

# SIPTAPI

**A TAPI service provider for SIP**  
**klaus.darilion@ipcom.at**

# Note

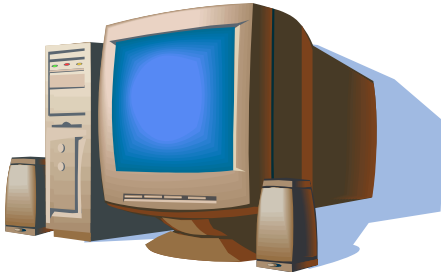
- If you can't get SIPTAPI to work, feel free to contact me, but:
- never ever contact me without reading all the READMEs, tutorials and other documentation included in the download package!!!!
- I do not have the time to tell people again and again all the things which is already written down in the docs
- Thus, if you still have problems, then I will assist you, if your problem description contains at least:
  - Operating System: XP, Vista...? 32/64 bit?
  - TAPI application: dialer.exe, Outlook, CRM...? 32/64bit?
  - The SIP phone you are using: SNOM, eyebeam, ...?
  - The SIP server you are using, e.g. Asterisk, sipX, Kamailio, or a hosted service (sipgate...)?

# Introduction

- SIPTAPI is a SIP based call-control client which can be used to initiate phone calls.
- SIPTAPI is not a full SIP client, thus a dedicated SIP client – a SIP softphone or hardphone – is needed
- SIPTAPI will instruct the SIP client to call a certain number
- SIPTAPI can be used with SIP proxies (Kamailio) or PBXs (Asterisk)
- If the SIP server is hosted by a service provider then SIPTAPI may not work → see alternative usage below

# Usage

1.



→ dial callee's number in a TAPI application (e.g. Outlook, Phoner, dialer.exe)

2.



→ caller's phone rings, pick up!

3.

→ SIPTAPI tells the caller's phone to dial the callee's number (SIP REFER request)

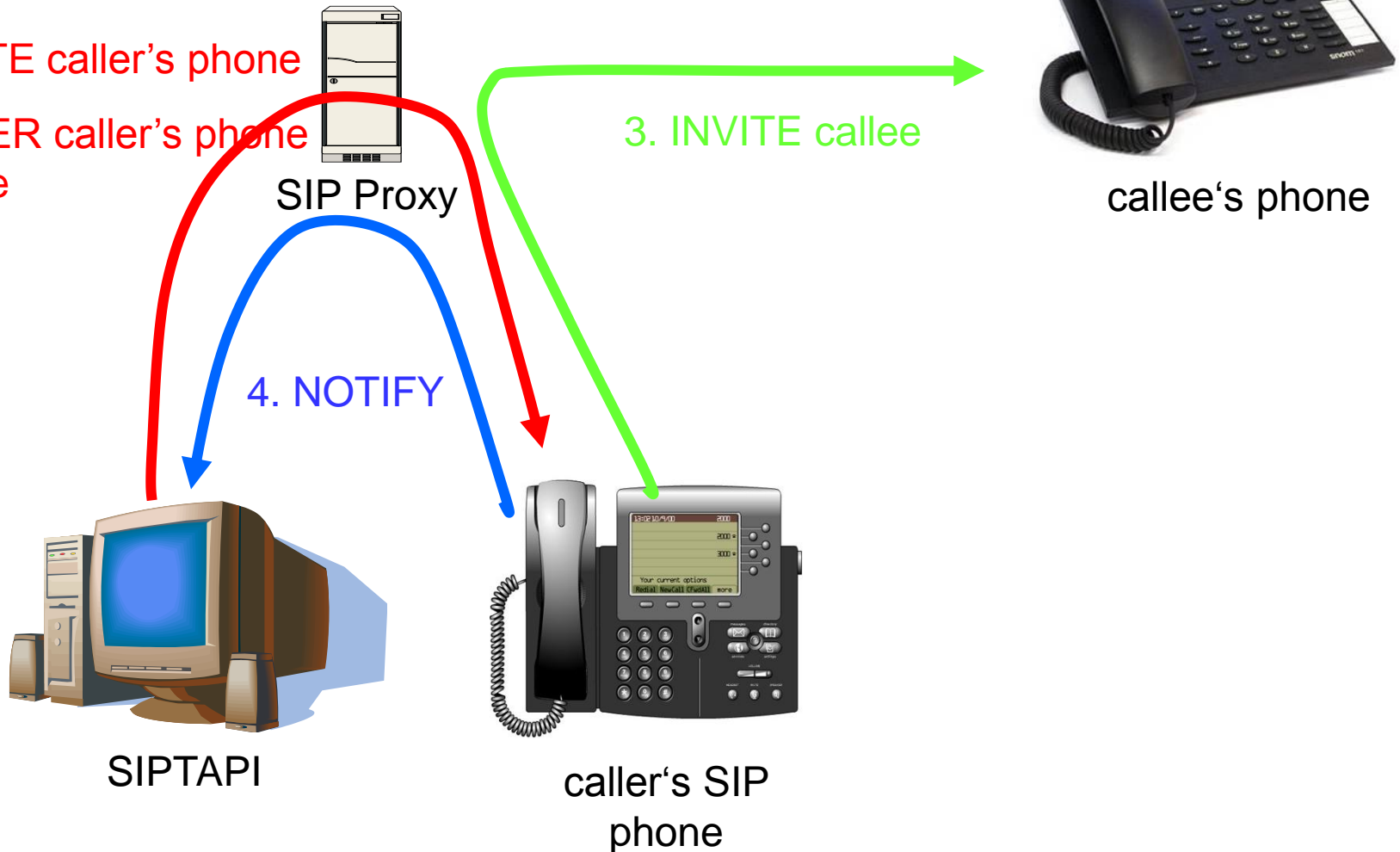
4.

