

Asterisk

~~the~~ general purpose Open Source Telephony platform



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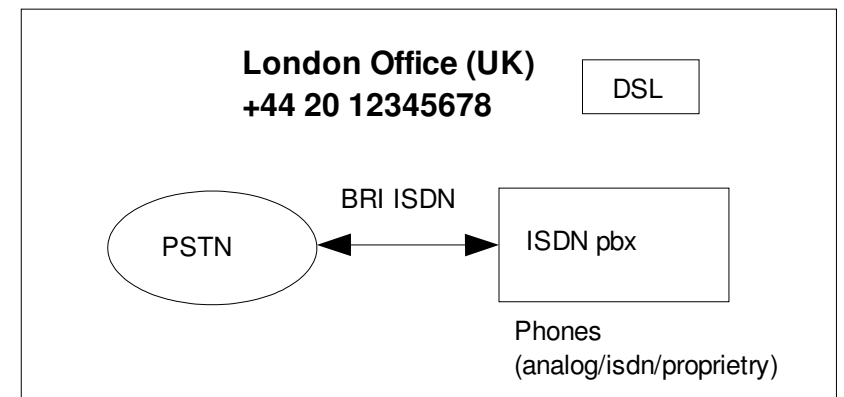
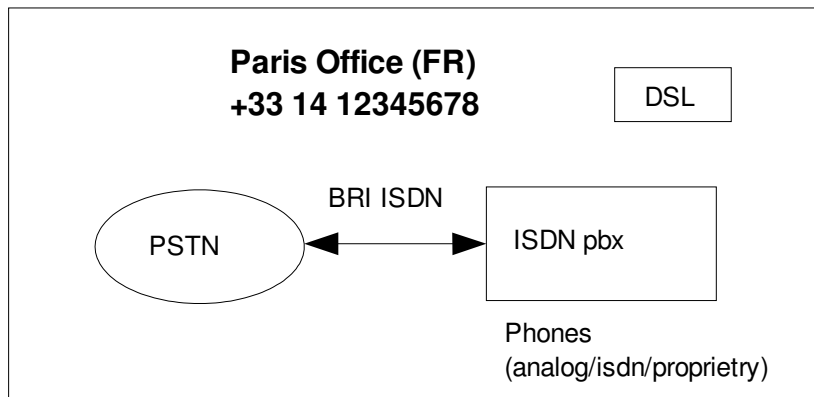
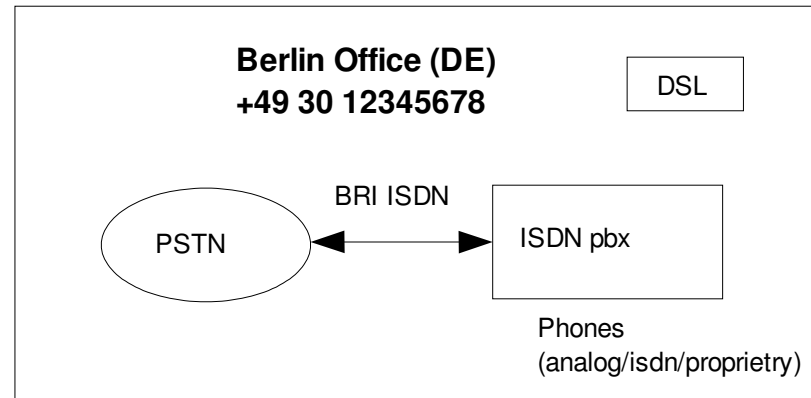
„Advanced Scenarios“

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<http://www.junghanns.net/asterisk/>

