



## **Avaya Solution & Interoperability Test Lab**

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# **Application Notes for Empix evolve Skypetophone with Avaya IP Office – Issue 1.0**

### **Abstract**

These Application Notes describe the compliance testing of Empix evolve Skypetophone with Avaya IP Office. Empix evolve Skypetophone is a Skype connectivity program which provides Skype access to local IP Office extensions, allowing them to make and receive calls from Skype endpoints.

Information in these Application Notes has been obtained through DevConnect compliance testing and additional technical discussions. Testing was conducted via the DevConnect Program at the Avaya Solution and Interoperability Test Lab.

## Table of Contents

1.	Introduction.....	3
1.1.	Interoperability Compliance Testing .....	3
1.2.	Support.....	3
2.	Reference Configuration .....	4
3.	Equipment and Software Validated .....	5
4.	Configure IP Office.....	6
4.1.	Licensing.....	7
4.2.	System.....	7
4.3.	Extensions .....	9
4.4.	Create Users .....	10
4.5.	Create SIP Line .....	11
4.6.	Create PRI Line.....	14
4.7.	Short Codes .....	16
4.8.	Incoming Call Routes .....	18
4.8.1.	PSTN Incoming Call Route .....	18
4.8.2.	SIP Trunk Incoming Call Route .....	19
5.	Configure Windows XP .....	21
6.	Configure Skype .....	26
7.	Configure Empix evolve Skypetophone .....	36
7.1.	Install Updates .....	37
7.2.	Install License .....	38
7.3.	Configure Call Routing.....	40
7.4.	Configure System Parameters.....	41
7.5.	Add Users.....	43
7.6.	Start Service .....	44
8.	General Test Approach and Test Results.....	47
9.	Verification Steps.....	47
10.	Conclusion .....	47
11.	Additional References.....	48

# 1. Introduction

Empix evolve Skypetophone is a member of the xtension **evolve** program suite. Empix evolve Skypetophone can be used alone, or together with the other xtension **evolve** components. Empix evolve Skypetophone is a PC-resident program which provides Skype access to local IP Office extensions, allowing them to make and receive calls from Skype client endpoints.

## 1.1. Interoperability Compliance Testing

The following tests were performed as part of the compliance testing:

- Incoming call from Skype client to local extension for basic call, call to busy device, and no answer.
- Outgoing call to Skype client from local extension for basic call, call to busy device, and no answer.
- Hold / Retrieve for Skype call.
- Transfer Skype call via local extension to local, PSTN, and Skype endpoint.
- Conference with local extensions and Skype endpoint.
- Call diversion from local extension to Skype endpoint for calls from local extension, PSTN endpoint, and a second Skype endpoint.
- Simultaneous calls from multiple local extensions to Skype endpoints.
- Recovery from power failure.
- Recovery network interruptions.

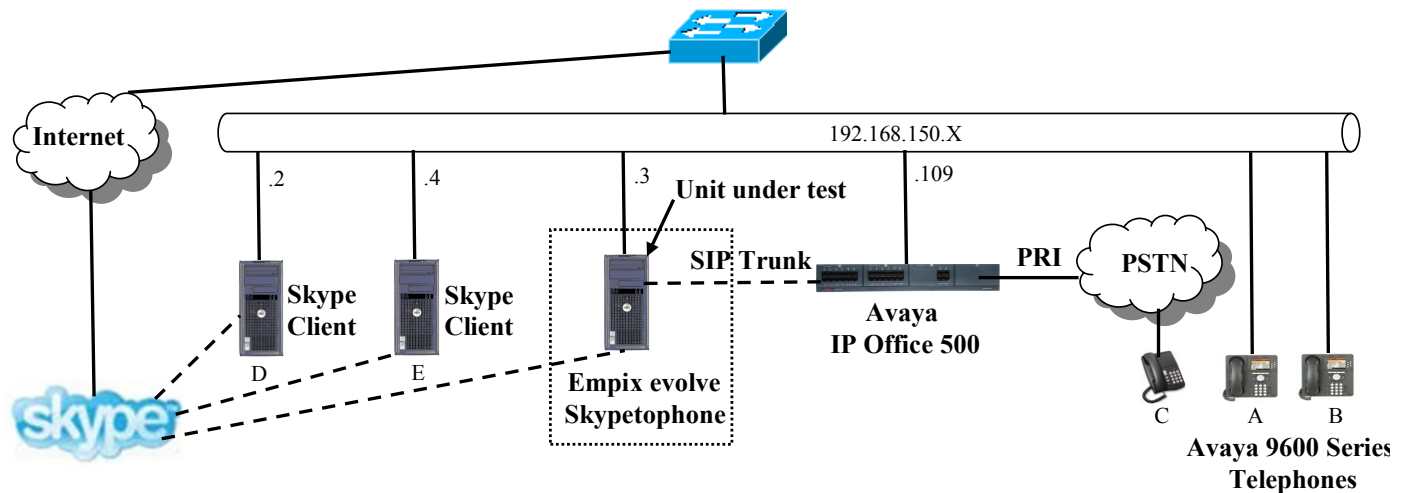
## 1.2. Support

Support is available from Empix evolve at:

- Empix evolve srl  
+39 0733 866 870  
[support@empixevolve.com](mailto:support@empixevolve.com)  
<http://www.empixevolve.com>

## 2. Reference Configuration

The following diagram illustrates the configuration which was used for testing.



**Figure 1: Empix evolve Skypetophone Configuration**

The following table shows the terminal types and extensions assigned to each of the telephone terminals used for these tests. Note that public telephone numbers have been obscured for security reasons.

Endpoint	Ext	Virt Ext	Speed Dial	PSTN	Skype ID	Type
A	201	601		+49 69 11111111 201	Avaya.201	Avaya 9640
B	202	602		+49 69 11111111 202	Avaya.202	Avaya 9650
C				+49 69 22222222		POTS Phone
D			512		Avaya.Client.D	Skype Client
E			513		Avaya.Client.E	Skype Client

**Table 1: Extensions Used for Testing**

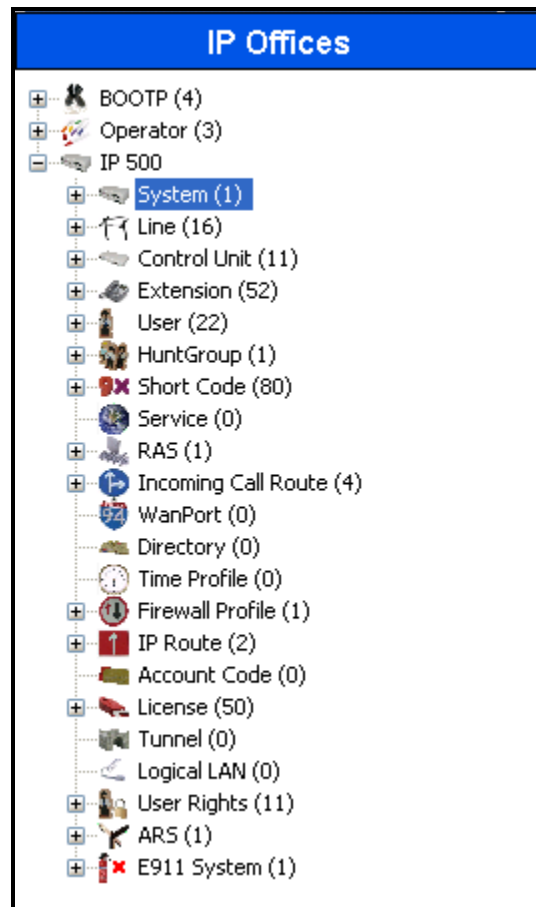
### 3. Equipment and Software Validated

Software Component	Version
Avaya IP Office	6 (8)
Avaya 9600 IP Telephones	S3.110B
MS XP Professional (xtension <b>evolve</b> Server platform OS)	SP2
Skype Client	4.2
Empix evolve Skypetophone	XTENSION EVOLVE 3.1 BUILD 20100802.013

**Table 2: Version Numbers of Equipment and Software**

## 4. Configure IP Office

The configuration and verification operations illustrated in this section were performed using the Avaya IP Office Manager program. When this program is started, a tree structure consisting of icons representing the configurable components of the system is displayed. When one of these icons is selected, the corresponding system component can be configured.



**Figure 2: IPO Manager Component Tree**

## 4.1. Licensing

IP Telephones included in the configuration each consume an **Avaya IP Endpoint** license, as described in reference [2].

One IP Office **SIP Trunk Channels** license instance is required for each active Skype call which is handled by IP Office.

## 4.2. System

Select the “System” icon shown in **Figure 2** and enter the parameters shown in the following table. Select the tab shown in the following table.

Item	Parameter	Usage
LAN1	IP Address	Specify the IP address to be assigned to IP Office.
	IP Mask	Enter the IP mask to be used by the LAN.
Telephony	Inhibit Off-Switch Forward / Transfer	Uncheck this box.

**Table 3: System Configuration Parameters**

The screenshot displays the 'IP 500' configuration window. The 'System' tab is selected, and within it, the 'LAN1' sub-tab is active. The 'LAN Settings' section contains the following parameters:

- IP Address:** 192 . 168 . 150 . 109
- IP Mask:** 255 . 255 . 255 . 0
- Primary Trans. IP Address:** 0 . 0 . 0 . 0
- RIP Mode:** None (dropdown menu)
- Enable NAT:** ☐
- Number Of DHCP IP Addresses:** 200 (spinner)
- DHCP Mode:** Server, Client, Dialin, Disabled (radio buttons, with 'Disabled' selected)

An 'Advanced' button is located at the bottom right of the configuration area.

