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SIPTAPI (scroll down for commercial SIPTAPI)

A TAPI service provider for SIP

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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…)?

Table of Content

- This README starts with the basic concept of TAPI and SIPTAPI. These principles are the same for the free and commercial SIPTAPI!
- At the end of the document, the extensions of the commercial SIPTAPI and SIPTAPI-ACD are described

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 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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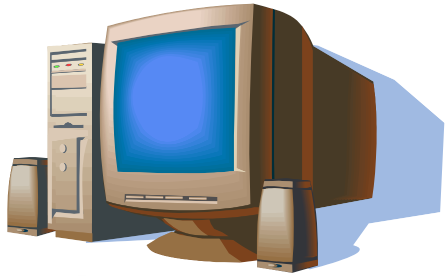
1.



➔ 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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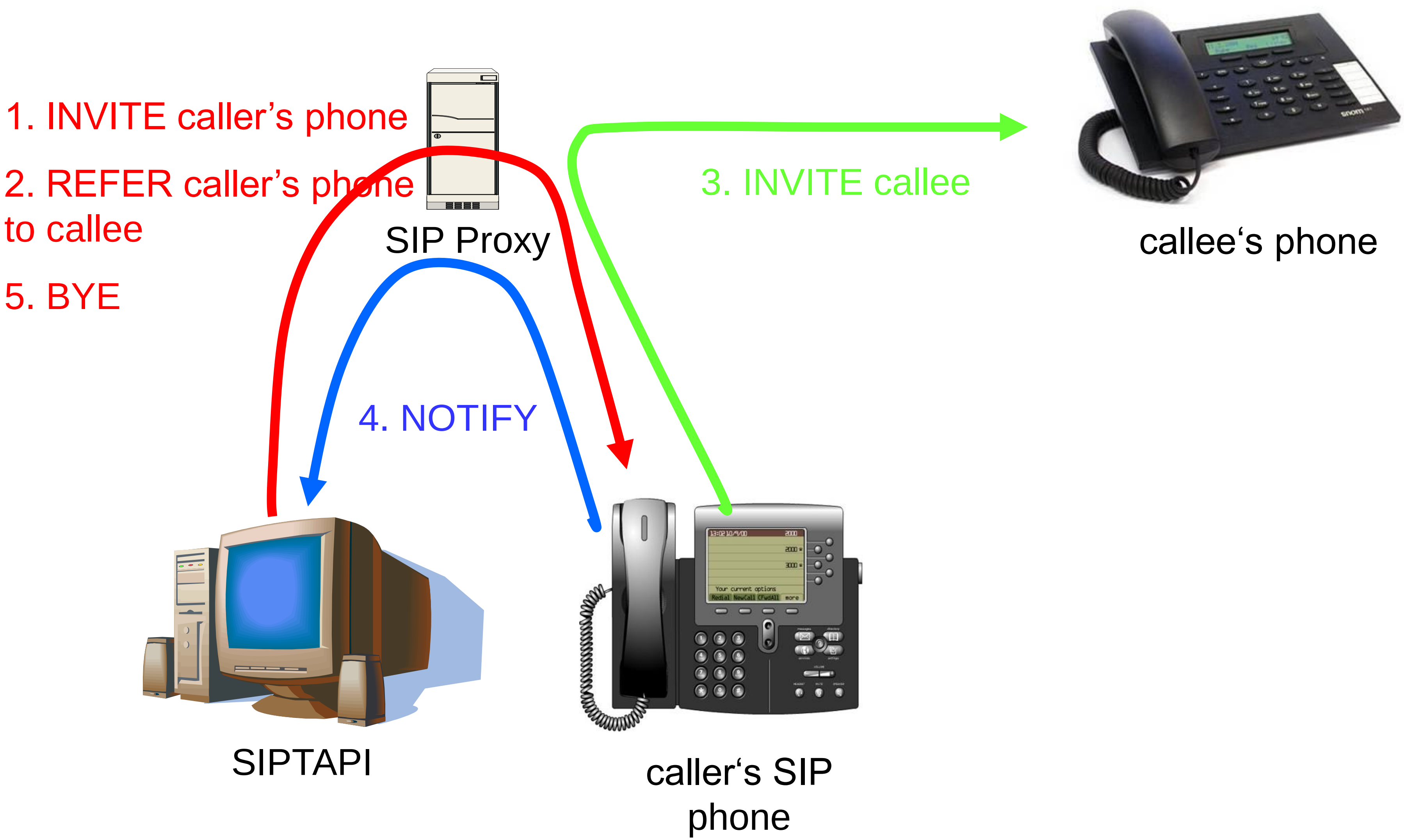

