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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

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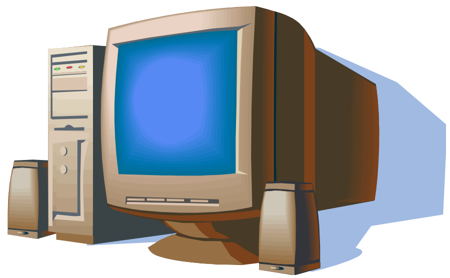
1.



➔ dial callee‘s numer 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)

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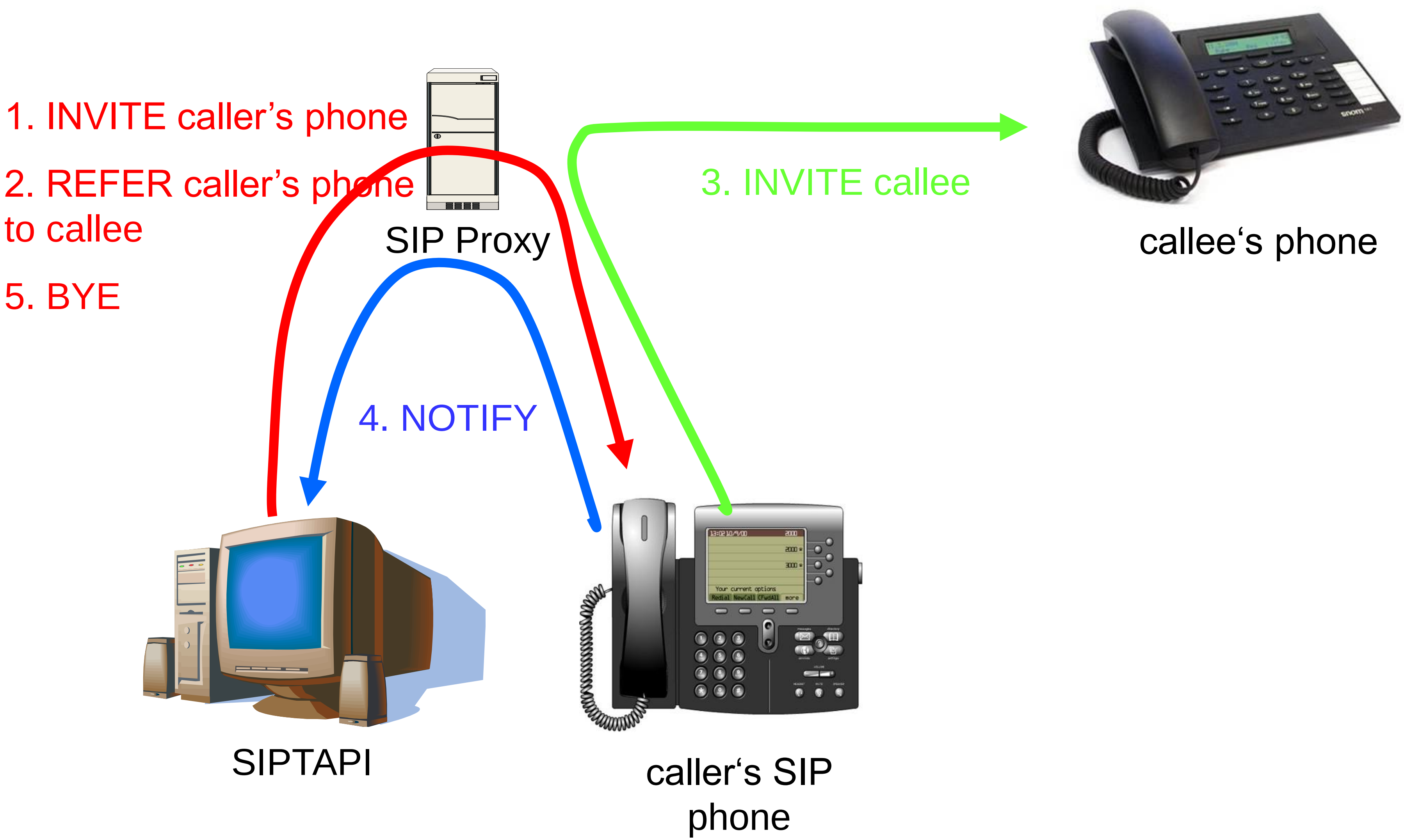