**Figure 3: System Parameters: LAN1**

IP 500

System

LAN1

LAN2

DNS

Voicemail

Telephony

Directory Services

System Events

SMTP

SMDR

Twining

Telephony

Tones & Music

Call Log

Analogue Extensions

Default Outside Call Sequence

Normal

Default Inside Call Sequence

Ring Type 1

Default Ring Back Sequence

Ring Type 2

Dial Delay Time (secs)

5

Dial Delay Count

0

Default No Answer Time (secs)

25

Hold Timeout (secs)

0

Park Timeout (secs)

300

Ring Delay (secs)

5

Call Priority Promotion Time (secs)

Disabled

Default Currency

EUR

Automatic Codec Preference

G.711 ALAW 64K

Companding Law

Switch

ULAW

ALAW

Line

ULAW Line

ALAW Line

DSS Status

Auto Hold

Dial By Name

Show Account Code

Inhibit Off-Switch Forward/Transfer

Restrict Network Interconnect

Drop External Only Impromptu Conference

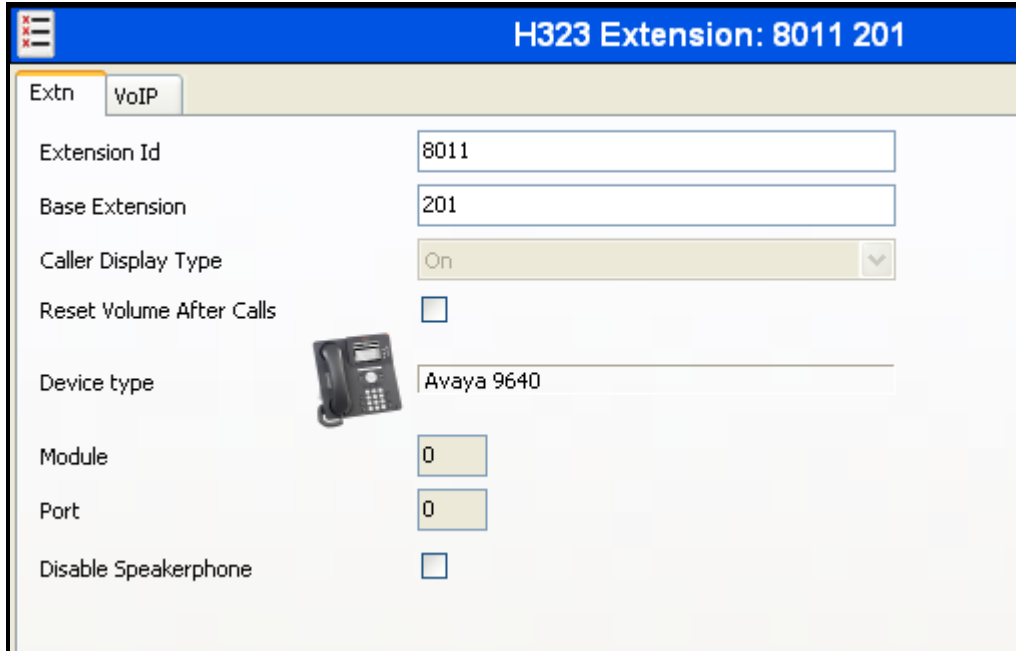
Visually Differentiate External Call

**Figure 4: System Parameters: Telephony**



### 4.3. Extensions

Select the “Extensions” icon shown in **Figure 2** and enter and click “new” to create an extension for each of the telephones A and B shown in **Table 1**. Set the “Base Extension” parameter to the extension to be assigned, and accept the default values for the remaining parameters.



**H323 Extension: 8011 201**


Extn VoIP

Extension Id 8011

Base Extension 201

Caller Display Type On

Reset Volume After Calls ☐

Device type  Avaya 9640

Module 0

Port 0

Disable Speakerphone ☐

**Figure 5: Extension Screen**

#### 4.4. Create Users

Select the “Users” icon shown in **Figure 2** and click “new” to create a user for the telephones A and B shown in **Table 1** using the parameters shown in the following table.

Parameter	Usage
Name	Enter a descriptive name to identify the user.
Extension	Enter the telephone extension to be assigned to the endpoint (which was allocated in <b>Figure 5</b> ).

**Table 4: User Configuration Parameters**

The screenshot shows a web-based user configuration interface. At the top, there's a blue header bar with the text "Ext201: 201". Below this is a row of tabs: "User", "Voicemail", "DND", "ShortCodes", "Source Numbers", "Telephony", "Forwarding", "Dial In", and "Voice Recording". The "User" tab is selected. The main area contains a form with the following fields:

- Name: Text input field containing "Ext201".
- Password: Text input field.
- Confirm Password: Text input field.
- Full Name: Text input field.
- Extension: Text input field containing "201".
- Locale: Dropdown menu.
- Priority: Text input field containing "5".
- System Phone Rights: Dropdown menu containing "None".
- Profile: Dropdown menu containing "Basic User".

The "Name" and "Extension" fields are highlighted with red rectangular boxes.

**Figure 6: User Screen**

## 4.5. Create SIP Line

This section contains a description of the configuration of the SIP trunk between IP Office and Empix evolve Skypetophone. Other types of PSTN trunks can be used as well. Select the “Line” icon shown in **Figure 2**, click “new”, and enter the parameters shown in the following table.

Tab	Parameter	Usage
SIP Line	ITSP Domain Name	Enter the IP address of the Empix evolve Skypetophone server.
	ITSP IP Address	Enter the IP address of the Empix evolve Skypetophone server.
	Use Network Topology Info	Select “None” from the drop-down menu.
	Send Port	Enter an available port number which matches the port number used in <b>Figure 39</b> .
SIP URI	Local URI	Enter “*”.
	Contact	Select “Use Internal Data” from the drop-down menu.
	Display Name	Select “Use Internal Data” from the drop-down menu.
	Registration	Select “None” from the drop-down menu.
	Incoming Group	Enter an available group number. This value must match the group number used in <b>Figure 15</b> .
	Outgoing Group	Use the same value as was used for “Incoming Group”
VoIP	Max Calls per Channel	This value must be sufficient to supply one channel each for the maximum number of simultaneous Skype calls, plus one channel each for the maximum number of simultaneous call diversions or call transfers for Skype calls.
	Compression Mode	Check the “G.711” box, and uncheck the others.

**Table 5: SIP Line Configuration Parameters**

SIP Line - Line 18

SIP Line

SIP URI

VoIP

T38 Fax

SIP Credentials

Line Number 18

ITSP Domain Name 192.168.150.3

ITSP IP Address 192 . 168 . 150 . 3

Prefix

National Prefix 0

Country Code

International Prefix 00

Send Caller ID None

Registration Required ☐

In Service ☒

Use Tel URI ☐

Check OOS ☒

Call Routing Method Request URI

Network Configuration

Layer 4 Protocol UDP

Use Network Topology Info None

Send Port 5062

Listen Port 5060

**Figure 7: SIP: SIP Line Configuration Screen**

**SIP Line - Line 18**

SIP Line | **SIP URI** | VoIP | T38 Fax | SIP Credentials

Channel	Groups	Via	Local URI	Contact
1	18 18	<... *		

Add...  
Remove  
Edit...

OK  
Cancel

**Edit Channel**

Via: <None>

Local URI: \*

Contact: Use Internal Data

Display Name: Use Internal Data

Registration: 0: <None>

Incoming Group: 18

Outgoing Group: 18

Max Calls per Channel: 3

**Figure 8: SIP: SIP URI Configuration Screen**

**SIP Line - Line 18**

SIP Line | SIP URI | **VoIP** | T38 Fax | SIP Credentials

Compression Mode: **Advanced**

Call Initiation Timeout (s): 4

DTMF Support: RFC2833

☒ G.711 ALAW 64K  
☐ G.711 ULAW 64K  
☐ G.729(a) 8K CS-ACELP  
☐ G.723.1 6K3 MP-MLQ

☐ VoIP Silence Suppression  
☒ Fax Transport Support  
☒ Re-invite Supported  
☐ Use Offerer's Preferred Codec

**Figure 9: SIP: VoIP Configuration Screen**

## 4.6. Create PRI Line

This section contains a description of the IP Office configuration for use with a PRI interface to the PSTN, which was used for conformance testing. Other types of PSTN trunks can be used as well. Select the “Line” icon shown in **Figure 2**, click “new”, and enter the parameters shown in the following table.

Parameter	Usage
Incoming Group ID	Assign the number to an available Group ID.
Prefix	Enter the prefix which is used to initiate a local external call via the PSTN.
National Prefix	Enter the prefix which is used to initiate a national external call via the PSTN.
International Prefix	Enter the prefix which is used to initiate an international external call via the PSTN.
Line SubType	Select “ETSI” from the drop-down list, as required for access to the PSTN in Europe.
Outgoing Group ID	Assign the number as was assigned to the Incoming Group ID.

**Table 6: Primary Rate Line Configuration Parameters**

PRI Line

Short Codes

Channels

Line Number

13

Line SubType

ETSI

Card

4

Port

P1

Telephone Number

TEI

0

Incoming Group ID

5

Outgoing Group ID

5

Prefix

0

Number of Channels

30

National Prefix

00

Outgoing Channels

30

International Prefix

000

Voice Channels

30

Data Channels

30

CRC Checking

☒

Clock Quality

Network

Line Signalling

CPE

Add 'Not end-to-end ISDN' Information Element

Never

Send Redirecting Number

☐

Supports Partial Rerouting

☐

Force Number Plan to ISDN

☐

Support Call Tracing

☐

Active CCBS Support

☐

Passive CCBS Support

☐

Cost Per Charging Unit

618

**Figure 10: Primary Rate Line Configuration Screen**

## 4.7. Short Codes

Select the “Short Code” icon shown in **Figure 2** and click “new” to allocate a short code to provide access to the PSTN via the PRI interface, as shown in the following table.

Usage	Parameter	Usage
PSTN Call	Code	Enter <b>0N;</b>
	Feature	Select <b>Dial</b> from the drop-down menu.
	Telephone Number	Enter <b>NSi</b> followed by the telephone number of the PSTN connection, then an <b>E</b> .
	Line Group ID	Enter the group number which was assigned to the PSTN interface in <b>Figure 10</b> .
Skype Call	Code	Enter the Empix evolve Skypetophone routing code configured in <b>Figure 38</b> , followed by “N;”.
	Feature	Select <b>Dial</b> from the drop-down menu.
	Telephone Number	Enter the Empix evolve Skypetophone routing code configured in <b>Figure 38</b> , followed by “N”, followed by “@<server IP>”, where <server IP> is the IP address of the Empix evolve Skypetophone server.
	Line Group ID	Enter the group number which was assigned to the SIP interface in <b>Figure 8</b> .

**Table 7: Shortcode Configuration Parameters**

The screenshot shows a configuration window titled "0N;; Dial\*". It has a tab labeled "Short Code". The form contains the following fields:

- Code:** 0N;
- Feature:** Dial (selected from a dropdown menu)
- Telephone Number:** NSi691111111E
- Line Group Id:** 5 (selected from a dropdown menu)
- Locale:** Germany (German) (selected from a dropdown menu)
- Force Account Code:** ☐

A red rectangular box highlights the Code, Feature, Telephone Number, and Line Group Id fields.