Call Flow with Asterisk

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TAPI structure

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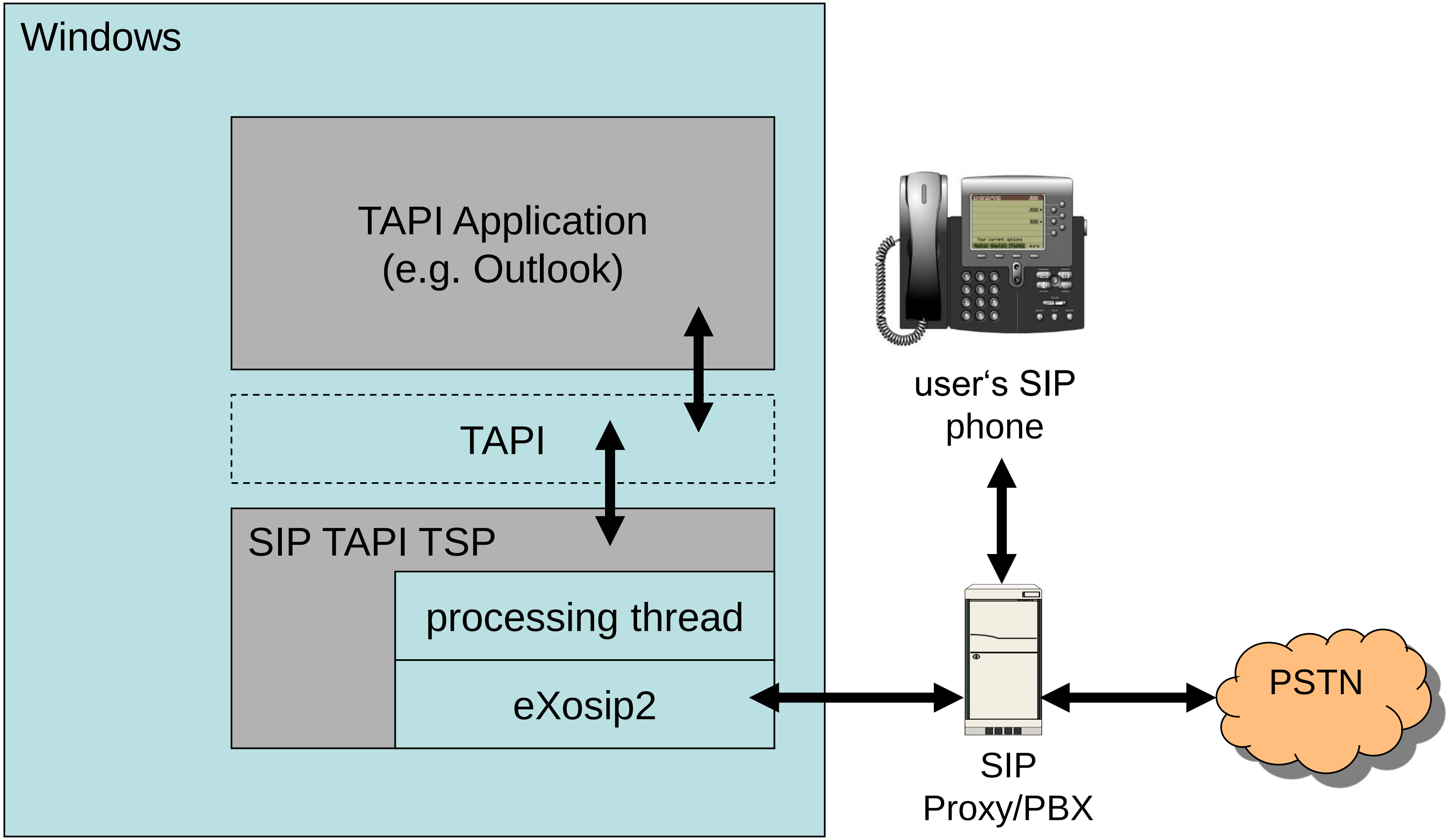
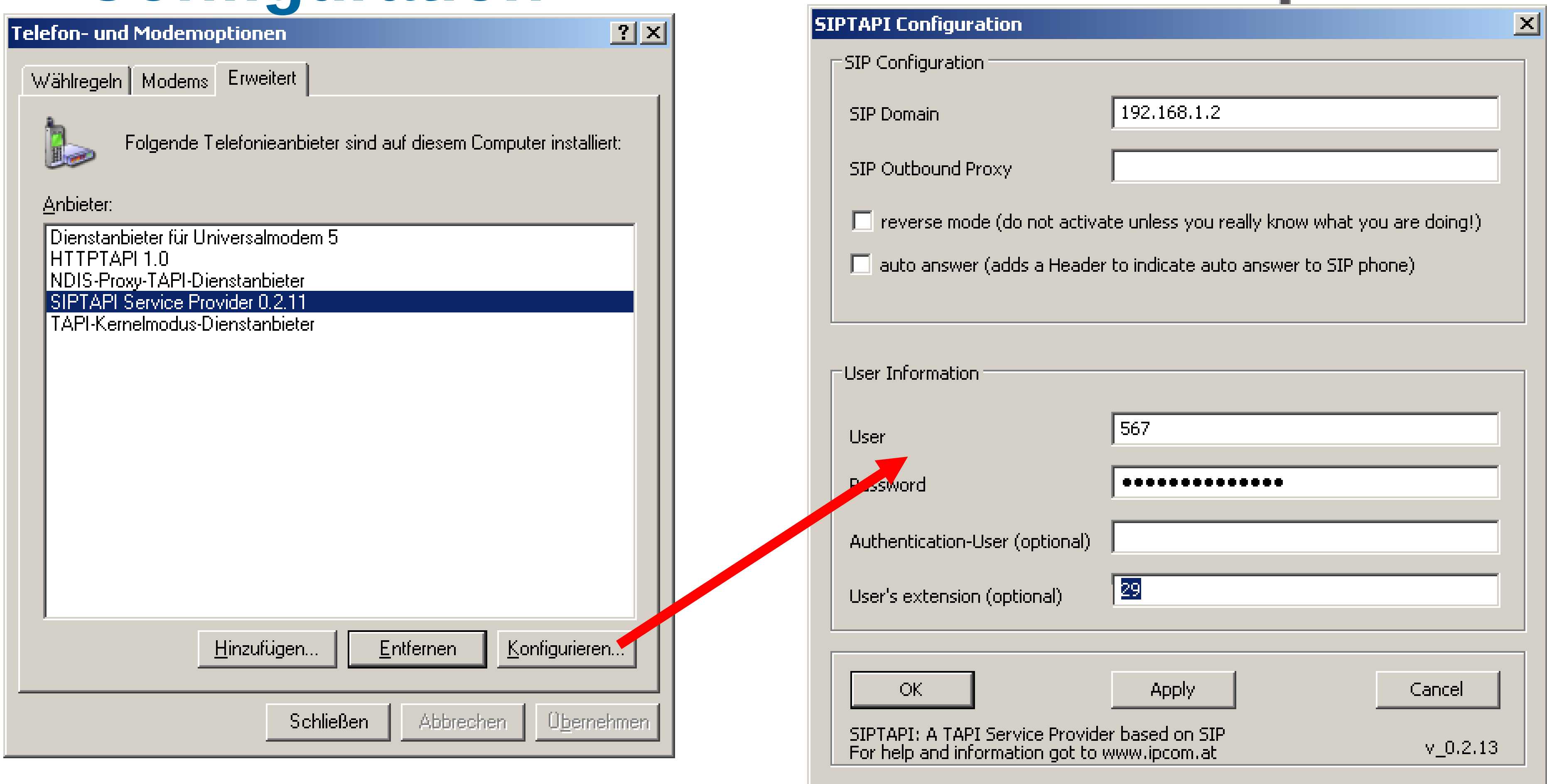
Find more details how TAPI works at http://www.ipcom.at/en/telephony/siptapi/tapi/



Configuration



- Note: On Windows Vista and Windows 7 you **must** configure SIPTAPI via the control panel and need administrator privileges!!!

TAPI dialing

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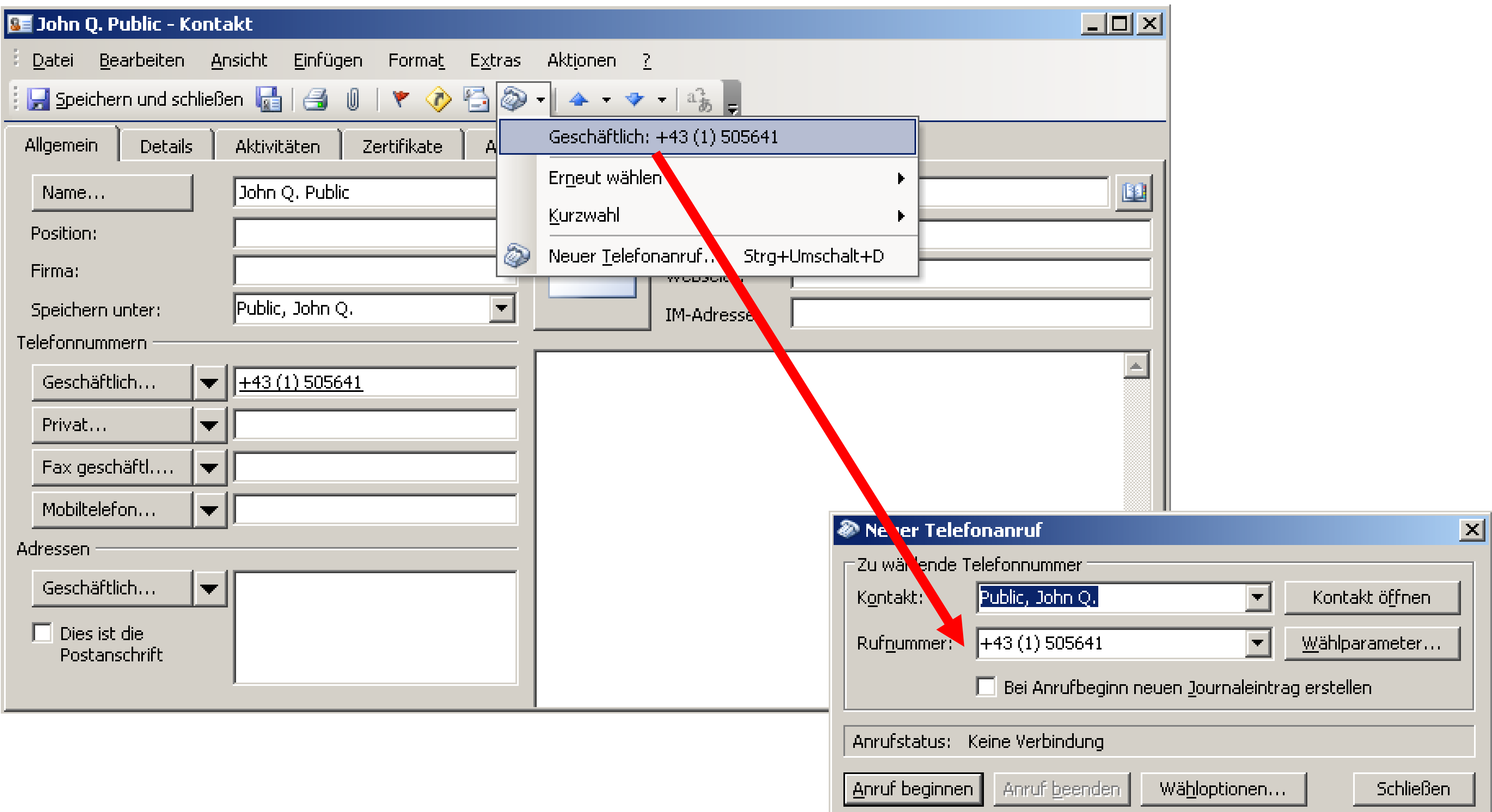
- e.g. Outlook



