MyIC setup and configuration
(with sample configuration for Alcatel Lucent test environment)

N.B. Goto MyIC Preferences in the System Toolbar.

Description:
this may be any appropriate description of the account

Enabled:
Tick the box 'Enabled' to activate the account.

Server address:
Is the address or domain name of the SIP server.

User:
This is the SIP username.

Password:

ACS address:
username + '@' + address of the ACS server.

Display name:
Username.

Authorization name:
Username.

External Line Prefix:
The number needed to access an outside line for internal SIP server's and/or PBXes.

Audio Codec:
It is recommended to use the default option of 'Choose Automatically' but the user may also select a codec depending on performance.

Voice Mail number:
Enter the number of the mailbox.

Preferences	
Server	Server
	Description: Stuttgart Showroom Enabled: ☑
Connection	Server address: 149.204.165.80
Audio	Username: 4277
GUI	Password: ●●●● Show
Privacy	ACS address: xnet5@149.204.165.94 Demo: ☐
Advanced	Display name: xnet5
	Authorization name: xnet5
	External line prefix:
	Audio codec: Choose automatically
	Voice mail number:
	Cancel OK

Listen RTP Port:
Use the default value of 10500 unless otherwise specified by the provider.

SIP expired time:
Default time is 3600 seconds but users may set another value if required.

STUN Server:
Allows NAT clients to setup phone calls to a VOIP provider hosted outside the local network.
There are many free STUN servers which may be found by a simple internet search.

SIP Proxy:
Basically the SIP Outbound Proxy is used to handle the signalling and media traffic across a firewall.
It is recommended not to use STUN and Outbound proxy together.

Protocol SIP or SIP-TLS:
For added security SIP-TLS may be chosen if it is supported.

Input: Select the device for incoming audio.
**Ring Tone:** Select your own ring tones.
Output: Select the device to outgoing audio.
Alternate Output: Select another device for incoming audio.
Usage Example: In the Call Window, this option may be selected to switch between the headset and the built-in speaker.
Ring Output: The device through which the ring tone will be played.
Acoustic Echo Cancellation: may be sued if the recipient hears an ECHO (Note: reducing the output volume may also help).
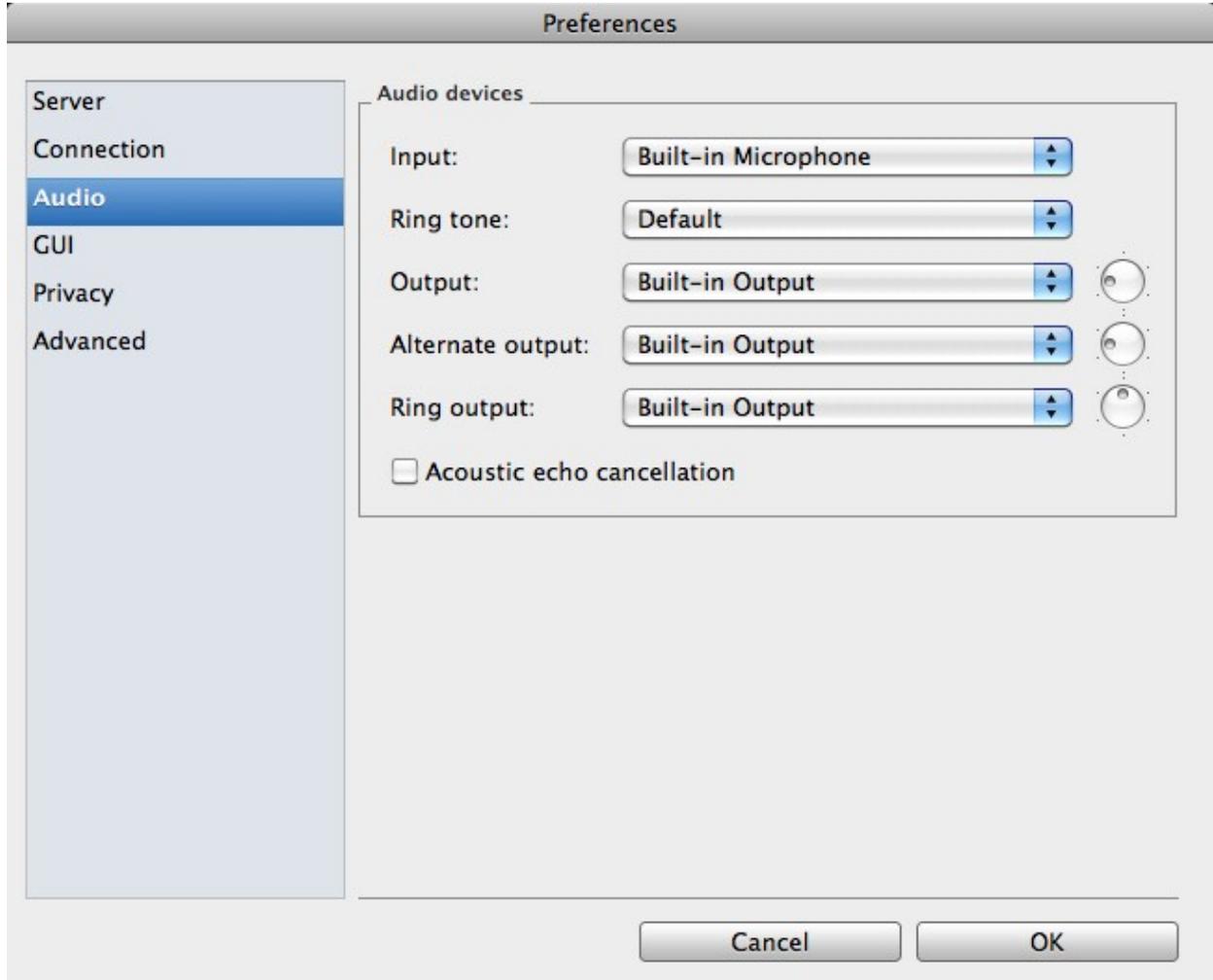
Dial Prefix Conversion:
Convert the leading '+' to '00' by selecting the option below.

