to Phonebook

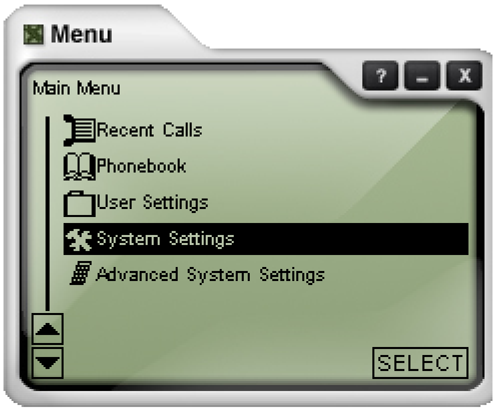
- ✓ XPRO allows you to import contacts stored in a CSV file into the Phonebook.
- ✓ When importing, choose the Proxy that you want the entries related to.
- ✓ Likewise, existing contacts from the Phonebook can be exported to a CSV file via **Export Phonebook**



import.

2.5 System Settings

Selecting System Settings from the Main Menu



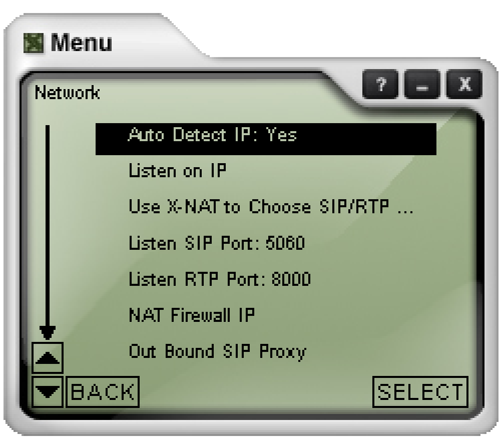
- ✓ The **System Settings** Menu is available from the Main Menu
- ✓ **System Settings** allows you to adjust your network, SIP Proxy, and audio device configuration.

2.5.1 System Settings - Network Settings

Select **Network** from the Systems Settings Menu

- ✓ Configure your network settings here

Available Network Settings



Auto Detect IP – select "Yes" to allow X-Lite/X-PRO's internal processes to dictate which network settings to use.

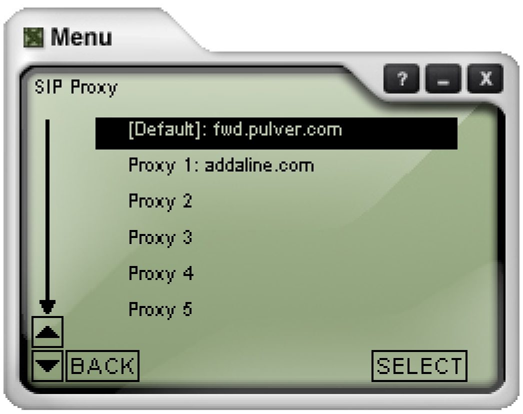
- ✓ Average users should leave this on. If you turn this off you will have to manually enter in all the network information that follows.
- ✓ **Use X-NAT to Choose SIP/RTP Ports** – Setting this option to "Default" or "Always" should be used when there may be several concurrent users behind the same firewall. "Default" allows the application to decide when to use this option.
- ✓ **Out Bound SIP Proxy** – Calls made via no specified Proxy will use this.
- ✓ **Primary/Secondary STUN Server**- XPRO will communicate with the STUN Server to determine what type of firewall it is behind.

2.5.2 System Settings – SIP PROXY

- ✓ Configure your SIP network settings here. The SIP network acts as your service provider.

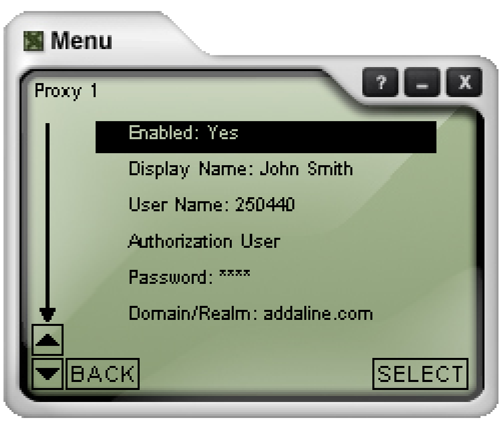
List of Configured SIP Proxies

- ✓ You are allowed 10 SIP network registrations within X-PRO.
- ✓ The first network field you configure will always be your default. To make calls on this network simply dial the SIP URL you are calling.
- ✓ Every subsequent network you configure will have a PROXY ID. To make calls using an alternate network besides your default you must type a "#" in front of the Proxy ID number you want to use.
- ✓ For details on how to place a call refer to **Section 3.1 Place a Call**

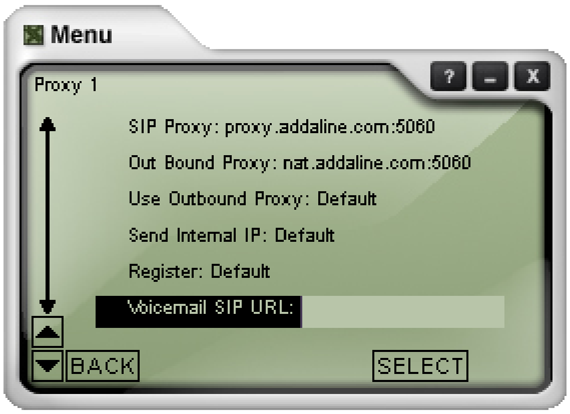


Configuring the SIP Proxy

- ✓ Set **Enabled** to "Yes" if you want to ensure that this Network is active in your X-PRO software.
- ✓ Enter your **Display Name** as you would others to see it in their Call ID.
- ✓ User Name is your VOIP service provider number (eg your addaline or FWD number)
- ✓ **Authorization User** can usually be left blank unless otherwise stated by your SIP network provider.
- ✓ **Password** is your VOIP service provider account password.
- ✓ **Domain/Realm** of your VOIP service provider.



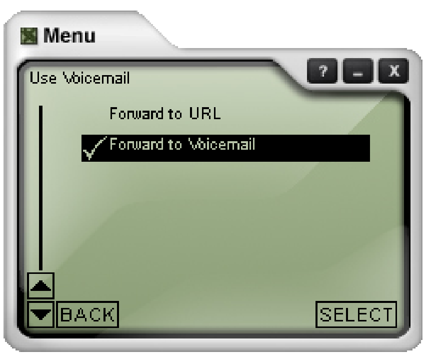
Configuring the SIP Proxy (Continued)



- ✓ **SIP Proxy** URL of your VOIP service provider. Port 5060 will be appended if do not specify a port.
- ✓ **Outbound Proxy** will be needed if you are traversing a firewall.
- ✓ **Use Outbound Proxy** set to "Default" to allow XPRO to determine whether it needs to use it or not. Outbound Proxy will be needed if you are traversing a firewall. ***Outbound When NAT IP:*** *The most commonly used setting for those using Private IP address or users behind firewalls.*
- ✓ **Send Internal IP** – needed when recipient's IP address is also "internal". Set this parameter to "Default" to allow XPRO to determine when this is needed or not. *If you are using Outbound when NAT IP for your Proxy Mode you will need to turn ON this feature.*
- ✓ **Register –** this notifies your SIP Proxy of your IP address, needed to receive incoming calls. This can be left as "Default".
- ✓ **Voicemail SIP URL** is used to divert call to voicemail, if you do not have a Voicemail SIP URL, leave this blank.