TAPI dialing

e.g. Outlook

1. choose SIPTAPI as TAPI line
2. make the phone call

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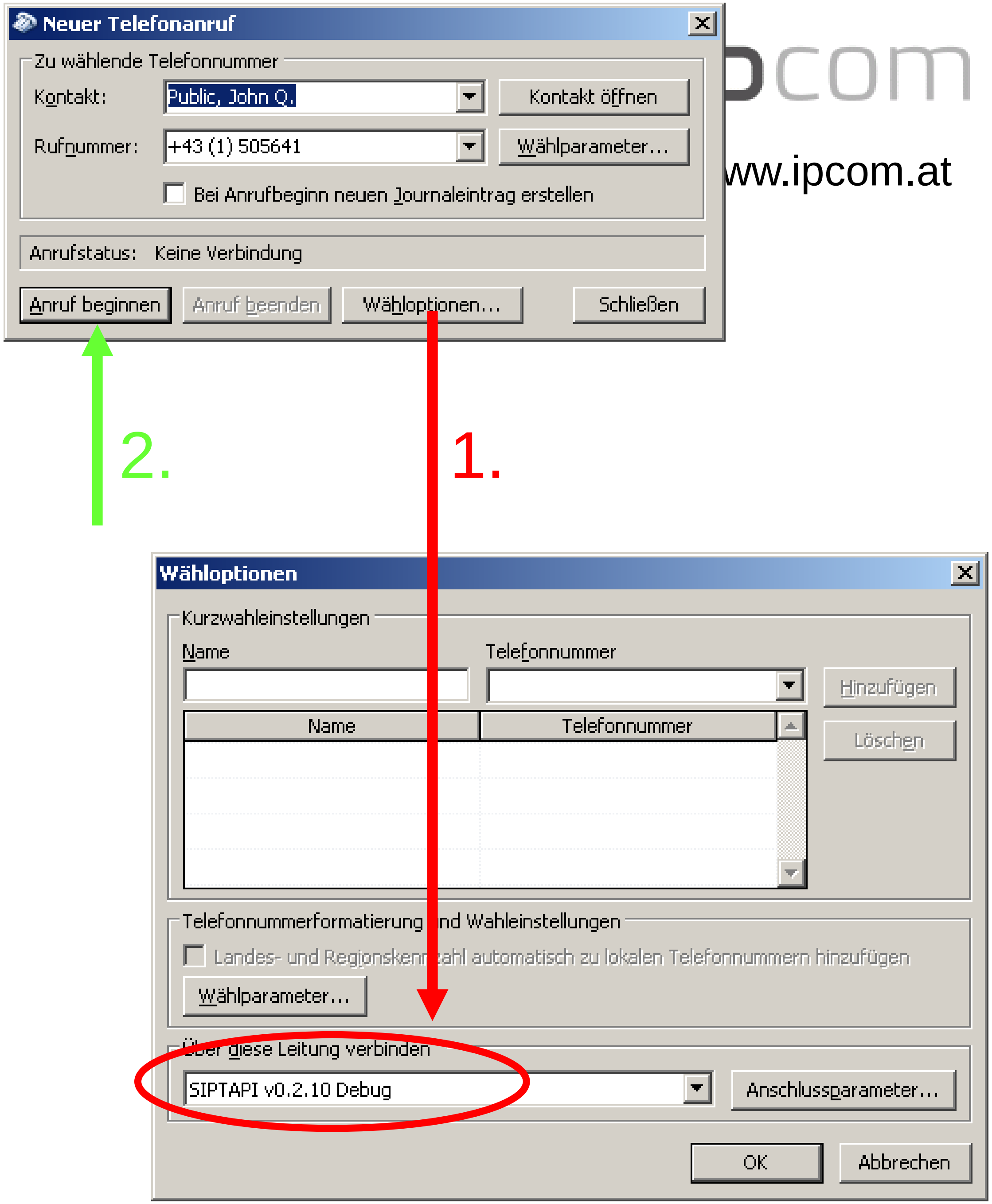

Configuration

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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 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
- Authentication User
  - Some systems use a dedicated username for authentication, which is not identical to the SIP username. Usually you can leave this field empty.
- 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 or phone-number must be used to dial to the SIP phone.
- reverse mode
  - In this mode SIPTAPI will first call the dialed number, and only if the target answers the call, the call will be transferred to the user's SIP phone. Use with caution - you can not hear any in-band messages or ringback tones in this mode!
- auto answer
  - In this mode SIPTAPI will add certain headers to the INVITE request to instruct the phone to answer the phone call automatically. This feature usually only works in the "alternative mode" or if the SIP server is a proxy. When using B2BUAs like Asterisk it won't work. It should work with Grandstream, Polycom and SNOM phones, but needs to be turned on in the web interface of the phone, e.g.: "Allow Auto Answer by Call-Info: Yes" for Grandstream phones.

