

iSoftPhone 1.3 and Swyx IP PABX

Compatibility Tests

Ver. 1.1 (2007-08-30)



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1. Introduction

These Application Notes describe the configuration steps for using the iSoftPhone 1.3 with Swyx PABX. General administration information can be found in the product documentation. Products WEB sites:

- iSoftPhone – <http://www.call4mac.com>
- SwyxWare – <http://swyx.com>

2. Configuration

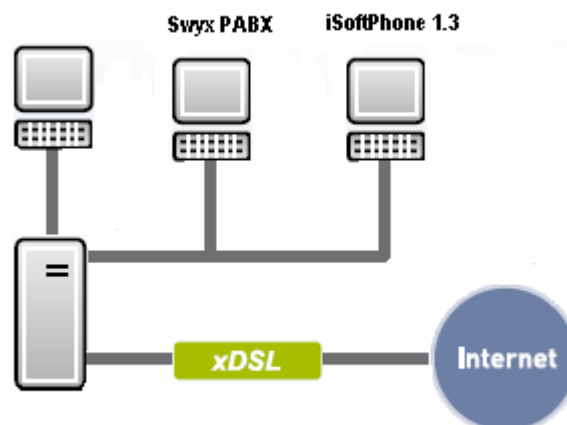
Two PC's:

- Swyx PABX host
- iSoftPhone 1.3 host

are connected to the same local network.

iSoftPhone 1.3 running on Apple operating system is configured to register as internal subscriber on the Swyx PABX.

The configuration used in the test is shown at the picture below.





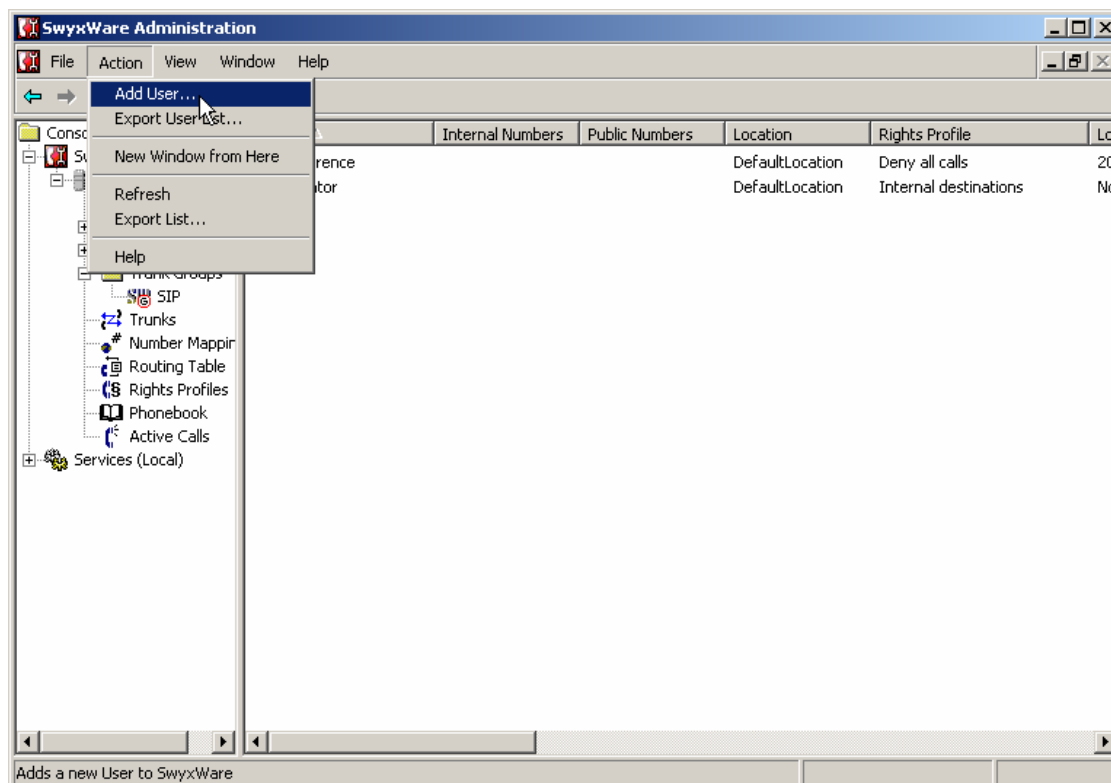
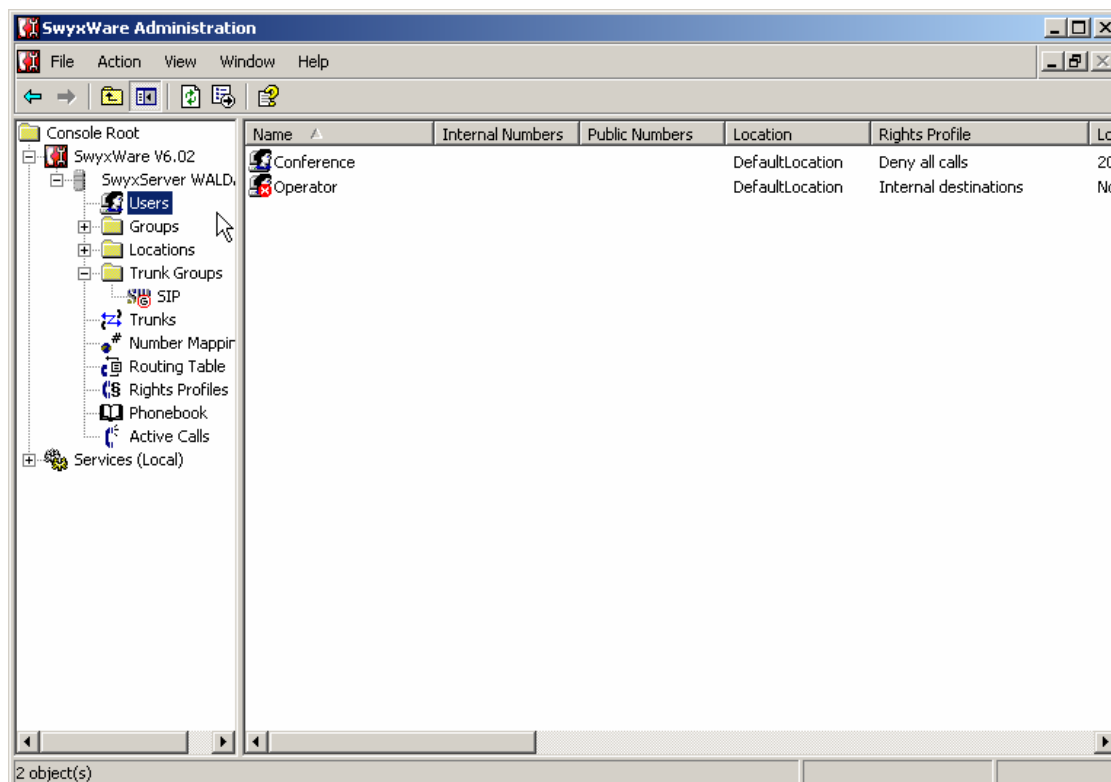
3. Equipment and Software

The following equipment and software were used in the configuration.