**Figure 11: PSTN Access Short Code**



5N;; Dial	
Short Code	
Code	5N;;
Feature	Dial
Telephone Number	5N"@192.168.150.3"
Line Group Id	18
Locale	
Force Account Code	<input type="checkbox"/>

**Figure 12: Skype Access Short Code**

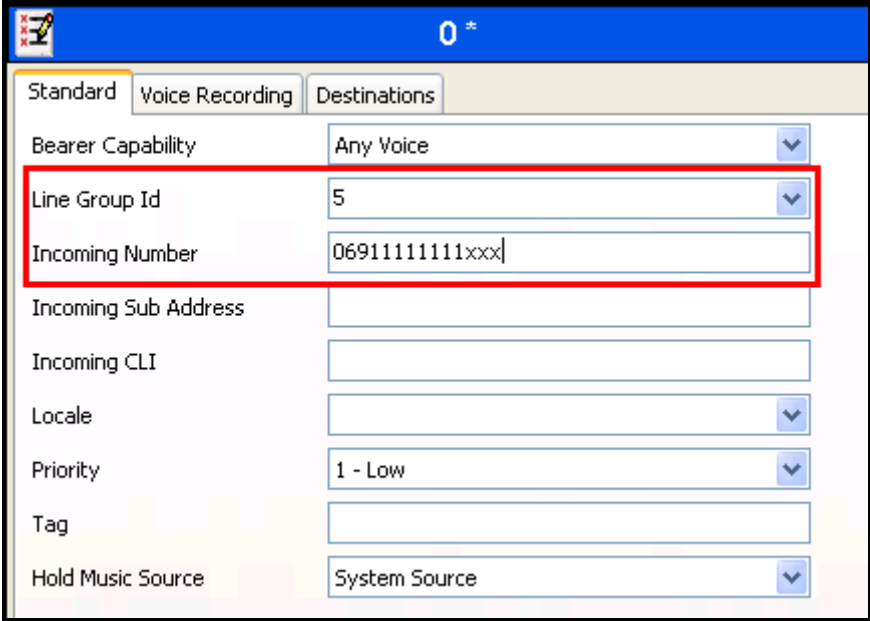
## 4.8. Incoming Call Routes

### 4.8.1. PSTN Incoming Call Route

Select the “Incoming Call Route” icon shown in **Figure 2** and click “new” to create an Incoming Call Route for routing calls from the PSTN to local extensions. Assign parameters to this call route as shown in the following table:

Tab	Parameter	Usage
Standard	Line Group Id	Enter the Group Id of the PRI line, as shown in <b>Figure 10</b> .
	Incoming Number	Enter “0” followed by the PSTN number assigned to the PRI interface, followed by the string “xxx” which matches the local extension
Destinations	Destination	Configure the destination to use the last four digits of the called party number by entering a value of “#”.

**Table 8: Parameters for Incoming Call Routes**



The screenshot displays the configuration interface for a PSTN Incoming Call Route. The 'Standard' tab is selected, showing various parameters. A red rectangle highlights the 'Line Group Id' (set to 5) and 'Incoming Number' (set to 0691111111xxx) fields. Other visible parameters include Bearer Capability (Any Voice), Incoming Sub Address, Incoming CLI, Locale, Priority (1 - Low), Tag, and Hold Music Source (System Source).

Parameter	Value
Bearer Capability	Any Voice
Line Group Id	5
Incoming Number	0691111111xxx
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

**Figure 13: PSTN Incoming Call Route - Standard Parameters**

TimeProfile	Destination	Fallback Extension
Default Value	#	

**Figure 14: PSTN Incoming Call Route - Destinations Screen**

#### 4.8.2. SIP Trunk Incoming Call Route

Select the “Incoming Call Route” icon shown in **Figure 2** and click “new” to create an Incoming Call Route for routing calls from the SIP trunk to local extensions. Assign parameters to this call route as shown in the following table:

Tab	Parameter	Usage
Standard	Line Group Id	Enter the Group Id of the SIP line.
	Incoming Number	Leave this field blank.
Destinations	Destination	Enter “.”.

**Table 9: Parameters for Incoming Call Routes**

Bearer Capability	Any Voice
Line Group Id	18
Incoming Number	
Incoming Sub Address	
Incoming CLI	
Locale	
Priority	1 - Low
Tag	
Hold Music Source	System Source

**Figure 15: SIP Incoming Call Route - Standard Parameters**

	TimeProfile	Destination	Fallback Extension
▶	Default Value	.	

**Figure 16: SIP Incoming Call Route - Destinations Screen**

## 5. Configure Windows XP

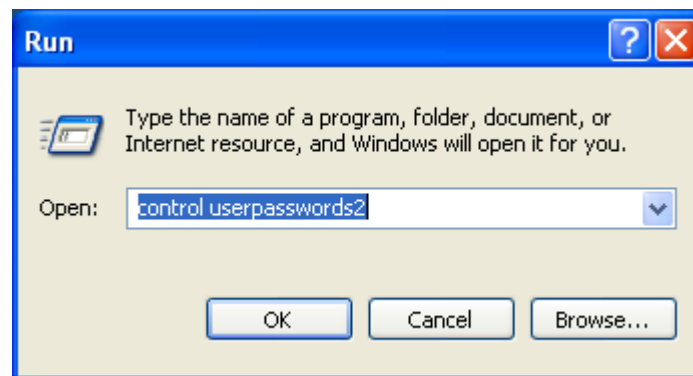
The procedure described in this section is optional. However, if this procedure is not performed, the Empix evolve Skypetophone program will not restart automatically after recovery from a power failure, and the Empix evolve Skypetophone application will have to be restarted manually.

However, this procedure has the disadvantage of disabling Windows XP login security. A designated user is logged in automatically, without entering a password, each time Windows starts and thus may pose too high a risk for some operational environments.

This procedure was configured for the server PC which was used for DevConnect compliance testing. This has been tested only with Windows XP.

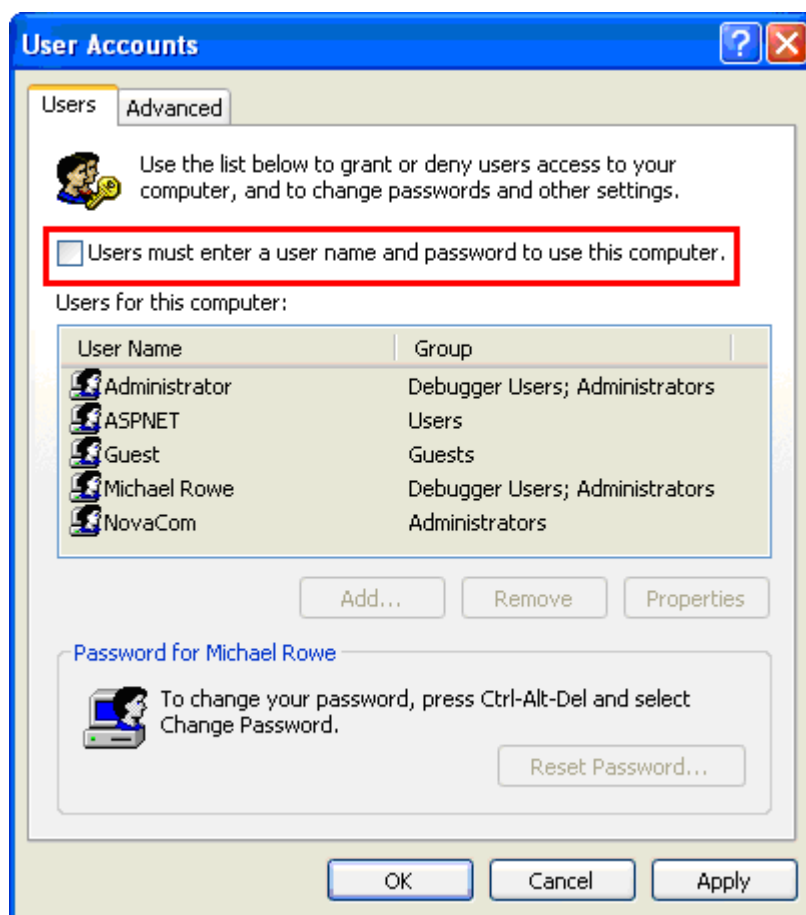
Since the Empix evolve Skypetophone service runs on the server PC as an application and not a system service, steps must be taken if it is required that the service start automatically after recovery from a power failure. This procedure causes the Windows user from which the Empix evolve Skypetophone application is run to be logged into Windows automatically when the server PC is started, and to auto-start the Empix evolve Skypetophone application at user login.

Start the Windows user account administration by starting the “Run” dialog from the Windows “Start” control, entering “control userpasswords2”, and clicking “OK”.



**Figure 17: Dialog to Start User Administration**

Uncheck the “Users must enter a user name and password to use this computer”, and click “OK”.



**Figure 18: User Account List Screen**

Enter the credentials for the user to be automatically logged in when the system starts, and click “OK”.

A Windows-style dialog box titled "Automatically Log On" with a blue header bar and a red close button. The main area has a light beige background. On the left, there is a yellow key icon. To its right, a message reads: "You can set up your computer so that users do not have to type a user name and password to log on. To do this, specify a user that will be automatically logged on below:". Below this, there are three input fields. The first is labeled "User name:" and contains the text "Michael Rowe". The second is labeled "Password:" and contains ten black dots. The third is labeled "Confirm Password:" and contains ten black dots. At the bottom right, there are two buttons: "OK" and "Cancel".

**Automatically Log On**

You can set up your computer so that users do not have to type a user name and password to log on. To do this, specify a user that will be automatically logged on below:

User name: Michael Rowe

Password: ●●●●●●●●●●

Confirm Password: ●●●●●●●●●●

OK Cancel

**Figure 19: Auto-login User Selection Screen**

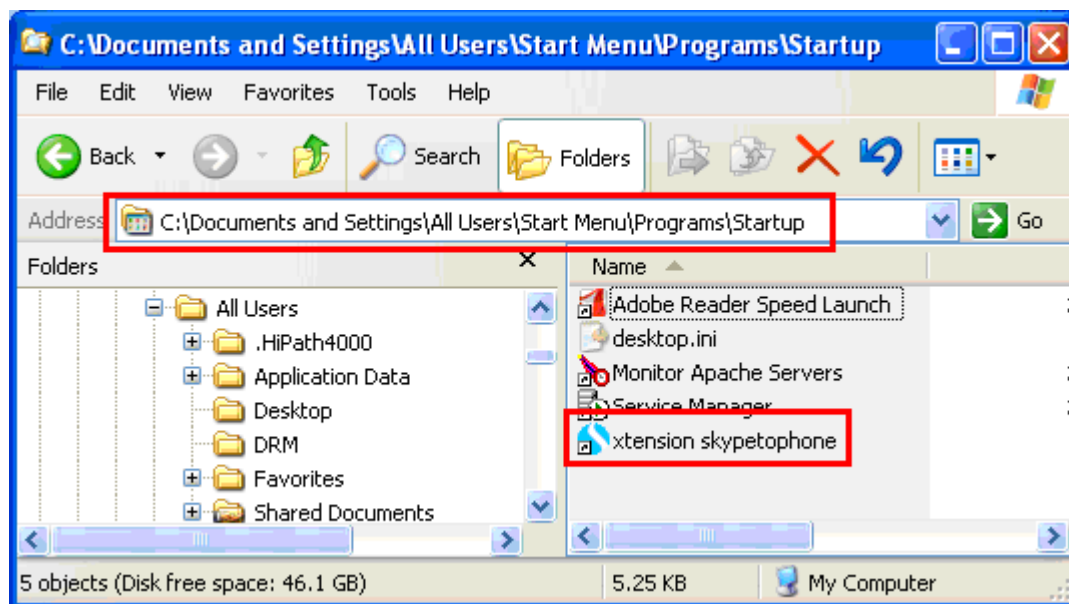
Select the “Advanced” tab and uncheck the “Require users to press Ctrl+Alt+Delete”, and click “OK”.