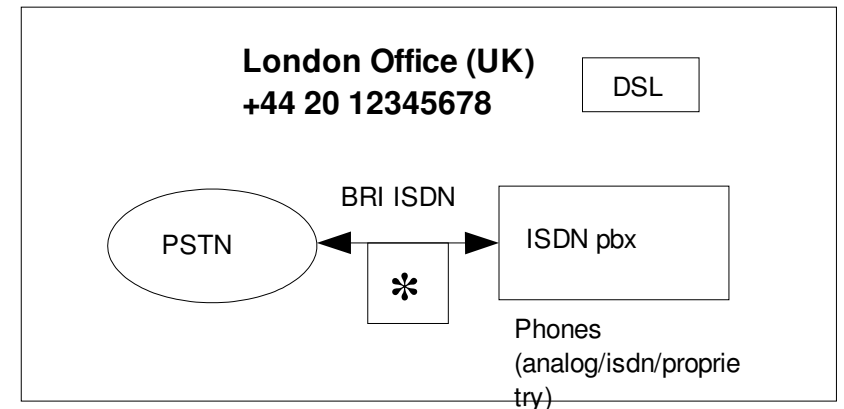
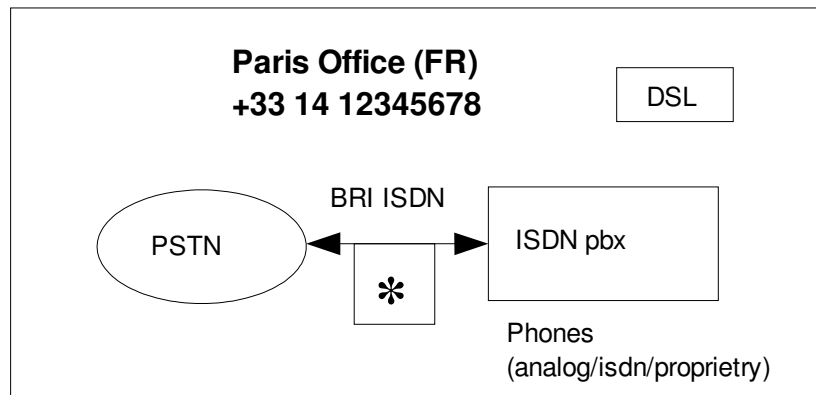
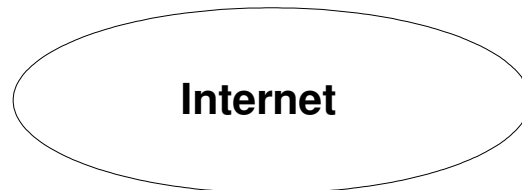
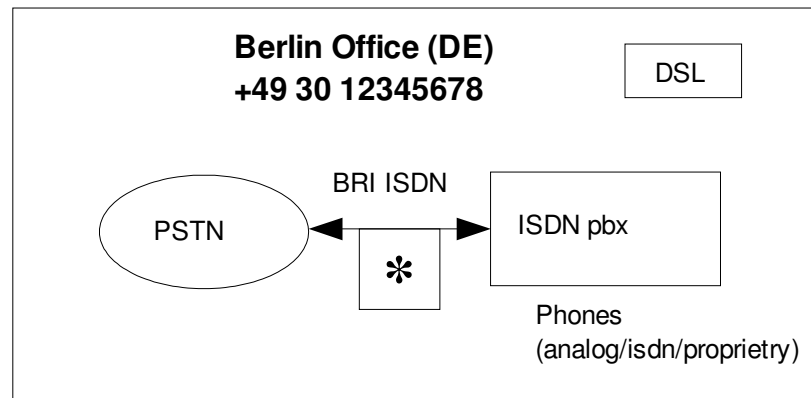
Traditional Setup of a 3 location company

- Company with 3 European locations
- Legacy ISDN PBXes, with BRI ISDN lines to the PSTN
- Analog, ISDN or propriatry PBX system phones (UP0...)
- Paying international long distance rates for company internal calls