Call Flow with Asterisk

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TAPI structure

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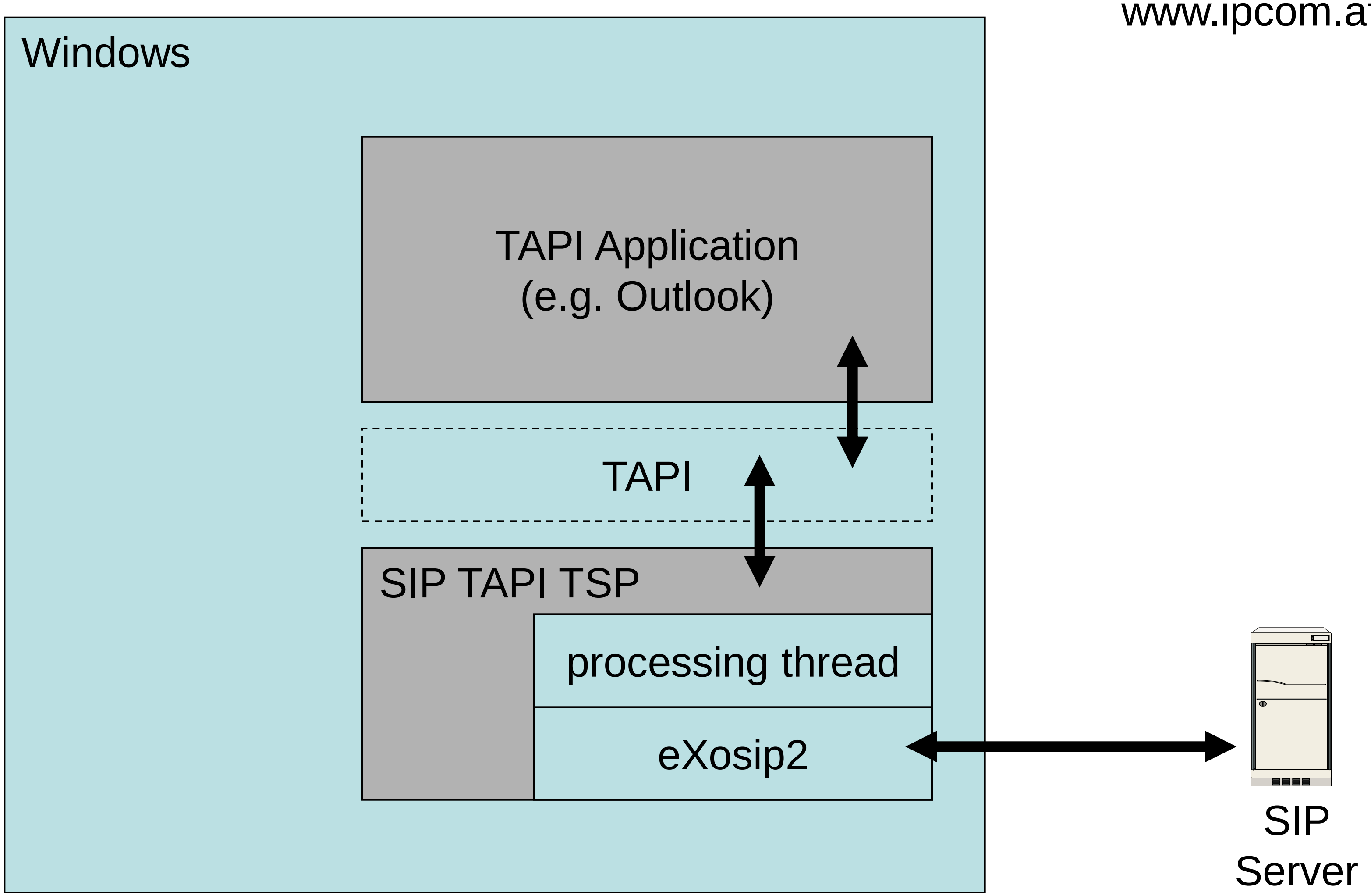