**Figure 20: User Account Advanced Tab Screen**



Insert a “shortcut” to the “xtension skypetophone” application in the Windows autostart directory.



**Figure 21: Windows XP Autostart Directory**

## 6. Configure Skype

Skype users are created via the Internet using a web browser by browsing to <http://www.skype.com> and clicking “Get Skype”. The procedure described in this section must be repeated for each extension which requires Skype access.

When the Skype Introductory Screen appears, click “Get Skype”.

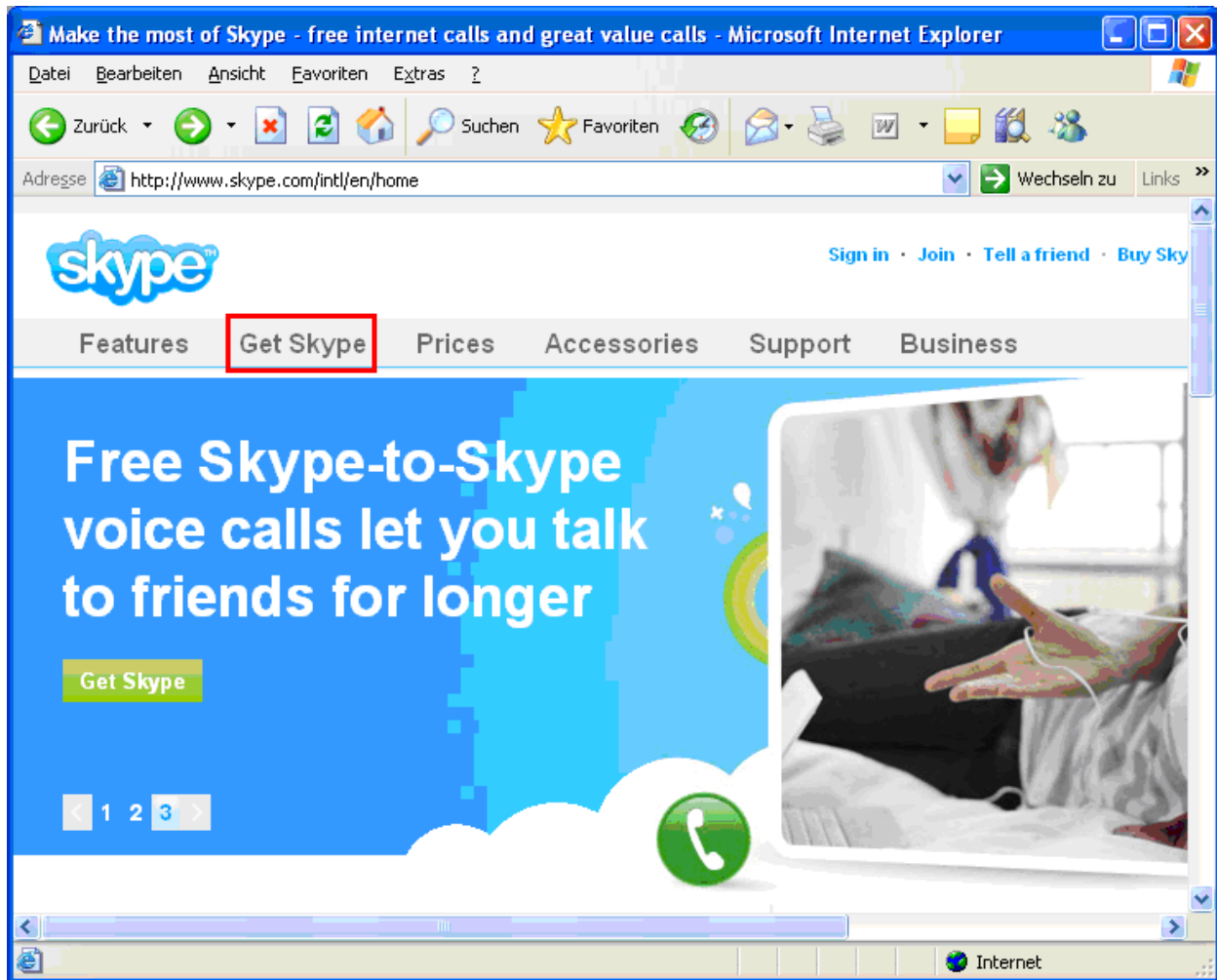
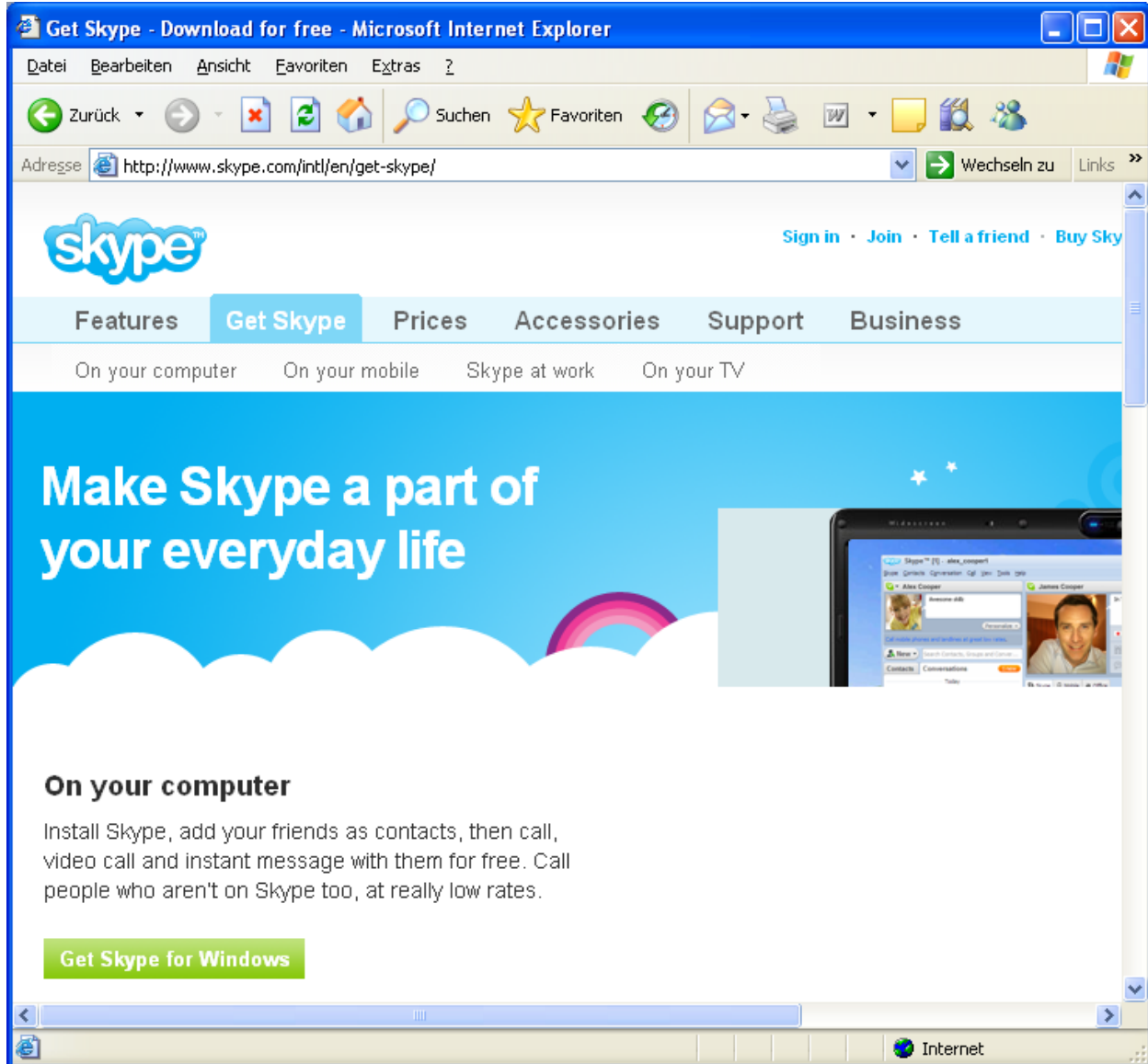


Figure 22: Skype Introductory Screen

Click “Get Skype for Windows”.



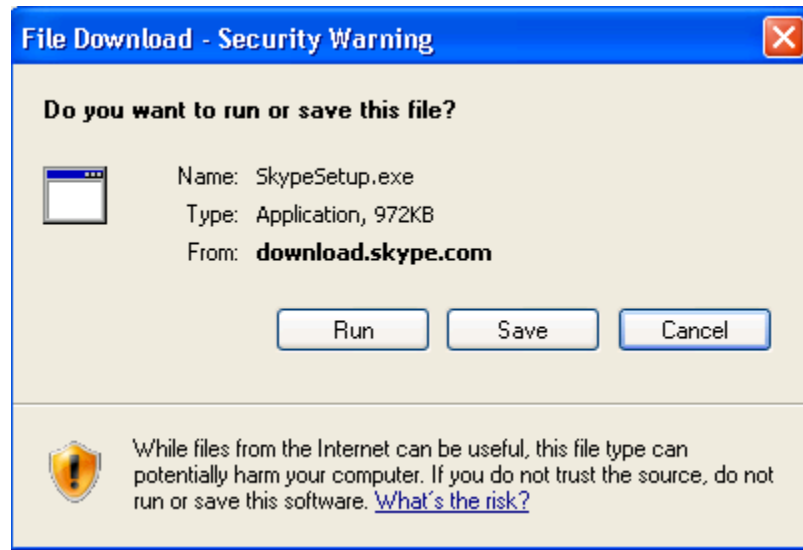
**Figure 23: Skype Platform Choice Screen**

Click “Download now”.