→ callee's phone rings



# Call Flow with a SIP Proxy (Kamailio)

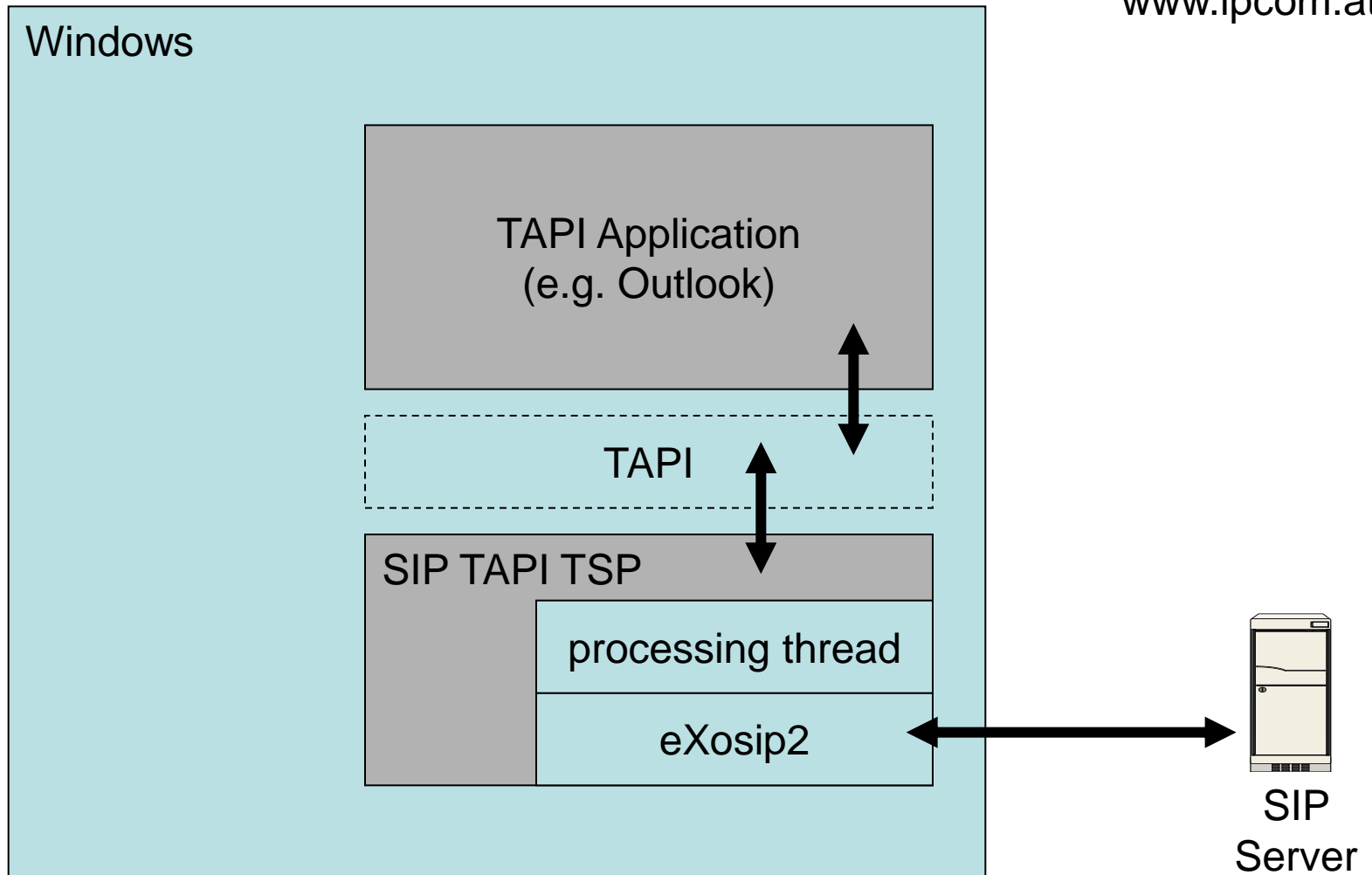
1. INVITE caller's phone
2. REFER caller's phone to callee
5. BYE