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

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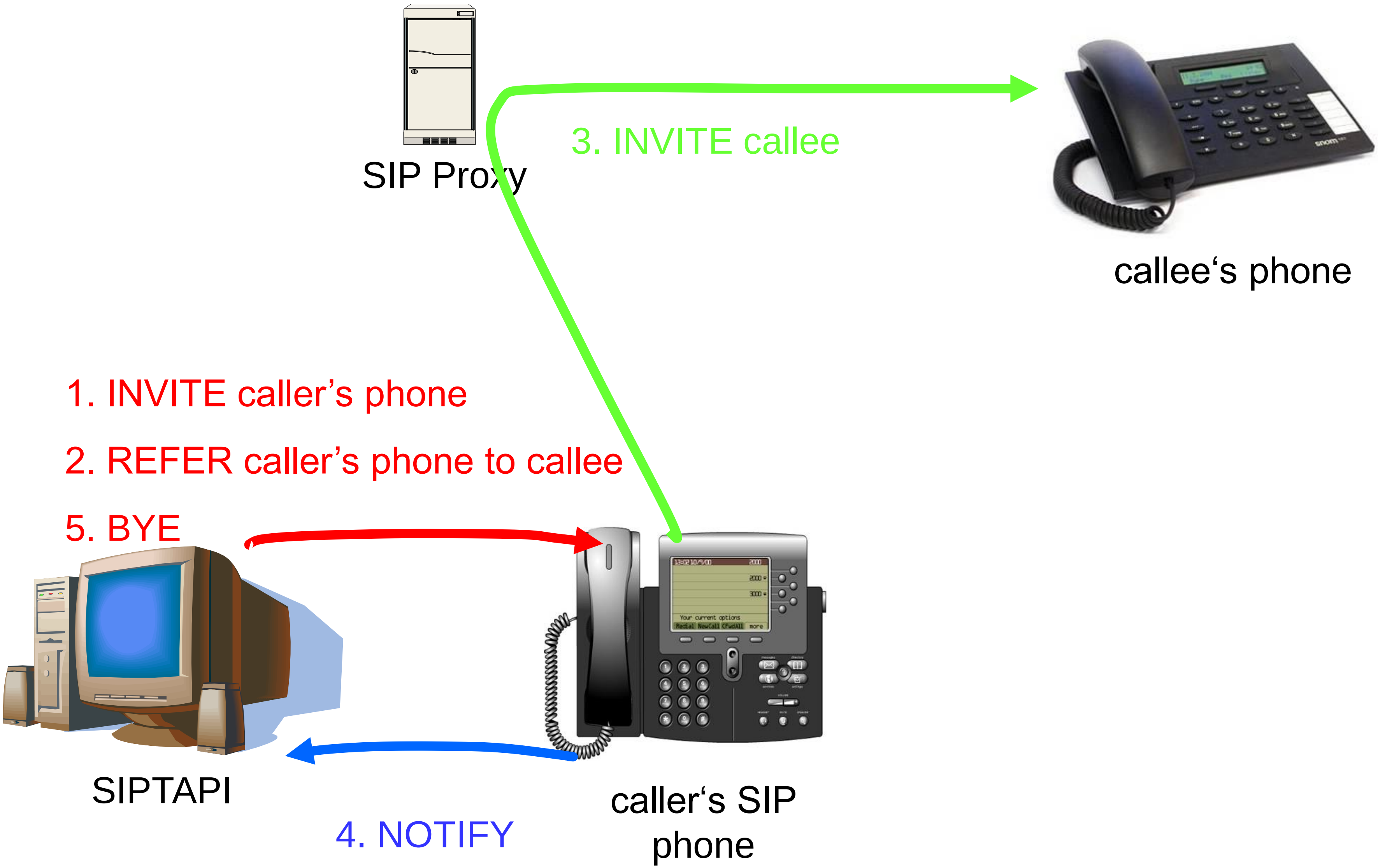

Alternative Configuration

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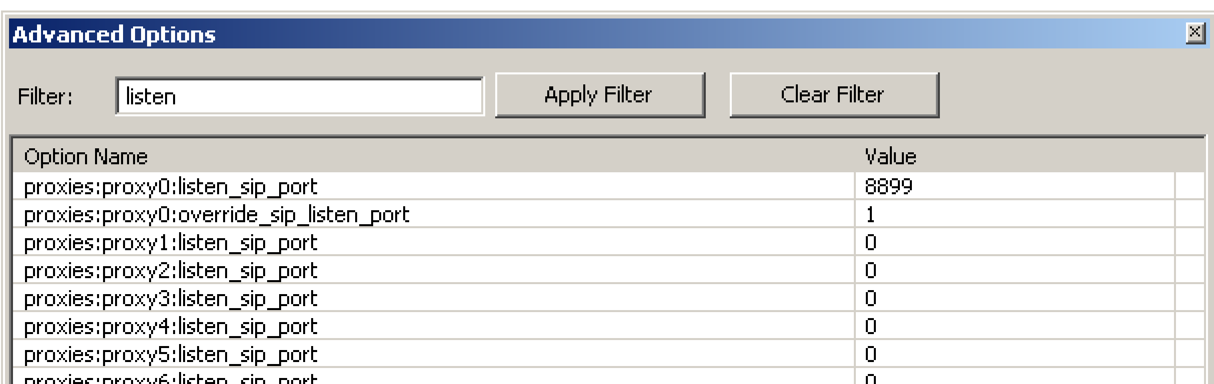
- Xlite per default uses a dynamic SIP port ☹
- How to get Xlite a static IP address?
  - Xlite uses the IP address of the PC. Thus make sure the PC has a static IP address
- How to get Xlite a static SIP port?
  - Start Xlite
  - Dial ***7469 → a pop window with advanced options appears. Filter for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

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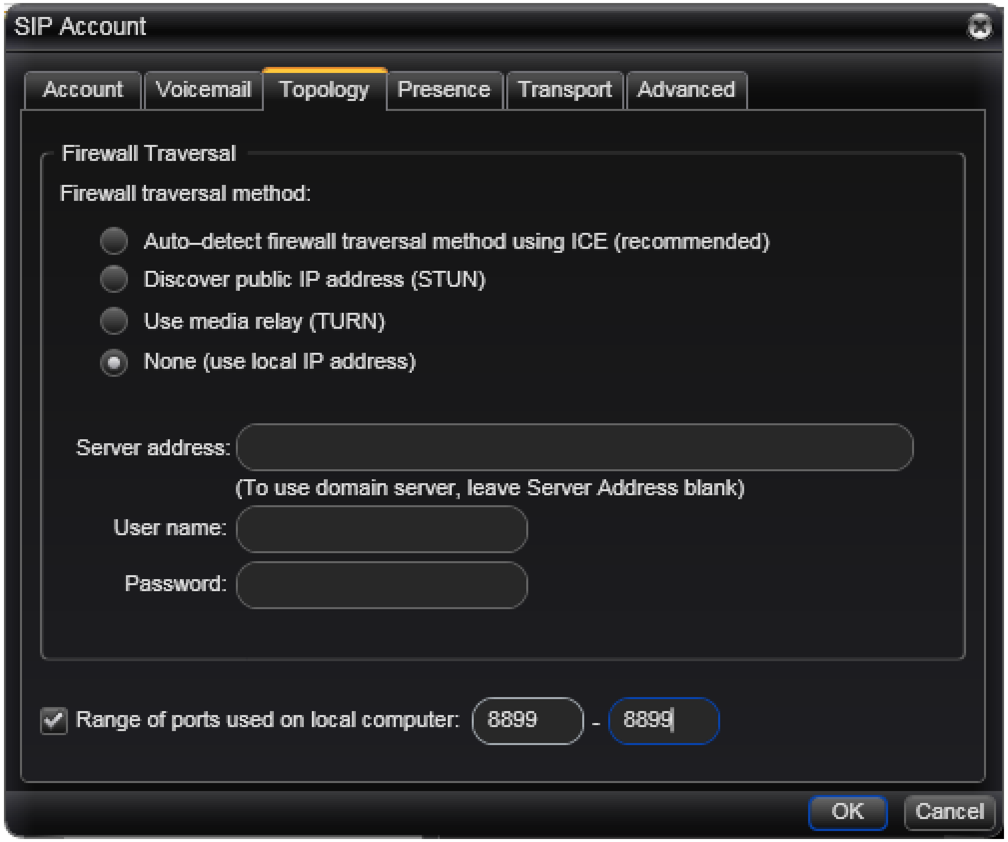
- How to get Xlite a static IP address?
  - Xlite4/Bria uses the IP address of the PC. Thus make sure the PC has a static IP address
- How to get a static SIP port?
  - limit the port range to a single port:



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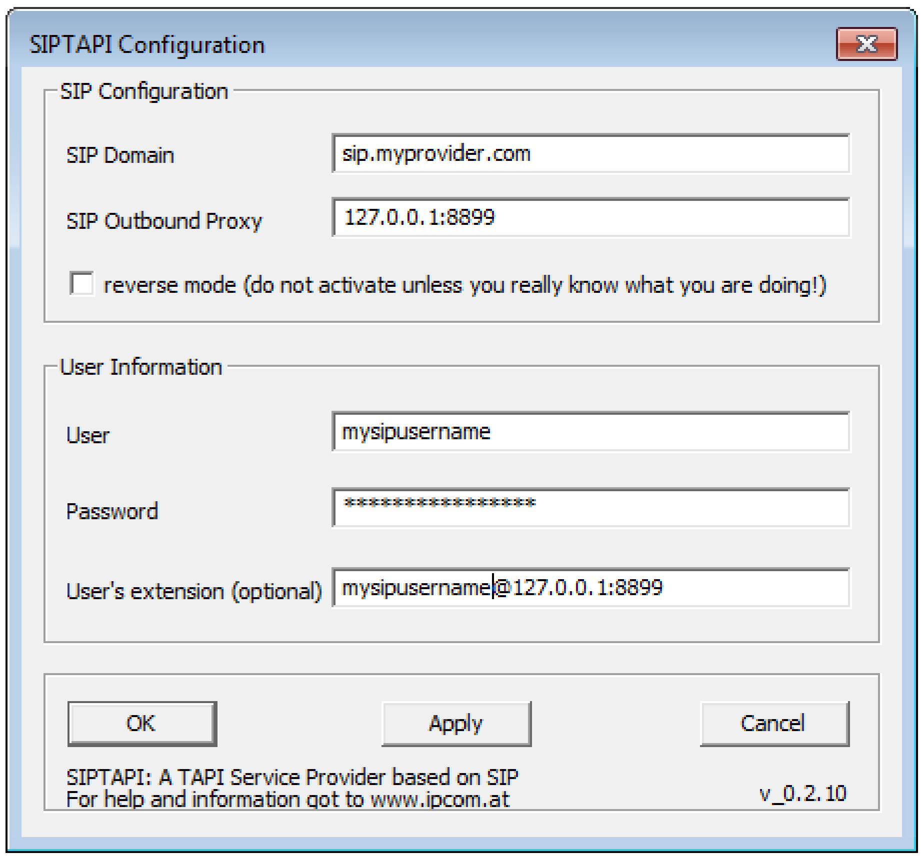
- SIPTAPI screenshot (e.g. client is fixed to port 8899)
- Note: this requires at least version 0.2.10!!!



Reverse Mode

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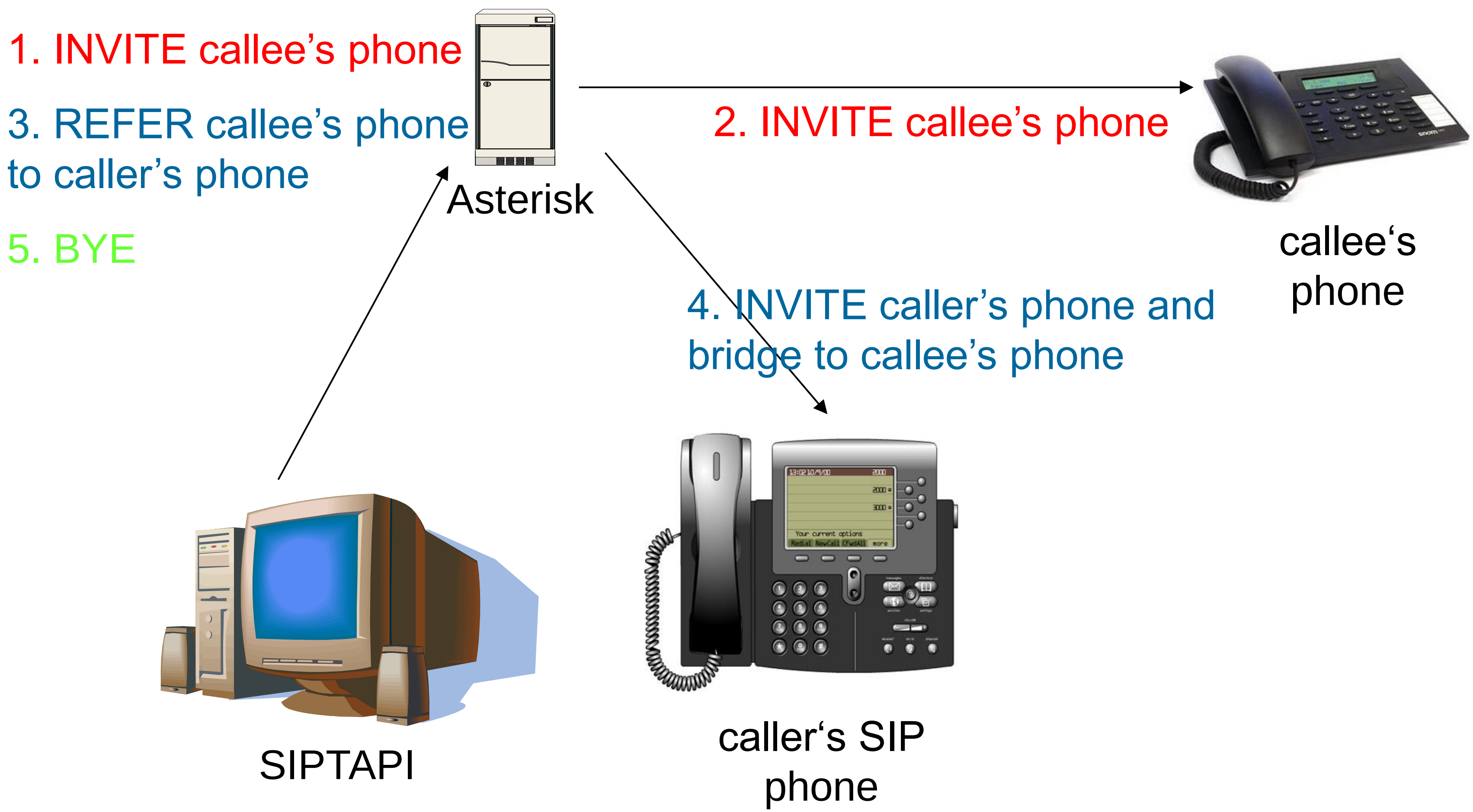
- Normally SIPTAPI calls the user's phone and then refers it to the real target
- In reverse mode SIPTAPI will call directly the target, and once the target answers, it gets refered to the user's phone.
- Note: In reverse mode you wont get any audible call progress indication (e.g. ringback, busy tone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

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- 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

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- 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

