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SIPTAPI

A click2dial application, based on asttapi and eXosip2

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Introduction

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- 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 “tell” the SIP client to call a certain number

Usage

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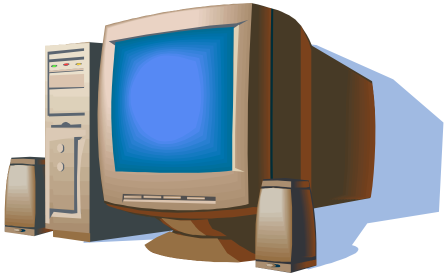
1.



➔ 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

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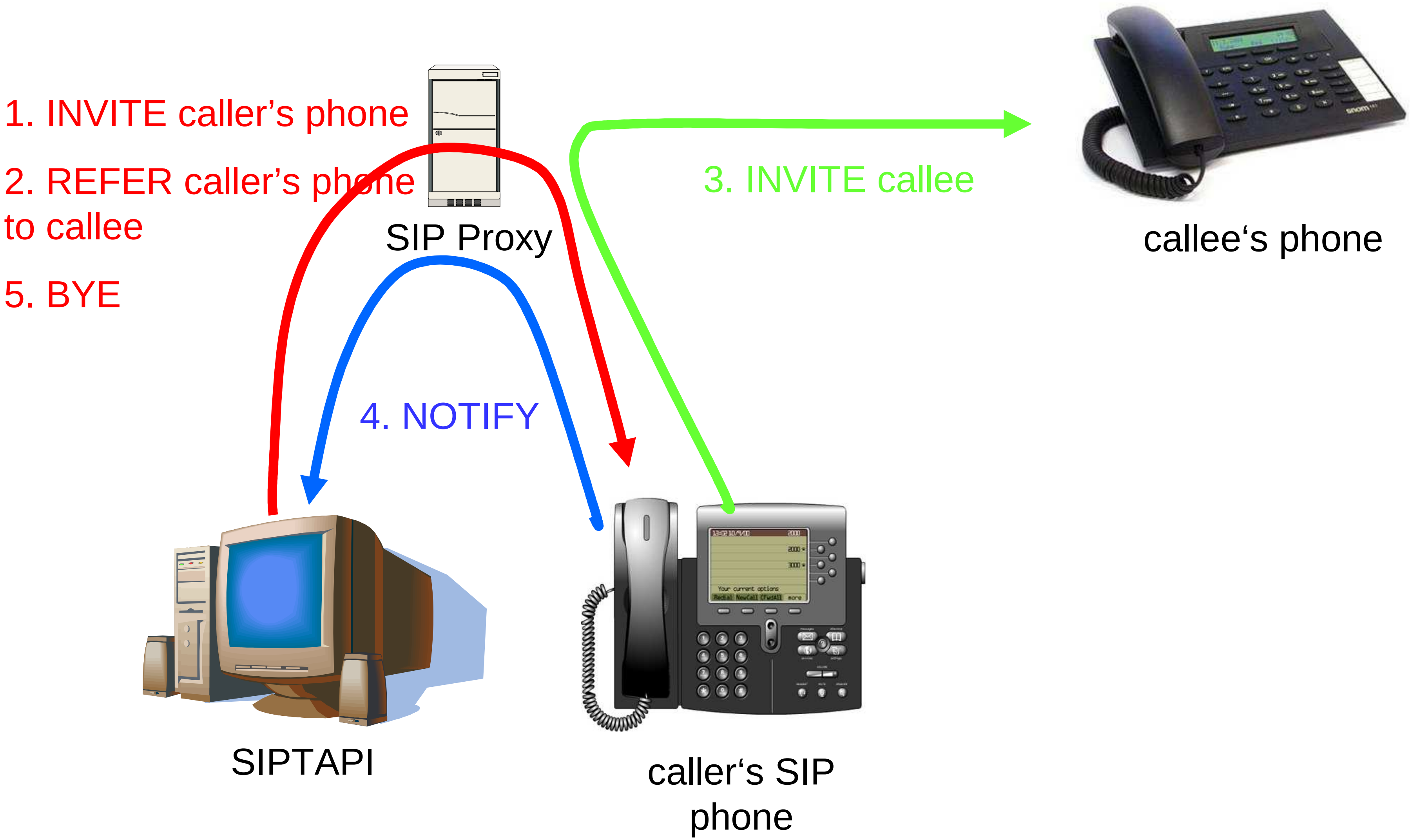
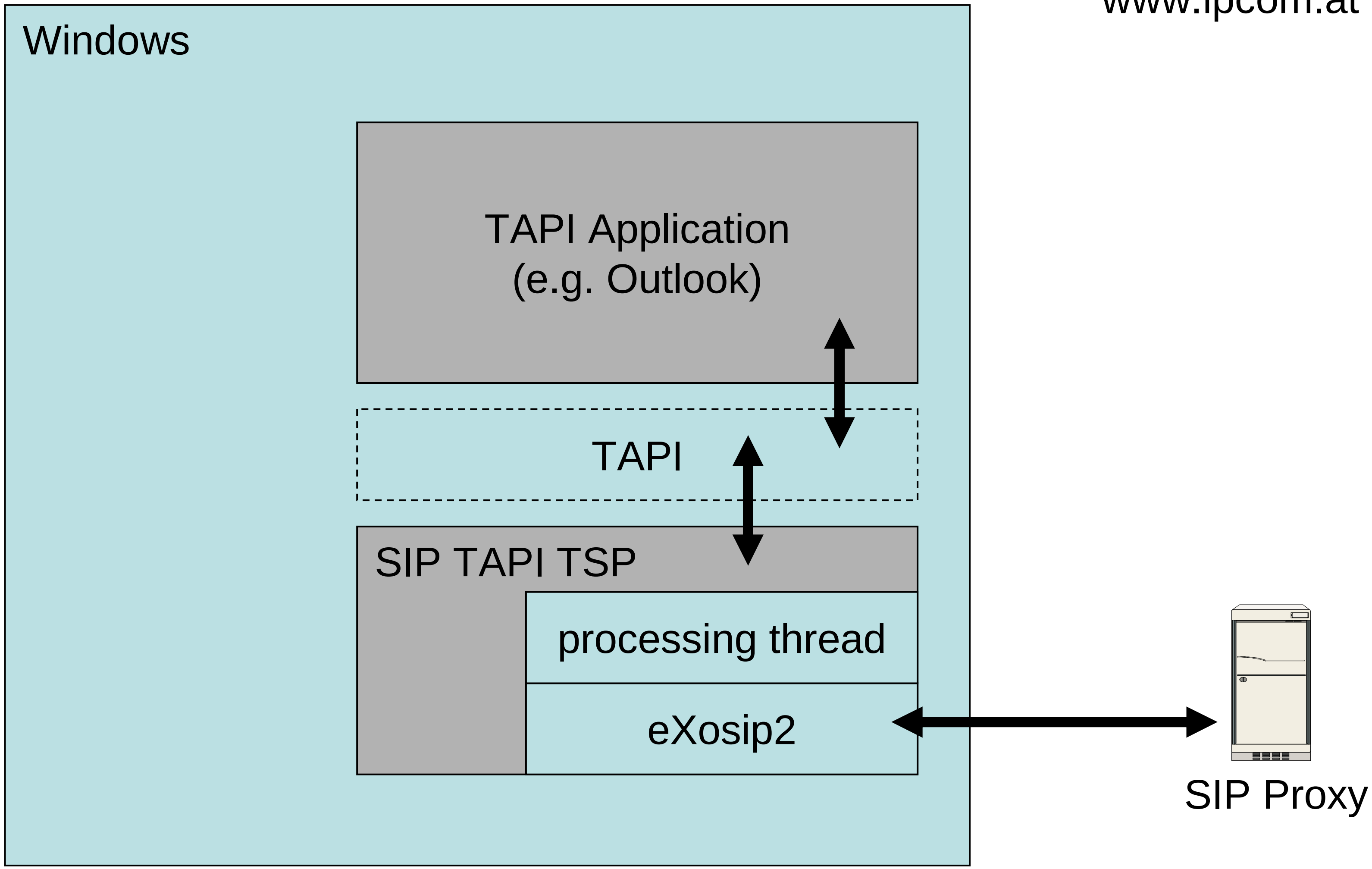
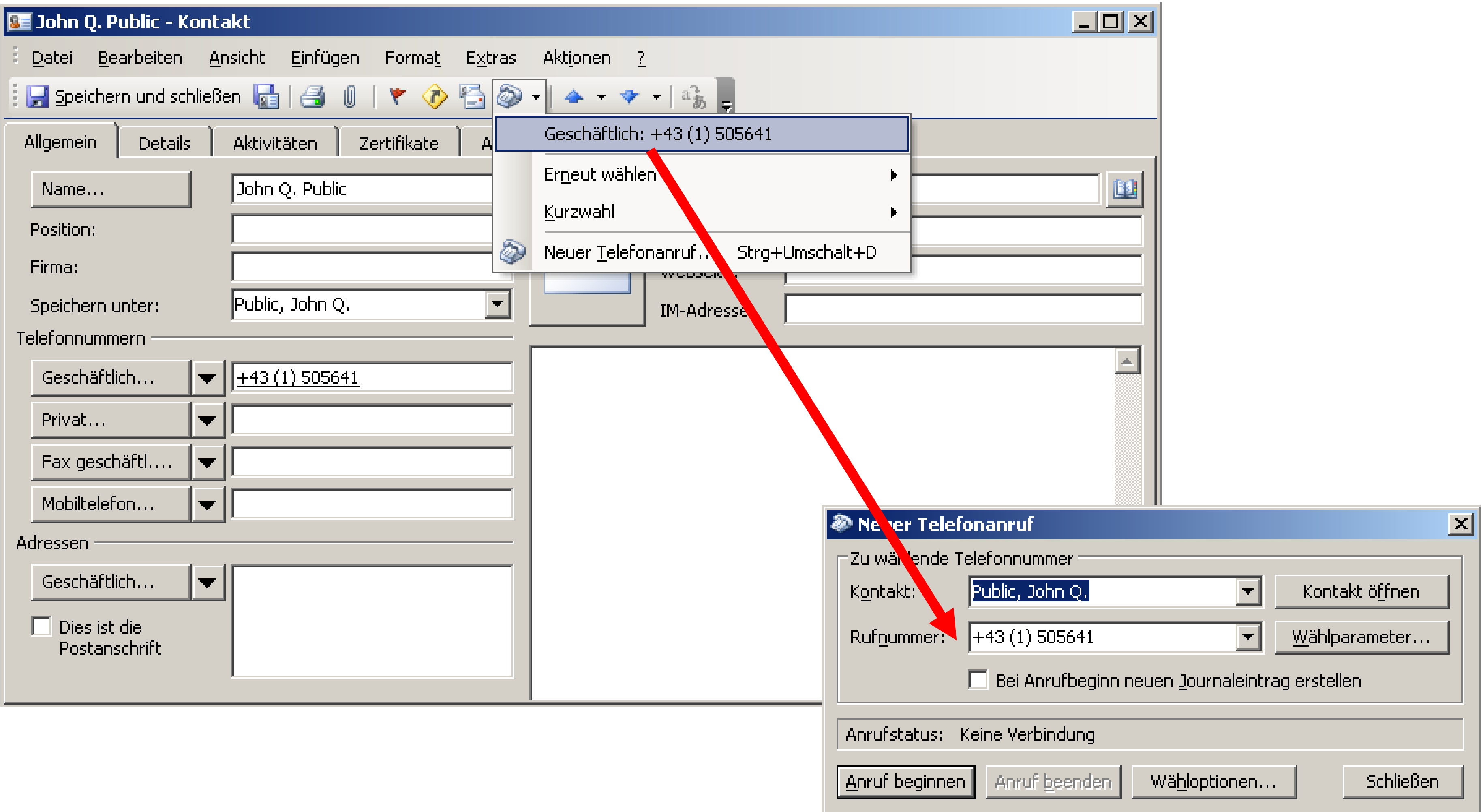