# Call Flow with Asterisk

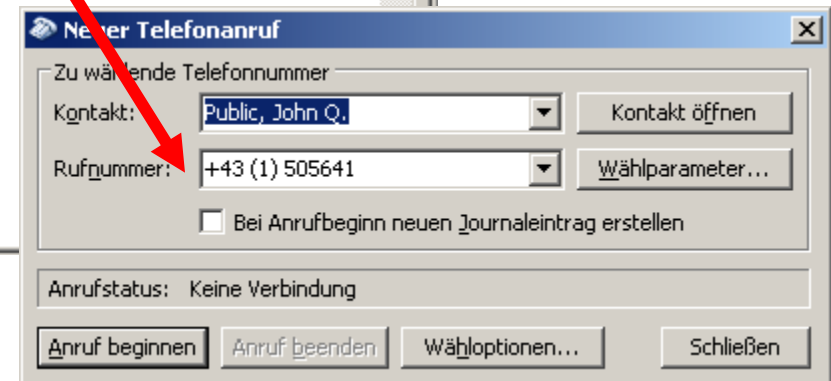
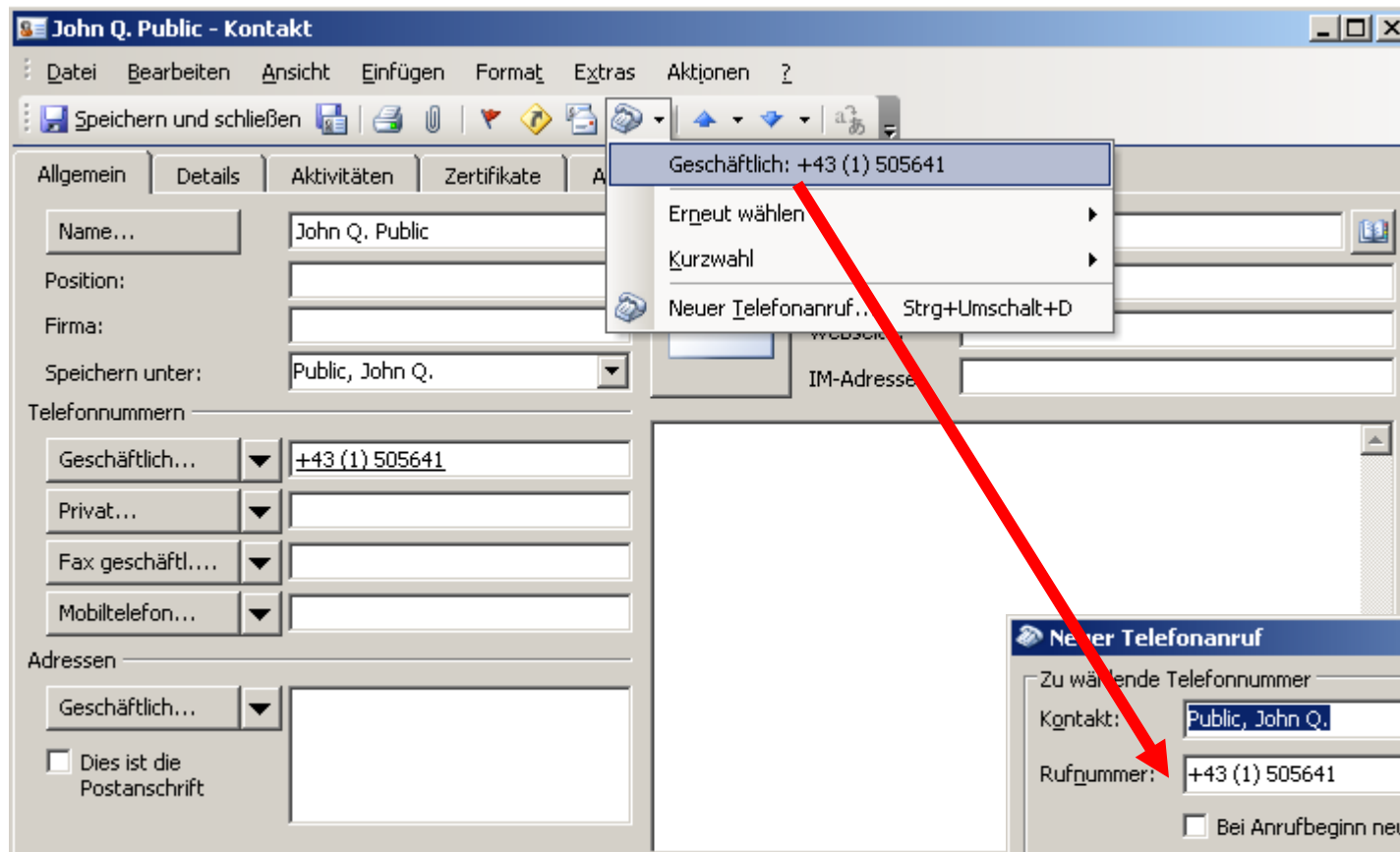