EQUIPMENT	SOFTWARE
SwyxWare Demo	6.02
Swyx Server PC	Win XP Pro SP2
iSoftPhone	1.3
Apple Macintosh	MacOS X 10.4.9

4. Creating a new internal VoIP subscriber in the SwyxWare Administration

To add a new internal subscriber in the SwyxWare Administration please open the “Users” tab and select “Add user...” from “Action” menu.





In the new window follow the steps shown in the pictures below.

Add new User [Close]

Name and type of the new User
Enter name and type of the new User.

An unambiguous name for the new User is required. The description is optional.

In case you would like to connect a fax, it can be defined under 'User type'.

Name:

Description:

Type:

< Back Next > Cancel

Add new User [Close]

Location of the new User
Please select a Location for the new User.

A Location within SwyxWare defines all location specific settings like the time zone, the required public access code, the country and area codes.

Please select one of the listed Locations which will be assigned to this User.

Location:

Description:

< Back Next > Cancel



Add new User [X]

Internal Number of the new User
Enter the Internal Number, under which the new User will be reachable.

To define a Internal Number for this User, enter the chosen number and click "Verify" for checking if it is already in use. By entering a number and clicking "Next unused" the system will suggest the next free number after the given.

Uncheck "Show in Phonebook" if you e.g. want to use the Internal Number for call routing purposes only.

New Internal Number:

Show in Phonebook

Add new User [X]

Internal Number mapping
Specify the Public Number to which the User's Internal Number will be mapped.

To permit calling this User directly from the public network, you have to associate the Internal Number to a Public Number.

To do so, choose one of the suggested Public Numbers from the drop-down list, or enter a Public Number (canonical format) or SIP URI manually.

Use the "Select..." button to obtain an overview of Public Numbers available in the entire system.

Internal Number: 98

Associated Public Number:



Add new User [X]

Terminals
Choose which terminals are used.

A User can make phone calls using different terminals. Check the terminals to be used by the new User. The required settings will be configured in the following dialogs.

- SwyxIt!
- SwyxPhone
- SIP compatible terminal
- H.323 compatible terminal
- Simple User account for call routing. No logins allowed.

< Back **Next >** Cancel

Add new User [X]

Windows user account
Enter the Windows user account for the new user.

For working with SwyxIt! the SwyxWare User needs an assigned Windows user account. If the account belongs to a Windows domain, enter '<Domain>\<Name>'.

Windows User Account:

< Back **Next >** Cancel



Add new User [X]

PIN for SwyxPhone
Enter the PIN and the confirmation.

For using SwyxPhone a PIN is required. Click on 'Create PIN' for assigning a new, unique PIN to the User.

Please inform the User about the created PIN.

You can change the PIN later on the User's 'Administration' property page.

SwyxPhone PIN:

< Back

Add new User [X]

Specify SIP parameters
Configure the UserID to enable SIP login and choose the authentication mode.

To logon via SIP it's necessary to specify a unique User ID for each User.
In case authentication is enabled you must enter a username and a password, too.

User ID:

Authentication Mode:

User Name:

Password:

< Back

'User Name' and 'User ID' should be the same to avoid registration (logon) problems.



Add new User [X]

Alias name for H.323 terminal
Enter the Alias name for the H.323 compatible terminal.

For H.323 compatible terminals a so called H.323 Alias name is required.
Enter this name and click 'Next'.

H.323 Alias Name:

< Back Next > Cancel

Add new User [X]

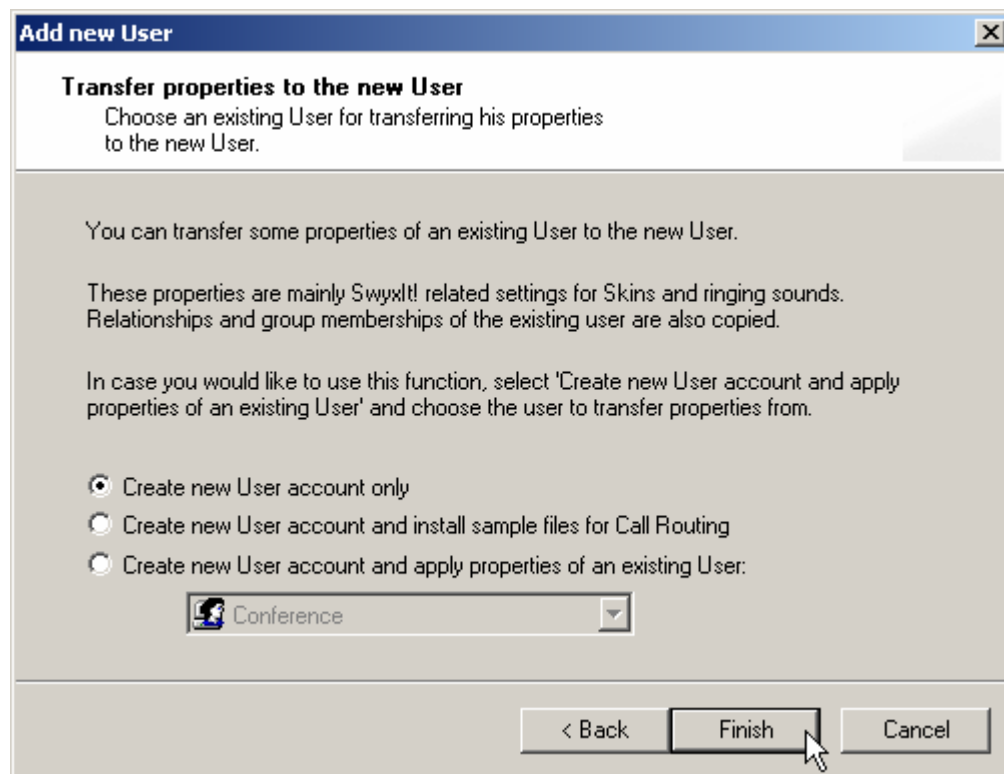
Rights Profile
Choose Rights Profile.

Rights Profiles represent individual call permissions or restrictions which can be assigned to a User.
Please select one of the listed Rights Profiles to define the call permissions of the User.

Rights Profile:

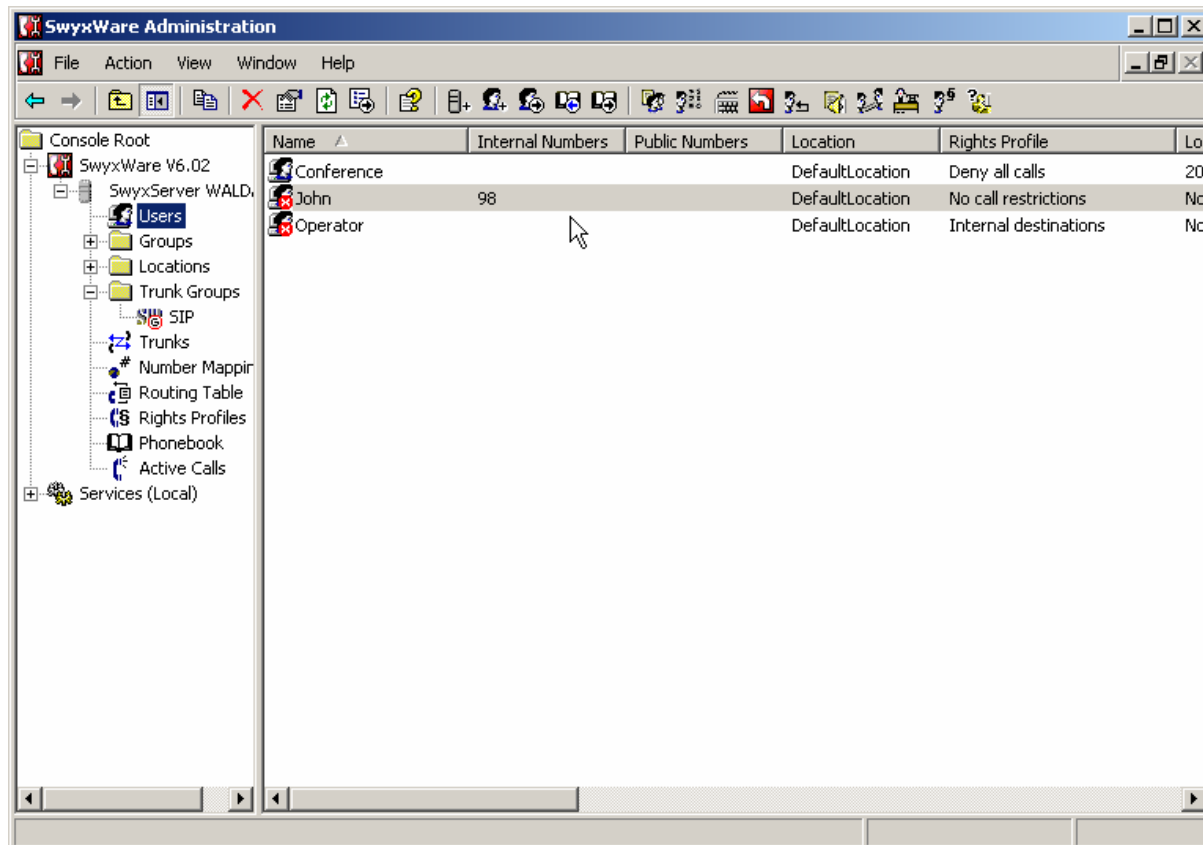
Description

< Back Next > Cancel





When you click “Finish”, new user (John, internal number 98) appears in the “Users” tab.






5. Configuration on iSoftPhone 1.3

Once iSoftPhone 1.3 is installed then SIP server settings must be configured.

5.1 Configuring SIP server account

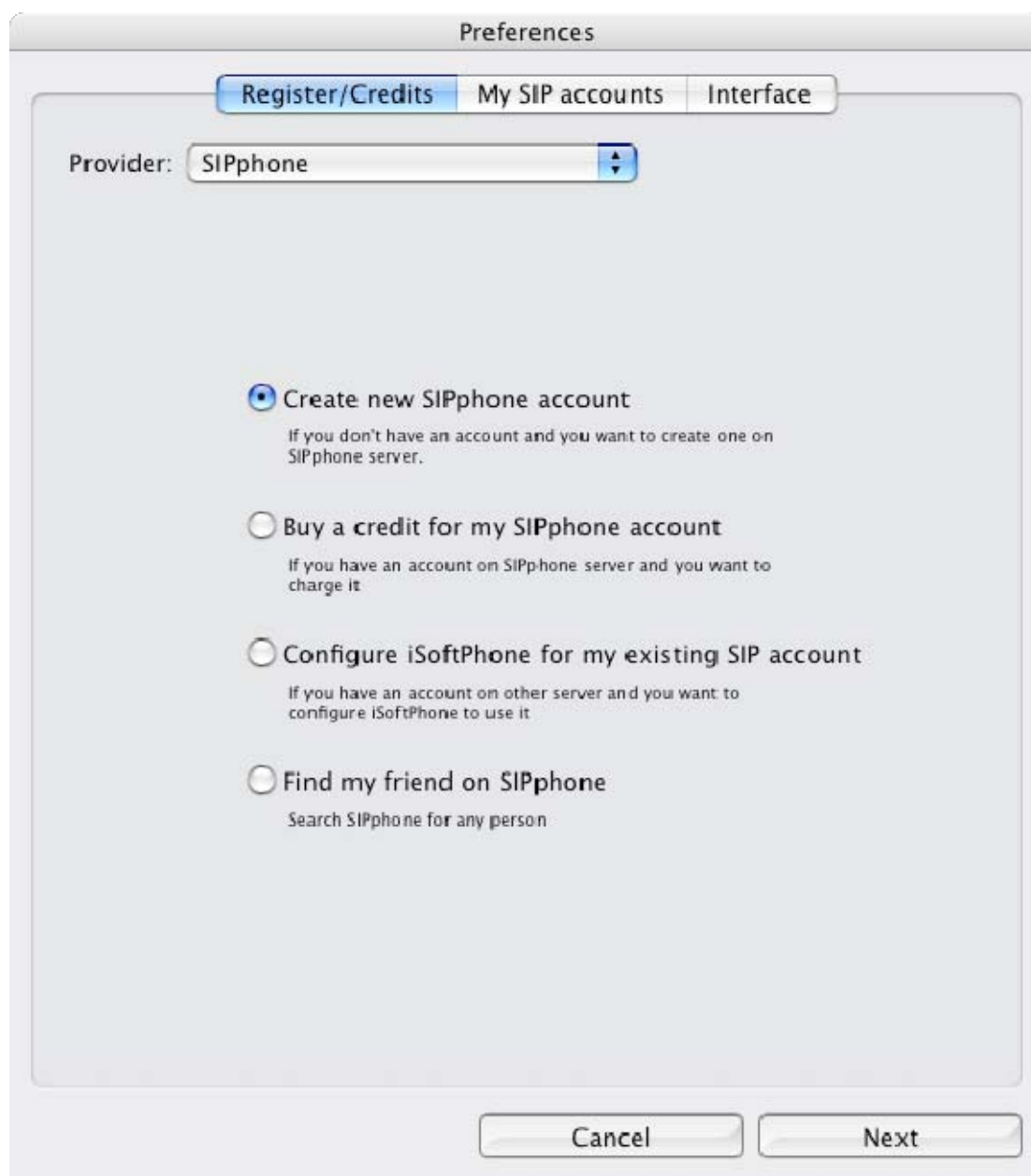
To configure SIP server account please press the icon  from the bottom toolbar of iSoftPhone.





