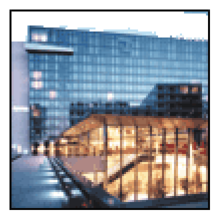


International SIP Conference,
Paris, January 21, 2004

The Disruption Caused by SIP/IP in the Telecom Industry

Henry Sinnreich

MCI

* The opinions expressed here may or may not be those of my company

Outline

- The integration of communications, applications and transaction
  - See the early birds
- New services enabled by SIP
- Endpoint versus network based services: Complexity that was not predicted
- How to preserve the goodness of end-to-end
- CPE complexity has not been predicted either
- Most common errors made by traditional telecom vendors and operators
  - Telecom vendors cannot let loose of central control
  - New providers – new errors
- IETF work on SIP – key directions
- QoS on the Internet
- Why the telecom disruption from SIP/IP is far from over

Integration of IP Communications with MS Office 2003

Mail folders

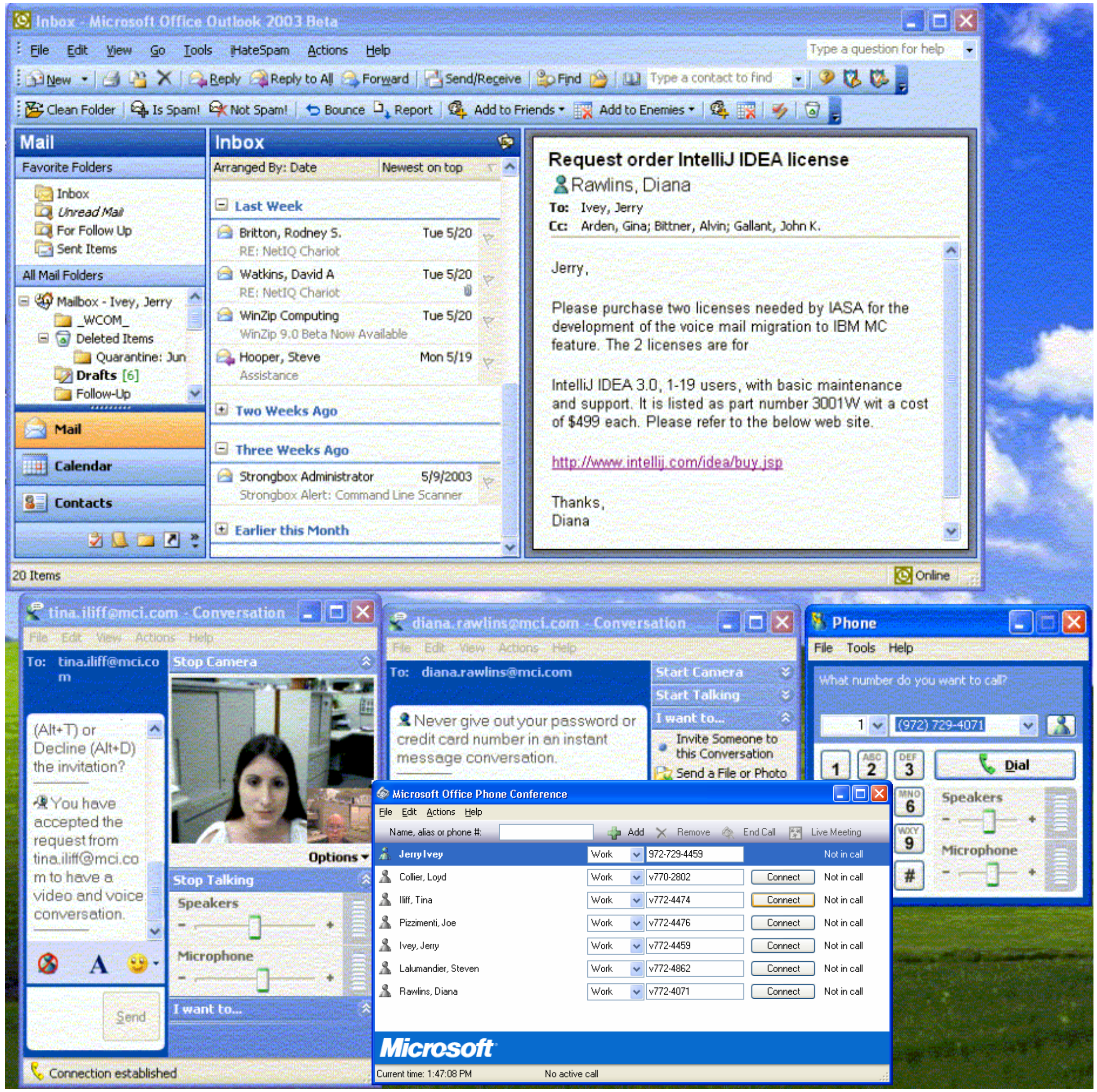
E-mail

Mail
Calendar
Contacts

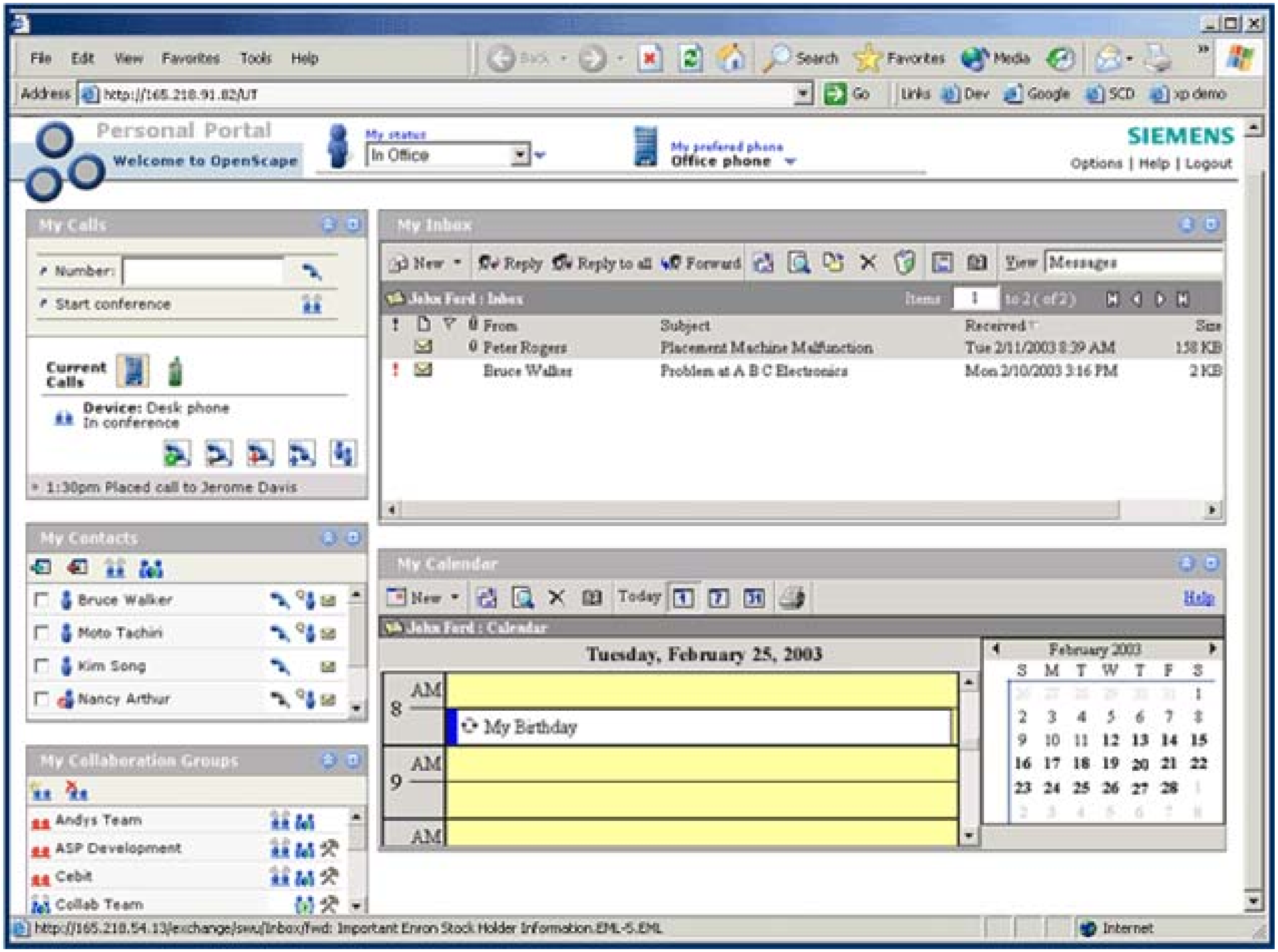
Phone call

IM, voice, video and data call

Office Phone Conference

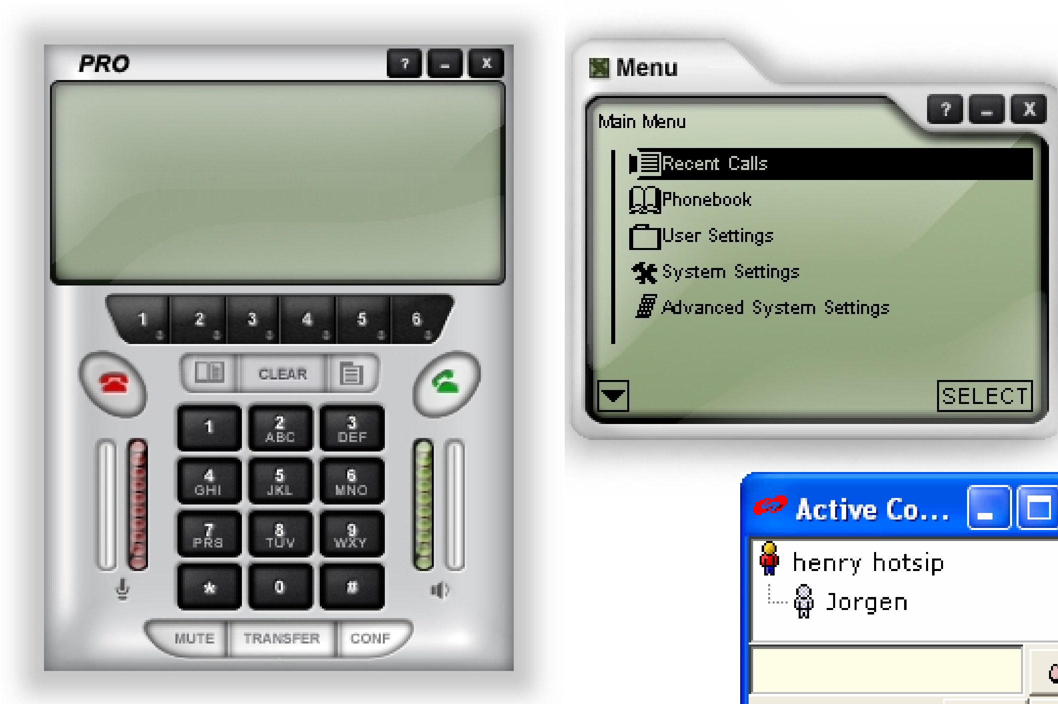


Integration: Siemens Openscape



Get control of your time, your tasks and your communications

WLANS are home for SIP



X-PRO for Pocket PC V2.0

HotSIP Active Contacts

http://xten.com/index.php?menu=products

http://www.hotsip.com/products/hotsip_active_contacts/hotsip_active_contacts_skins6.asp

The Value Proposition of IP Communication Services

Higher service resilience than PSTN – proven on 9/11/03 and 8/15/03 E.C. black-out,

More than one service provider – see above

Better voice quality than PSTN, new

Multimedia: Text, voice, video, data, new

Mobility for all communication services -new

Presence based services - new

Event based communications - new

Integration of voice mail, e-mail, IM, SMS

Multiple conferencing models and media - new

Call routing heaven + ENUM - new

Secure communications

User preferences and control for all of the above- new

Integration with the Web (new!):

Communication, information, productivity apps, entertainment, transactions

Gateways to PSTN, mobile telephony, paging networks, ISDN, H.323, etc.