# TAPI structure



# TAPI dialing

- e.g. Outlook





# TAPI dialing

- e.g. Outlook



**Neuer Telefonanruf**

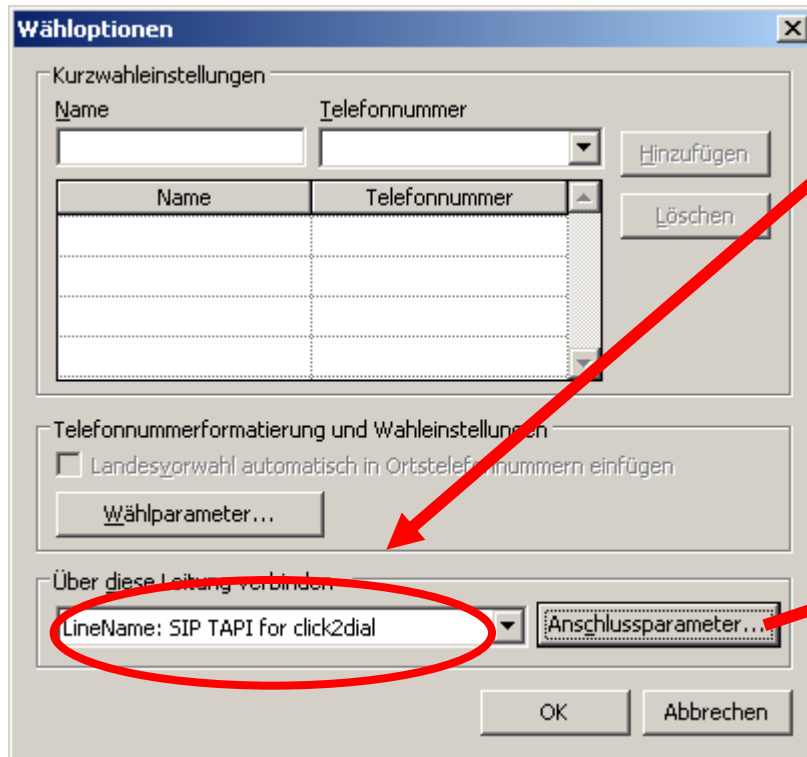
Zu wählende Telefonnummer

Kontakt:

Rufnummer:

☐ Bei Anrufbeginn neuen Journaleintrag erstellen

Anrufstatus: Keine Verbindung



**Wähloptionen**

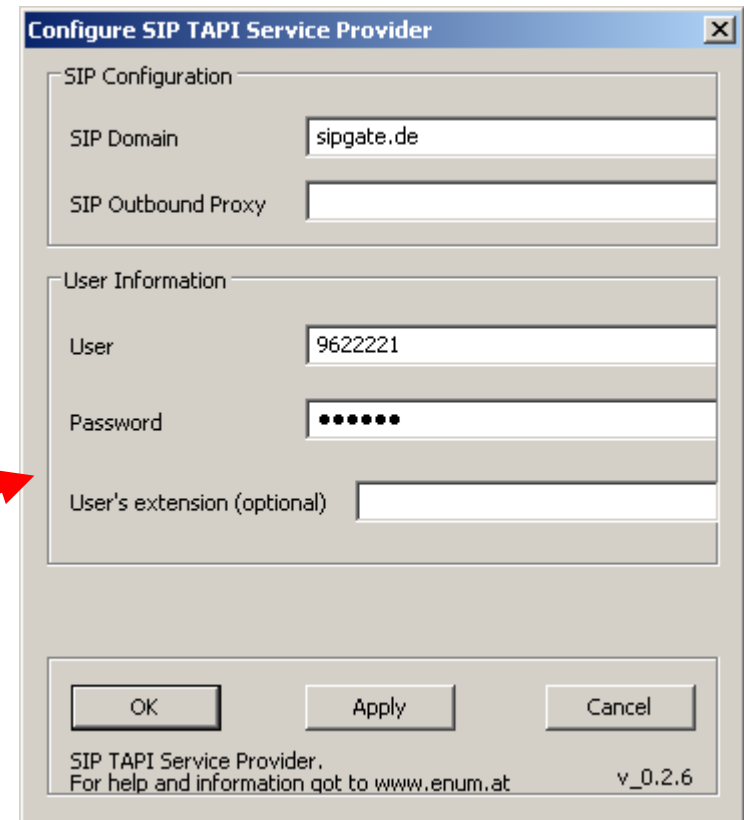
Kurzwahleinstellungen

Name	Telefonnummer
<input type="text"/>	<input type="text"/>
<input type="button" value="Hinzufügen"/>	
<input type="button" value="Löschen"/>	
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>
<input type="text"/>	<input type="text"/>

Telefonnummerformatierung und Wahleinstellungen

☐ Landesvorwahl automatisch in Ortstelefonnummern einfügen

Über diese Leitung verbinden:



**Configure SIP TAPI Service Provider**

SIP Configuration

SIP Domain:

SIP Outbound Proxy:

User Information

User:

Password:

User's extension (optional):

SIP TAPI Service Provider.  
For help and information go to [www.enum.at](http://www.enum.at) v\_0.2.6

# Configuration

**Configure SIP TAPI Service Provider**

SIP Configuration

SIP Domain: sipgate.de

SIP Outbound Proxy:

User Information

User: 9622221

Password: .....