5.2 Preferences “Register/Credits”

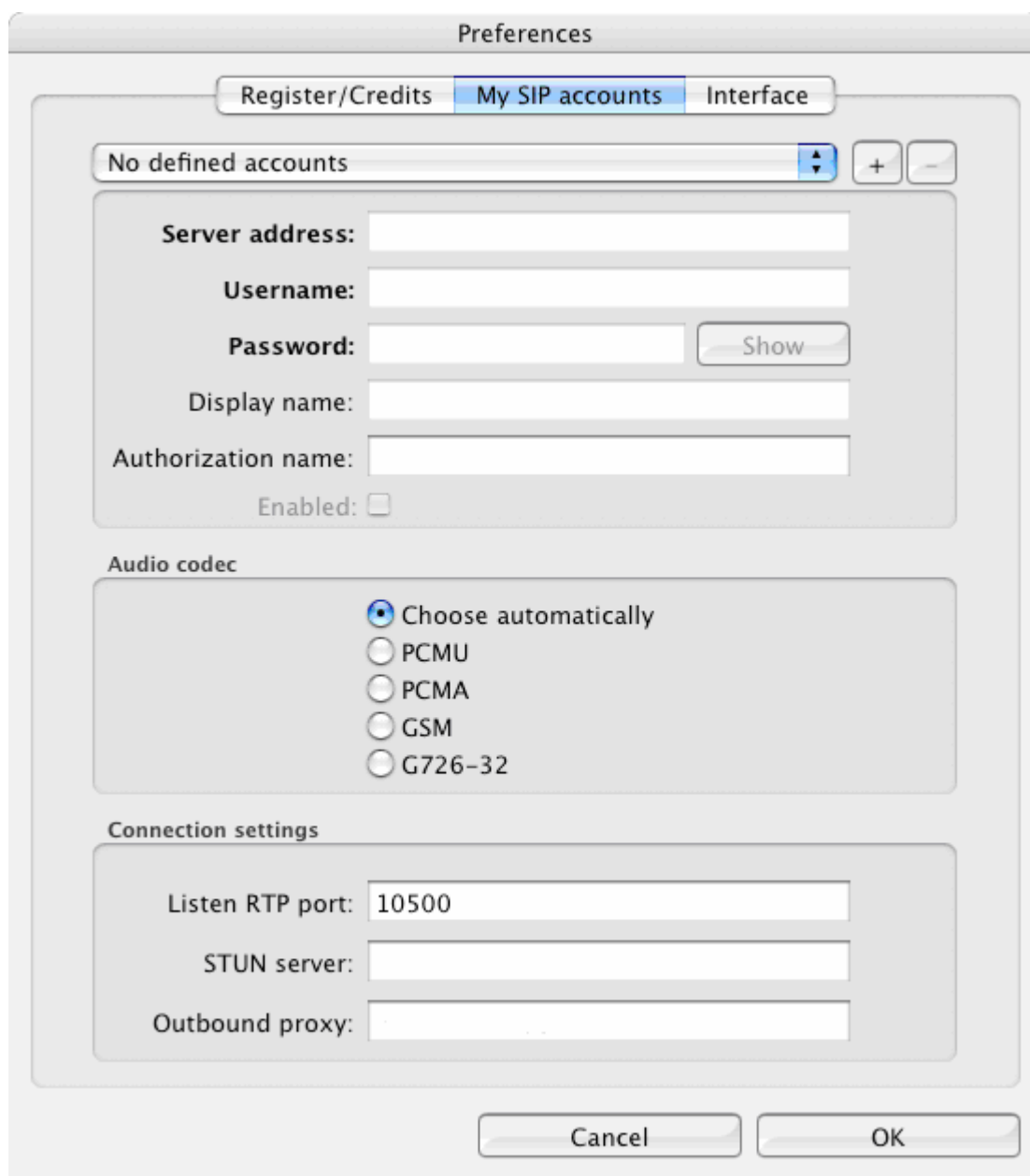
In the new appeared “Preferences” window please select “My SIP accounts” tab.






5.3 Preferences “My SIP accounts”

The empty “My SIP accounts” tab was shown below.





To configure SIP account that was created on elmeg ICT PABX please press the icon  and fill up parameters as was shown below.

Preferences

Register/Credits My SIP accounts Other

John@192.168.200.27 + -

Server address: 192.168.200.27

Username: John

Password: Show

Display name: John

Authorization name: John

Enabled:

Audio codec

Choose Automatically Automatic choice

PCMU

PCMA

GSM

G726-32

iLBC

The codec will be negotiated with a SIP server.

Connection settings

Listen RTP port: 10500

STUN server:

Outbound proxy:

Cancel OK

After all please confirm settings by pressing "OK" button.

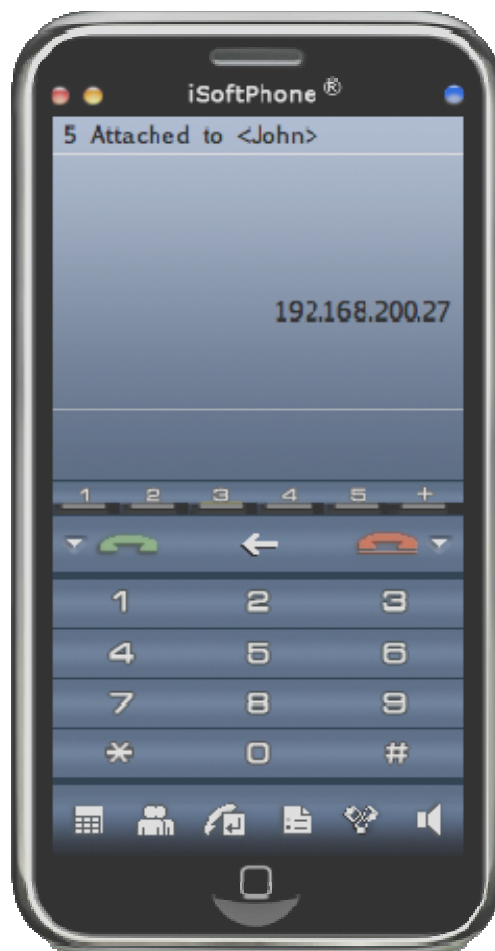


5.4 Registration on SIP server account

Once SIP account settings were configured then iSoftPhone will try to connect and register on Swyx PABX. The process was shown below.



When all parameters were configured properly and iSoftPhone successfully registered on Swyx then application window will look as shown below.