Send DTMF:
two methods are supported, may be selected depending on the server config.

Play Dial Tone:
option to turn the dial tone on/off.
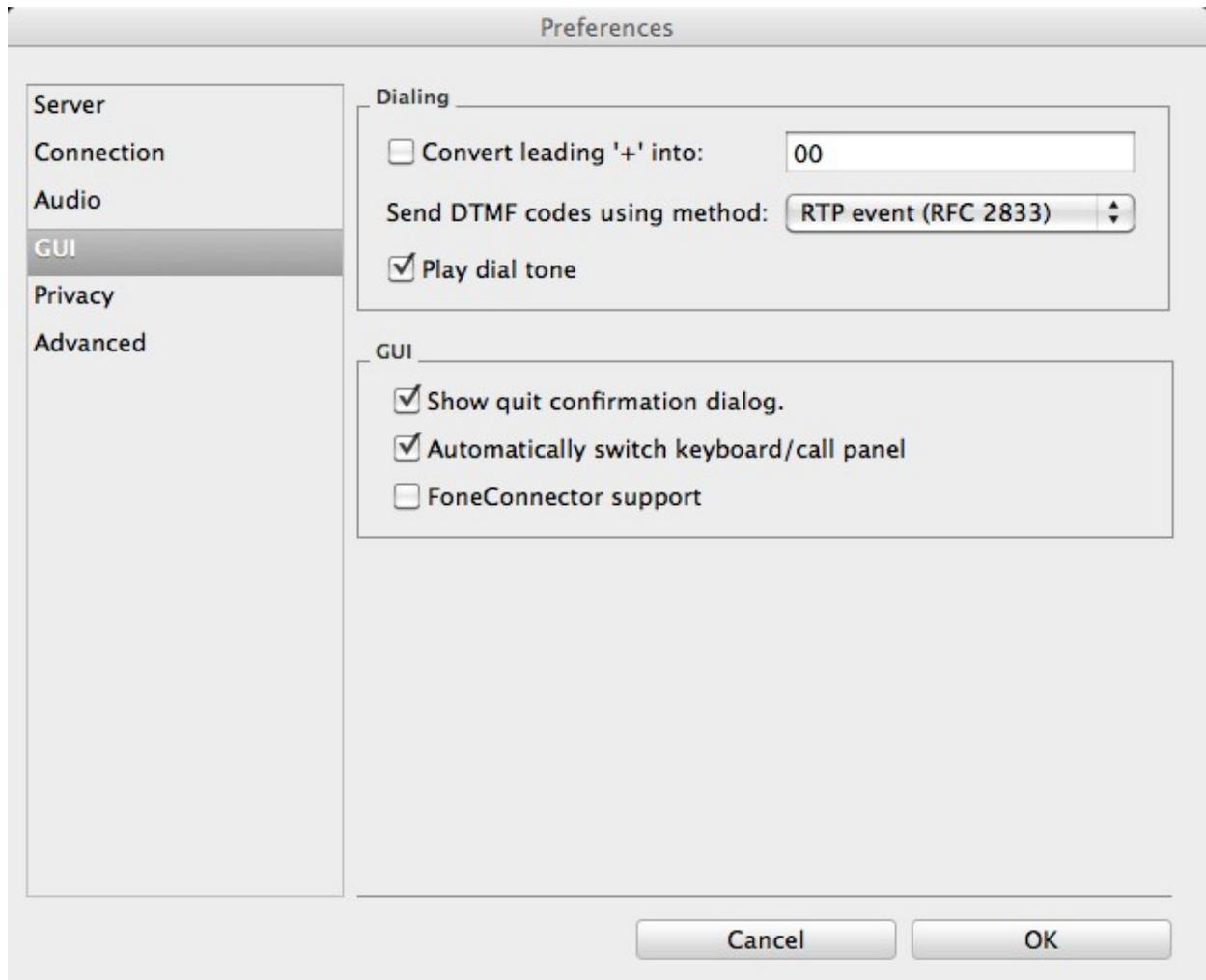
Show Quit confirmation:
Option to confirm application closure to prevent accidentally closing the application.