TAPI dialing

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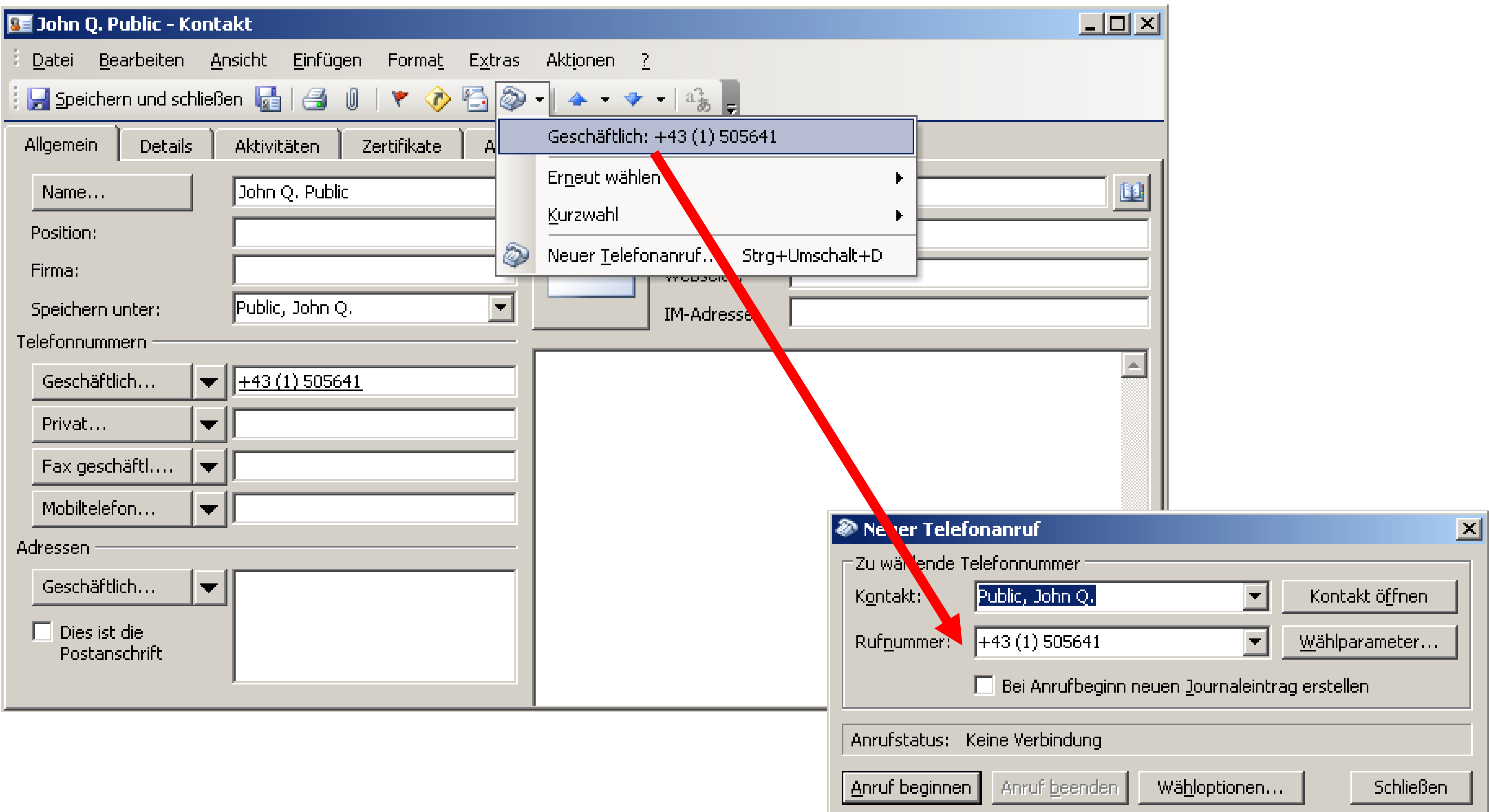