TAPI structure



TAPI dialing

- e.g. Outlook



TAPI dialing

- e.g. Outlook

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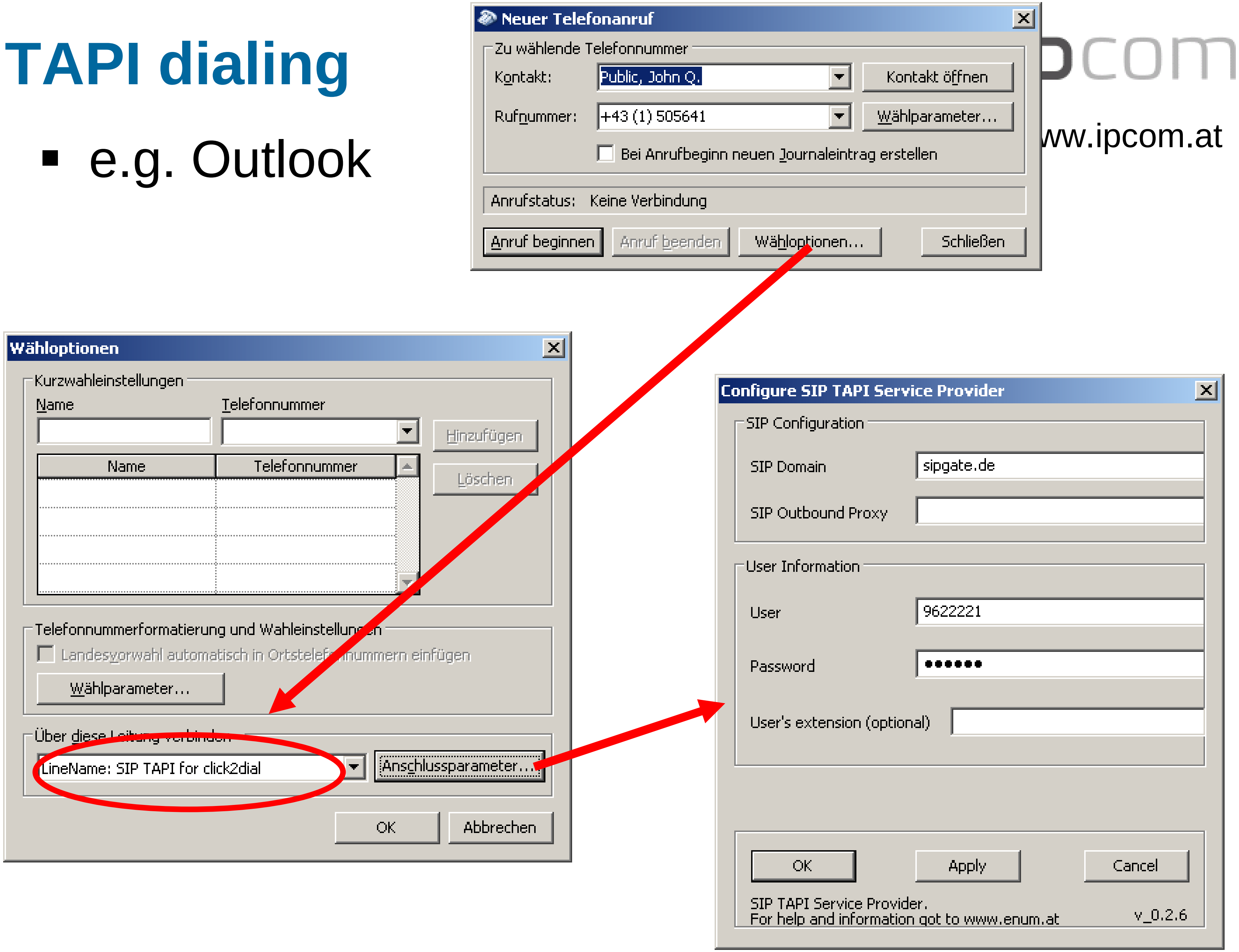
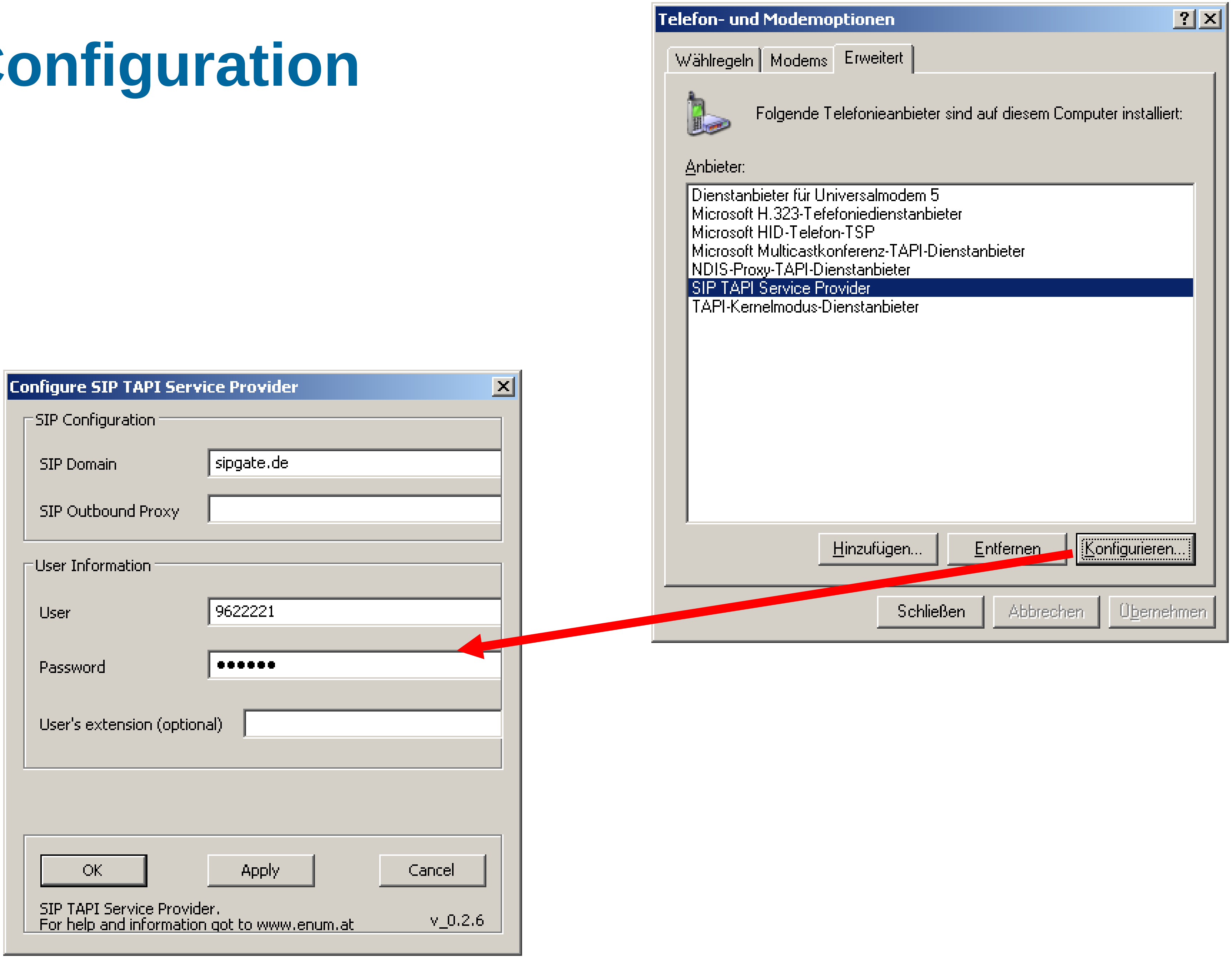