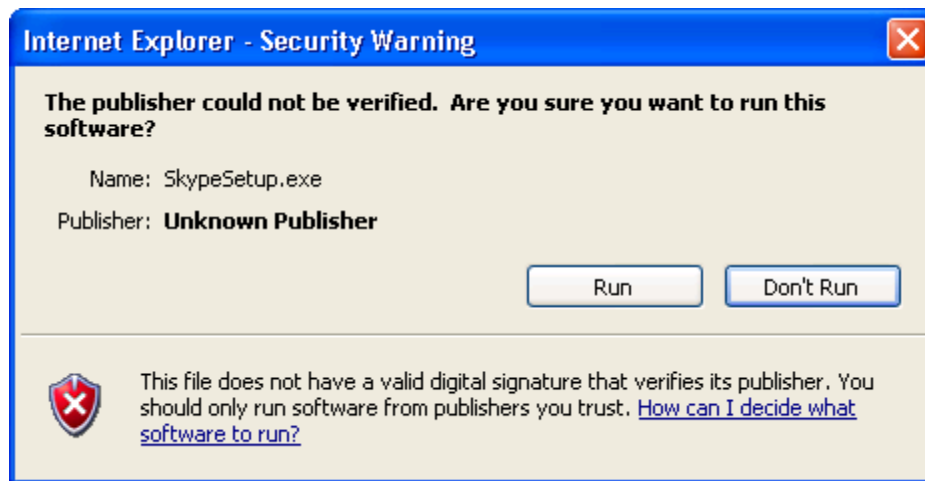
Figure 24: Skype Download Confirmation Screen

Click “Run”.



**Figure 25: Skype Download Dialog**

Click “Run”.



**Figure 26: Skype Security Warning**

Click “Run”.



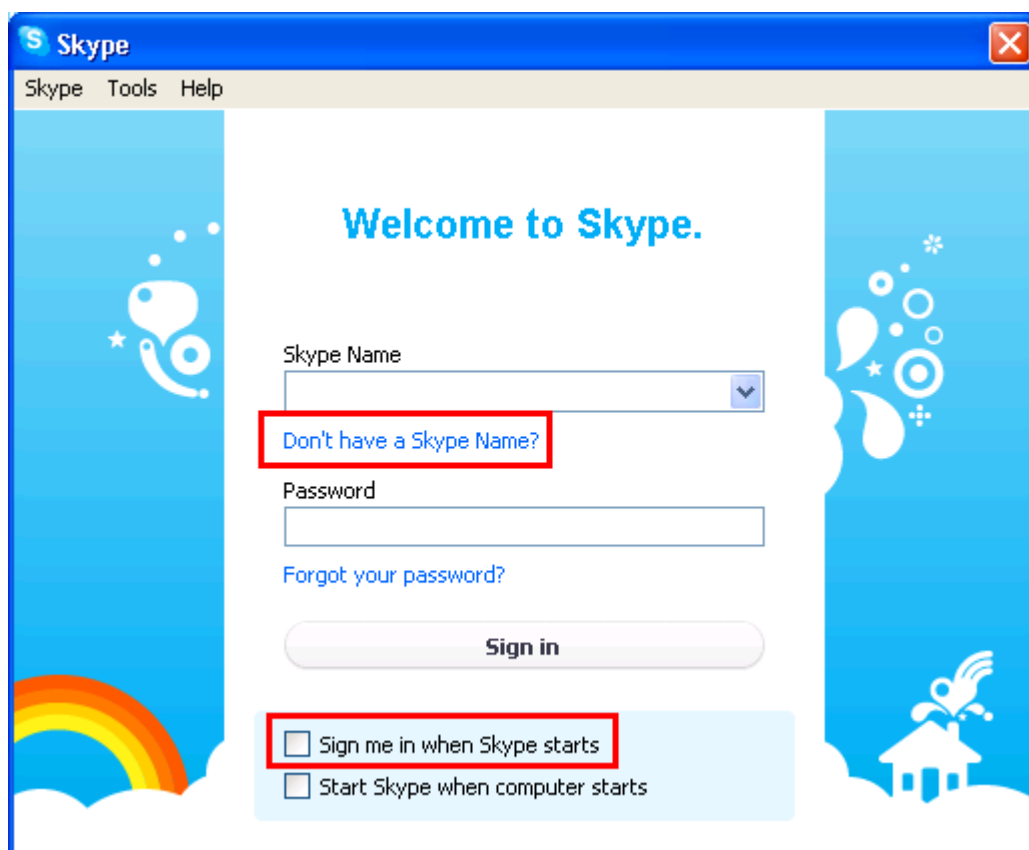
**Figure 27: Skype Security Confirmation Screen**

Click “I agree – install”.



**Figure 28: Skype License Screen**

Uncheck the “Sign me in when Skype starts” box, and click “Don’t have a Skype Name”.



**Figure 29: Skype Welcome Screen**

Enter the parameters shown in the following table and click “I agree – create account”.

Parameter	Usage
Full Name	Enter a descriptive name to identify the user.
Create Skype Name	Enter the Skype user ID. This must match the value entered in <b>Figure 40</b> .
Password	Enter an appropriate password. This must match the value entered in <b>Figure 40</b> .
Email	Enter the email address at which the user can be reached.
Send Promotions	Uncheck this box if you do not require promotional material.

**Table 10: User Configuration Parameters**

Skype™ - Create account

Create a new Skype account

Already have a Skype account? [Sign in](#) [Get help](#)

**Full name**  
Extn201

**Create Skype Name**  
Avaya.201

Note: anyone can see this information. Please fill in all fields.

**Password**  
\*\*\*\*\* ✓

**Repeat password**  
\*\*\*\*\* ✓

**Email**  
mrowe@avaya.com ✓

**Repeat email**  
mrowe@avaya.com ✓

☐ Yes, send me Skype news and promotions

Note: only you can see this information. Please fill in all fields.

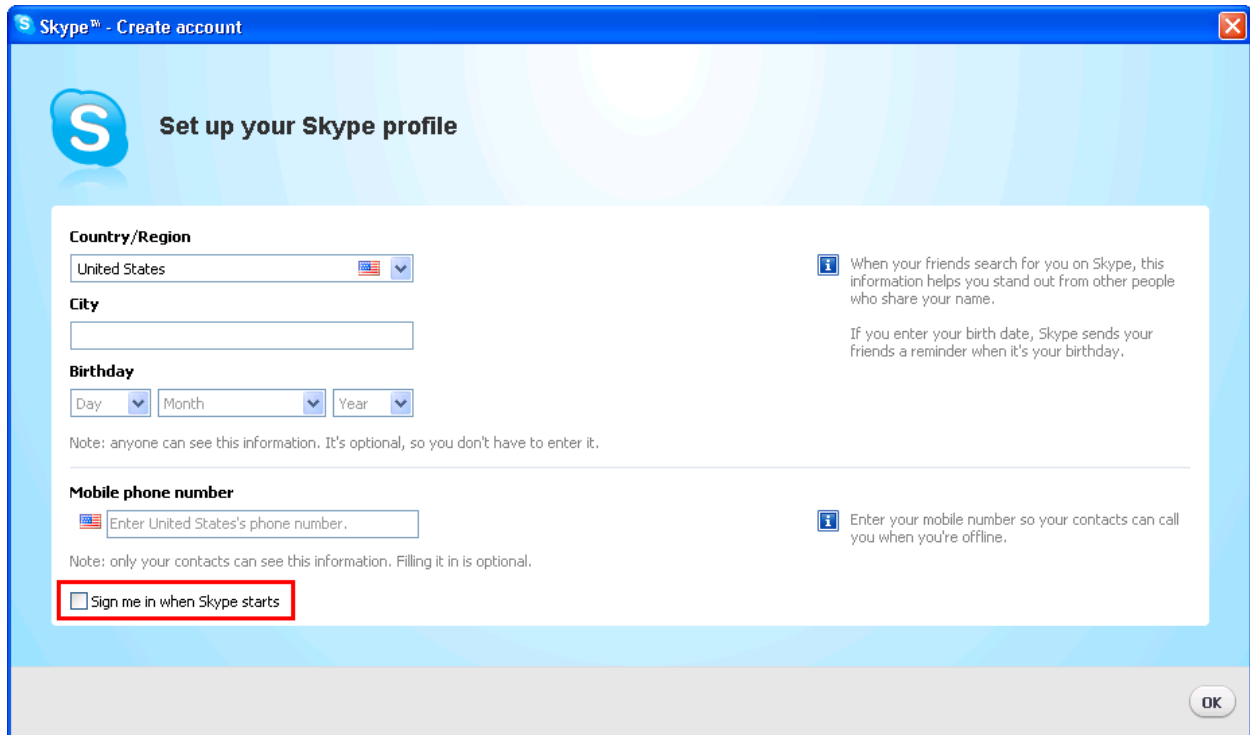
Yes, I have read and I accept the [Skype End User License Agreement](#), the [Skype Terms of Service](#) and the [Skype Privacy Statement](#)

[I agree - create account](#) [Cancel](#)

**Figure 30: Skype Account Dialog**



User identification information can optionally be entered into this dialog. Uncheck the “Sign me in when Skype starts” box and click “GO”.



The image shows a Windows-style dialog box titled "Skype™ - Create account". The main heading is "Set up your Skype profile". The form contains several sections: "Country/Region" with a dropdown menu set to "United States"; "City" with an empty text field; "Birthday" with three dropdown menus for "Day", "Month", and "Year"; and "Mobile phone number" with a text field containing the placeholder "Enter United States's phone number.". To the right of the form, there are two informational messages: one about search visibility and another about birthday reminders. At the bottom left, there is a checkbox labeled "Sign me in when Skype starts" which is currently unchecked and highlighted with a red rectangle. At the bottom right, there is an "OK" button. The dialog box has a blue title bar and a light blue background with a subtle pattern.

Skype™ - Create account

**S** Set up your Skype profile

**Country/Region**  
United States

**City**

**Birthday**  
Day Month Year

Note: anyone can see this information. It's optional, so you don't have to enter it.

**Mobile phone number**  
Enter United States's phone number.

Note: only your contacts can see this information. Filling it in is optional.