100% open standards based, multi-vendor interoperable- new

Service development is easy and fast - new

Bottom line: Lowest overall cost and highest functionality combined

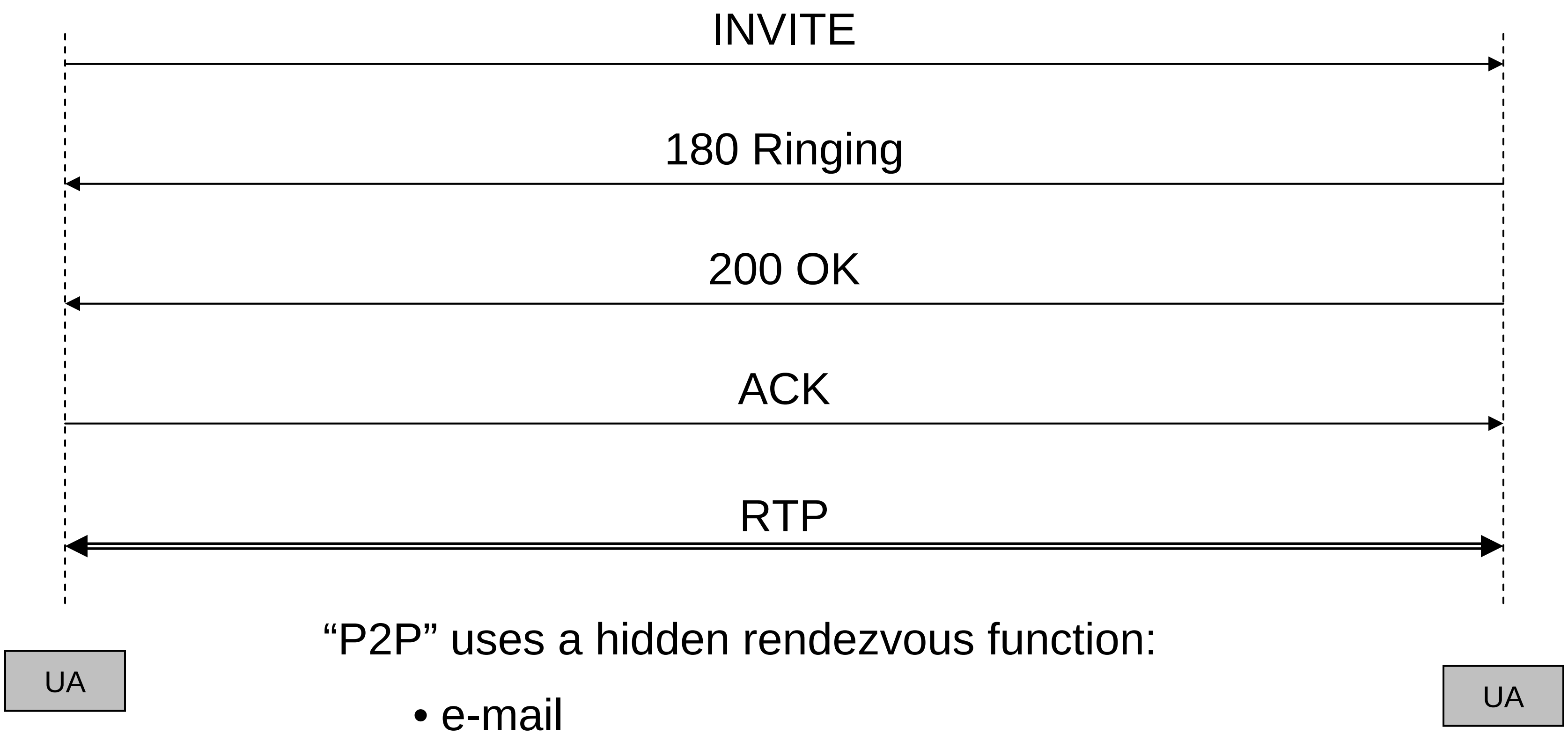
Endpoint versus Network Based SIP Services

SIP and Internet communications have quickly developed from the simple e2e model to multi-network and multi-application interoperability*

Is the complexity of Internet communications following the path of circuit based telecoms?