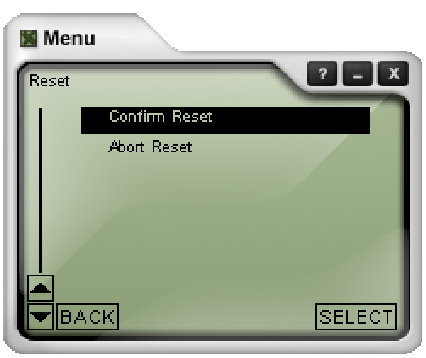
TIP: To set up a simple voicemail system. Use an ATA-186/188 from Cisco and attach it to an answering machine. Assign a SIP URL to the ATA and reference that URL as your Voicemail SIP URL in X-PRO.

- ✓ **Forward SIP URL** operates in a similar fashion to that of Voicemail SIP URL. The difference being that when a SIP URL is placed in this field your X-PRO will always forward to this URL.



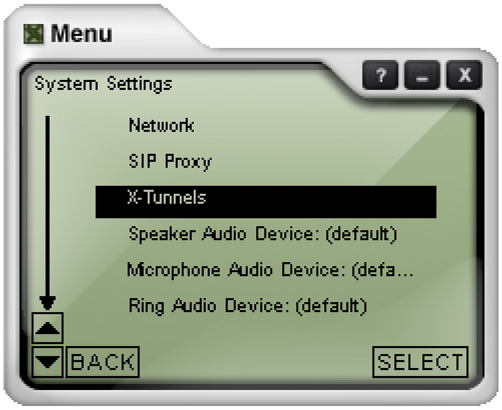
- ✓ **Use Voicemail**: Toggles which URL you want your calls to be forwarded to when using DND or IGNORE.
- ✓ Double-click on the item to reveal the choices of **Forward to URL** or **Forward to Voicemail**.

- ✓ **Direct Dial IP**: Normally left as "No". (set it as "Yes" when SIP Proxy is not present)



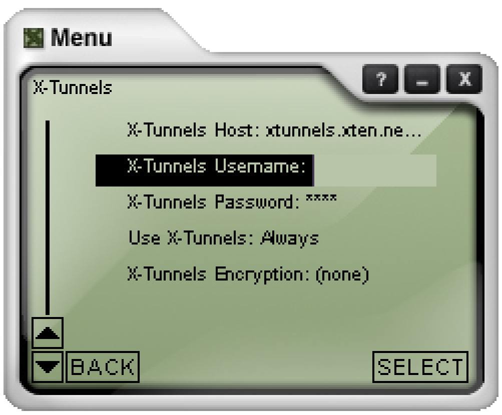
- ✓ The final field is **Reset** which allows you to clear all entered values for that proxy and start fresh.
- ✓ Double-click on the item to bring up the following sub-menu allowing you to **Confirm** or **Abort** the operation.

2.5.3 System Settings – X-Tunnels



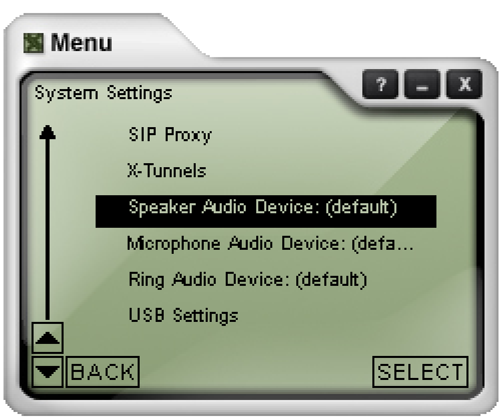
- ✓ The **X-Tunnels** setting is used to configure X-PRO to work with an X-Tunnels Server.
- ✓ The X-Tunnels Server solution allows users behind a firewall to initiate and successfully make calls to other users who may sit behind a firewall as well.

X-Tunnels Configuration Panel

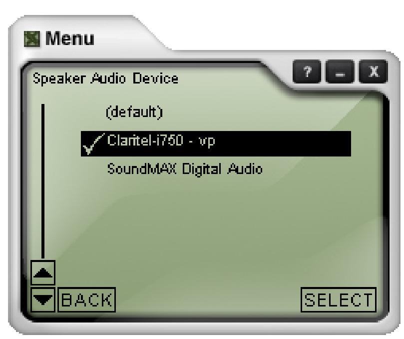


- ✓ **X-Tunnels Host**: Use this to specify the URL of your X-Tunnels Server
- ✓ **X-Tunnels Username**: the user name assigned by your X-Tunnels administrator needed to logon to the server
- ✓ **X-Tunnels Password**: the password associated with your user name
- ✓ **Use X-Tunnels**: setting this parameter to "Default" allows X-PRO to decide when the X-Tunnels server is needed and when it can be bypassed
- ✓ **X-Tunnels Encryption**: allows for specification of an encryption algorithm to be used to encrypt data sent between the SIP client and the X-Tunnels server

2.5.4 System Settings –Audio Devices and USB Settings



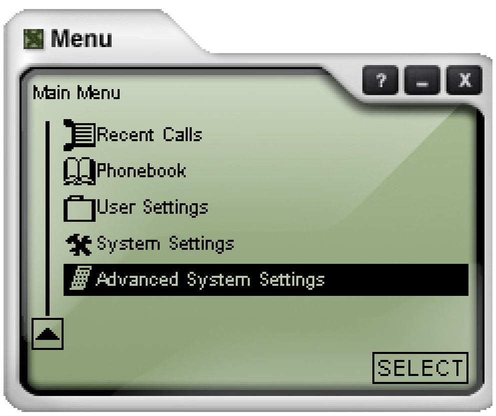
- ✓ **Speaker Audio Device:** Select the Audio device you would like to use as your speaker
- ✓ Repeat this procedure for **Microphone Audio Device** and **Ring Audio Device**.



USB Settings Panel

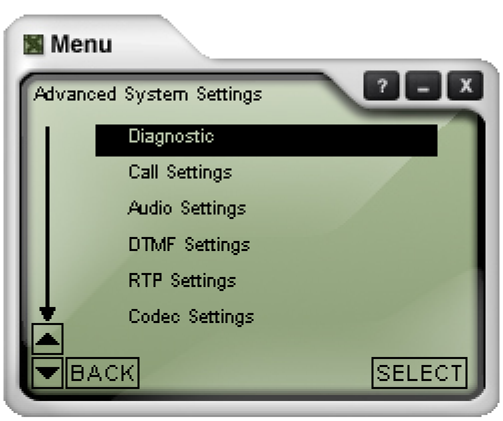
- ✓ **Preferred USB Phones:** Select your preferred device.
- ✓ **USB Off Hook Pop to Top**: Options are “Yes” or “No”.
- ✓ **USB On Hook Minimize**: Options are “Yes” or “No”

2.6 Advanced System Settings



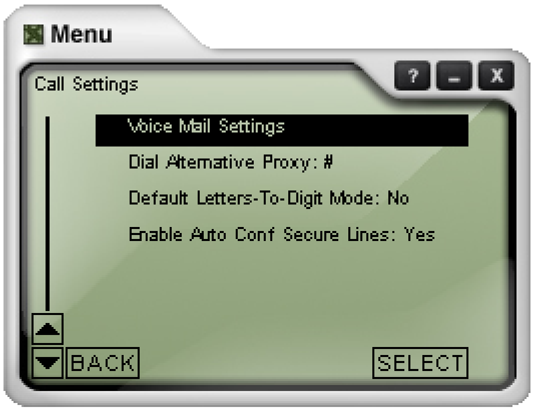
- ✓ **Advanced System Settings:** This section provides access to some of the more sensitive parameters. Only experienced users need adjust these settings.