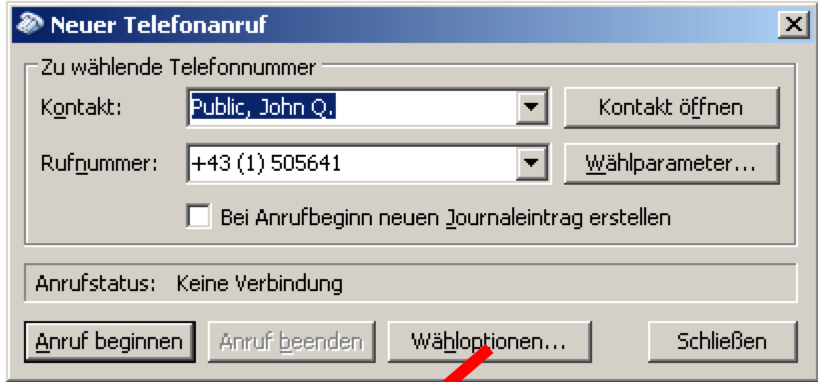
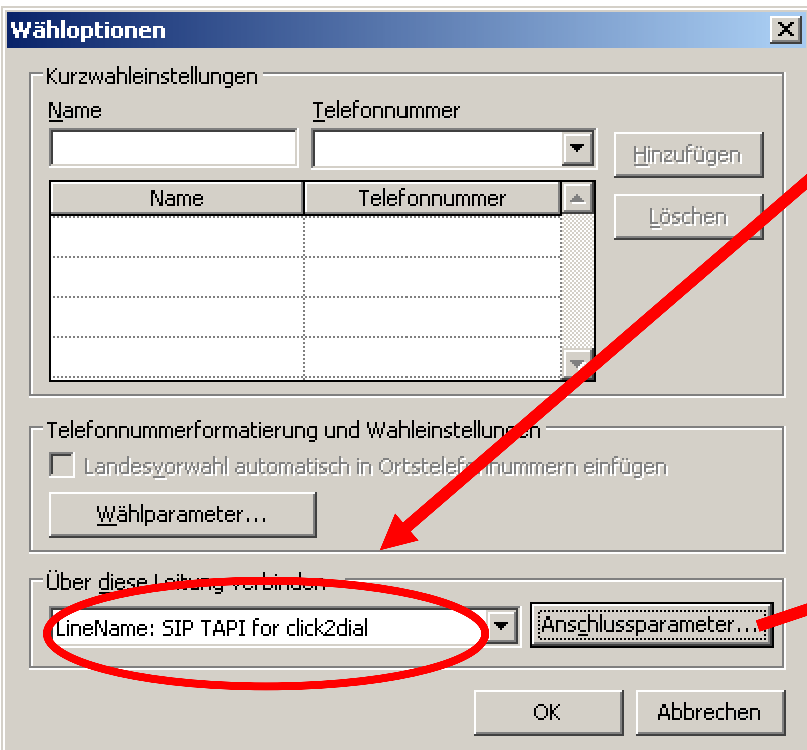
- e.g. Outlook