Commercial SIPTAPI

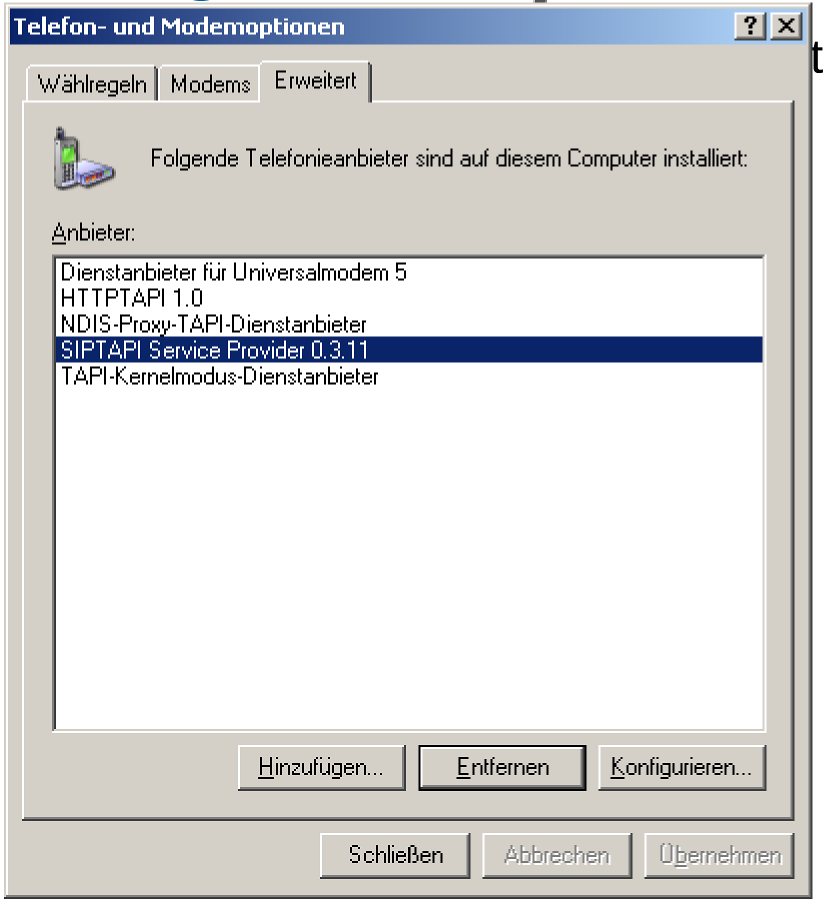
Note: The following slides do only apply to the commercial SIPTAPI

Commercial SIPTAPI

- The commercial SIPTAPI additionally supports
  - multiple lines → can be used in terminal server (e.g. Citrix) environments
  - incoming call indication
  - ACD mode: to be used by ACD dialer applications

Commercial SIPTAPI Configuration

Control Panel → Phone and Modem Options



- Note: On Windows Vista and Windows 7 you **must** configure SIPTAPI via the control panel and need administrator privileges!!!

Commercial SIPTAPI Configuration

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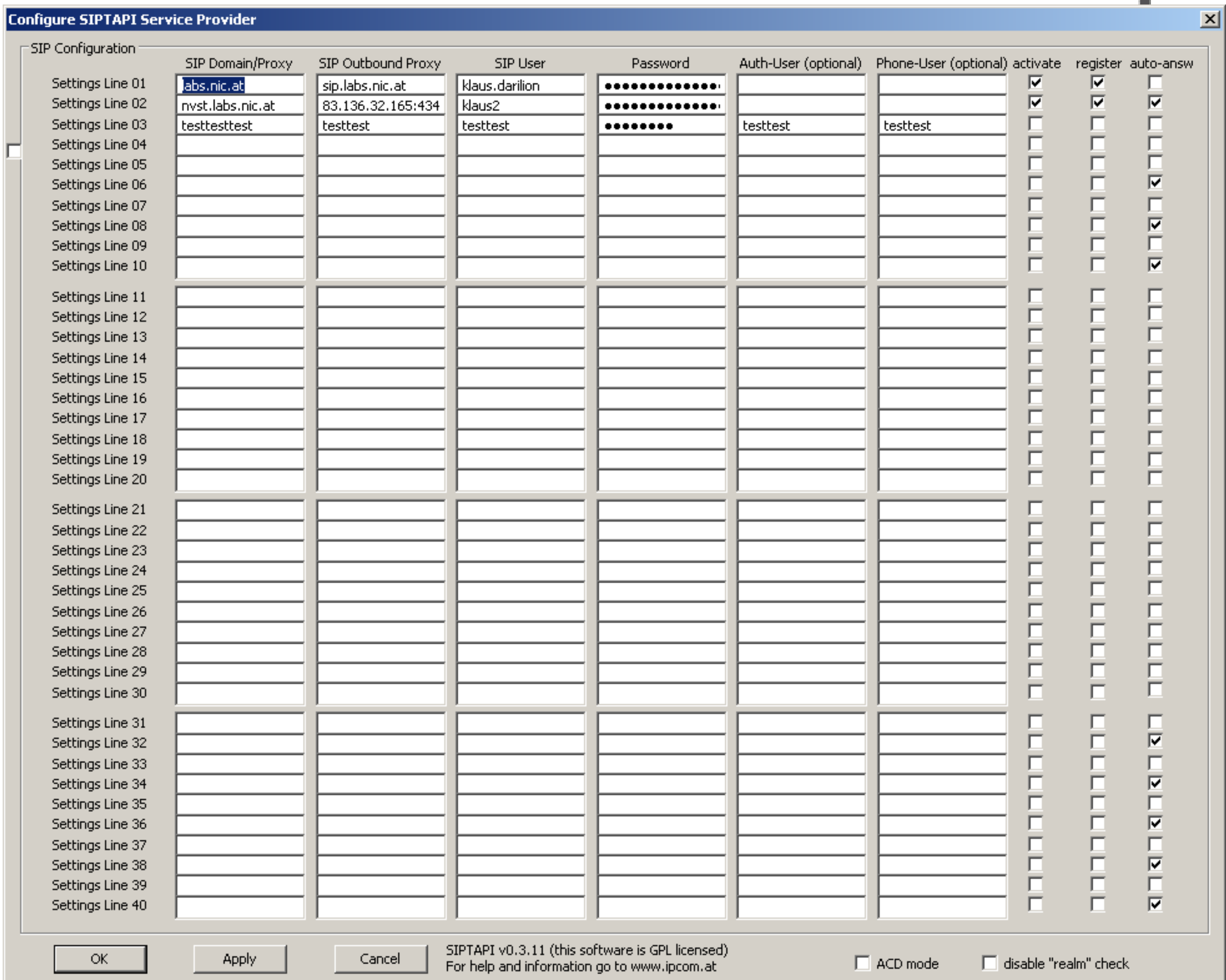

supports up to 40 totally independent lines

Commercial SIPTAPI Configuration

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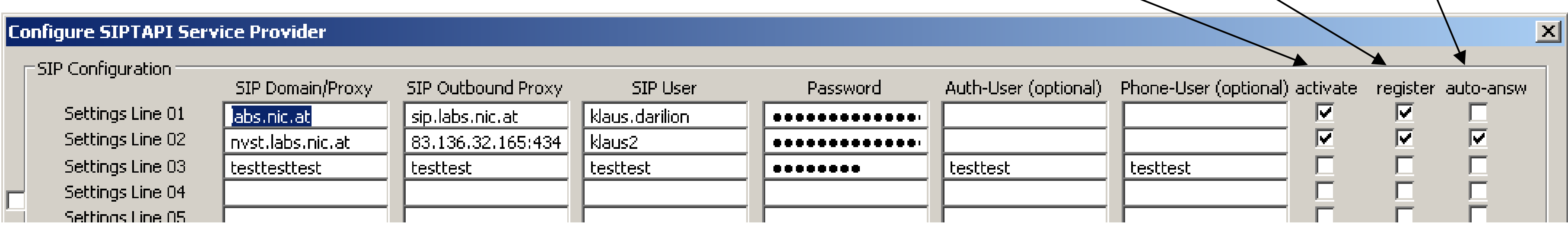
indicate to the phone that it should auto-answer the call