2.6.1 Diagnostic Settings



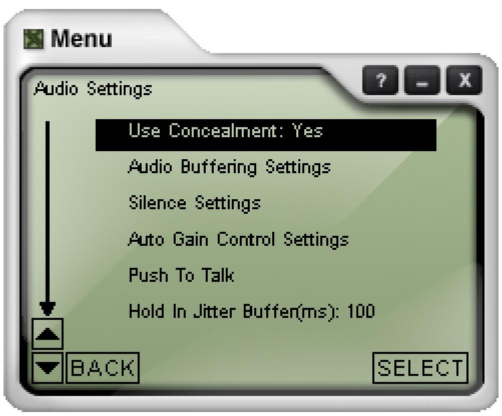
- ✓ SIP Protocol specific information is output to the **Diagnostic Log**. This information is sometimes needed when network issues are involved in preventing the establishment of a proper call.
- ✓ The **Diagnostic** settings are used to manage the viewing of this information.
- ✓ **Diagnostic Log**: Select this item to bring up the X-PRO Diagnostic Window.
- ✓ Select **Clear Diagnostic Window** to clear the contents of the window to start fresh.

2.6.2 Call Settings



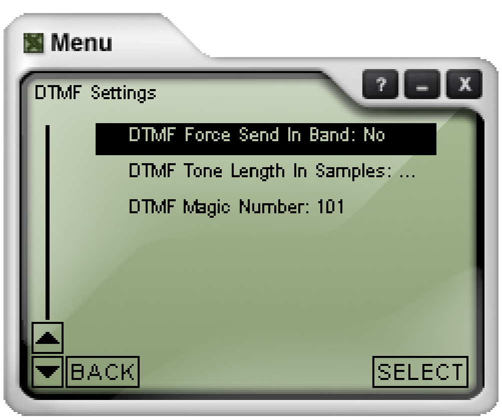
- ✓ **Call Settings** are used to configure your voice mail and other call-related options.
- ✓ Select **Voice Mail Settings** to fine tune your voice mail.
- ✓ **Letters to Digit Mode:** Converts letters to digits. E.g. 1-800-callusa would actually convert the letters in callusa to digits.

2.6.3 Audio Settings



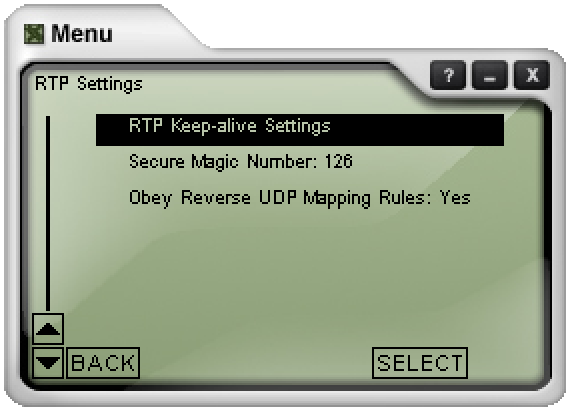
- ✓ **Audio Settings** are used to fine-tune audio quality and should not be adjusted unless necessary.
- ✓ **Push to Talk:** For Pocket PC Users to enable the Voice Record button as a Push-to-Talk button.

2.6.4 DTMF Settings

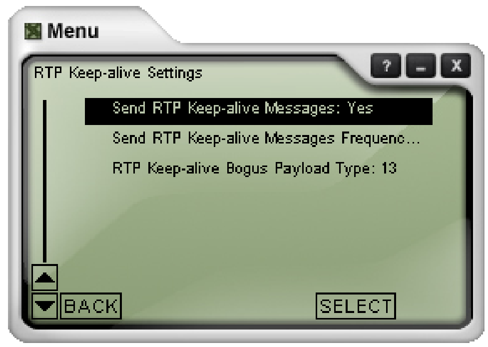


- ✓ **DTMF Settings** are used to adjust the way dial tones are generated. These settings should only be modified by SIP Service Provider Administrators.
- ✓ **DTMF Force Send In Band** instructs XPRO to send DTMF tones "in band" needed for those networks which are not RFC 2833 compliant.

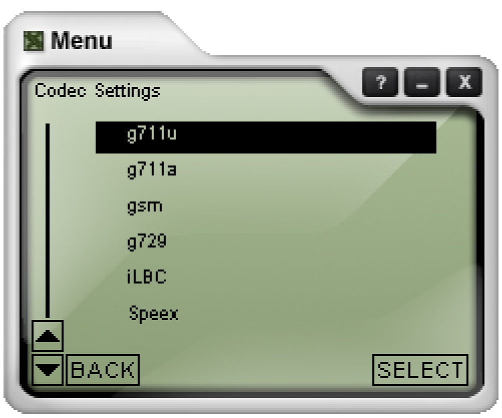
2.6.5 RTP Settings



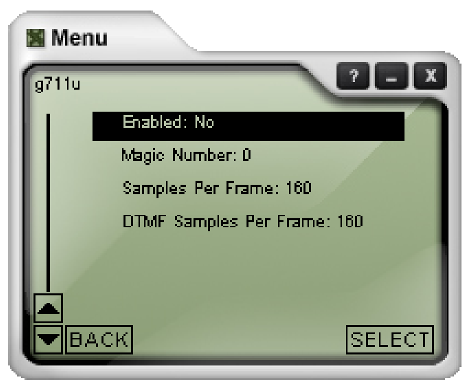
- ✓ **RTP Settings** include **RTP Keep-alive Settings** which can be used to configure the way that RTP Keep-alive messages are sent.
- ✓ RTP Keep-alive messages are used to ensure that the UDP ports used in the RTP-transfer of audio are not closed by the firewall.
- ✓ Normally, **Send RTP Keep-alive Messages** should be set to "Yes"



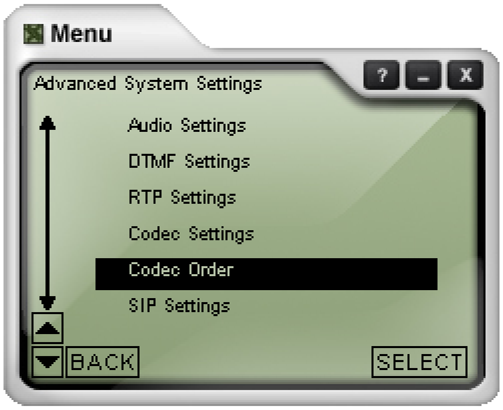
2.6.6 Codec Settings



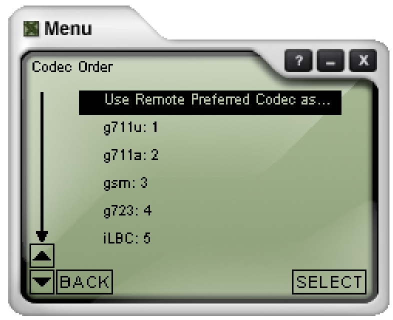
- ✓ **Codec Settings** allow the user to configure any of the available codec's to be used for compression of the audio stream.