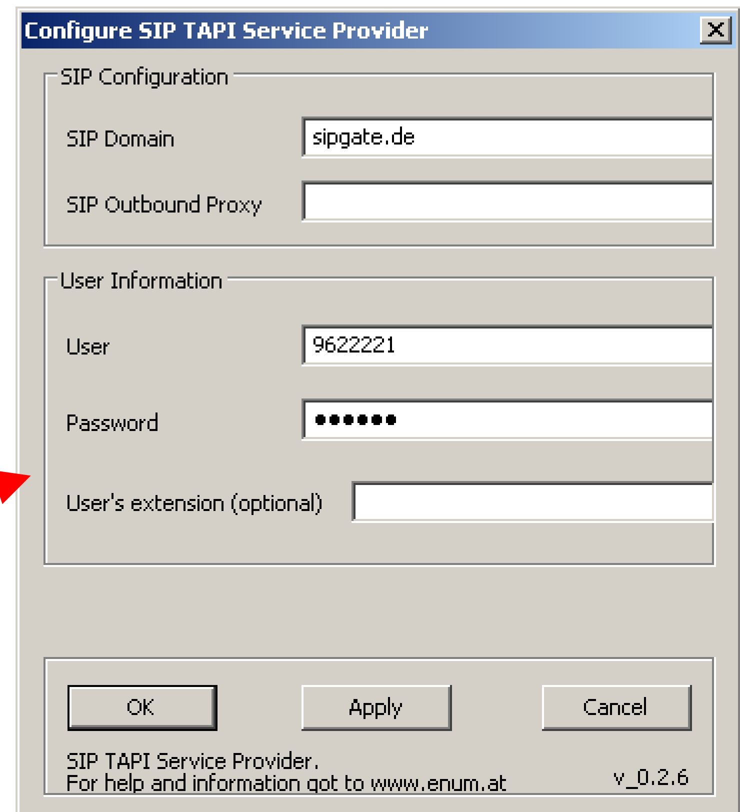


TAPI dialing

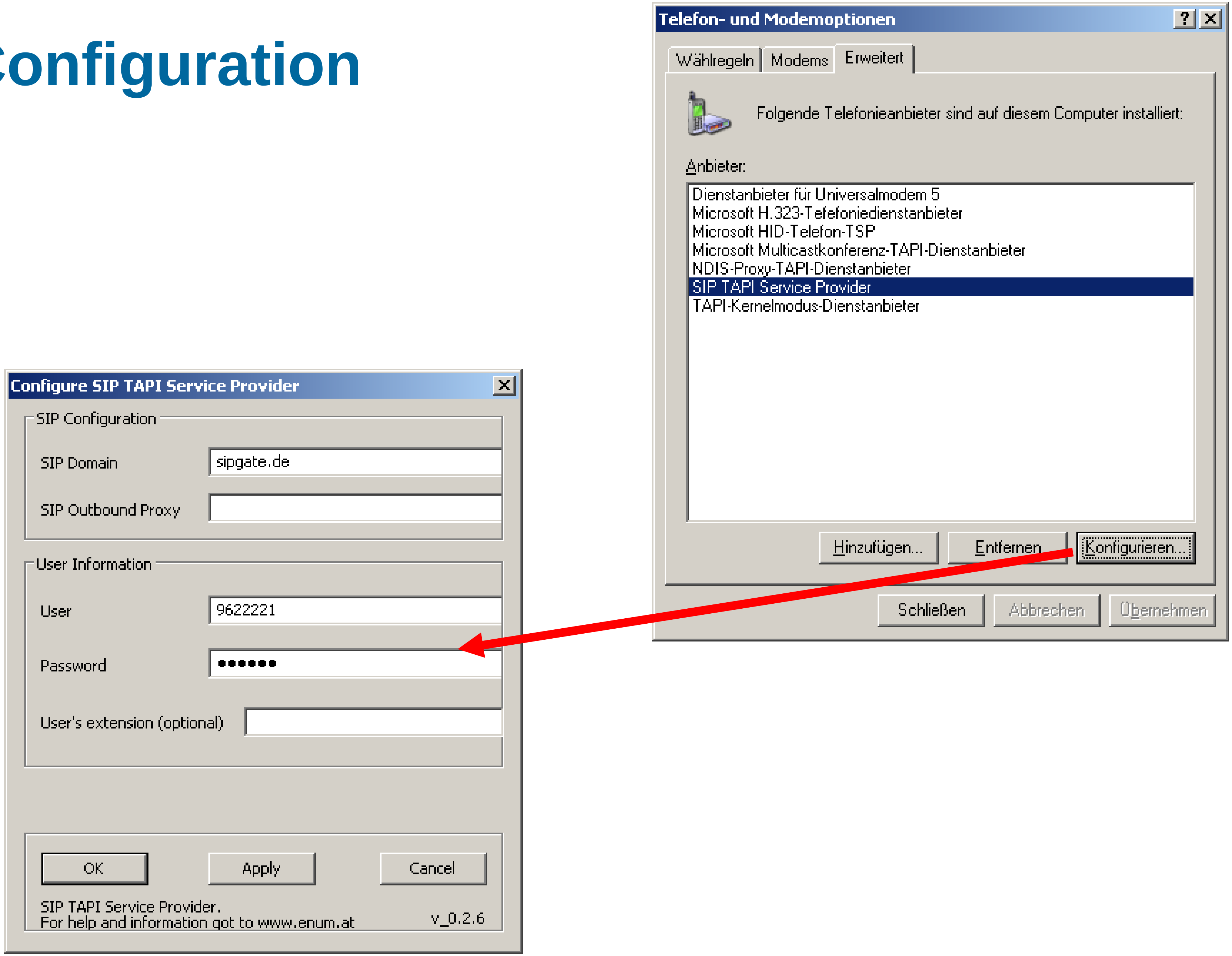
- e.g. Outlook







Configuration



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

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- 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