☐ Sign me in when Skype starts

OK

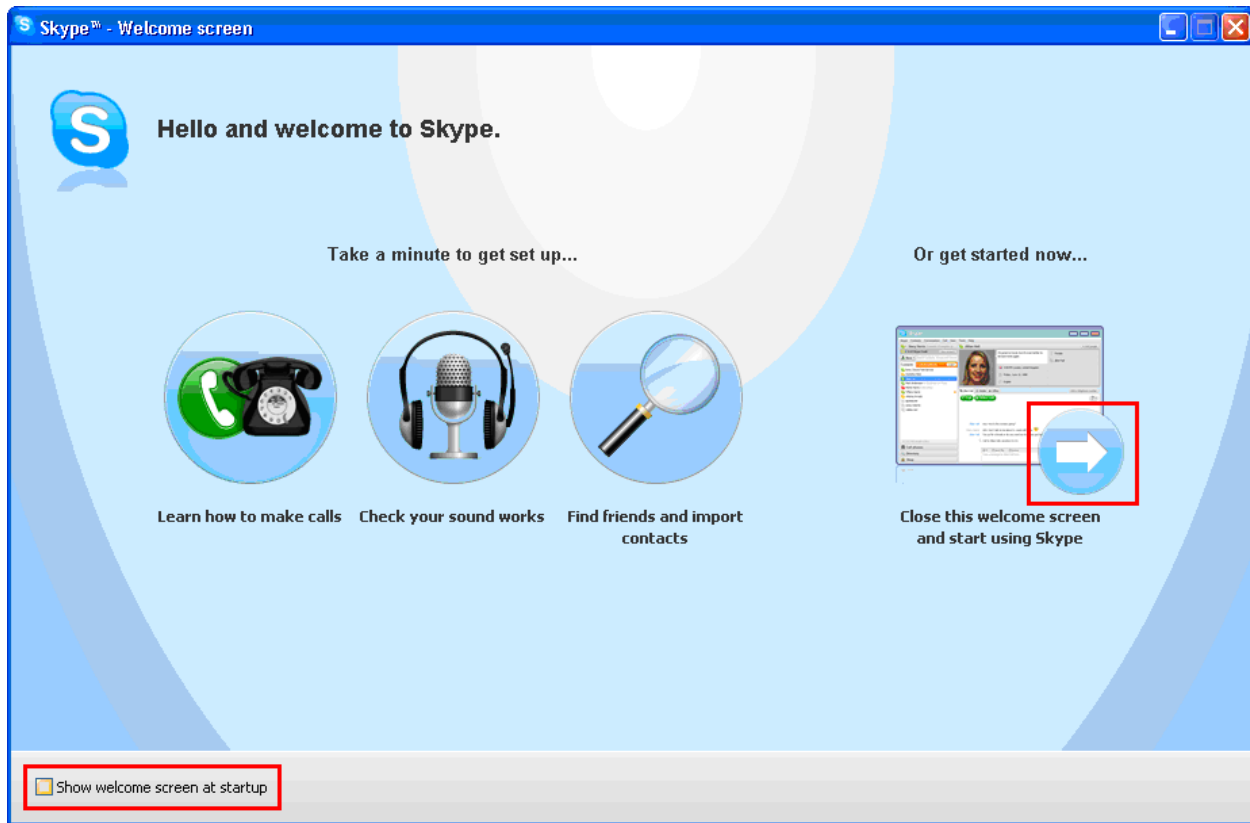
When your friends search for you on Skype, this information helps you stand out from other people who share your name.

If you enter your birth date, Skype sends your friends a reminder when it's your birthday.

Enter your mobile number so your contacts can call you when you're offline.

**Figure 31: Skype Profile Screen**

Uncheck the “Show welcome screen at startup” box and click the right arrow button.



**Figure 32: Skype Startup Screen**

Click the red “X” in the upper right corner to exit the program.

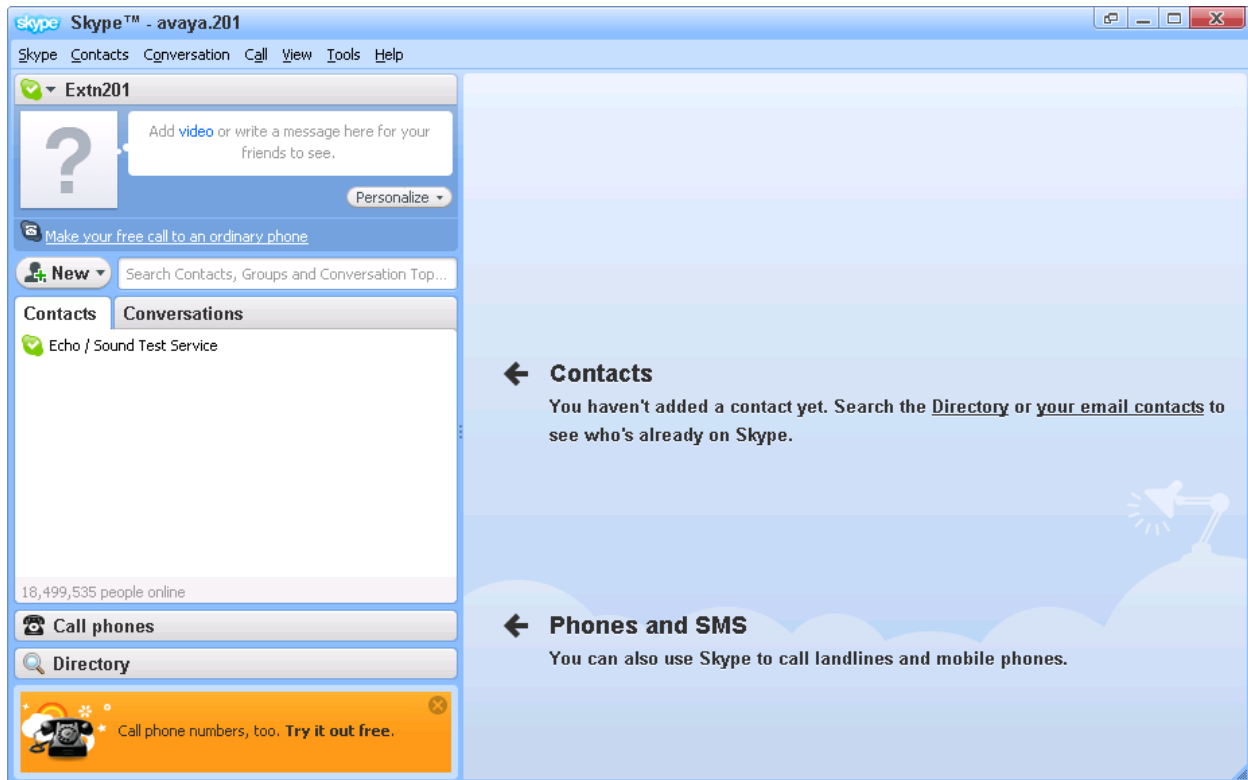
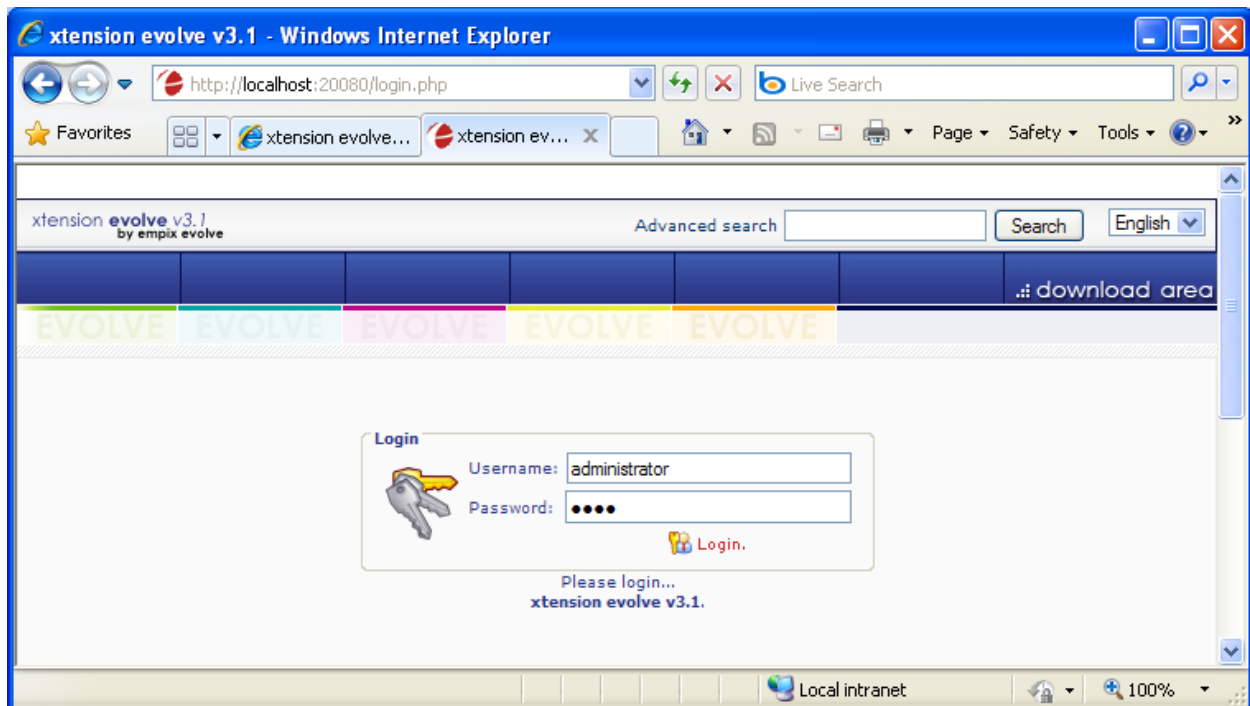


Figure 33: Skype Program Screen

## 7. Configure Empix evolve Skypetophone

The Empix evolve Skypetophone installation process also installs MySQL and the Apache server. The installation procedure is beyond the scope of this document. However, the installation used for compliance testing was done using the prompted default values.

Empix evolve Skypetophone uses a Web-based tool for configuration, which can be accessed from the server via <http://localhost:20080/login.php>. The initial login credentials are for Username “administrator, with Password “1234”. These values should be changed subsequently for security reasons.



**Figure 34: Xtension evolve Login Screen**

## 7.1. Install Updates

Navigate to “Administrative tools” → “Updates”. Click the “Check Updates” button to check for new updates, and the “Download & install” button to install needed updates.