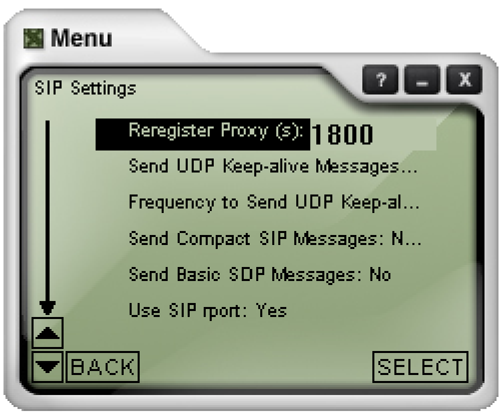
2.6.5 Codec Order



- ✓ **Codec Order** allows you to set the preference of codec to be used for a call. Codec's with a lower number will be given higher priority.
- ✓ **Use Remote Preferred Codec as Local Preferred Codec** allows you to use the codec preferred by the other calling party.



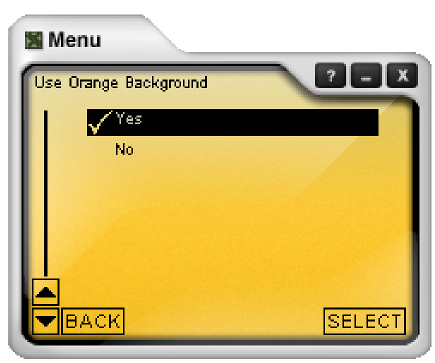
2.6.5 SIP Settings



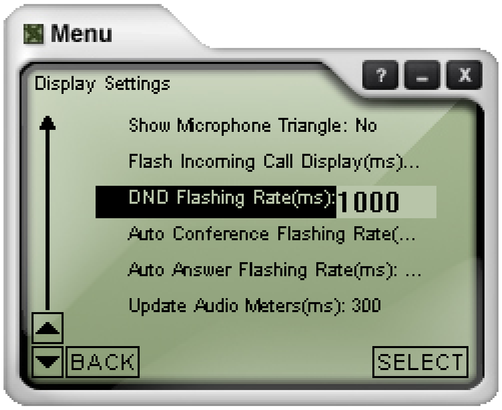
- ✓ **SIP Settings** allow an advanced user to fine tune various elements of the SIP protocol implementation.

2.6.5 Display Settings

- ✓ **Display Settings** allow you to customize the presentation of the user interface.
- ✓ **Main Window Always on Top:** X-PRO will always be on top of other applications. Default is NO.
- ✓ **Use Orange Background:** Toggle this setting to "Yes" to change the color of your LCD screens to Orange.



Display Settings (continued)



- ✓ **MIC Audio Meter:** Displays the MIC LEDs, default is "Yes".
- ✓ **Speaker Audio Meter:** Displays Speaker LEDs, default is "Yes"
- ✓ **Show Microphone Triangle:** Silence Threshold indicator, default is "No"
- ✓ **Flash Incoming Call Display:** The rate at which the call display flashes.
- ✓ You can also adjust the rate at which the **DND, Auto Answer,** and **Auto Conference** indicators on the Call Display area flash.

2.7 Call Data Display

The Call Data display provides detailed information about the call in progress and the previous call completed.

The following can be seen from the Call Data display:

- ✓ **Caller ID [SIP]:** The Caller ID displays the SIP number actually sent to server, including the domain (e.g. 21290@fwd.pulver.com) and on inbound calls the Caller ID shows the "display name" provided by the calling SIP User Agent.
- ✓ **Call Timer:** The call timer gives the exact number in hours, minutes and seconds, which have transpired since the call achieved the "connected" status.

Last Call Data Display: For the last completed call the above Caller ID and Call Timer data is moved to the portion on the display screen just below the active call data where the previous information is archived to support the user call tracking requirements. The data is displayed in the following manner:

- ✓ **Last Caller ID:** The Last Call ID displays the SIP number actually sent to server, including the domain (e.g. 21290@fwd.pulver.com) or, if the last call was inbound, shows the Caller ID "display name" provided by the calling SIP User Agent.
- ✓ **Duration:** The length of the last call time in hours, minutes and seconds.

Call Status
Shows *Call Incoming*

Call Timer
Set to zero, starts when call is picked up

Caller ID
Shows SIP ID of caller
And allows for IGNORE, which sends caller a BUSY signal or SEND TO VOICEMAIL which send the call directly to Voicemail.

Click on the Line Button that is flashing to answer.

2.8 Dial Pad Audio Controls and Meters

The Dial Pad Main Display has both controls and meters for to adjust the audio quality of **X-Lite/X-PRO**. These Dial Pad Audio features are: *(a Tool tip is displayed for items on the User Interface when the mouse is moved and left over the item)*

- ✓ **Microphone Volume Slider:** Adjust the slider to increase or decrease the maximum input level of your selected microphone device.
- ✓ **Microphone Volume Meter:** Indicates input sound level for the current session.
- ✓ **Speaker Volume Slider:** Adjust this to increase or decrease the maximum output level of your selected speaker device.
- ✓ **Speaker Volume Meter:** Indicates output sound level for the current session.
- ✓ **Microphone Silence Threshold Slider** is activated, by clicking the round **Microphone Icon** located below the microphone meter. The triangular slider/pointer can be moved with the mouse, and should be adjusted according to the ambient sound environment of the microphone pickup. X-Lite/X-PRO does not send IP packets during periods of silence, which reduces the network bandwidth load.
  <u>**NOTE**</u>: If this feature is activated and set too high, fragments of your speech may be lost.

3.0 How-to

3.1 Place a Call

With **X-Lite/X-PRO** you can make calls by dialing on the Dial Pad, by using the speed dial list, or by using the computer keyboard.

At the **Dial Pad,** enter a phone number by clicking on the dial pad number buttons or by typing directly on your PC keyboard. Click the **Call** button, or press **Enter.**

You will note the **Caller ID** display line shows the actual number and domain being dialed. **X-Lite/X-PRO** status line displays the *Trying* message while the system processes the number you dialed and contacts the phone dialed. Once the system establishes contact, **X-Lite/X-PRO** starts timing the call and the status line displays the *Connected M*essage. Once the call starts, just speak into the microphone. When you're ready to hang up, click **Hang-up**. While dialing, you may need to clear a phone number from the display, this is done clicking on the **Clear** button found above the **Dial Pad**.

3.2 Answer a Call

You need to have **X-Lite/X-PRO** running to answer an incoming call. When your computer receives a call, you will hear a ringing sound. Simultaneously, the line 6 button will began flashing and the Caller ID will display the identification of the calling party. X-PRO has an automatic answer feature, which is outlined in Section 4.8.

The **Do Not Disturb (DND) (X-PRO only)** mode can be also turned on and off on the display control menu. To activate DND click the indicator once and it will display in dark letters. To deactivate DND click the indicator once more and it will become grayed. If **X-PRO** is in the**" Do Not Disturb**" mode, all incoming calls will be automatically sent directly to Voicemail, or given a "Busy" indication if no voicemail SIP URI has been configured.

3.3 Placing a Call on Hold

While you're on a call in progress, you can place a call on hold simply by selecting another Line button. The active call is automatically placed on hold and the Line button flashes to indicate the call is on hold. To resume the call, re-select the line button where the active call is located. Alternatively, you may place a call on hold by clicking its Line button and resume the call by selecting the Line button again.