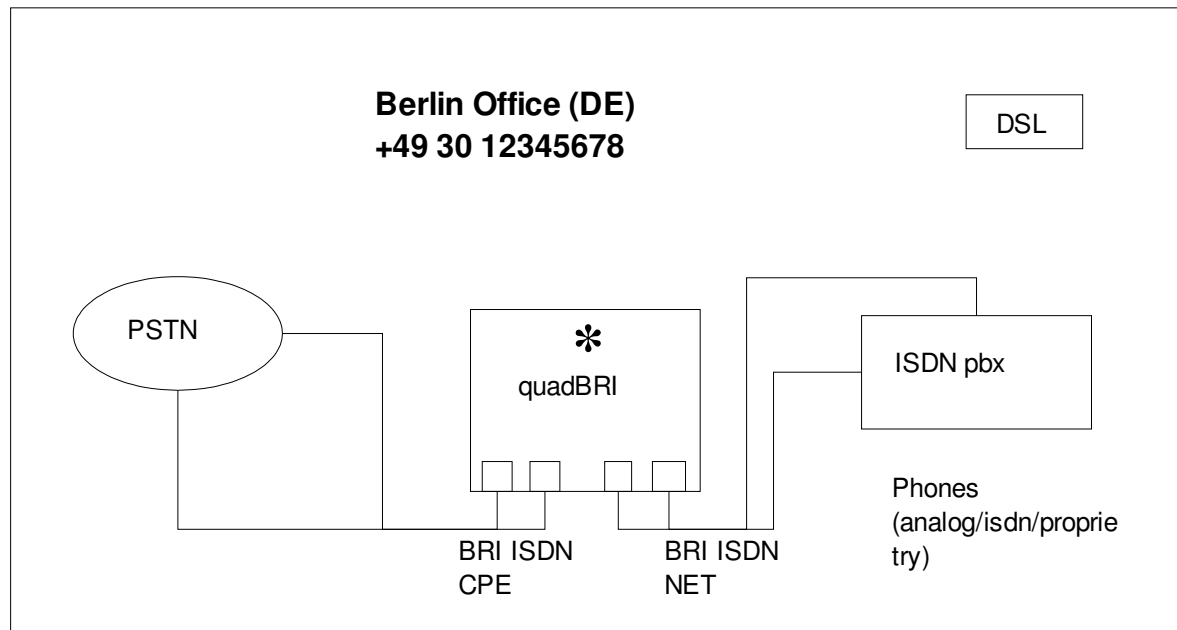


Asterisk enabled Setup

- Asterisk Server is placed inbetween the telco and the legacy ISDN PBX at each location
- The configuration of the legacy ISDN PBXes are NOT touched!
- Outgoing calls to the PSTN are routed transparently through Asterisk (no 9 for an outside line)
- Company internal calls are routed over the public internet between the location



Technical details for a 2 BRI ISDN line example



- Asterisk Server is diskless (fanless) standard PC, e.g. Mini-ITX based VIA C3 with 1 quadBRI card installed
- 2 BRI ports running in CPE mode to connect to the telco ISDN BRIs
- 2 BRI ports running in Network mode to connect to the legacy ISDN PBX
- Ethernet card connect to local network with Internet access
- All phones of the legacy ISDN PBX are now Asterisk extensions
- If the hardware of the Asterisk server fails the legacy ISDN PBX can be easily reconnected directly to the telco BRI ISDN lines (because the configuration of the PBX was not changed)

Key features of the Asterisk enabled Setup

- Geographical LCR (outbound calls that are local to one of the locations will be routed through that location), that way a call from Berlin to a destination in Paris (which was an international call before) now becomes a local call
- Transparent LCR, make use of ISDN call-by-call providers or VoIP-to-PSTN providers
- CTI support for all legacy phones, you can easily integrate incoming calls into your CRM solution (e.g. a small popup window that shows the result of a database lookup based on the callerID)
- Drop incoming calls that have no callerID (Telemarketers!) directly into Voicemail
- All phones of the legacy ISDN PBX are now VoIP enabled, with a proper configuration all phones can be called from outside (over the Internet) with VoIPs (SIP, MGCP, H323, IAX2)
- A professional grade Voicemail System (with Email notification, forwarding of messages, etc)
- Detailed CDRs for inbound and outbound calls allowing you to verify your telcos bill and gathering statistics
- Freelancers or partner companies can be integrated into the PBX with VoIP
- Inbound calls can be routed to a different geographical location (e.g. if all call agents are busy, are out for lunch or the office is in a different timezone and it's 2 am there), your customers will always reach someone!