* Slides 7-17 were jointly developed with Alan Johnston/MCI

SIP started as Endpoint based e2e SIP and RTP

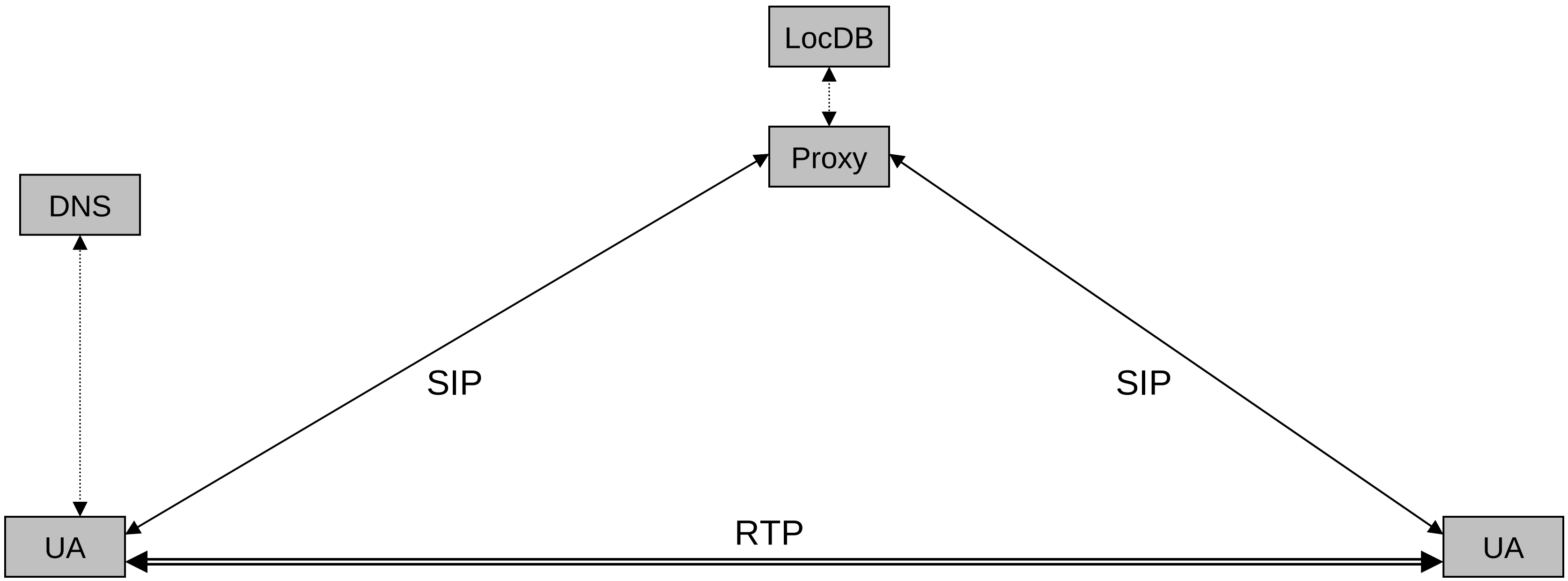


"P2P" uses a hidden rendezvous function:

- e-mail
- phone
- DNS
- some other server

P2P is also not scalable, but is a nice try (Skype)