3.4 Ending a Call

As with traditional phones, a call is brought to a close when either you or the person you are talking to actively ends the call. To end a call, simply click the red **Hang Up button** at the left of the **Dial Pad**.

3.5 Start a Conference (X-PRO Only)

X-Lite supports connections with multiple respondents, but you may talk only to one respondent at a time. To talk to several respondents simultaneously in a conference call **X-PRO** supports this feature and allows mixing both Internet (IP) and regular telephone (PSTN) calls in a single conference.

You may initiate the conference by calling the parties or you may allow the conference parties to call you. As you answer and/or make the calls individually, the **Line buttons** become populated.

To initiate the conference simply click the **Conference button** below the **Dial Pad** and ALL of the lines which are active will be placed on hold as indicated by flashing Line buttons. Click the **Conference button** once again and all active lines will be involved in the conference.

Additional callers may be selectively added to the conference. For example, if a caller calls the conference host, the host must put the entire conference on hold when he answers the call. At this point, the other parties will not be able to communicate while on hold. The host can then select the incoming call, speak privately to the individual, and may choose to add the party to the conference by clicking the conference button. This will put all active lines on hold, at which point the host clicks the conference button again to re-establish the conference with all lines including the new calling party.

Conferences may be multi-node, meaning each **X-PRO** phone user can add parties to the conference, utilizing their **X-PRO** softphone as a node. Conferences can also be multi-server, meaning the conference nodes may utilize any of the SIP servers configured within **X-PRO** to add parties to the conference. Thus a conference might be composed of users from various SIP networks including PCs, Pocket PCs, regular telephones, cellular telephones, etc.

NOTE: The major limitation for host nodes of the conference tends to be bandwidth considerations. Each conference node requires sufficient bandwidth to provide audio encoded in the audio compression codec selected for each participant. Failing to provide for enough bandwidth causes a severe drop in audio quality in the conference.

For more details regarding conference calling, refer to Section 4.7 **Call Conferencing**.

3.6 Transfer a Call

If you are on an active call, that call may be transferred to another SIP phone. Employing the SIP REFER method (RFC 3515), **X-Lite/X-PRO** allow for two different types of call transferring, **Blind Call Transfer** and **Supervised Call Transfer**.

3.6.1 Blind Call Transfer

To initiate a transfer without informing the third party that you are sending a call their way, follow these steps:

1. With an active call, inform the other party that you are going to transfer them to a third party and then click the **Transfer button.**
2. Dial the number you wish to transfer the call to.
3. Click the **Transfer button** again and the call is sent to the number dialed.

3.6.2 Supervised Call Transfer

To conduct a transfer in which you first call and inform the third party that you are sending a call their way, perform the following steps after you have an active call in progress.

1. Select a different line and call the third party.
2. Inform the third party that a transferred call is about to be received and press the **Transfer** button.
3. Select the line with the original party (original caller to be transferred).

After this, the original caller will call the third party while you remain on hold with both of them. The third party’s X-PRO should auto-answer the original caller’s call and switch to talking with them while hanging-up with you.

After this you will notice that the original caller has been transferred to the third party and you are free to hang up on the original caller. This will result in the original caller talking with the third party without you as part of the conversation.

4 Extended Calling Features
The **X-PRO** softphone includes many extended calling features. These features include the following and their use and configuration is described in the sections to follow:

- ✓ Voice Mail
- ✓ Call Forwarding
- ✓ Inbound Call Options
- ✓ Call Conferencing
- ✓ Automatic Answering
- ✓ Automatic Conferencing

4.1 Configuring Voicemail
You can configure **X-PRO** to transfer unanswered calls to a voicemail server system by specifying the **Voicemail SIP URL** found under **System Settings** -> **SIP Proxy** -> <Proxy ID> in the menu system. For detailed illustration of how to navigate the **X-Lite**/**X-PRO** menu system, refer to section 2.

4.2 Configuring Call Forwarding
You can configure **X-PRO** to transfer unanswered calls to a SIP URL by specifying the **Forward SIP URL** found under **System Settings** -> **SIP Proxy** -> <Proxy ID> in the menu system.

4.3 Inbound Call Options
X-PRO gives the user several control functions to manage incoming or inbound calls. The three discussed below enhance that control based upon the user's preferences and the real-time situation.

4.3.1 Do Not Disturb (DND)
The **Do Not Disturb (DND)** feature is controlled by the DND indicator on the display menu. To activate DND click the indicator once and it will display in dark letters. To deactivate DND click the indicator once more and it will become grayed.

When DND indicator is activated, all incoming calls will be automatically sent directly to Voicemail as configured under section 4.1 or given a "Busy" indication if no voicemail SIP URI has been configured.

4.3.2 Call Ignore
When an inbound call rings, a new icon appears just below the Call ID display indicating **IGNORE.** If the user is hosting a conference or otherwise unable to accept the call, a click on the **IGNORE** icon will give a "Busy" indication to the caller.

4.3.3 Send to Voicemail
As shown in the above screen capture, when a call rings, a new icon appears just below the Call ID display indicating **SEND TO VOICEMAIL.** The user may choose not to accept the call, and click on the **SEND TO VOICEMAIL** icon to immediately transfer the caller to the Voicemail URI as configured under Section 4.1.

The IGNORE icon and the SEND TO VOICEMAIL icon are activated by an incoming call.

4.4 Call Conferencing

X-PRO supports multiple SIP Proxy Servers, multiparty conferences from both the PC and Pocket PC platforms.

As mentioned previously, lines are populated as you answer and/or make the calls individually to participants, place each call on **HOLD**. To initiate the conference simply click the **Conference button** on the **Dial Pad** and *ALL* active lines will be joined in the conference.

Callers may be selectively added to the conference, for example, if a caller calls the conference host, the host must put the entire conference on Hold, when he answers the call. The other parties will not be able to communicate while on hold. The host can select the incoming call, add the party to the conference by simply clicking the conference button putting all active lines on hold and then clicking the conference button again to re-establish the conference with all lines including the new calling party. If the conference audio is choppy, all participants should disconnect and switch to the lesser bandwidth G.723 codec, by disabling other audio compression codecs. The number of participants is normally limited by the upload bandwidth at host, see section 5.1.

Please see the chart below for procedures to add, remove and manage a conference:

ACTION	PROCEDURE	COMMENTS
Add a call to Conference – by dialing new party	Place conference on HOLD by selecting available LINE. Dial party and advise. Click CONFERENCE BUTTON –all on HOLD Then Click CONFERENCE button to join all	Conference parties will not be able to hear one another during HOLD periods. Dial out call is private and conf parties cannot hear conversation.
Add a call to Conference – new party called in	Place conference on HOLD by selecting available calling LINE. Advise calling party. Click CONFERENCE BUTTON –all on HOLD Then Click CONFERENCE button to join all	Conference parties will not be able to hear one another during HOLD periods. Incoming call is private and conf parties cannot hear conversation.
Remove call from Conference	Place conference on HOLD by Clicking CONFERENCE BUTTON select LINE of party for removal. Advise party. Click HANGUP, Then Click CONFERENCE button to join remaining parties back in Conference	Conference parties will not be able to hear one another during HOLD periods. Selecting single line makes call private and conf parties cannot hear conversation
Conference Party Hangs Up and all LINES began Flashing	Click CONFERENCE button to join remaining parties back in Conference	Sometimes if the last party to join the conference hangs up it will place the entire conference on HOLD.
Conference Party advises you he is Hanging Up	Place conference on HOLD by Clicking CONFERENCE BUTTON, When LINE clears, Click CONFERENCE button to join remaining parties back in Conference	Sometimes if the last party to join the conference hangs up it will place the entire conference on HOLD.
Add more parties to conference but no available LINES	Identify existing X-PRO conference participant with sufficient bandwidth, and request the additional parties be added utilizing that participant as a second host.	Multiple /conference Host nodes are permitted, with little, if any effect on conference quality.
Add Pocket PC as a Conference Node	The Pocket PC may act as a Conference Node limited only by the bandwidth to the device. The G.711 codec is used on Pocket PC, so, verified throughput is imperative.	If 802.11b is used empirical test indicate that hosting be limited to 3 person conferences
Host looses conference participant(s), while performing a procedure	It is important the soft buttons, HOLD function and Conference initiation be given 3 seconds to process. Rapid button action will cause inadvertent disconnections.	SIP protocol messages allow the hold, conference and associated functions. Time must be allow for all conference client to respond.