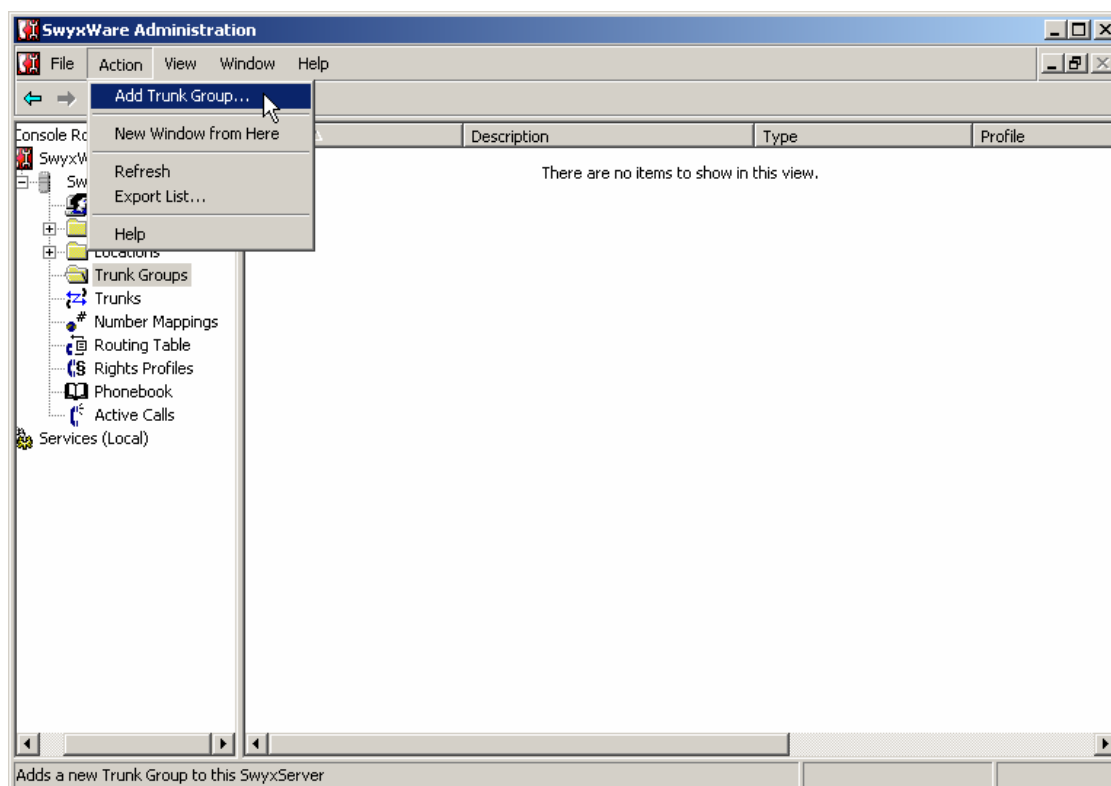
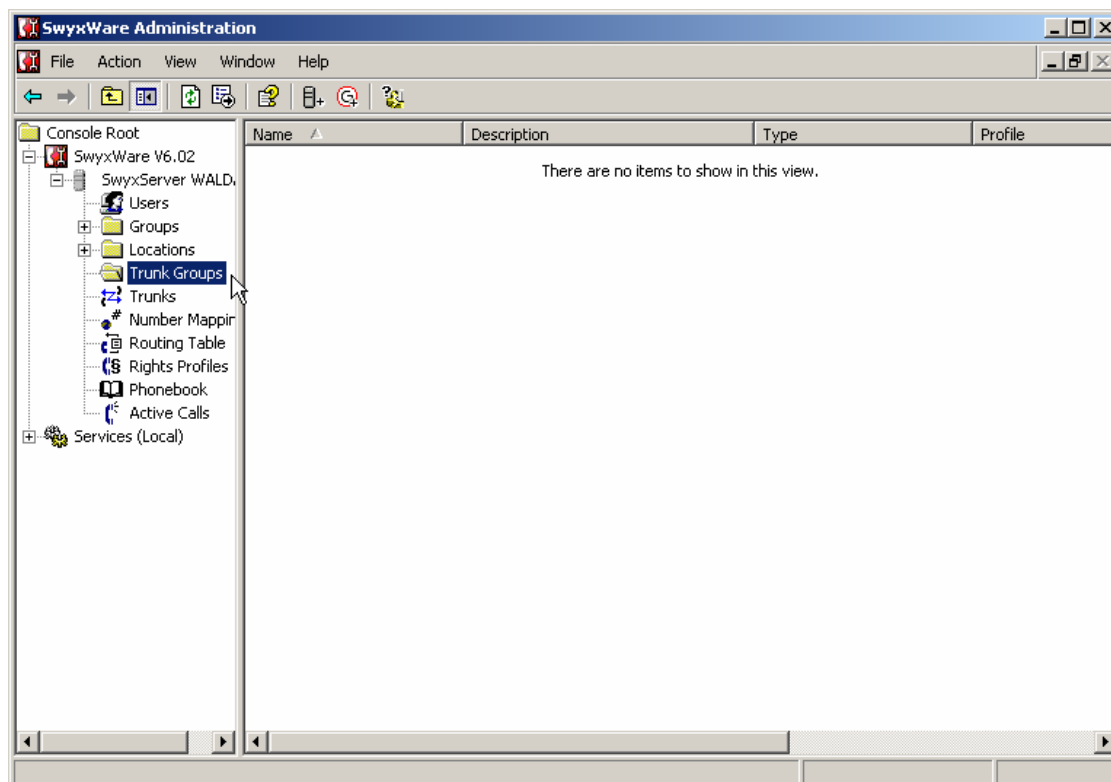


Congratulations! You have successfully configured iSoftPhone with Swyx PABX.

6. Configuration of the SIP account at SIP Provider (for external calls)

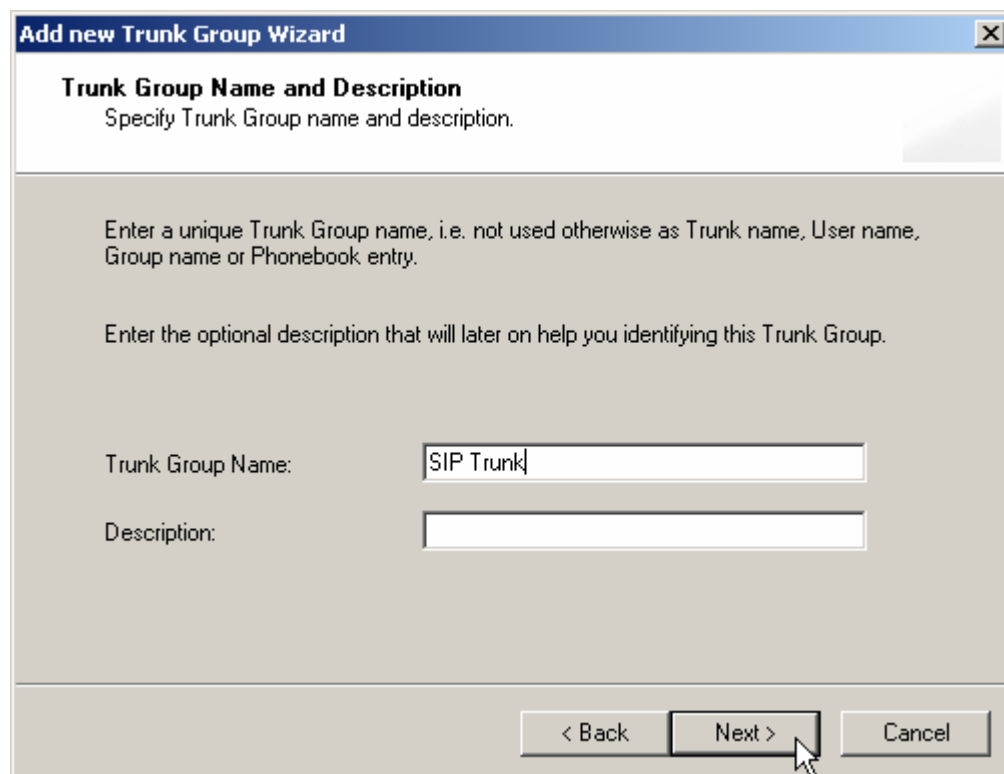
6.1 Creating a new SIP Trunk group

To add a new Trunk group in the SwyxWare Administration please open the “Trunk Groups” tab and select “Add Trunk Group...” from “Action” menu.





In the new window follow the steps shown in the pictures below.