Automatically switch keypad/call panel:
if selected the Call Panel will automatically appear during a call.

FoneConnector Support:
Allows iSoftPhone to automatically provide Daylite CRM data on demand with the FoneConnector plugin.

Blocking Online Status 'Presence':
iSoftPhone can send information about your online status to other users. iSoftPhone provides the option this allow or block users from seeing when you are online.

Allow all to see my Online Status:

Option Selected:
With this option selected, you shall not be prompted when another contact adds you to their address book.
All users may see when you are online.

Option Unselected:
With this option unselected you shall be informed every time another contact adds you to his/her address book.
You are also provided with the option to add them to your address book and allow them to view your online status.

Blocking Calls and Messages:
There is also the option to block new or old contacts from communicating with you.
Users who are added to the Blocked list may not contact you. To allow a contact to resume contact with simply remove them from the blocked list.

Address Book:
Contacts who are online and available to be contacted will be highlighted in the address book.

Blocking an existing Contact:
A user that has been approved may be blocked at any time in the future.
To do this open Contacts, right click on the user and select the option to 'Block' them.
The user is now blocked from:
1. Viewing your online status
2. Calling you
3. Messaging you
Note: You may allow blocked contacts to communicate with you at any time by simply removing them from the list of blocked users.
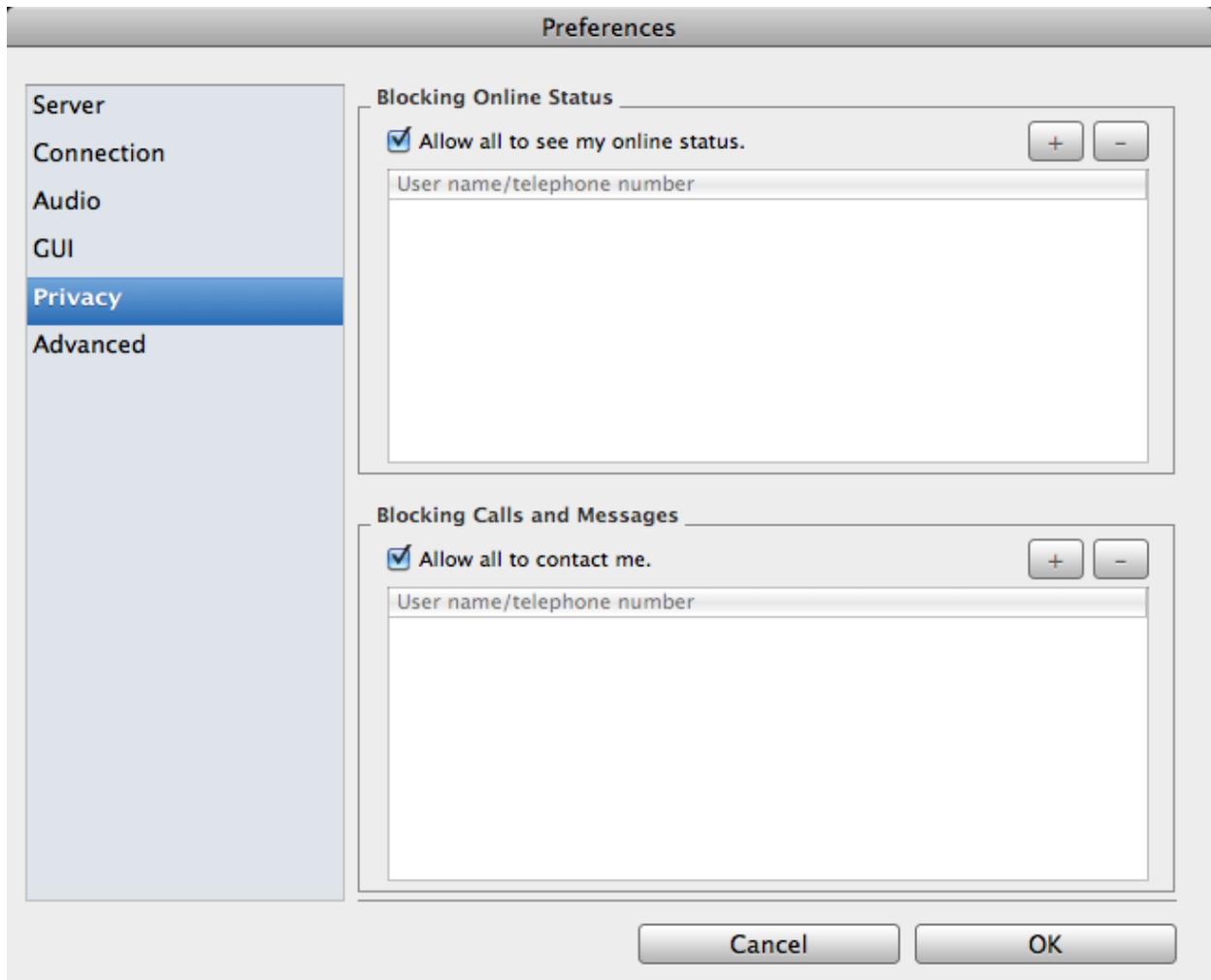
Local SIP Port (Listen SIP Port):
The default port for SIP is 5060, users may set an alternate port number when using multiple SIP devices or use the value '0' to choose a random port.

Send 101 Dialog Establishment Message:
Some SIP servers do not accept a Dialog Establishment message. In such a case turn this option off. iSoftPhone restart is needed to apply the change.

Use DNS for every message:
Turn on this option when you can register to SIP server but are not able to call any number.

Send SIP keep-alives:
Sends keep-alive packets to keep connection alive (needed for some routers).

Use rport:
Requests that the server send the response back to the source IP address and port from which the request originated, instead of the port written into the topmost 'Via' header field value of the request.

Send PUBLISH:
Sends PUBLISH request as an additional method for Presence.

Confirm Callback request:
If the Callback is implemented in a 3rd part application or website, iSoftPhone will notify of all callback requests.

SRTP Voice encryption:
SIP Support for SRTP - Secure Real-Time Transfer protocol (SRTP).
Ensures the integrity of RTP and Real-Time Control Protocol (RTCP) packets providing authentication, integrity, and encryption of media packets between two SIP endpoints.