The Conference Host may identify the participants of the conference by clicking with the mouse (or stylus [Pocket PC]) on the respective Line button and the Caller ID will show the Display Name of the associated conference participant.

4.5 Automatic Answer

X-PRO supports Automatic Call Answer. The screen shot below shows the activated Auto-Answer function.

Auto-Answer

The Auto-Answer icon is an “AA” displayed at the lower left side of the Call Display Screen. The “AA” icon shows a blinking red status when auto-answer is activated to alert the user of the fact that all incoming calls will be automatically answered. The icon is gray when de-activated.

X-PRO, by default, waits 6 seconds before auto-answering however this parameter is adjustable by way of the Menu/System Settings.

The Auto Answer feature is useful in numerous situations but is most powerful when coupled with the Auto-Conference feature outlined in section 4.6.

4.6 Automatic Conferencing

X-PRO supports Automatic Call Conferencing. The end point functioning as the Conference Host has control of how the participants will be admitted to the conference.

Auto-Answer

Auto-Conference

The Conference Host has four levels of control:

Control profile	Conference Process
Manual conference	Conference Host manually accepts calls and adds each to conference.
Auto-answer, manual add to conference	**X-PRO** automatically answers calls and the host manually adds each to conference
Individual answer, auto add to conference	Host answers each call and each is automatically added to conference
Conference Server Mode (auto-answer/auto-conference)	**X-PRO** acts as a fully automated conference serve

Calls may be selectively answered, selectively added to the conference, or the Conference Host may automate as much or as little of the teleconferencing process as desired. When the teleconference process is fully automated by setting **X-PRO** to both auto-answer and auto-conference, **X-PRO** functions as a standalone, unattended, conference server, and it can run on either a local machine or a remote machine.

A conference may include up to 7 individuals (the Conference Host plus a participant on each of the 6 lines, limited only by the upload bandwidth available to the conference host machine. Using more than one conference node can further expand the number of participants. (e.g. more than one machine acting as a conference server, in conference with other **X-PRO's**).

5 Advanced Features

The following features are meant for advanced users only. You need to be familiar with the details of the SIP protocol, audio compression codecs, and other technical details in order to use them correctly.

5.1 Audio Compression Codecs

To select an audio compression codec, click on the codec format at the bottom of the call display window. A particular codec format is enabled if it is darkened in the call display window. Disabled codec formats will appear grayed-out in the call display window. Individual codecs can also be enabled or disabled by selecting **Advanced System Settings** -> **Codec Settings** in the menu system.

Attention! Please be advised you can severely degrade **X-Lite/X-PRO's** performance with improper audio compression codec settings.

When two VoIP systems are establishing a call, they negotiate an audio compression codec they are going to use. Which audio compression codec to choose depends on many factors: which audio compression codecs are installed on both systems, bandwidth limitations, desired sound quality, etc. During the negotiation, **X-Lite/X-PRO** offers to the remote system the first audio compression codec from the list. If the remote system rejects the audio compression codec, **X-Lite/X-PRO** offers the next one from top to bottom until they both accept the audio compression codec.

The current **X-lite /X-PRO** versions support 6 different audio compression codecs:

G.711A and G.711U provide good sound quality but low voice compression. Both audio compression codecs require 64 Kbps bandwidth and may be recommended for wide bandwidth networks, such as LANs.

G.723 and GSM provide poorer sound quality but good voice compression. G.723 requires 6.4 Kbps and GSM requires 13 Kbps. These audio compression codecs may be recommended if network bandwidth is limited, for example for long-distance calls or for dial-up connections down to 28.8 Kbps. *Speex [SPX]* and *iLBC* are also provided.

The current Pocket PC 2000/2002 versions of **X-Lite/X-PRO** support only 5 audio compression codecs: G.711A and G.711U codecs, plus the GSM, iLBC, and SPX codecs. This is primarily to avoid the more CPU intensive high compression audio codecs.

5.2 Multiple SIP Server Configuration (X-PRO only)

Sometimes you may have several SIP proxy servers you would like to have your SIP phone registered with, including providers with PSTN gateways. Each SIP proxy server has users and clients, which you would like to contact and allow those users to be able to contact you without the process of manual reconfiguration.

This **multiple SIP proxy server** feature is employed in both **X-PRO** and **X-Lite**. You also need to specify each SIP proxy server profile you would like to use under an associated Proxy ID. The Proxy ID is given as a number from 1 to 9. To specify a given SIP proxy to be used to make or receive calls, follow **System Settings** -> **SIP Proxy** in the menu system. **X-PRO** allows up to 10 different SIP proxies to be configured.

For each SIP proxy server, under the Proxy ID (Proxy 1, Proxy 2, etc) you must configure the complete information to register **X-PRO** on that server as outlined in the **System Settings** section of this manual. The Default Proxy does not require you to specify a proxy ID to make an outgoing call over its network.

To place a Call using a SIP Server other than the default server, at the **Dial Pad,** enter the pound sign followed by the Proxy ID (i.e. #1, #2, #3 etc.), followed by the number. Similarly, you can right-mouse-click over the call display window for a list of the proxies you have already configured. Select the proxy you would like to use by left-mouse-clicking over it in the list. Finally enter the SIP URL you are calling and click the **Call** button, or press **Enter**. Please verify the proper phone number is being dialed on the **Caller ID** display line, which shows the actual number and domain being dialed, including country and area codes.

For Example, if Deltathree.com is configured as SIP Proxy 3, then to call the PSTN number 1-604-555-1212 using the Deltathree SIP server, dial #316045551212 and click Dial. The Caller ID line would read "Call ID: 16045551212@deltathree.com". Always confirm the number being dialed utilizing this data.

Using this **multiple SIP proxy server** feature in **X-PRO** and **X-Lite** most calls will now involve a server/registrar prefix followed by the SIP number and can be dialed directly from the Dial Pad or your keyboard requiring fewer characters.