Configuration example for geographical LCR

extensions.conf:

```
[fromPBX] ; we are in Berlin
exten => _0044X.,1,Dial(IAX2/london/0${EXTEN:4}) ; route UK calls through the * in london
exten => _0044X.,2,Dial(ZAP/g1/${EXTEN}) ; dial local if london is down
exten => _0044X.,102,Busy ; if the called party is busy

exten => _0033X.,1,Dial(IAX2/paris/0${EXTEN:4})
exten => _0033X.,2,Dial(ZAP/g1/${EXTEN}) ; dial local if paris is down
exten => _0033X.,102,Busy

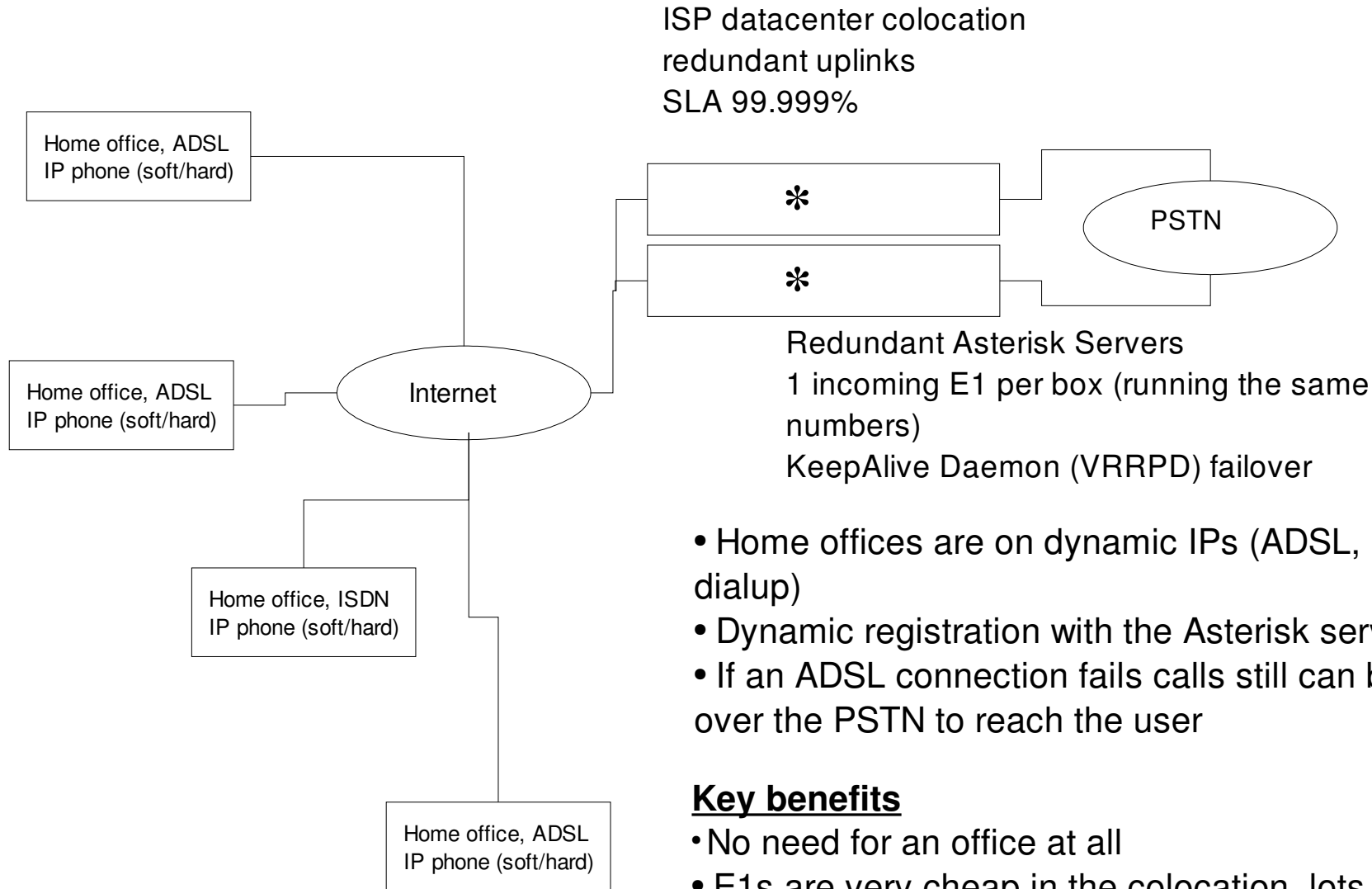
exten => _[1-9]XXXX.,1,Dial(ZAP/g1/${EXTEN}) ; local calls start without a 0
exten => _[1-9]XXXX.,102,Busy

exten => _0[1-9]XXXX.,1,Dial(ZAP/g1/${EXTEN}) ; long distance
exten => _0[1-9]XXXX.,102,Busy
```

iax.conf:

```
[london]
type=peer
username=berlin
secret=supersecret
qualify=200 ; if the latency is more than 200ms
host=london.mydynamicdns.net ; something is wrong
```