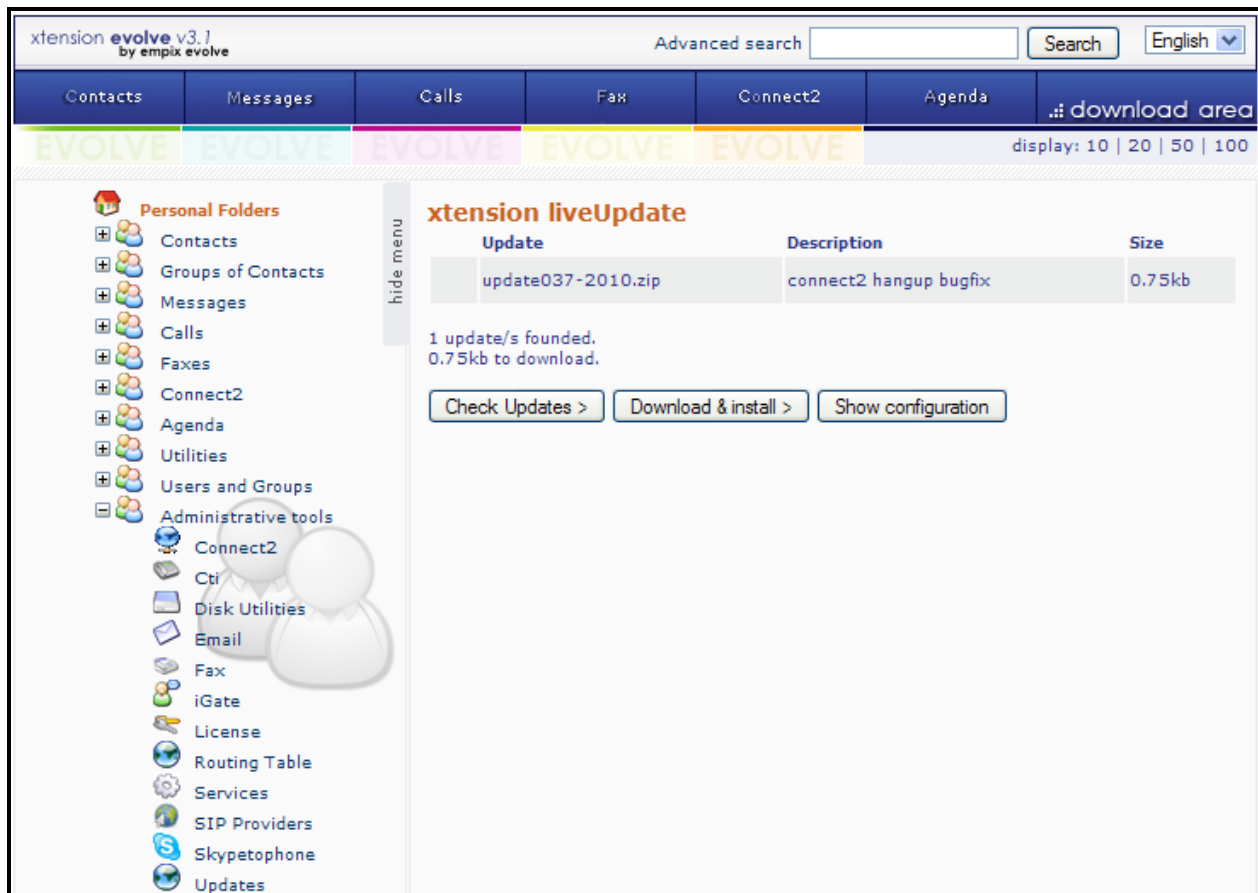


Figure 35: Xtension evolve Update Screen

## 7.2. Install License

Navigate to “Administrative tools” → “License” and enter the appropriate identification information. If a permanent license is available, enter the license code into the “Serial Number” field and click the “Activate” button. If no license is available, a thirty day trial license can be obtained without charge by clicking the “Request try’n buy” button.

The screenshot shows the 'Xtension evolve v3.1 by empix evolve' web interface. The top navigation bar includes 'Contacts', 'Messages', 'Calls', 'Fax', 'Connect2', 'Agenda', and a 'download area'. Below this is a 'display: 10 | 20 | 50 | 100' selector. The left sidebar lists 'Personal Folders' (Contacts, Groups of Contacts, Messages, Calls, Faxes, Connect2, Agenda, Utilities, Users and Groups) and 'Administrative tools' (Connect2, Ctl, Disk Utilities, Email, Fax, iGate, License, Routing Table). The 'License' option is selected. The main content area is titled 'License' and has two radio buttons: 'Simple' (selected) and 'Advanced'. The 'License' form contains the following fields: Name (DevConnect), Address (Kleyerstr 94), Postal Code (12345), City (Frankfurt), Province (FR), Country (Germany), empixevolve Login (mrowe@avaya.com), and Serial Number. At the bottom of the form are two buttons: 'Activate' and 'Request try'n buy'.

Figure 36: Xtension evolve License Activation Screen

Click the “Activate” button to activate the license.

xtension **evolve** v3.1  
by empix evolve

Advanced search  Search English

Contacts Messages Calls Fax Connect2 Agenda download area

EVOLVE EVOLVE EVOLVE EVOLVE EVOLVE display: 10 | 20 | 50 | 100

**Personal Folders**

- Contacts
- Groups of Contacts
- Messages
- Calls
- Faxes
- Connect2
- Agenda
- Utilities
- Users and Groups
- Administrative tools
- Connect2
- Cti
- Disk Utilities
- Email
- Fax
- iGate
- License
- Routing Table
- Services
- SIP Providers

**License**

☒ Simple ☐ Advanced

Client fax	Client cti	Canali ip
15	15	8

Name: DevConnect

Address: Kleyerstr 94

Postal Code: 12345 City: Frankfurt Province: FR

Country: Germany

empixevolve Login: mrowe@avaya.com

Serial Number: E22X-0FX7-SM61-MONK-2GTX

Warning: you are using a try'n buy license.  
You license will expire in 30 days on 02/09/2010.

Activate Request try'n buy

**Figure 37: Xtension evolve Try’n Buy License Activation Screen**

### 7.3. Configure Call Routing

Navigate to “Administrative tools” → “Routing Table” and enter the parameters shown in the following table.

Parameter	Usage
skypetophone	Enter the leading digit which is used to route Skype calls. This should be the same value configured in <b>Figure 12</b> .
connect2 Faxserver SIP gateway Recorder	If any of these services are used, this field should be set appropriately. If the service is not used, a digit should be entered which does not correspond to any numbers in the IP Office dialing plan, as none of these fields can be left blank.
Digits number	Enter the number of digits to be used for Skype extensions.

**Table 11: User Configuration Parameters**

The screenshot displays the 'Routing Table' configuration page in the Xtension evolve application. The left sidebar shows a tree view of system components, with 'Administrative tools' expanded to show 'Routing Table'. The main content area is titled 'Routing Table' and includes a 'Simple' radio button selected. Below this, the 'Calls Handler' is set to 'Use GnuGK to handle calls'. The 'System prefix' section contains a table with the following values:

skypetophone	5
connect2	6
faxserver	7
SIP gateway	8
Recorder	9

Below the table, the 'Digits number' is set to 3. Other fields include 'Address PABX', 'PABX ext', 'PSTN PABX', 'Country code', 'International prefix', 'Number for outside calls', and 'Replace with'. The 'Input numbers syntax' is set to '\*[127.0.0.1]' with a legend explaining the symbols.

**Figure 38: Xtension evolve Routing Table**



## 7.4. Configure System Parameters

Navigate to “Administrative tools” → “Skypetophone” and enter the parameters shown in the following table.

Parameter	Usage
SkypeOut prefix	Prefix to be used to make calls using Skype minutes; by default it's set by the system to Empix evolve Skypetophone's leading digit set on the routing table followed by a “0”.
Country code	Enter the country dialing code preceded by a “+” character.
International prefix	Enter the prefix used to dial international numbers.
Skype path	This field should contain the path of the Empix evolve Skypetophone server. If this field is valid, a green check icon should appear to the right of the field.
Maximum number of channels	This value must be sufficient to supply one channel each for the maximum number of simultaneous Skype calls, plus one channel each for the maximum number of simultaneous call diversions or call transfers for Skype calls. This value should match the “Match Calls per Channel” field in <b>Figure 8</b> .
Interface (advanced)	Enter “*.” followed by the port number used to receive SIP messages. Set this value to match the “Send Port” parameter in <b>Figure 7</b> .

**Table 12: User Configuration Parameters**



## 7.5. Add Users

Click “Add” within the “Skype Instances” box to add a Skype user, and enter the parameters shown in the following table. Repeat this for each of the endpoints in **Table 1**.

Parameter	Usage
Skype User (simple)	Enter the username for the Skype account. This must match the value from <b>Figure 30</b> .
Skype Password (simple)	Enter the password for the Skype account. This must match the value from <b>Figure 30</b> .
Forward to (simple)	Enter “sip:”, the user’s local extension, “@”, IP address of IP Office SIP trunk configured in <b>Figure 7</b> .
Reserved for (simple)	Enter the user’s local extension (optional field: if this is set, the Skype instance will be used only by the extension.).

**Table 13: User Configuration Parameters**

The screenshot displays the 'Skypetophone configuration' window in the Empix evolve application. The 'Simple' configuration tab is selected. The 'General' section includes fields for 'SkypeOut prefix' (50), 'Country code' (+49), 'International prefix' (00), 'Skype path' (C:\Program Files\Phone\Skype.exe), 'Maximum number of channels' (3), 'Default profile mood text', and 'Default chat auto answer'. The 'Skype instances' section shows a single instance named 'avaya.201'. The 'Forward to' field is highlighted with a red box, showing the value 'sip:201@192.168.150.109'. The 'Reserved to' field shows the value '201'.

**Figure 40: Empix evolve Skypetophone Simple Configuration**

## 7.6. Start Service

Navigate to “Administrative tools” → “Services” and click the “Start xtension evolve service” button. The Empix evolve Skypetophone application must subsequently be started manually. It can be found at Start→Programs→xtension evolve.