User's extension (optional):

OK Apply Cancel

SIP TAPI Service Provider.  
For help and information got to [www.enum.at](http://www.enum.at) v\_0.2.6

**Telefon- und Modemoptionen**

Wählregeln Modems Erweitert

Folgende Telefonieanbieter sind auf diesem Computer installiert:

Anbieter:

- Dienstanbieter für Universalmodem 5
- Microsoft H.323-Telefoniedienstanbieter
- Microsoft HID-Telefon-TSP
- Microsoft Multicastkonferenz-TAPI-Dienstanbieter
- NDIS-Proxy-TAPI-Dienstanbieter
- SIP TAPI Service Provider**
- TAPI-Kernelmodus-Dienstanbieter

Hinzufügen... Entfernen Konfigurieren...

Schließen Abbrechen Übernehmen

# Configuration

- **SIP Domain**
  - This is the SIP domain, either a fully qualified domain name or the IP address of the SIP server. Examples:
    - sipgate.de
    - 1.2.3.4
- **SIP Outbound Proxy**
  - This field is optional and usually not needed. It should only be used if your SIP provider requires you to use an outbound proxy. In doubt leave this field empty. Examples:
    - 1.1.1.1
    - 2.2.2.2:6060
- **User**
  - The SIP username (SIP-ID) assigned to you by your SIP provider. This is just the userpart without the domain. Examples:
    - klaus.darilion
    - 00431234567
- **Password**
  - The SIP password
- **User's extension**
  - This field is optional and usually not needed. If a user's extension is specified, the SIPTAPI will call this extension instead of its own extension. This is sometimes needed with Asterisk installations where you have a dedicated SIP account for the SIPTAPI and a dedicated SIP account for the SIP phone, or if the SIP username can not be used for dialing and an extension must be used.

# Alternative Usage

- Sometimes the SIP provider/PBX does not apply correct routing to REFER requests, or just denies them
- If the following items are fulfilled, the alternative approach can be used
  - the SIP phone has a static IP address and uses a static port
  - there is direct IP connectivity between the SIPTAPI PC and the SIP phone

# Alternative Call Flow

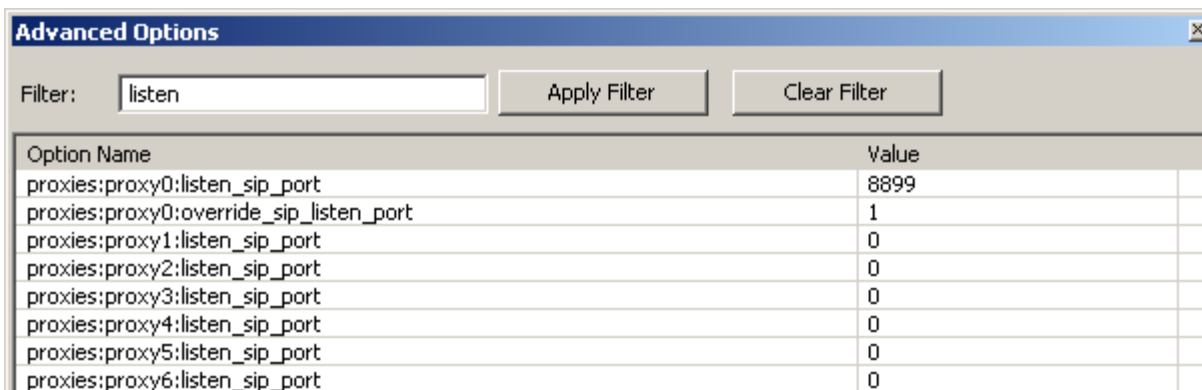


# Alternative Configuration

- Example
  - SIPTAPI is on PC with IP address 192.168.1.2
  - SIP phone uses static IP address 192.168.1.3
  - SIP phone uses static SIP port 8899
- SIP Domain
  - the SIP domain specified by your service provider
- SIP Outbound Proxy
  - 192.168.1.3:8899
- User
  - the SIP user name specified by your service provider
- Password
  - the SIP password specified by your service provider
- User's extension
  - leave this field empty