REGISTER to the SIP server (needed for incoming call indication)

enable/disable a line



Commercial SIPTAPI Configuration

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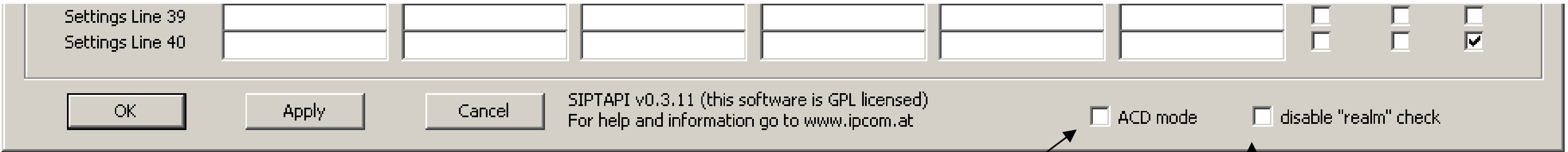

Activate ACD mode (to be used by ACD dialer software for outbound call campaigns)

Many SIP server use an authentication realm which does not match the SIP domain. In this case you have to disable the "realm" check.

Commercial SIPTAPI Operation Modes

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- Standard Mode
  - call local user, automatically transfer the local user to the to-be-dialed number
- ACD Mode (Dialer Mode)
  - call the to-be-dialed number, wait until TAPI application transfers the call to a local agent/user

Commercial SIPTAPI Configuration

- SIP Domain/Proxy:
  - 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
- SIP 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
- Auth-user
  - if the SIP authentication username is different to the SIP username you can specify it here. Otherwise it should be empty. If you are unsure just leave it empty.
- Phone-user
  - 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.

Commercial SIPTAPI Configuration

- Activate
  - Here you can disable lines without loosing the configuration settings.
- Register
  - when checked, this line REGISTERs to the SIP proxy/PBX. This is necessary to receive incoming call indication
- Auto-answer
  - In this mode SIPTAPI will add certain headers to the INVITE request to instruct the phone to answer the phone call automatically. This feature usually only works in the "alternative mode" or if the SIP server is a proxy. When using B2BUAs like Asterisk it won't work. It should work with Grandstream, Polycom and SNOM phones, but needs to be turned on in the web interface of the phone, e.g.: "Allow Auto Answer by Call-Info: Yes" for Grandstream phones.
- Enable "realm" checking
  - If this option is checked, then the SIP authentication also verifies the authentication realm provided by the SIP server. Usually it is not disabled, as may providers do not set the authentication realm properly. Checking this feature is necessary if you use the same user under multiple domains (e.g. sip:john.doe@example.com and sip:john.doe@mydomain.com)

Commercial SIPTAPI Configuration

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- Normal mode
  - In normal mode, SIPTAPI calls first the local phone. On answer, the local phone gets transferred to the actual target.
- Reverse mode
  - In reverse mode, SIPTAPI calls first the actual target. On answer, the actual target gets transferred to the local phone. Note, in this mode you can not detect errors with in-band signaling (early media announcements)
- ACD mode
  - This mode is intended for outgoing ACD dialer applications. In this mode, SIPTAPI calls first the to-be-dialed target, and then waits for a blindTransfer request to transfer the call to the local agent. To be used with ACD-TAPI-Dialers, e.g. Overdialer from Grutzeck. Note, in this mode you can not detect errors with in-band signaling (early media announcements)

Alternative Usage

- same technique as in free SIPTAPI version
  - does not work in ACD mode
  - not compatible with incoming call indication

ACD Mode

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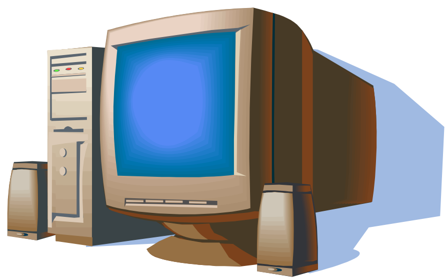
1.



➔ the ACD dialer TAPI application dials a new target number (e.g. Grutzeck‘s Overdialer)

2.

➔ target‘s phone rings, pick up!

3. ➔ the ACD dialer application requests a blind transfer to the agent → SIPTAPI instructs the PBX to transfer the target to a local agent (a certain extensions or a calling queue)

4.

➔ agent‘s phone rings

ACD Mode - Signaling

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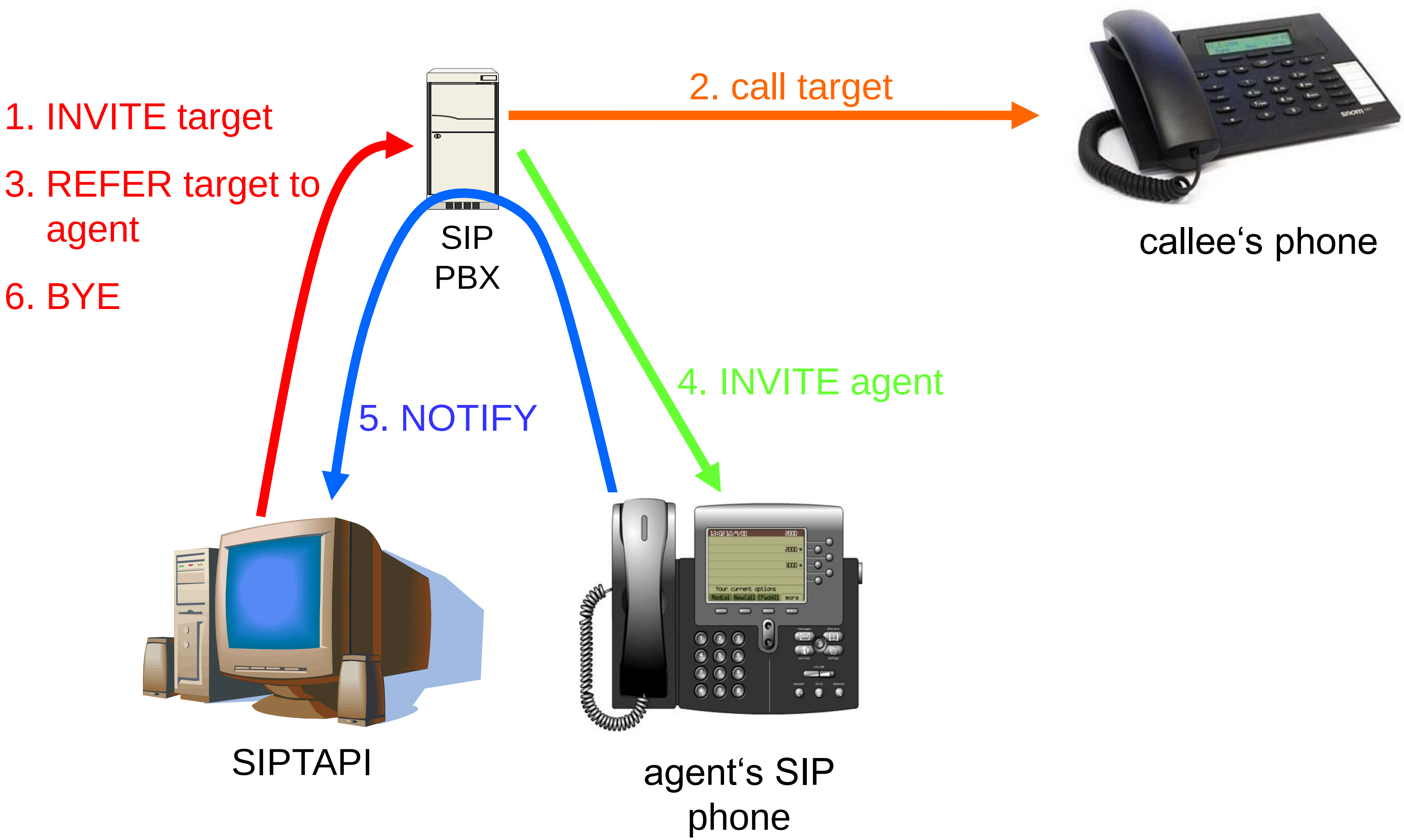