Add new Trunk Group Wizard [X]

Trunk Group Type
Specify the type of the Trunk Group and select the appropriate profile.

Select the Type of Trunk Group to be added from the first list and choose the applicable profile from the second list. If you are uncertain, which profile is applicable for your installation, consult the SwyxWare Administration documentation.

If you want to add a Trunk Group for a non-listed SIP service provider, select the Profile "Custom". This will allow entering all required parameters in subsequent steps.

Trunk Group Type:

Profile:

< Back Next > Cancel

You can choose predefined SIP Provider from the list or create your own.

Add new Trunk Group Wizard [X]

SIP settings
Please specify whether SIP registration is enabled for this Trunk Group.

The subsequently prompted information must have been supplied by your SIP service provider.

If your service provider requires a SIP registration (usual case), enable the checkmark and enter the registrar's name or IP address and port.

The SIP account specific information must be entered when you add a Trunk to the Trunk Group you are currently creating.

Enable SIP registration

Registrar: Port:

Re-registration interval: seconds

< Back Next > Cancel



Add new Trunk Group Wizard [X]

SIP Settings
Specify SIP settings for this Trunk Group.

The SIP Proxy is the service provider's interface for call control. Therefore its name or IP address and port (usually 5060) must have been provided.

The SIP realm is part of the SIP addressing mechanism, i.e. it is used for SIP URI composition. The parameter "DTMF Mode" determines how a user's keypad input is passed to the provider.

Proxy: Port:

Realm:

DTMF Mode:

< Back Next > Cancel

Add new Trunk Group Wizard [X]

STUN Server Settings
Specify STUN Server Settings.

A STUN server can be used to traverse non-symmetric NAT firewalls, in order to access another SIP proxy. The STUN server must be located in the public Internet.

Please enter the name or IP address of the STUN server and the STUN service port (usually 3478). A publicly available STUN server is e.g. "stunserver.org".

Enable STUN support

STUN Server: Port:

< Back Next > Cancel



Add new Trunk Group Wizard [X]

Definition of Routing
Specify for what calls this Trunk Group is supposed to be used.

Depending on your choice, initial Routing Records will be created. Public Numbers should be added in canonical format (e.g. "+4930123456"), ""*"" can be used as a wildcard.

Use Trunks of this Trunk Group...

- for all external calls
- for all external calls to the following Called Party Number or SIP URI only:
- for all external calls and all unassigned Internal Numbers
- for Internal Numbers:

< Back Next > Cancel

Add new Trunk Group Wizard [X]

Rights Profile
Select a Rights Profile to be used for this Trunk Group.

The Right Profile of a Trunk Group determines, where incoming calls from this Trunk Group are allowed to be routed to.

Please select one of the listed Right Profiles which will be assigned to this Trunk Group.

Rights Profile:

Description:

< Back Next > Cancel



Add new Trunk Group Wizard [X]

Location Profile
Select the applicable Location Profile for this Trunk Group.

A Location within SwyxWare defines all location specific settings like the time zone, the required public access code, the country and area codes.

Please select one of the listed Locations which will be assigned to this Trunk Group.

Location:

Description:

< Back Next > Cancel

Add new Trunk Group Wizard [X]

You have successfully completed the Add Trunk Group Wizard.

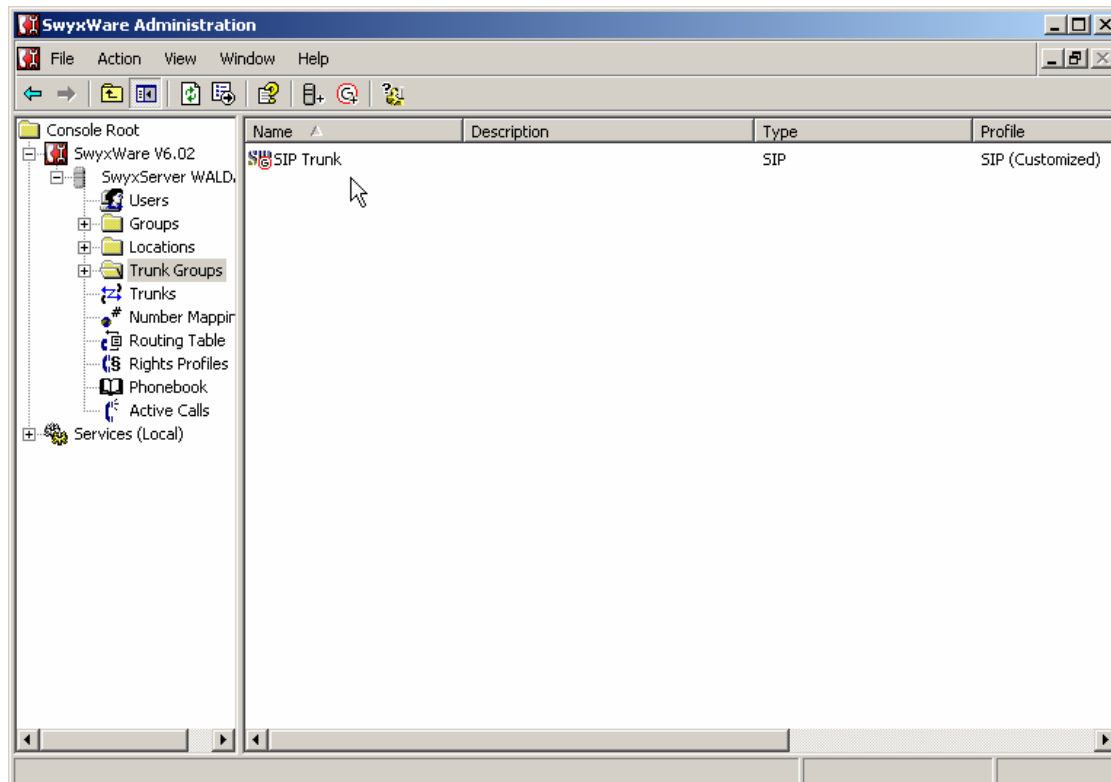


After finishing this Wizard you can assign Trunks to the created Trunk Group.