For example, let's demonstrate this feature to dial the SIP URI examples mentioned earlier:

SIP URL Dial examples:

•A host computer: joe.mynet.com (192.168.0.32)
Type in the **Dial Pad:** sip:joe.mynet.com or sip:192.168.0.32

•A user johndoe at iptel.org Type in the **Dial Pad:** sip:johndoe@iptel.org

•A phone number1-604-555-1212 through a SIP gateway deltathree.com Type in the **Dial Pad:**
sip:16045551212@deltathree.com

First assume, the default SIP server is *Xten.net*, SIP Server #1 is *iptel.org* , SIP Server #2 is *Deltathree.com* and SIP Server #4 is *mynet.com.* Now using **multiple SIP proxy server** to dial the same Examples we have:

•A host computer: joe.mynet.com (192.168.0.32)
Type in the **Dial Pad:** #4joe.mynet.com or #4192.168.0.32
To verify note that the Call ID line would read "Call ID: joe@mynet.com or 192.168.0.32 @mynet.com

•A user johndoe at iptel.org Type in the **Dial Pad:** #1johndoe@iptel.org
To verify note that the Call ID line would read "Call ID: johndoe@iptel.org.

•A phone number1-604-555-1212 through a SIP gateway deltathree.com Type in the **Dial Pad:**
#216045551212@deltathree.com

To verify note that the Call ID line would read "Call ID: 16045551212@deltathree.com"

5.2.1 Configuring for Direct IP Dialing

If you would like to directly dial to another IP address without traversing a SIP Proxy, **X-Lite/X-PRO** allow for this option. To configure Direct IP Dialing, you must go to the SIP Proxy configuration panel by following **System Settings** -> **SIP Proxy** in the menu system.

Configuring for Direct IP Dialing is similar to configuring an actual SIP Proxy except that you only have to specify the following fields:

- ✓ **Enabled**: choose "Yes"
- ✓ **Display Name**: your calling display name
- ✓ **Domain/Realm**: the network your client resides upon
- ✓ **Direct Dial IP**: choose "Yes"

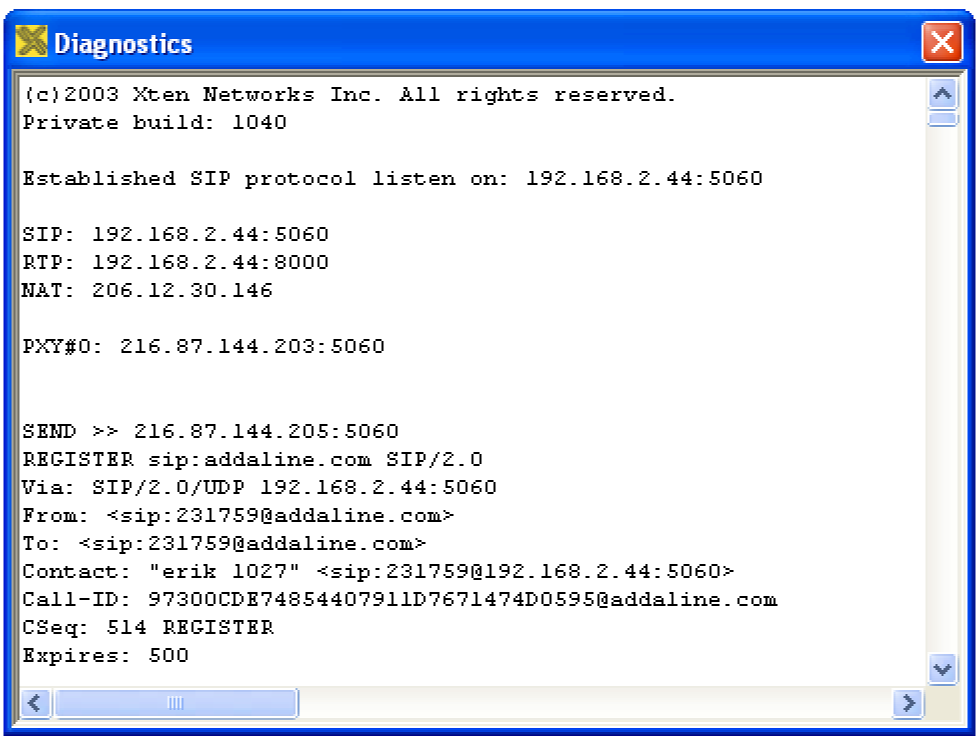
To make a call, simply specify the Proxy ID which you configured for Direct IP dialing and then specify the IP address you would like to call. For example, you may specify #2192.168.79.10 as the number to call IP 192.168.79.10 where Proxy 2 was configured for Direct IP dialing.

5.3 Diagnostics Screen

Xten Diagnostic Window:

Xten Networks Software Version and Build Number: This shows the operating software version information.

SIP: This line identifies the SIP Control Port being used by **X-Lite/X-PRO**
RTP: Port for **X-Lite/X-PRO** UDP protocol communications
NAT: IP address of local NAT Firewall being used in SIP messages
PXY#0: This line identifies the IP address and port of the SIP proxy; #0 indicates that the default proxy (Proxy ID 0) is being used.



5.4 SIP User Agent Ports

X-Lite and X-PRO uses the following ports:

Port	Protocol	Description
5060	TCP	SIP User Agent Control
5060	UDP	SIP User Agent Control
8000	UDP	RTP multimedia streaming
8001	UDP	RTCP multimedia streaming
8000-8020 (multi-call recommendation)	UDP	RTP /RTCP

Please be advised that the above ports are the default ports utilized. For each additional simultaneous line in operation the **X-Lite/X-PRO** opens an additional 2 ports. (e.g. first line in use uses ports 8000 and 8001, second line in use employs line 8002, and 8003, etc). Hence, for conferencing operations and multiple lines in use a small range of ports should be opened. For additional **X-Lite/X-PRO** systems behind a firewall unique ports must be opened for each.

6.0 Windows Hot Keys

The following section gives a list of short cut key-strokes available for **X-PRO** and **X-Lite** for Windows users.

- ✓ **Line 1** - F1 - Ctrl + 1
- ✓ **Line 2** - F2 - Ctrl + 2
- ✓ **Line 3** - F3 - Ctrl + 3
- ✓ **Line 4** - F4 - Ctrl + 4
- ✓ **Line 5** - F5 - Ctrl + 5
- ✓ **Line 6** - F6 - Ctrl + 6
- ✓ **Mute** - F11 - Ctrl + u
- ✓ **Default SIP Proxy Settings** - F8 – Ctrl + 8
- ✓ **Diagnostics window** - F9 – Ctrl + 9
- ✓ **Calibrate audio device** - F7 – Ctrl + 7
- ✓ **Push-to-talk** - F12 – Ctrl + k (*PPC Users)*
- ✓ **Transfer** – CTRL + t
- ✓ **Conference** - CTRL + f
- ✓ **Address** - CTRL + b
- ✓ **Menu** – CTRL + m
- ✓ **Dial** - Enter
- ✓ **Hang-up** - CTRL + h
- ✓ **Help** - CTRL + F1
- ✓ **Minimize** - F23
- ✓ **Restore** - CTRL + F23
- ✓ **Clear** - Backspace
- ✓ Auto complete number to dial - Tab
- ✓ **DND** - CTRL + d
- ✓ **Auto Answer** - CTRL + a
- ✓ **Auto conference**- CTRL + f
- ✓ **Ignore** - CTRL + i
- ✓ **Send to voice mail** - CTRL + s

7.0 Mac OS X Users

Other than a few exceptions, the versions of **X-Lite** and **X-PRO** for Mac OS X should behave exactly as those for Windows. The following differences, however, do exist for installs of the software on Mac OS X.

1. For Windows users, the **Mute** function is toggled by the Ctrl + U combination. For Mac users the Mute function is toggled by the space bar.

