Hosted PBX

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The Future of VoIP, 14 Oct 2005, Den Haag

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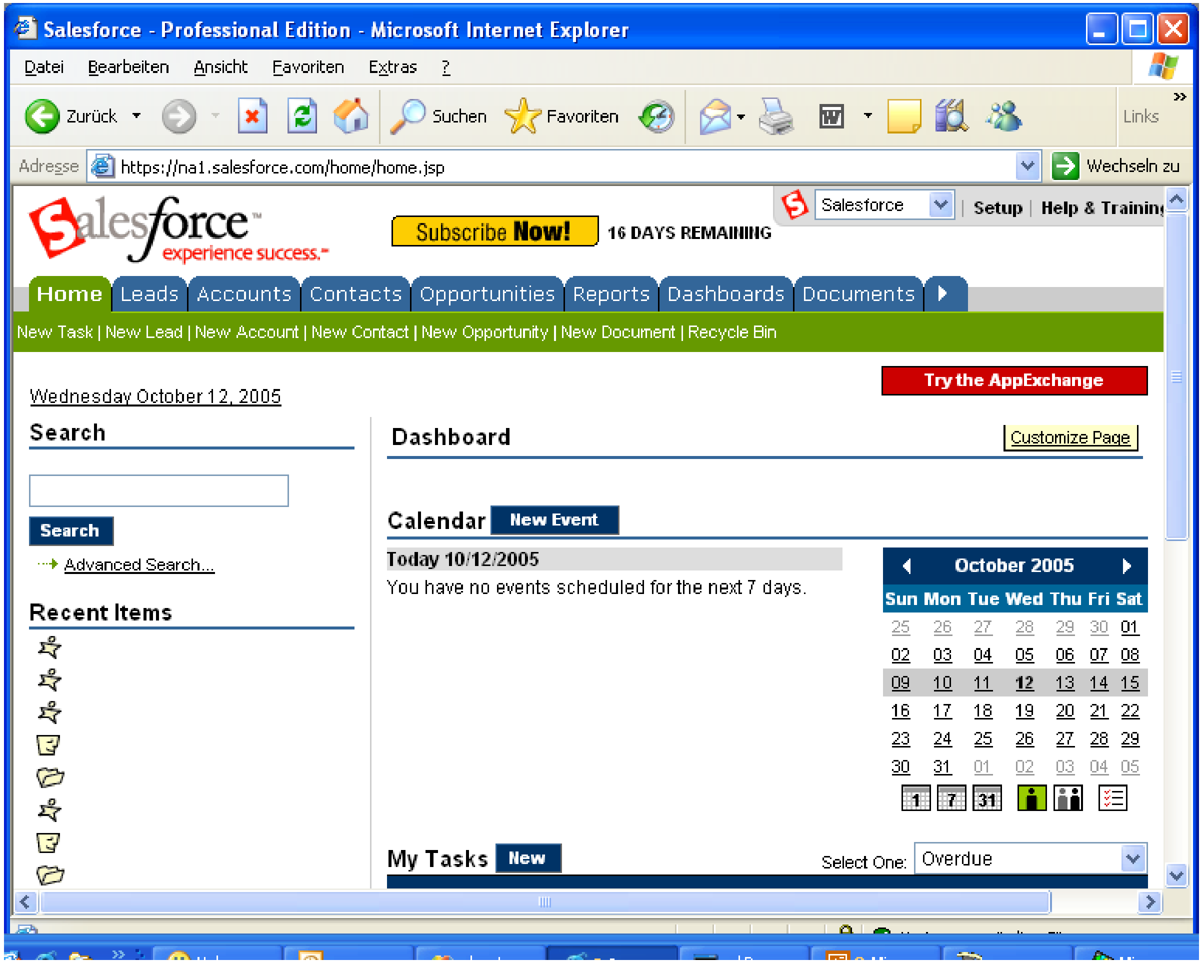
Why are so many people talking about hosted XXX?