Configuration



Configuration

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- 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 this case make sure to add the ";lr" parameter to indicate loose-routing. In doubt leave this field empty. Examples:
    - 1.1.1.1;lr
    - 2.2.2.2:6060;lr
- 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
- 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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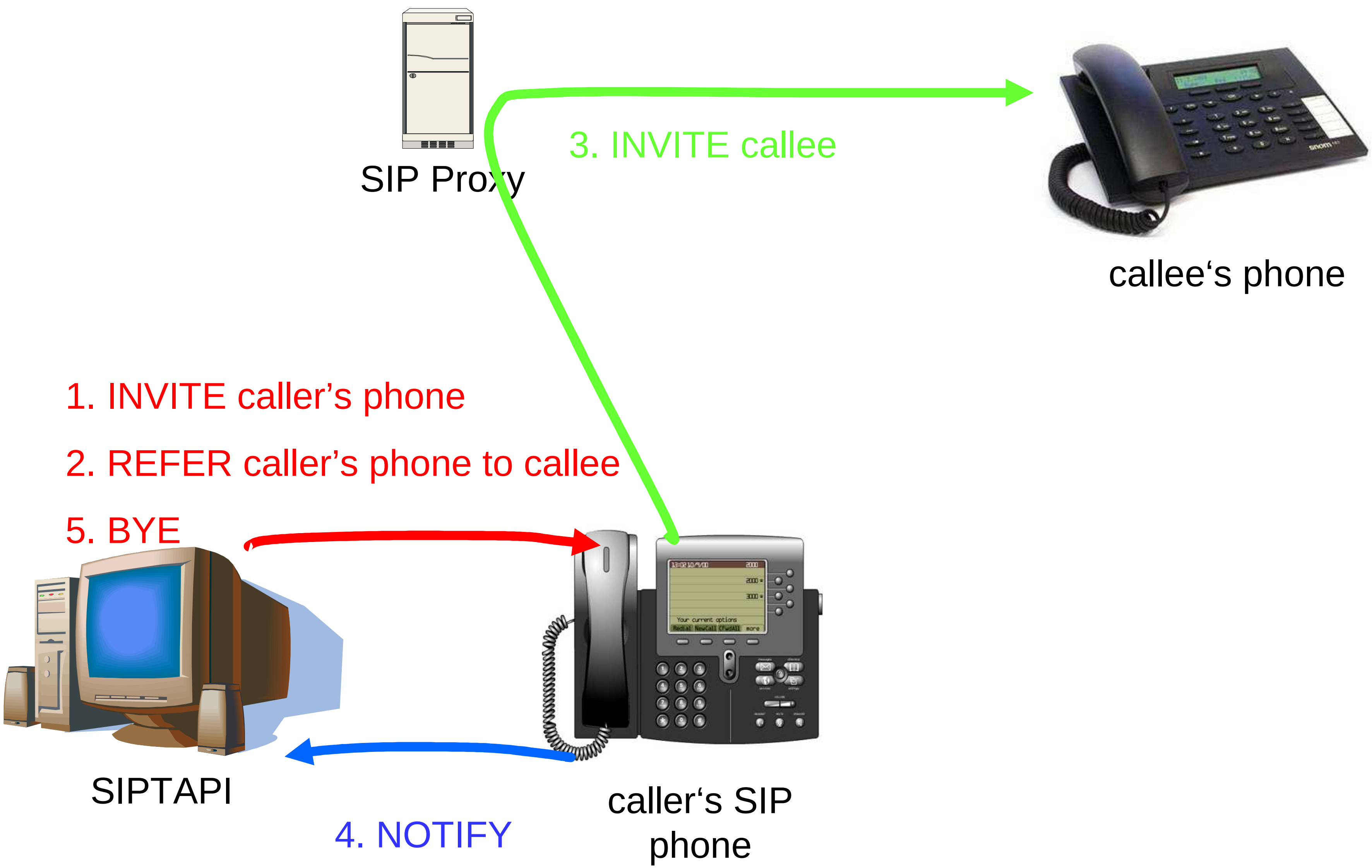

Alternative Configuration

- 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 8879
- SIP Domain
  - the SIP domain specified by your service provider
- SIP Outbound Proxy
  - 192.168.1.3:8879;lr
- 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

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

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- no STUN support ➔ NAT traversal requires SIP proxy with NAT traversal or outboundproxy (rport supported)
- no SRV support, resolves only A records (no problem if the outboundproxy performs SRV)
- probably buggy TAPI implementation
- some TAPI applications require strange phone number formats, e.g. Outlook: +43 (1) 505641
  ➔ *+country code (local area code) number*

ToDo

- STUN
- SRV
- ENUM
- check sipfrag response code in NOTIFY
- allow regexp applied to phone number
- incoming call indication

how to get it

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- https://sourceforge.net/projects/siptapi/
- Website: http://www.ipcom.at/index.php?id=561
- 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: http://www.sysinternals.com/ntw2k/freeware/debugview.shtml
  - use Wireshark to capture the SIP packets