2. For Mac users running under OS X 10.3 "Panther", the F9 key, normally used for showing and hiding the Diagnostic Window, is used by the Exposé system service. The hot key to show the Diagnostic Window in this case is Command-D.

3. The "WindowShade X" system extension from Unsanity, which collapses windows to their title bars when minimized, does not work well with X-PRO's windows, as they have no title bars. If you are running this system extension, we recommend that you add X-PRO to the "Exclude List" found in WindowShade X's System Preferences configuration pane.

8.0 Troubleshooting

This section describes some common remedies in case you experience difficulties while using **X-Lite/X-PRO**.

8.1 Sound Quality Issues

There are several potential factors that can affect sound quality that are not controlled by **X-Lite/X-PRO**. These include the quality of the network connection, LAN/WAN card quality, headset model, and the distance between **X-Lite/X-PRO** and a wireless access point.

Sound has pops at loud points in voice

Popping sound is a symptom of a problem with over modulation. Ask the party to decrease the microphone slider and/or adjust the microphone gain.

Sound has bothersome background noise

Significant background noises and sound is a symptom of a problem with the setting of the silence threshold. Have the party click on the microphone icon. Both a silence threshold slider (triangular icon to the right of the microphone meter) and meters for the microphone volume and the speaker volume will appear. The silence threshold should be set approximately 10% above the ambient noise as shown on the microphone meter.

Conference has choppy audio

Poor conference audio is frequently caused by upload bandwidth limitations at the host, not download bandwidth. For example, some sites have 1 Meg download, and 128K upload bandwidth. In these conditions, try multiple host sites for the conference or have all participants disconnect and switch to a lesser bandwidth audio compression codec such as G.723.

Poor Sound

Poor or scratchy sound is a symptom of a problem with your network connection, network performance, or network card and access point drivers.

- If you're using a wireless internet connection, you may be out of range of an access point.
- Radio Frequency (RF) interference originating from multiple access points on similar channels or other devices nearby can cause sound quality problems. To prevent this, you or your network administrator must configure access points so they are separated by at least five channels. Be careful when setting up Xten SIP User Agent in lab areas where there tends to be higher RF interference.
- There may be packet loss on the IP network. Check the network performance, latency, and throughput. A 200 millisecond round trip latency as determined from a ping should deliver good quality audio.
- Network cards or access points may have performance problems. Make sure you are using the latest Drivers

There is an Echo.

- Try using a headset with your sound card. Another suggestion is the use a USB-enabled headset which bypasses system sounds because it acts as a sound device. However, if you have network problems (such as an access point that is being used at or beyond its capacity), installing a USB-enabled headset will not help, and you or your network administrator needs to troubleshoot the network.
- Are you using a boom type headset? It has been noted that some boom headsets that sit on the ear can create feedback between the boom microphone and the headset.
- You may be too close in distance to the person you're speaking to. If you're too close you will hear both the voice being delivered from their microphone, and your microphone picking up the person's voice in the same room or down a hallway. Together, it sounds like an echo.
- The microphone balance is not muted in the playback section of Windows audio controls. When you are configuring sound, you may need to examine the volume control setting.

- You, as the caller will hear echo if your microphone can pick up your speaker sounds, such as when not using a headset.
- Microphone level may be too high or amplifier is turned on in the generic Windows audio controls.

On the **Advanced Controls** dialog box, there is an optional setting to increase the sensitivity of the microphone. Using this setting (generically “Enable microphone booster”) can set the microphone level too high.

8.2 Network Connection Issues

There are several potential factors that can affect getting online that are not controlled by **X-Lite/X-PRO**. However, **X-Lite/X-PRO** provides some features to help you troubleshoot the network connection when calls do not connect or when you’re experiencing general networking problems. This section describes both the tools that **X-Lite/X-PRO** provides to help you or your network administrator troubleshoot and some possibilities you can check, grouped in the following categories: access points, IP addresses, firewalls, battery power.

Login Messages and Problems: Observing *Login* messages in the **Login** Status **Display** can often provide clues to troubleshoot the connection problem you are experiencing.

Login message	Explanation	Solution
Discovering Firewall…	**X-PRO** is discovering the type of firewall that it is behind	Allow X-PRO to discover the firewall type which will be needed to auto-configure some network settings.
Connecting X-Tunnels…	**X-PRO** is connecting to the X-Tunnels server	If status does not change, X-Tunnels server may be down or your login ID may not be valid.
Logging in…	**X-Lite/X-PRO** is attempting to register with SIP Proxy Server(s)	Wait to verify success. If multiple servers all are in the process
"Logged in - Enter Phone Number"	**X-Lite/X-PRO** has successfully registered with ALL configured SIP servers	**X-Lite/X-PRO** is ready
"Login timed out! Contact Network Admin."	One or more SIP registrations failed	If only one registrar, verify settings and network screen. Then verify proper connection thru firewall ports to internet. Uncheck autodiscovery in network settings.
"Enter Phone Number"	**X-Lite/X-PRO** has successfully registered with ALL configured SIP servers	If test calls fails, exit **X-Lite/X-PRO** and restart to verify proper login status
"Call failed: 480 Temporarily unavailable"	Dialed SIP number is not on-line at this time	Try a test call to a known working number to verify operability
"Trying" (long period)	SIP Call being established by SIP message protocol	Manually configure network setting dialog screen; autodetect has failed to detect proper settings
"Connected"	Called party has answered and is ready to converse.	Talk to party

Working with Access Points

- Is your system associated to the access point? Check your 802.1X card driver to determine signal strength and the name of the SSID.
- If you are set for dynamic IP address and not connected to an access point, the network driver may automatically generate an address of 169.x.x.x, which is a default IP address. Make sure you are associated to an access point and that your computer's IP is being routed.
- Check to see if you are connected to an access point and that it is the correct SSID. Verify signal strength and connection speed.

Working with IP Addresses

- If you don't have an IP address that works on the network, then make sure you have selected DHCP or picked the right static address configuration for your network (including netmask, gateway and DNS).
- If you are not connecting, but are seeing "Login timed out! Contact network administrator", check the IP address assigned. If it is in the 169.x.x.x range, you are probably not connected. Check to see if you are connected to the correct access point SSID. On Windows XP systems, look in the **Tools** menu, **Networking**, **Profiles** tab to see if Xten SIP User Agent has specific instructions to correct your problem.
- Check the status page to determine if the IP address in use makes sense.

Firewalls

- If you are behind a NAT firewall, make sure that you are using a NAT traversal server or that ports 5060, 8000 to 8020 are open for TCP and UDP. In addition, verify that the SIP proxy server being used supports NAT Firewall SIP messages.
- If you experience one-way audio or no audio upon connection it is likely the UDP ports are not open or your Out Bound Proxy is not set properly.

System Power

- If you're running a portable personal computer on low batteries, you may see intermittent connectivity to the access point as the card power decreases. To correct this, begin recharging the battery.
- On newer Windows systems, you may have the option to control the amount of power that goes to different devices with power profiles/schemes. For example it is possible to set up a power profile that turns off the power for your wireless LAN card after a period of time, thus causing connectivity problems. Check this in the **Windows Control Panel**, access **Power Options**.

Other Tips

- Test network connectivity by using a web browser or by some other means, such as the ping command.

  If you don't detect IP connectivity, check for access point signal strength. Try reinserting the NIC (network interface card) if possible, or switching on and off.