- **No hardware required**
- **No hassle with software updates**
- **No hassle with backup**
- **Usually no investment costs**

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While http is clearly the choice for hosted web services, this is not so clear for telephony services

H323

- The first (?) serious protocol for signaling VoIP
- ASN.1 limited the friendship to this protocol
- H.225, H.245 rose questions if H.323 was maybe a little bit too complicated
- H.450 was and is mind-boggling
- Still used, especially between operators

ISDN via Ethernet

- Why invent another protocol?
- ISDN does the telephony part pretty well
- Tunnel the ISDN stuff trough IP
- Most “IP” PBX vendors did it that way and still do it
- And what about presence, IM and other cool stuff?
- Keep the “good old” business rules…
- Americans never understood it, anyway

MGCP

- Goal: replace the cable with a stupid protocol
- Keep the endpoints cheap and stupid
- ASN.1 also in the game, but not mandatory for the endpoints
- Pretty open, but community looked quite “static”
- MEGACO, etc
- Still not 100 % dead

SIP

- Hey, lets copy HTTP!
- Very simple in the beginning (1996?)
- “Made in Berlin”!
- Became quite difficult over time
- Big players say that’s it

Yahooo!!!

- Try the latest yahoo messenger
- All the promises of SIP work there in reality!
- Proprietary
- Unlikely support from the “rest” of the industry

Sk(h)ype

- Here we go again
- Solves NAT!!!
- More details unknown
- Proprietary power 3

IAX

- Are you kidding…

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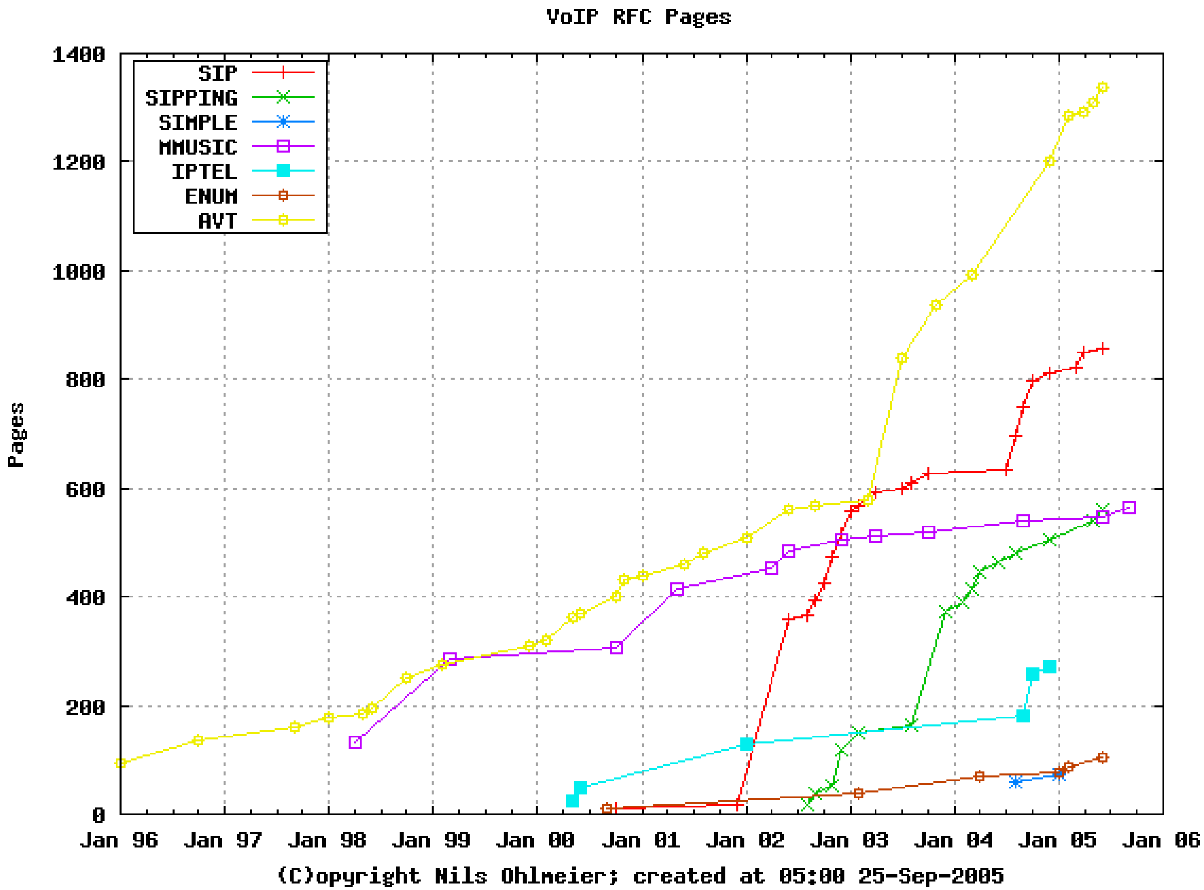
SIP went through a long ripe process already