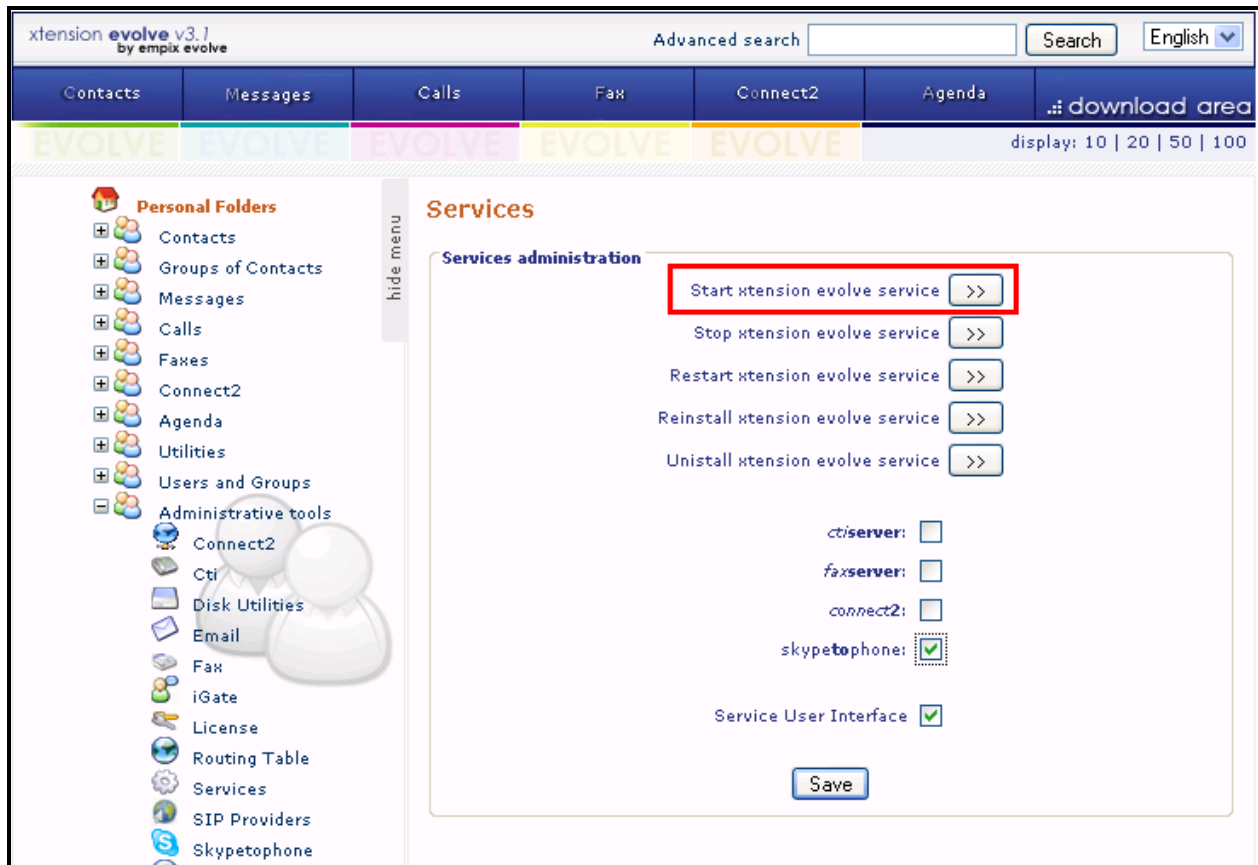
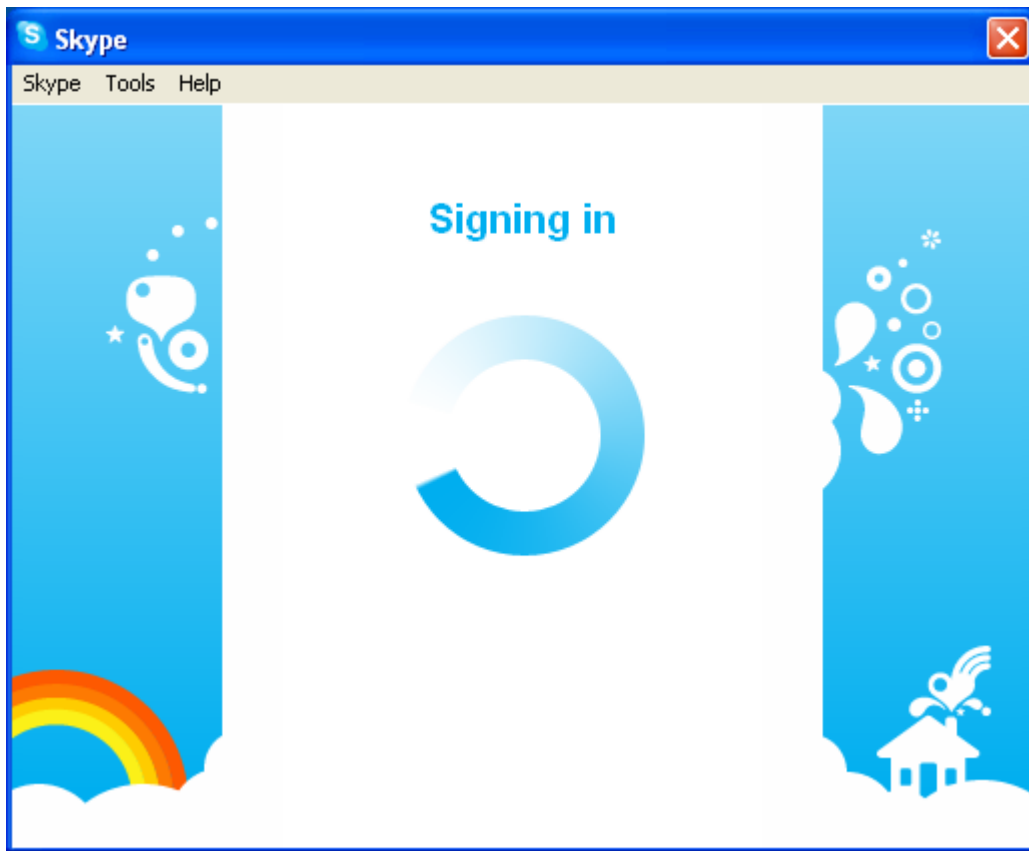


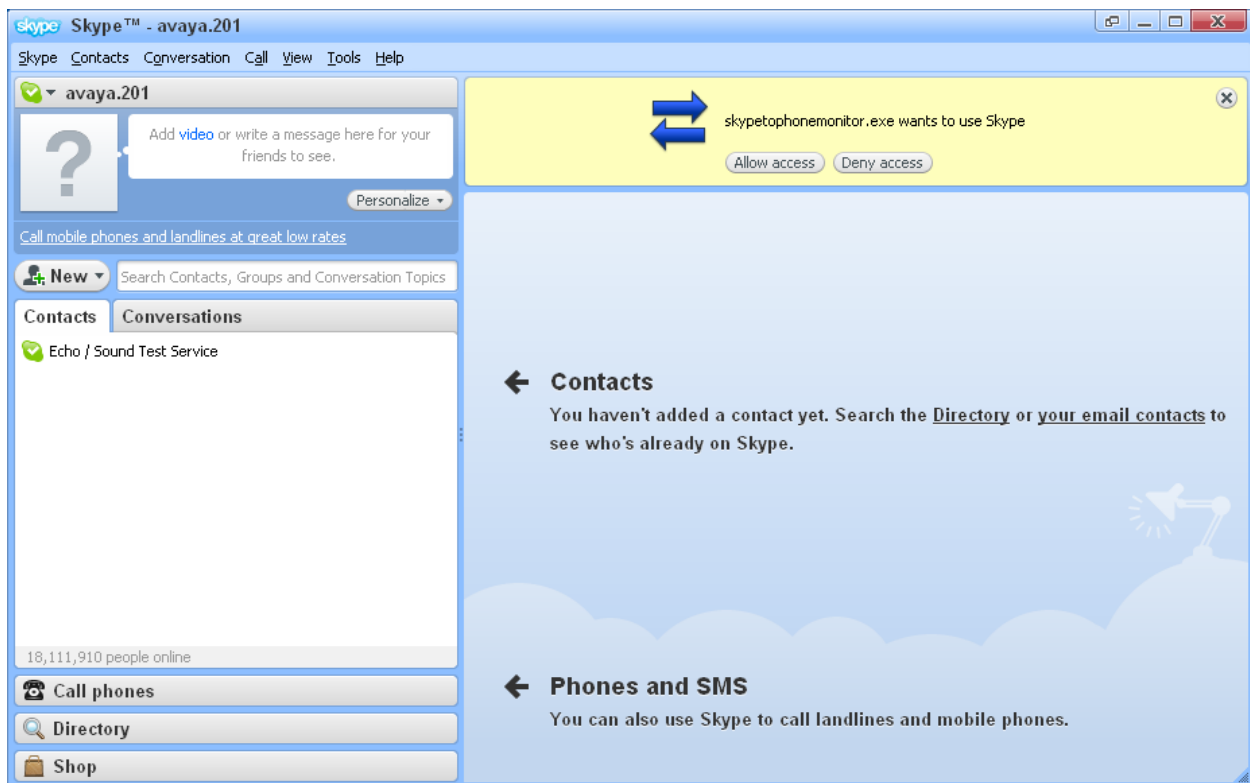
Figure 41: Services Screen

At this point a Skype instance will be started for each Skype client that has been configured and they will be automatically logged in by the system. This will appear only briefly, and disappear on its own.



**Figure 42: Skype Login Indication Screen**

The first time that a client is logged into Skype the dialog screen shown below will appear on the server. Click “Allow access” to complete the Skype user registration.



**Figure 43: Skype Login Indication Screen**

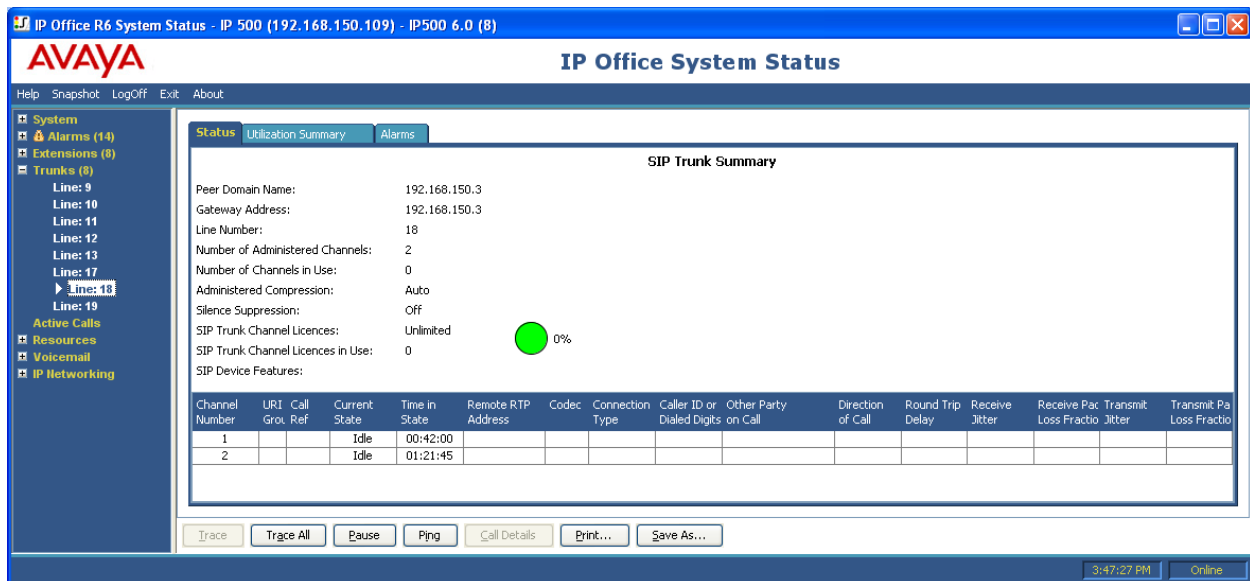
## 8. General Test Approach and Test Results

All tests were performed manually. Only functional testing was performed: no performance testing was done. All tests produced the expected result.

Since Empix evolve Skypetophone runs as an application and not as a Windows service, it can only recover automatically after interruption due to power failure by dispensing with Windows login security.

## 9. Verification Steps

Examine the status of the SIP trunk which connects IP Office to the Empix evolve Skypetophone application with the IP Office System Status utility, and verify that the allocated trunk members are in the “idle” state.



## 10. Conclusion

These Application Notes describe the compliance testing of the Empix evolve Skypetophone server with Avaya IP Office. The various features of the Empix evolve Skypetophone server which involve interaction with telephony were tested. A detailed description of the configuration required for both the Avaya IP Office and the Empix evolve Skypetophone is documented within these Application Notes.

## 11. Additional References

This section references documentation relevant to these Application Notes. The Avaya product documentation is available at <http://support.avaya.com>.

- [1] *Avaya Office 6.0 Manager 8.0*, May 2010, Document Number 15-601011
- [2] *Avaya IP Office Release 6 H323 IP Telephone Installation*, March 2010, 15-601046
- [3] *Installing and configuring xtension evolve*, July 2010, ver 2.2



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