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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

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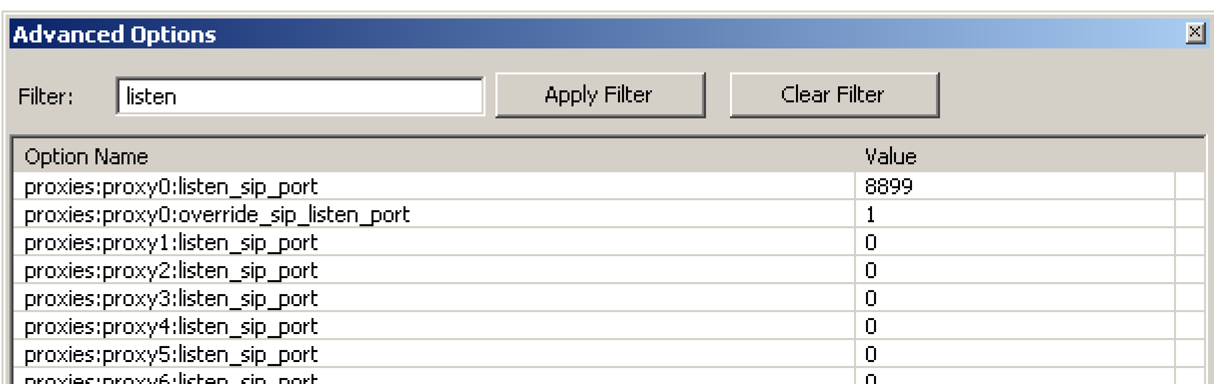
- Xlite per default uses a dynamic SIP port ☹
- How to get Xlite a static IP address?
  - Xlite uses the IP address of the PC. Thus make sure the PC has a static IP address
- How to get Xlite a static SIP port?
  - Start Xlite
  - Dial ***7469 → a pop window with advanced options appears. Filter for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

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- 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

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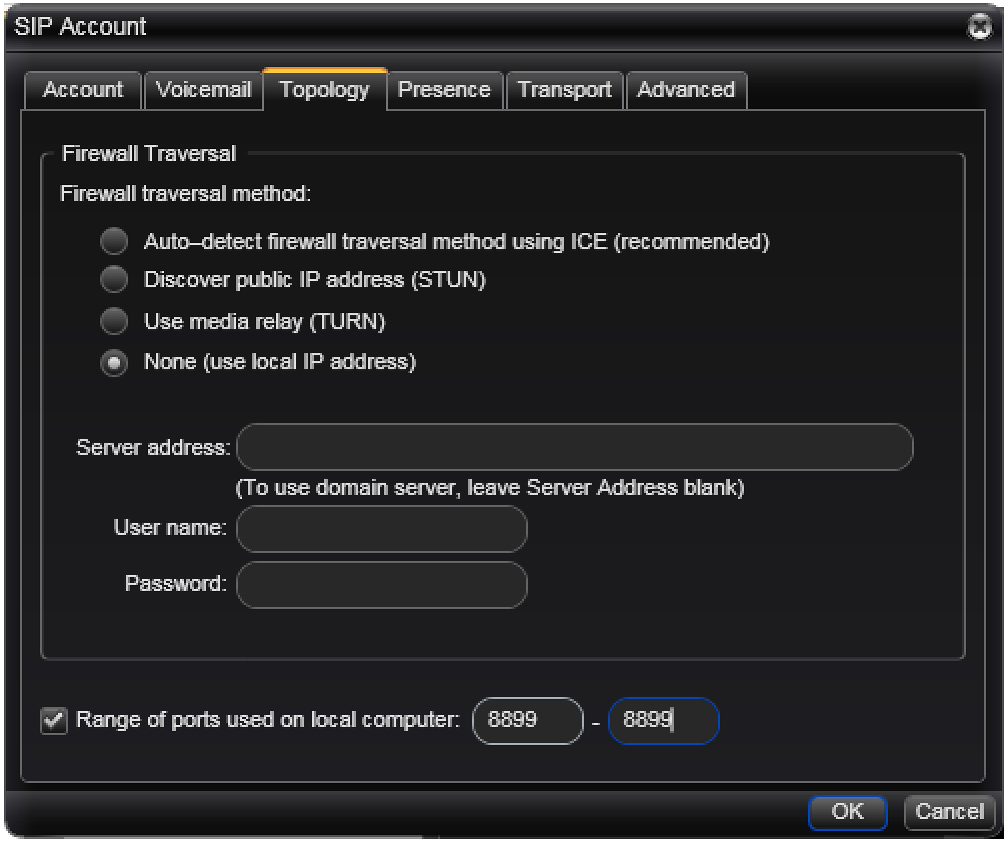
- How to get Xlite a static IP address?
  - Xlite4/Bria uses the IP address of the PC. Thus make sure the PC has a static IP address
- How to get a static SIP port?
  - limit the port range to a single port:



Alternative Configuration for Bria and Xlite 4

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- Softphone → Account Settgins → 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

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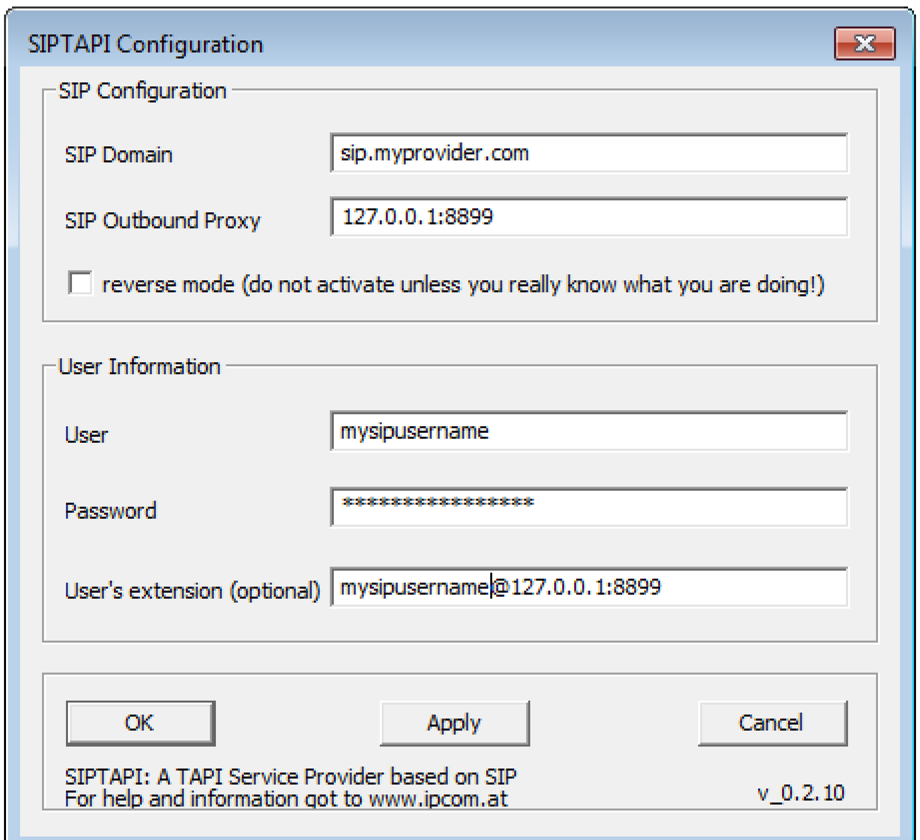
- SIPTAPI screenshot (e.g. client is fixed to port 8899)
- Note: this requires at least version 0.2.10!!!