Taking it a few steps further.... A Virtual Global Company

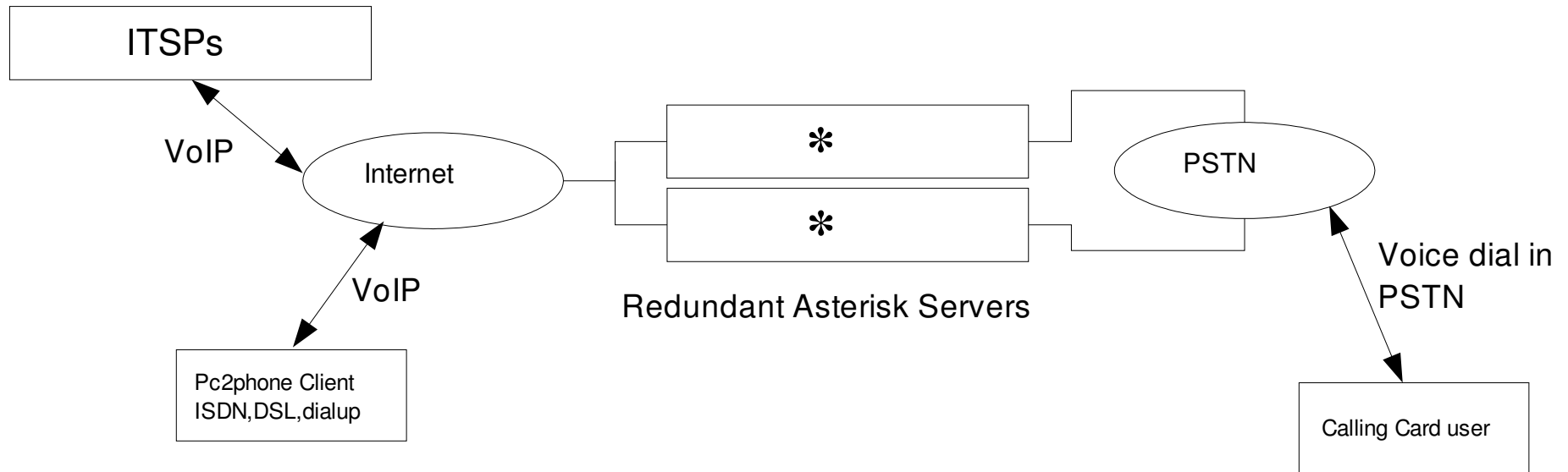


- Home offices are on dynamic IPs (ADSL, ISDN, dialup)
- Dynamic registration with the Asterisk server
- If an ADSL connection fails calls still can be routed over the PSTN to reach the user

Key benefits

- No need for an office at all
- E1s are very cheap in the colocation, lots of telcos to choose from
- Very big conferences can be made with participants from the PSTN or VoIP channels
- Possibility to share the system with several companies (virtual hosting)

Setting up a calling prepaid card application / pc2phone application



- PC2phone users and Calling Card users are authenticated at the Asterisk server
- Calls are routed through several ITSPs (LCR)
- The remaining minutes on the card/account for the particular destination are calculated and a call timeout is set when the call is connected (limited to 1 channel per card/account)
- A realtime billing solution is needed to allow more than 1 channel per card/account, which could be implemented using the Asterisk Manager Interface (over a TCP connection)

Live demonstration of VoIP-to-PSTN and PSTN-to-VoIP termination

- Your PSTN number can be routed to wherever you are without being noticed by the caller.

+49 30 79705392

- ITSPs can provide you with PSTN numbers in foreign countries, so your customers can reach you without paying international long distance rates.

+44 870 1244692

- ITSPs provide affordable international and long distance rates. Sample call routing through global-gateway.net using IAX2.

Try yourself!

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