# Alternative Configuration for Xlite 3 and Eyebeam

- Xlite per default uses a dynamic SIP port ☹
- How to get Xlite a static IP address?
  - Xlite uses the IP address of the PC. Thus make sure the PC has a static IP address
- How to get Xlite a static SIP port?
  - Start Xlite
  - Dial \*\*\*7469 → a pop window with advanced options appears. Filter for "listen":



Option Name	Value
proxies:proxy0:listen_sip_port	8899
proxies:proxy0:override_sip_listen_port	1
proxies:proxy1:listen_sip_port	0
proxies:proxy2:listen_sip_port	0
proxies:proxy3:listen_sip_port	0
proxies:proxy4:listen_sip_port	0
proxies:proxy5:listen_sip_port	0
proxies:proxy6:listen_sip_port	0

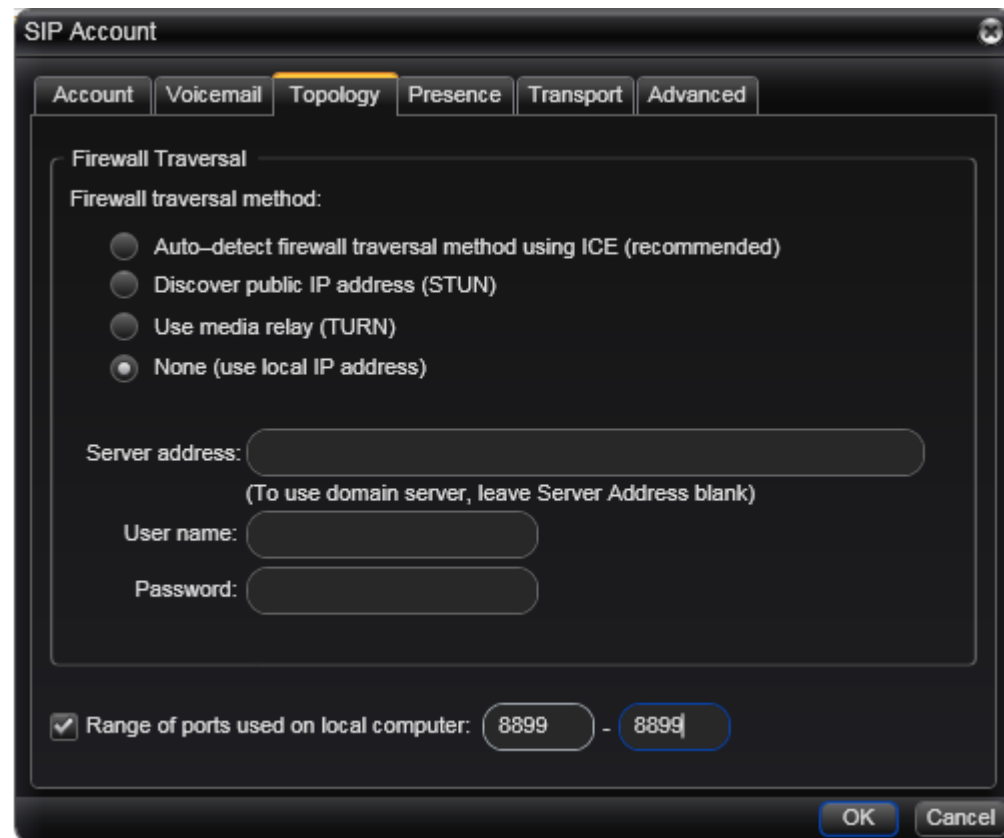
# Alternative Configuration for Xlite 3 and Eyebeam

- change **listen SIP port** for the respective SIP account to a certain SIP port, e.g.: **8899**
- set **override\_sip\_listen\_port** to **1**
- restart Xlite
- configure in SIPTAPI outbound proxy:  
**127.0.0.1:8899**
- Configure “user extension” with the Sip clients username and 127.0.0.1:port as hostpart, e.g:  
**klaus.darilion@127.0.0.1:8899**



# Alternative Configuration for Bria and Xlite 4

- How to get Xlite a static IP address?
  - Xlite4/Bria uses the IP address of the PC. Thus make sure the PC has a static IP address
- How to get a static SIP port?
  - limit the port range to a single port:

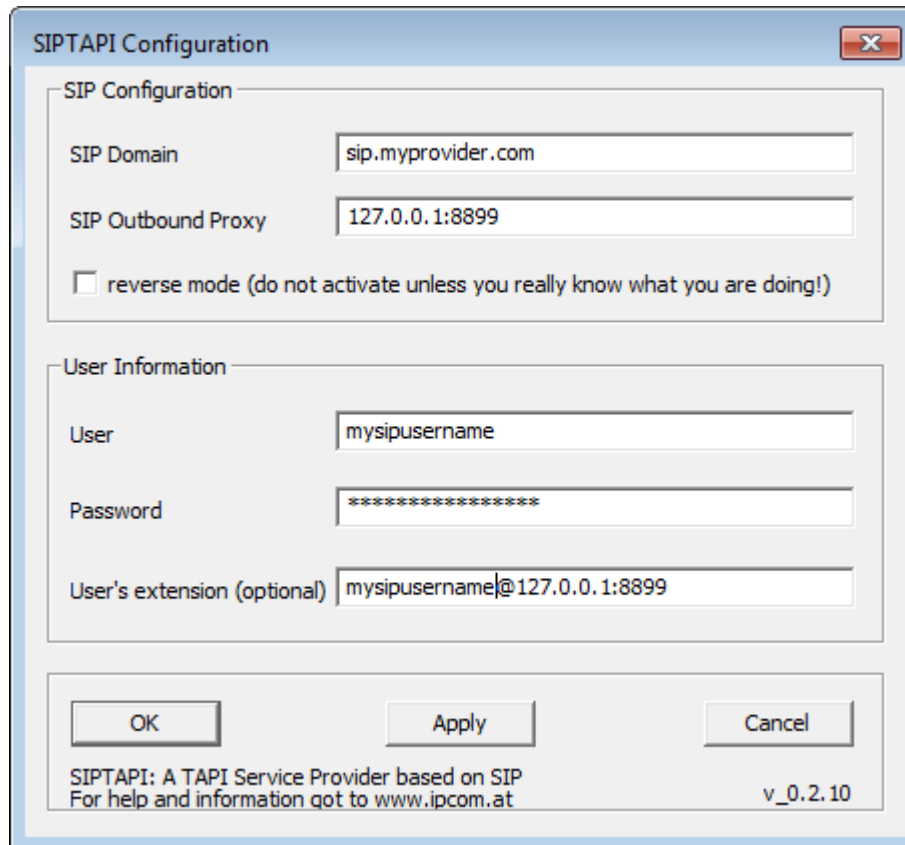


# Alternative Configuration for Bria and Xlite 4

- Softphone → Account Settings → choose the respective account → Topology → Range of ports ... → limit to one port, e.g: **8899-8899**
- restart Xlite4/Bria
- configure in SIPTAPI **outbound proxy:**  
**127.0.0.1:8899**
- Configure “**user extension**” with the SIP clients’ username and 127.0.0.1:port as hostpart, e.g:  
**klaus.darilion@127.0.0.1:8899**

# Alternative Configuration for Bria/Xlite/Eyebeam

- SIPTAPI screenshot (e.g. client is fixed to port 8899)
- Note: this requires at least version 0.2.10!!!



The image shows a screenshot of the 'SIPTAPI Configuration' dialog box. It has a title bar with a close button. The dialog is divided into two main sections: 'SIP Configuration' and 'User Information'. In the 'SIP Configuration' section, there are text boxes for 'SIP Domain' (containing 'sip.myprovider.com') and 'SIP Outbound Proxy' (containing '127.0.0.1:8899'). Below these is a checkbox for 'reverse mode (do not activate unless you really know what you are doing!)' which is currently unchecked. The 'User Information' section contains text boxes for 'User' (containing 'mysipusername'), 'Password' (containing a series of asterisks), and 'User's extension (optional)' (containing 'mysipusername@127.0.0.1:8899'). At the bottom of the dialog are three buttons: 'OK', 'Apply', and 'Cancel'. Below the buttons, there is a footer line that reads 'SIPTAPI: A TAPI Service Provider based on SIP For help and information got to www.ipcom.at' and the version number 'v\_0.2.10'.

SIPTAPI Configuration

SIP Configuration

SIP Domain: sip.myprovider.com

SIP Outbound Proxy: 127.0.0.1:8899

☐ reverse mode (do not activate unless you really know what you are doing!)