Reverse Mode

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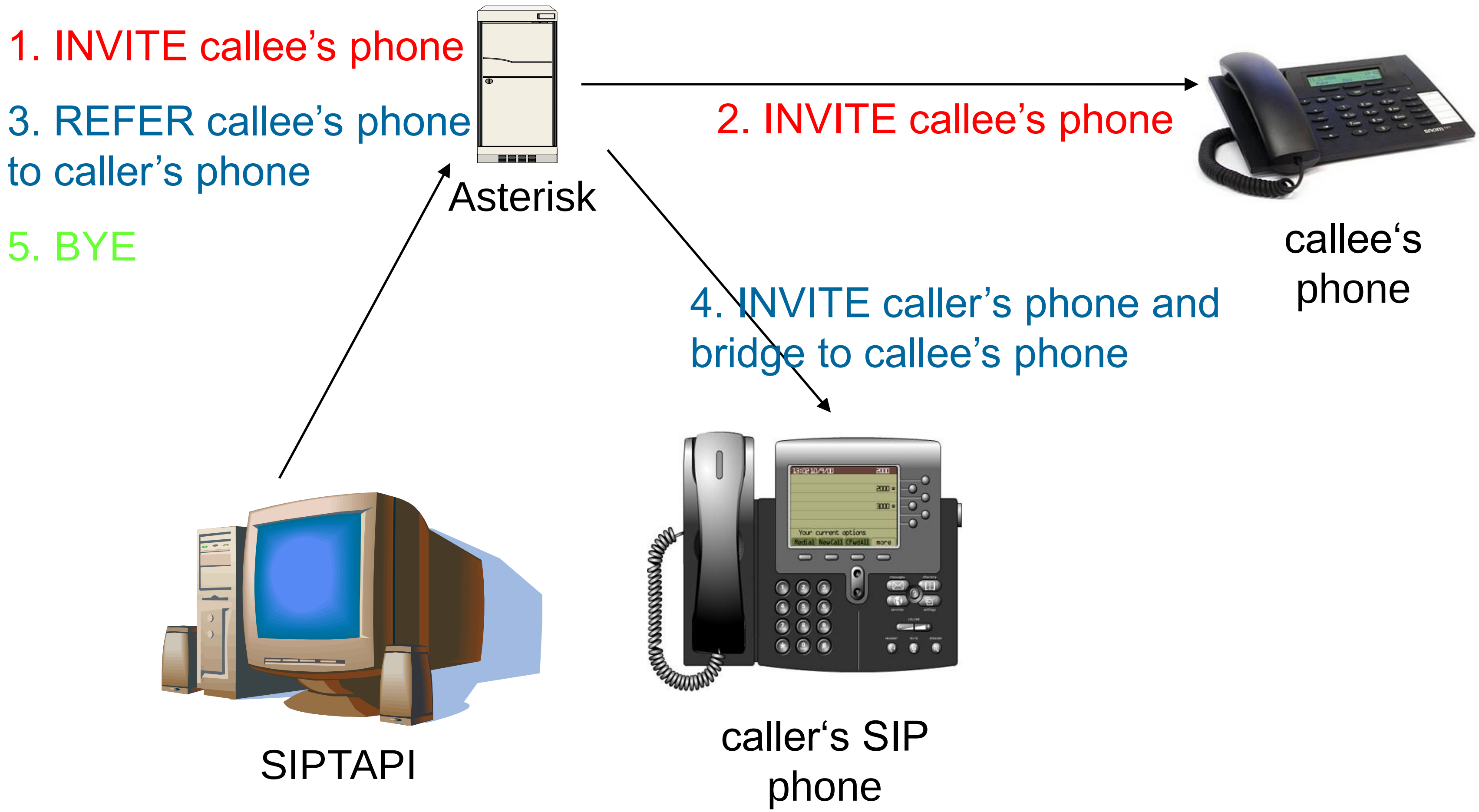
- Normally SIPTAPI calls the user's phone and then refers it to the real target
- In reverse mode SIPTAPI will call directly the target, and once the target answers, it gets refered to the user's phone.
- Note: In reverse mode you wont get any audible call progress indication (e.g. ringback, busy tone ...)
- 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

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- 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