- Probably more standards than H323
- AVT still pretty active
- Core SIP seems to stabilize
- Biggest change was in 2002

Source: Nils Ohlmeier, www.rfc3261.net

Some optimistic assumptions in SIP are creating a lot of problems

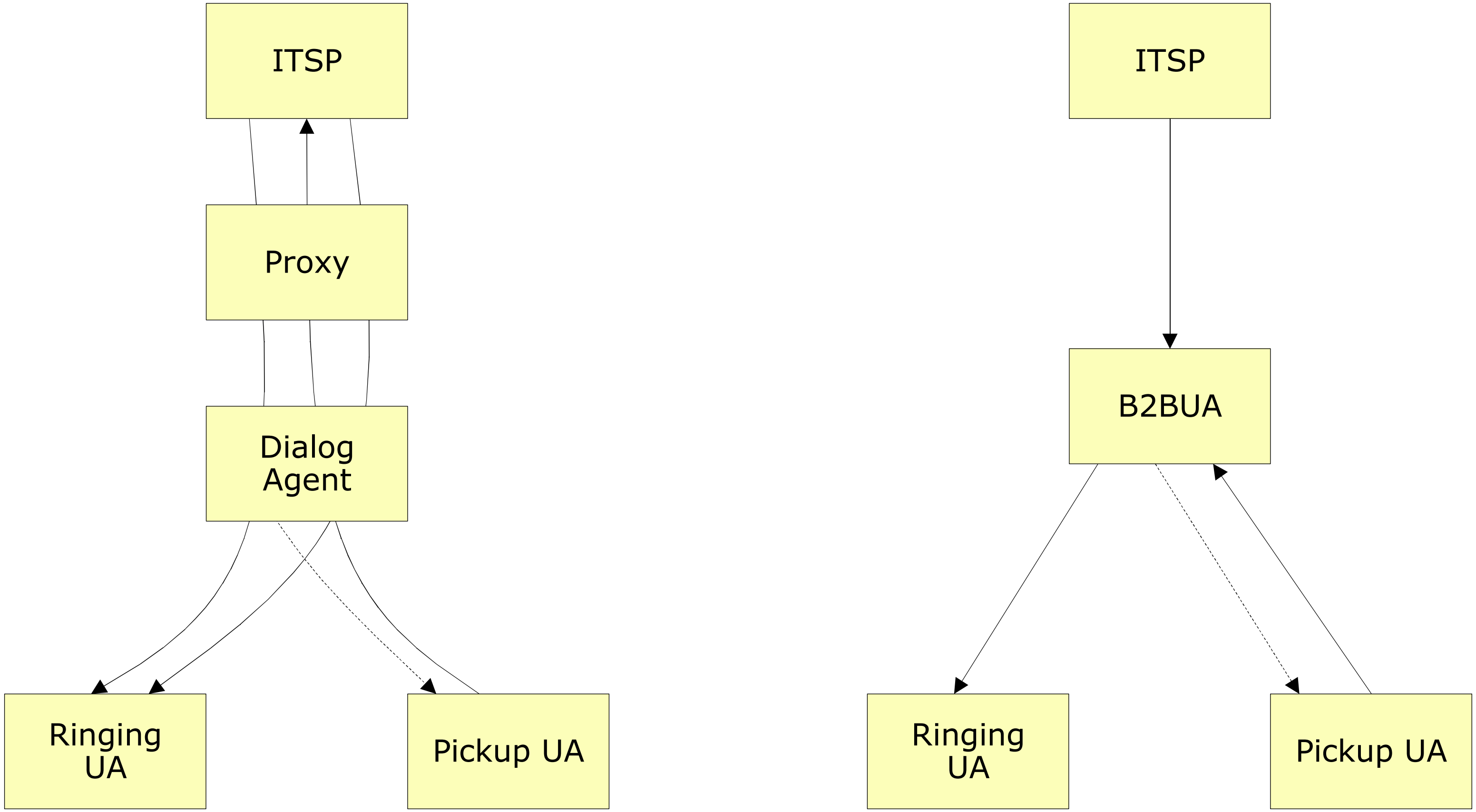
NAT and UDP Assumptions

- **Everybody can talk to everybody**
- **Firewalls are not existing**
- **UDP fragmentation does not exist and packets can become huge**