< Back Finish Cancel



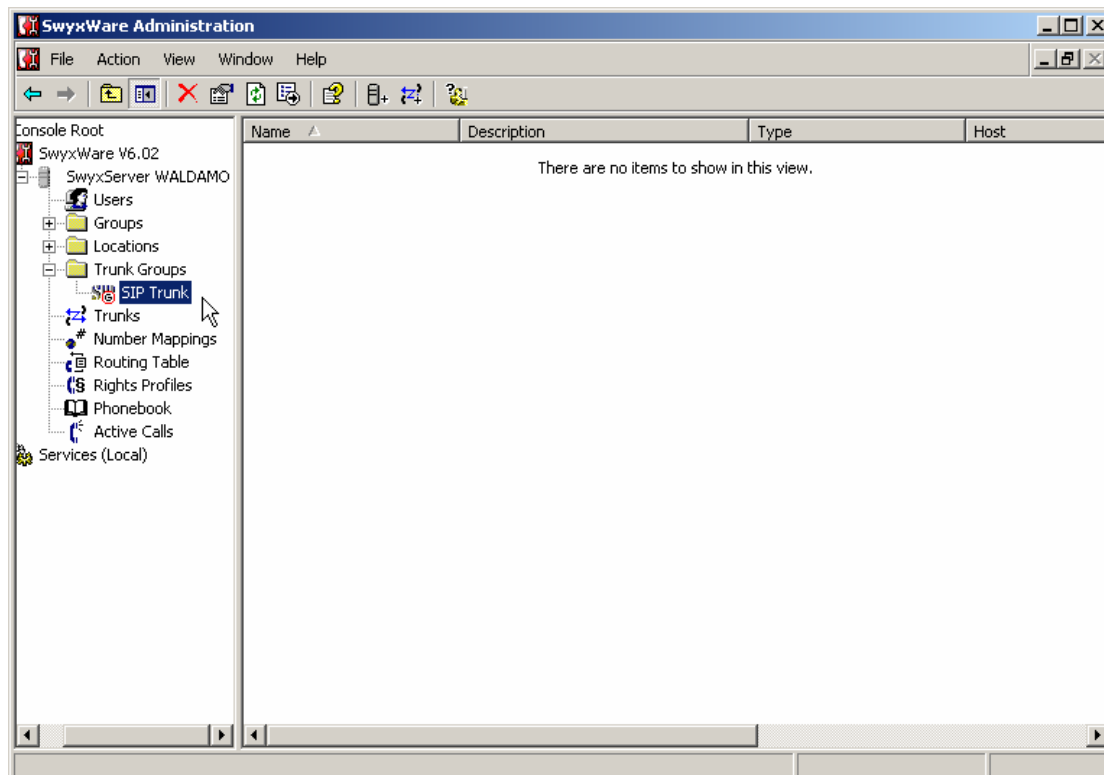
When you click “Finish”, new trunk group (SIP Trunk) appears in the “Trunk Groups” tab.

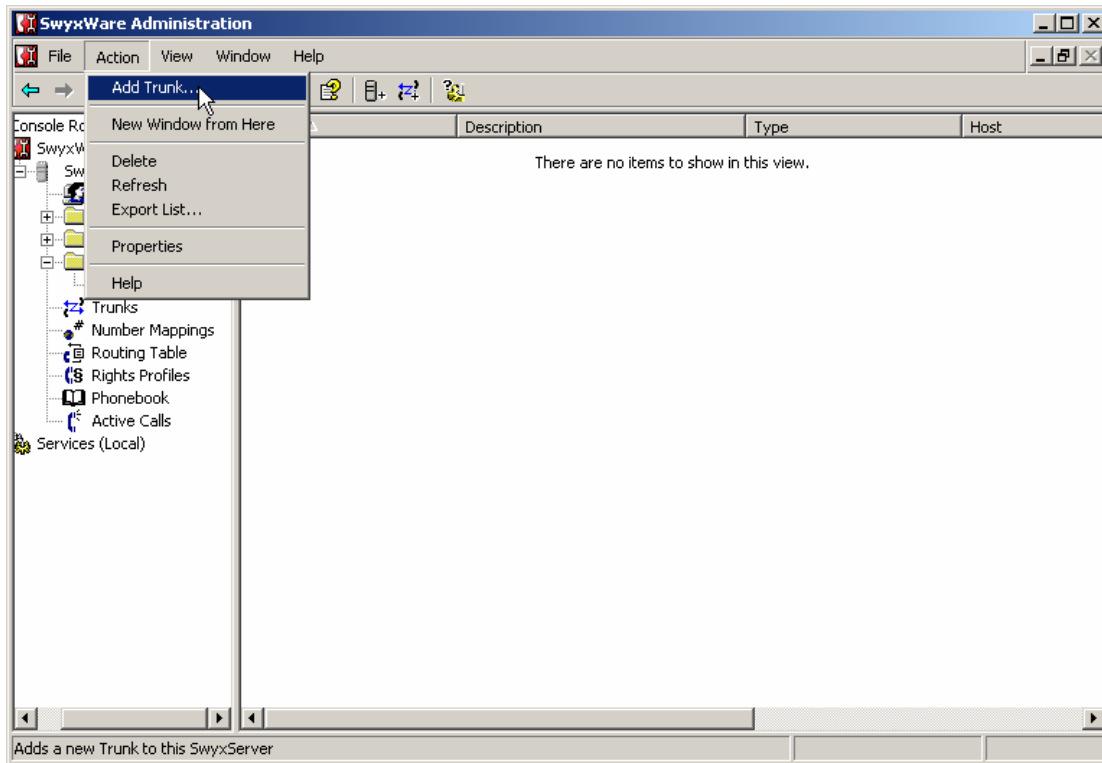




6.2 Adding new trunk

Now you need to configure the account provided to you by your SIP Provider (follow the steps shown in the pictures below).







Add new Trunk Wizard [X]

Trunk Name
Choose an unique name for the new Trunk.

Enter a unique Trunk name, i.e. not used otherwise as Trunk Group name, User name, Group name or Phonebook entry.

Enter the optional description that will later on help you identifying this Trunk.

Trunk Name:

Description:

< Back Next > Cancel

Add new Trunk Wizard [X]

Trunk Group Selection
Assign the new Trunk to an existing Trunk Group or create a new Trunk Group to which the new Trunk will be assigned.

The chosen Trunk Group determines the type of Trunk (ISDN/Analogue/SIP Gateway, SIP or ENUM Trunk, SwyxLink) and defines several common properties.

Furthermore Trunk Group settings specify if a Trunk is considered for routing outbound calls.

Trunk Group:

< Back Next > Cancel



Add new Trunk Wizard [X]

SIP Trunk Provider / User Data
Specify your account data.

Enter the user identification data as provided by your SIP service provider. The user ID will be used to compose your SIP address while user name and password will be used for authentication.

SIP Provider:

User ID:

User Name:

Password:

< Back **Next >** Cancel

Add new Trunk Wizard [X]

Subscriber Numbers
Specify Subscriber Numbers.

Enter the subscriber number part of the Public Numbers that are terminated by this Trunk.

If your set of subscriber numbers is incoherent enter only the first subscriber number and add the other subscriber numbers later via the Trunk's properties.

If this Trunk does not add any Public Numbers to the system, leave all fields empty and click 'Next'.

Note: Country Code and Area Code have been pre-determined by the Trunk Group's location.

Country Code	Area Code	First Subscriber Number	Last Subscriber Number
<input type="text" value="48"/>	<input type="text" value="61"/>	<input type="text" value="4545454"/>	<input type="text" value="4545459"/>

< Back **Next >** Cancel



Add new Trunk Wizard [X]

SIP URI
Specify SIP URI.

If this Trunk is supposed to handle non-numeric SIP URIs (e.g. assigned by your SIP service provider), please enter these below.

SIP URIs have the following format:
sip:<name1> @ <name2>
with <name1> reflecting the user's name and <name2> the realm.

For convenient input "*" can be used as wildcard so that "*@company.com" would address all users in the realm "company.com". The realm field shown below is pre-filled with the configured realm in the SIP properties but may be overwritten case by case.

URI: sip: @

< Back **Next >** Cancel

Add new Trunk Wizard [X]

Codecs
Select the codecs to be used for data transmission.