User Information

User: mysipusername

Password: \*\*\*\*\*

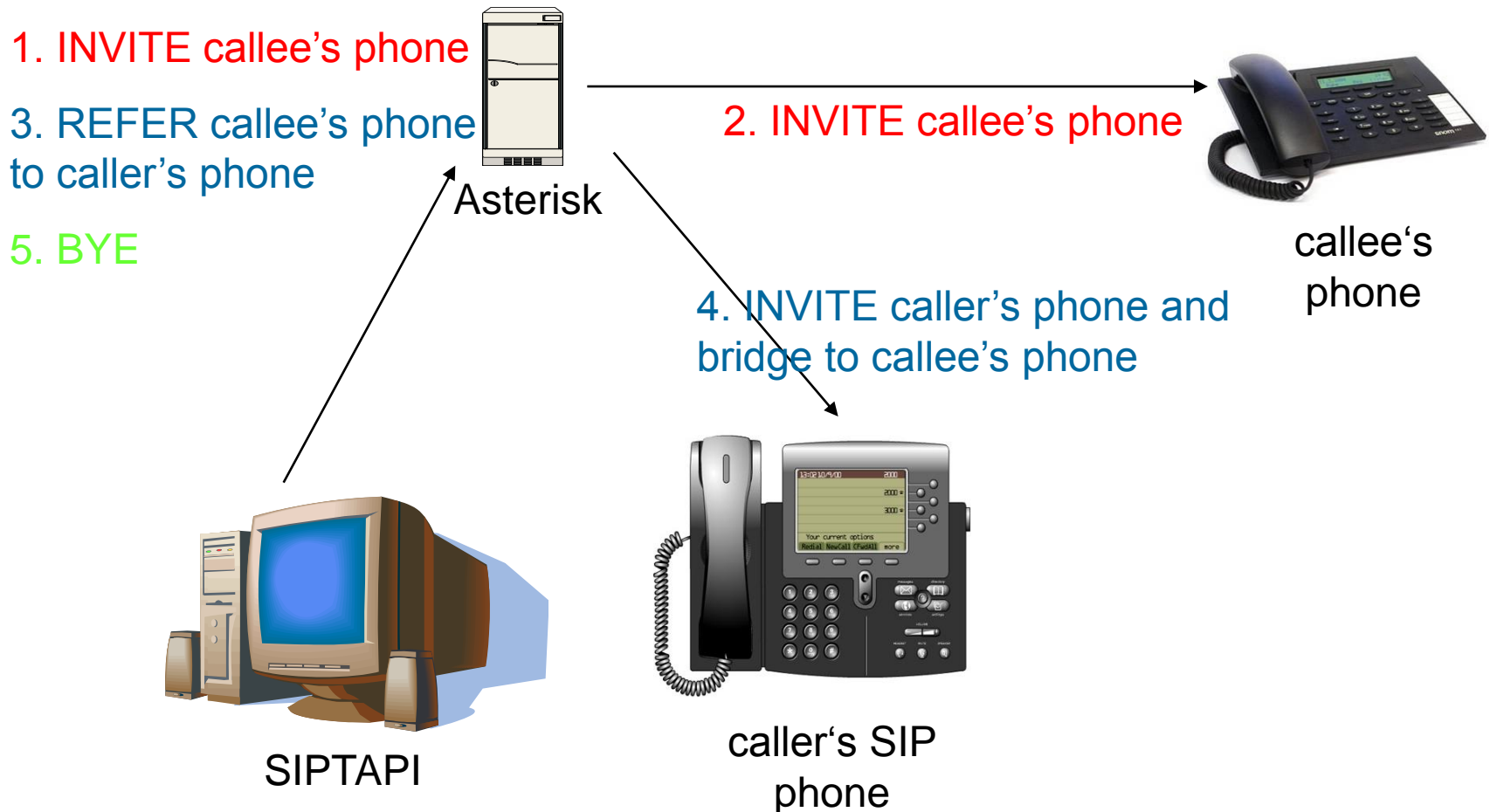
User's extension (optional): mysipusername@127.0.0.1:8899

OK Apply Cancel

SIPTAPI: A TAPI Service Provider based on SIP  
For help and information got to [www.ipcom.at](http://www.ipcom.at) v\_0.2.10

# Reverse Mode

- Normally SIPTAPI calls the user's phone and then refers it to the real target
- In reverse mode SIPTAPI will call directly the target, and once the target answers, it gets referred to the user's phone.
- Note: In reverse mode you won't get any audible call progress indication (e.g. ringback, busy tone ...)
- Note: "reverse mode" does not work with the "alternative configuration"



# Technology

- Microsoft TAPI  
<http://msdn.microsoft.com/library/default.asp?url=/library/en-us/dnanchor/html/tapitspimspi.asp>
- asttapi – TAPI provider for asterisk  
<http://sourceforge.net/projects/asttapi/>  
© Nick Knight
- eXosip2/osip – SIP stack + high level API  
<http://www.gnu.org/software/osip/osip.html>  
<http://savannah.nongnu.org/projects/exosip/>  
© Aymeric Moizard

# Limitations

- no STUN support → NAT traversal requires SIP proxy with NAT traversal or outboundproxy (rport supported)
- some TAPI applications require strange phone number formats, e.g. Outlook: +43 (1) 505641  
→ *+country code (local area code) number*

# how to get it

- <https://sourceforge.net/projects/siptapi/>
- Website: <http://www.ipcom.at/en/telephony/siptapi/>
- License: GPL
- Installation:
  - copy siptapi.tsp into the windows\system32 directory
  - ControlPanel → PhoneAndModemOptions → Advanced → Add → SIP TAPI Service Provider
  - SIP TAPI Service Provider → Configure...
  - configure your username, password, SIP domain and outboundproxy

# Debugging

- use Sysinternals DebugView to capture log messages (run as Administrator on Vista and above):  
<http://www.sysinternals.com/ntw2k/freeware/debugview.shtml>
- use Wireshark to capture the SIP packets
- Log files
  - Free version: c:\siptapi\_0.2.log
  - Commercial version: c:\siptapi.log and c:\siptapi\_osip.log