ACD Mode - Configuration

- Fixed: 40 lines
- Every line needs a dedicated SIP Account at the SIP PBX (e.g. Asterisk)
- disabled lines are still reported to TAPI as lines, but using them triggers LINERR_UNAVAIL error

ACD Mode with SIPTAPI

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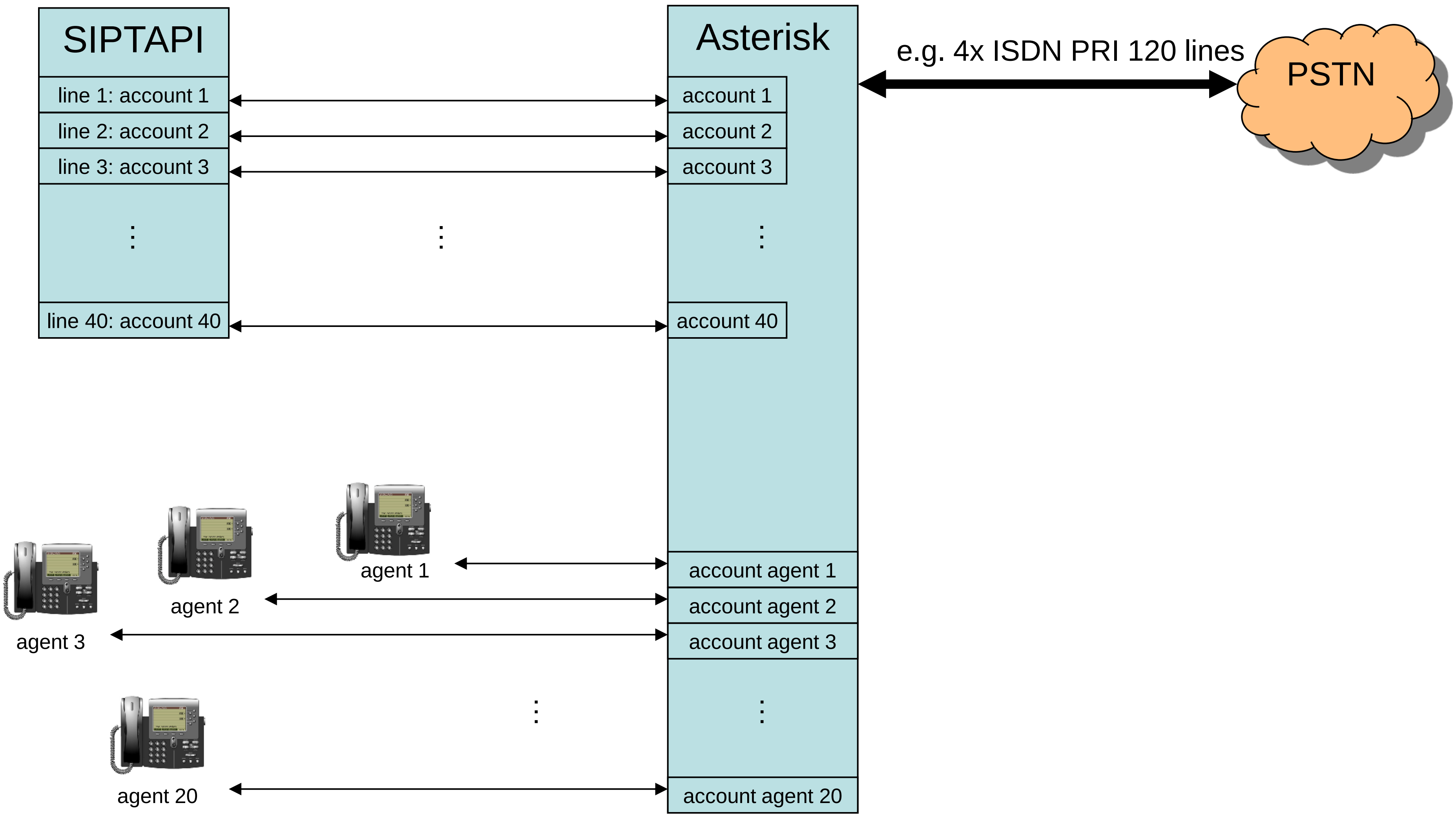
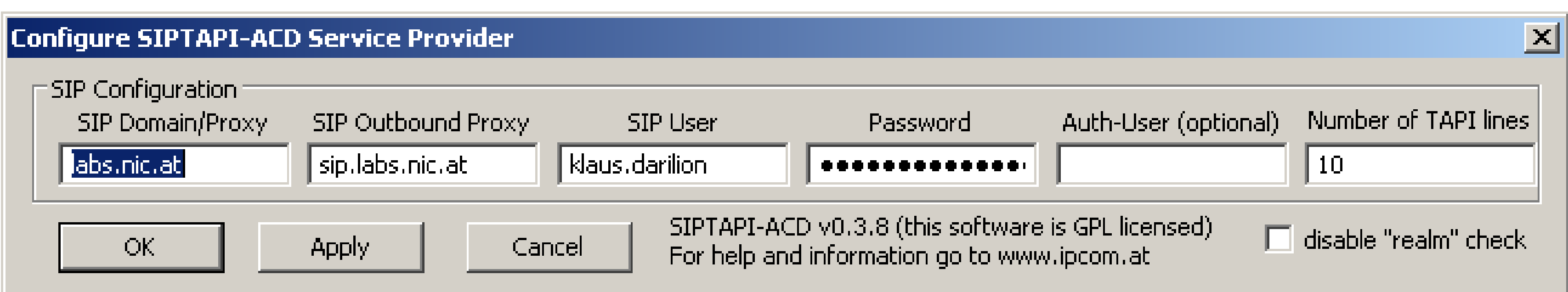


SIPTAPI-ACD

- Based on SIPTAPI, but optimized for ACD Dialers
- Number of TAPI lines configureable: 1 - 999
  - changing the numbers requires reinitializing of the TSP)
- Uses a single SIP account for all lines
  - Asterisk: call-limit = 999
- No support for incoming call indication!!!



SIPTAPI-ACD

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