The selected codec defines the type of compression for calls using this Trunk. Therefore the selected codec has an impact on the used bandwidth and the quality of the call.

Codecs

- G.711 (approx. 84 kBit/s per call)
- G.729 (approx. 24 kBit/s per call)
- Fax over IP (T.38, approx. 20 kBit/s per call)

< Back **Next >** Cancel



Add new Trunk Wizard [X]

Number of Channels
Select number of Channels to be used by this Trunk.

The number of concurrent calls via a specific Trunk is usually limited by the Trunk's physics, the available bandwidth or by a provider limitation.

Furthermore the number of simultaneous calls can artificially be limited to reserve (e.g. ISDN) channels or bandwidth for other applications.

Usually ISDN BRI interfaces would allow to make up to 2 simultaneous calls, while ISDN PRI interfaces allow up to 30 calls.

Number of simultaneous calls on this Trunk:

< Back **Next >** Cancel

Add new Trunk Wizard [X]

Computer Name
Define the computer name where the Trunk is hosted.

The Trunk may be hosted on another computer than the SwyxServer. In this case, the computer name must be provided here, otherwise keep the proposed default.

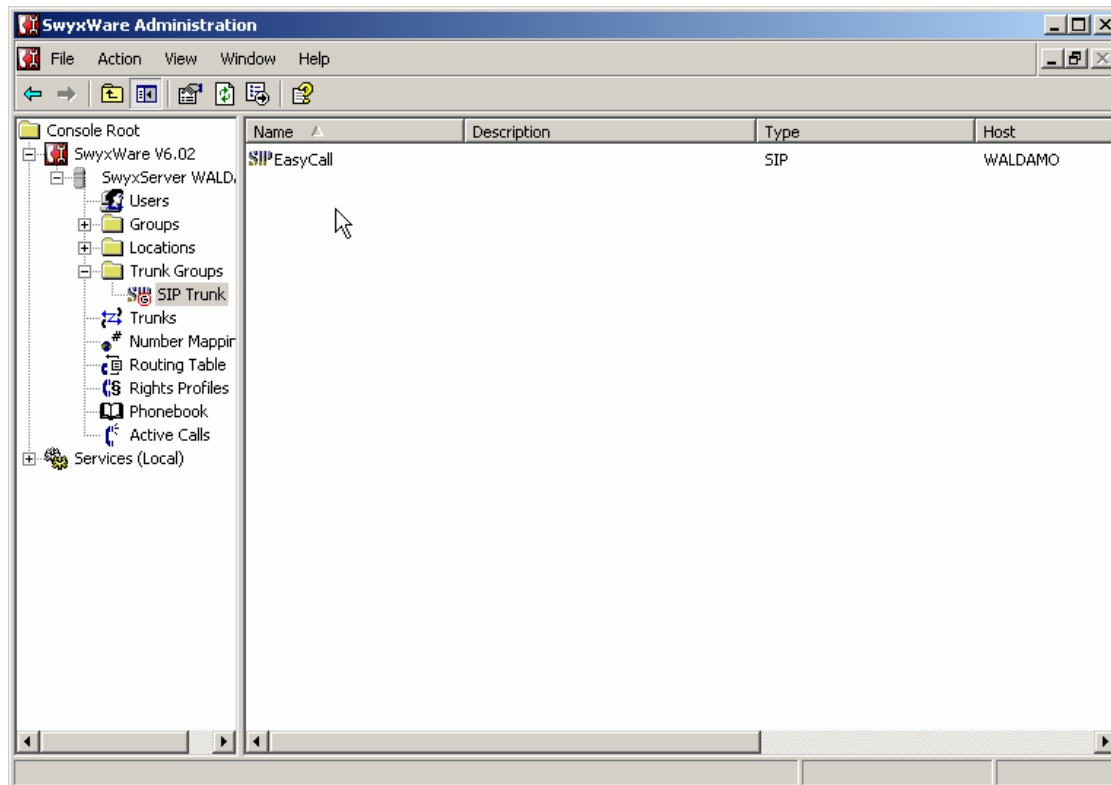
Please enter the computer name as it is given in the Windows Server's system properties.

Computer Name:

< Back **Finish** Cancel




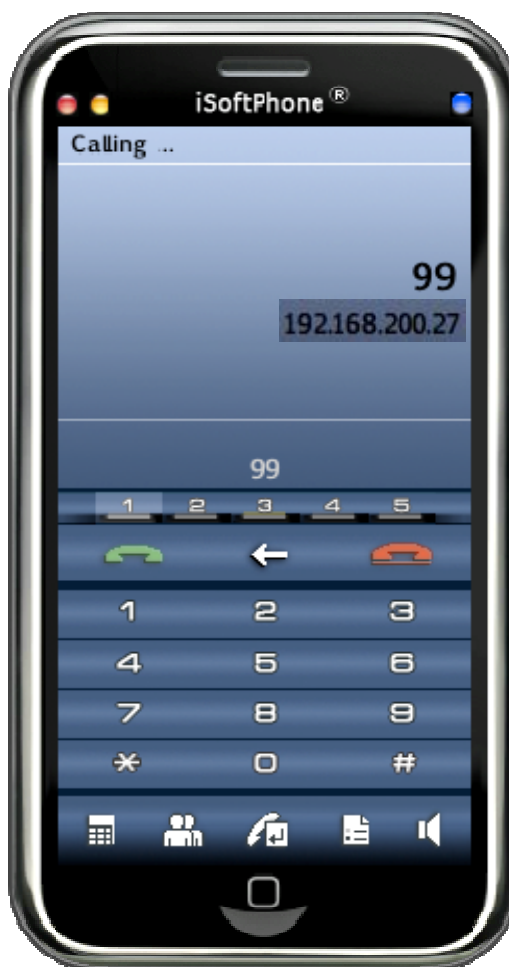
When you click “Finish”, new trunk (EasyCall) appears in the “Trunk Groups” tab.





7. Making calls


Making calls from iSoftPhone is easy! Just enter the number to dial and hit <ENTER> or press  button. Below was shown sample call process for dialed number 99.





Once the connection is established then iSoftPhone display's will look like shown below.



To finish the connection just hit <ESC> or press  button. Once the connection is finished then iSoftPhone display's will look like shown below.



8. Verification steps

The following steps can be used to verify and/or troubleshoot installations in the field.

1. Configure internal subscriber on Swyx PABX for the iSoftPhone. After configuring the iSoftPhone, verify that the “Logged in...” message appears in the upper left corner of the display, indicating that registration has occurred.
2. Verify that the extension shown in the brackets (in our example <John> is the desired value.
3. If “Not logged to server.” is displayed,



use the following to troubleshoot the problem: Check in Preferences “My SIP accounts” settings if the correct IP address of Swyx PABX host is entered. If “Connecting...” is displayed for a long time



please check that username and password is correct entered for this account in Preferences “My SIP accounts”.

4. Verify basic feature by making calls to other phones. If audio cannot be heard from an iSoftPhone, check the firewall/NAT settings to make sure that ports for VoIP communication are not blocked.

9. Support

For technical support of iSoftPhone please contact:

Internet: <http://www.call4mac.com/isoftphone/support.html>

E-mail: mac_support@xdsnet.de