Proxy Assumptions

- **Forking Proxy add value**
  - Messy handling of 2xx and other return codes
- **Record-Routing with strict and loose routing is easy to implement**
  - Why are most implementations buggy then?
- **The contact is the address of the user agent vs. the route is the real address**
  - How do you indicate the transfer destination then?
  - Try to fix that by GRUU

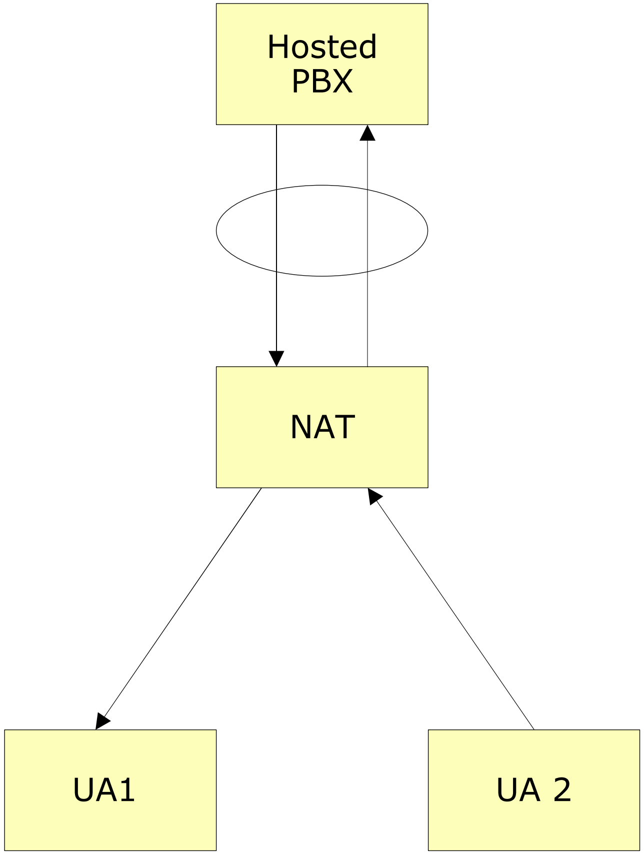
An example of how the IETF call flows make life very hard and how it can be solved the PBX way



The SIP PBX has no other choice than using B2BUA architecture

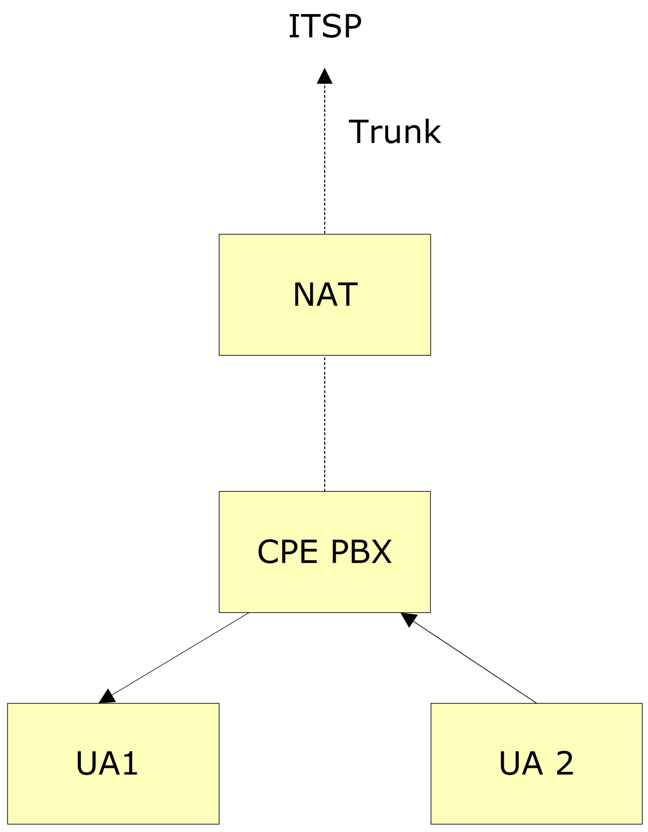
- **Solves the NAT/UDP problem**
  - NAT is more or less under control for the basic call
  - Short routing paths keep packets small (UDP fragmentation)
- **Transfer and other features become trivial**
- **Features do not depend on the other side of the call**
- **Problem: Media Relay**
  - Can be avoided by re-negotiation of the SDP
  - However renegotiation takes time and causes additional interoperability problems
- **The most simple (=working) solution is media relay with transcoding**

The no-trouble hosted PBX must relay its media



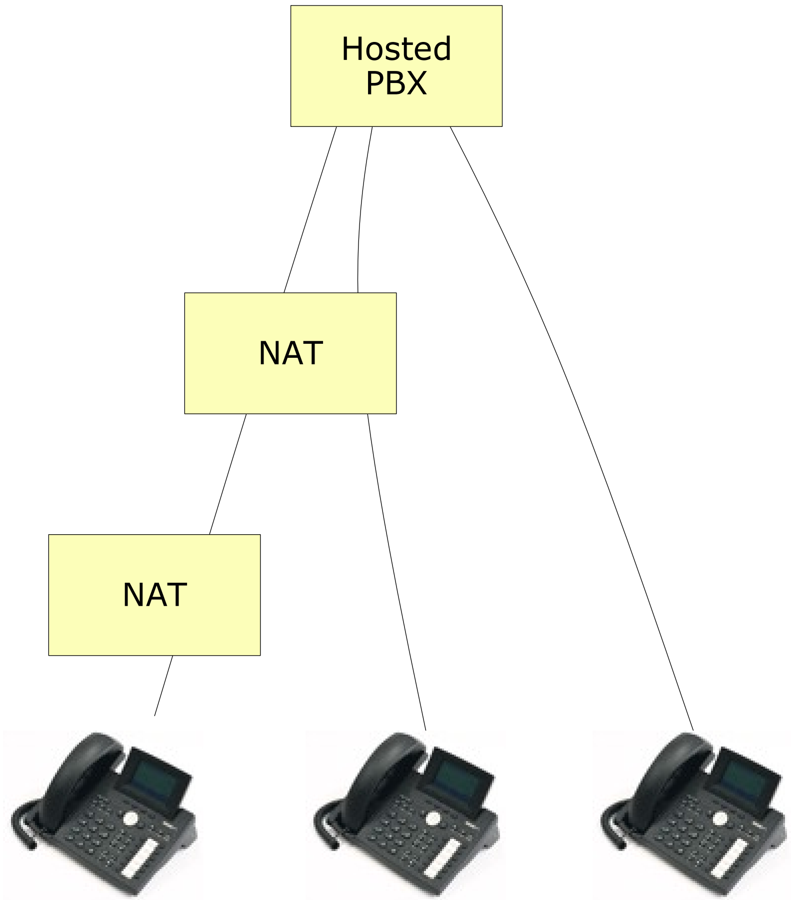
- **No customer premises equipment necessary**
- **Features like recording make the necessity for relay obvious**
- **Double Bandwidth**
  - Like incoming call and outgoing call
- **Double roundtrip time**
  - Like a call to a foreign domain
- **Makes sense when the probability for internal calls is low**

Customer premises PBX solves all problems, but means extra equipment



- **Media relay only inside the local network**
- **Media leaves the organization only trough a trunk**
- **The price tag is the customer premises equipment necessity**
  - Backup, failure prevention
  - However remote management is possible

You must think about security if you are thinking about enterprise communications



- **Permanent TLS connections solve a couple of problems**
  - No NAT problems
  - Security between phone and PBX is guaranteed
  - Key exchange for SRTP can be done in plain text
- **PBX must guarantee that security policy is kept**
  - What about trunks to PSTN?
  - No "End-to-End" security
- **UA must support TLS**
  - This limits the choice for hard phones to snom
- **Limit on number of TCP connections!**

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Examples for PBX that could be used for hosted PBX services

- **Asterisk**
  - Pragmatic implementation of the B2BUA approach with media relay
  - Poor SIP implementation, no TCP/TLS
- **BroadSoft**
  - “Carrier-Grade” B2BUA that avoids media relay
  - Good SIP Implementations
- **Kapsch**
  - “Carrier-Grade”, uses SBC for media path optimization
  - Not offered as product as such, comes with the service
- **pbxnsip**
  - PBX that can be used in CPE and hosted environment
  - Good SIP implementation, supports TCP/TLS and SRTP
- **Sylantro**
  - “Carrier-Grade” with advanced PBX and Centrex functionality
  - Extensive Star code features, no TLS and SRTP.

The Bottom Line

- Hosted PBX is possible today
- Call flows must be kept simple
- Tradeoff between delay for hosted PBX and equipment cost for CPE PBX

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