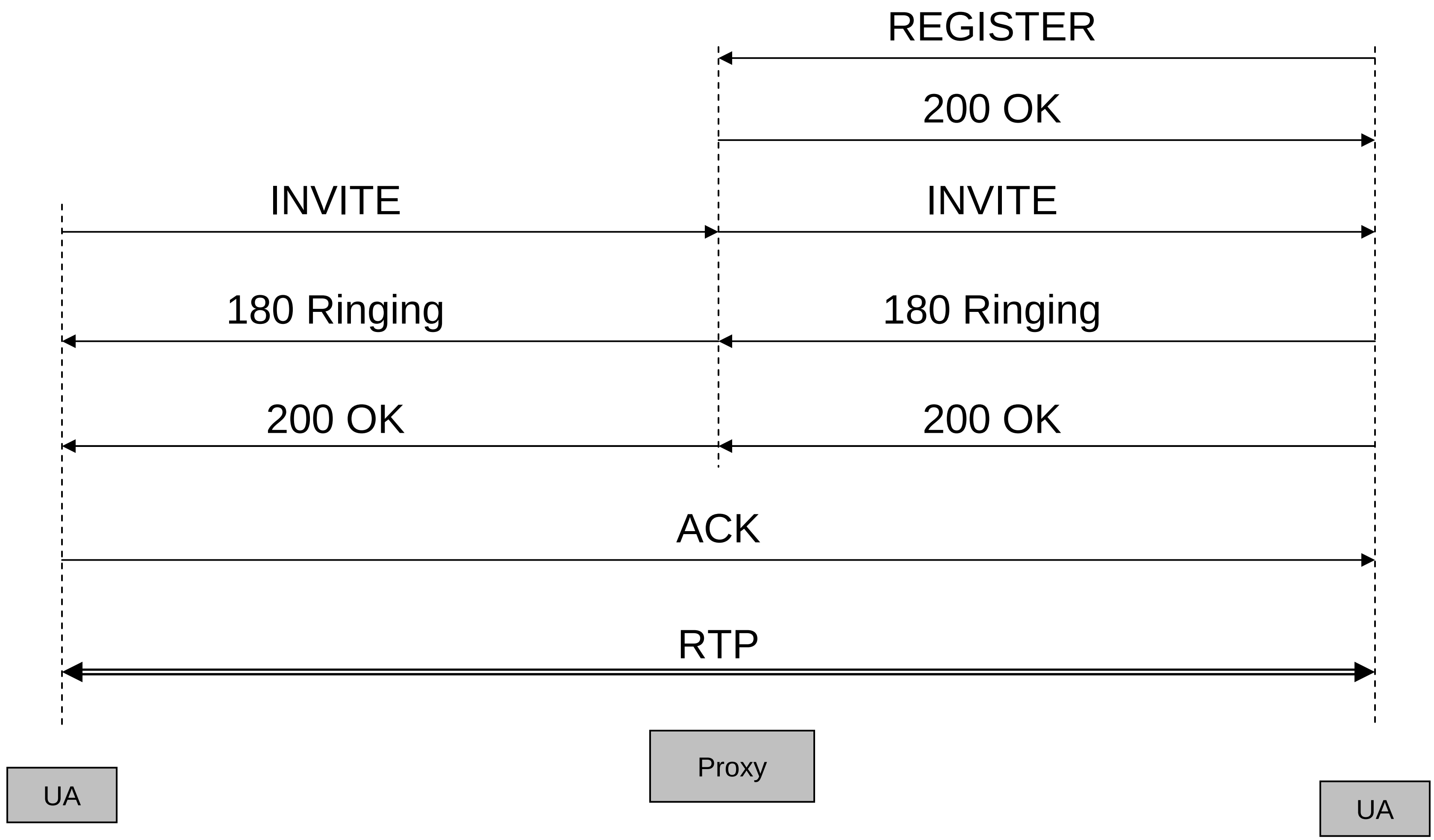
SIP Proxy Servers and REGISTER solve the rendezvous problem



Endpoints register with a proxy server and use an AOR URI to reach each other.

Basic SIP allows proxy to drop out of dialog starting with the ACK

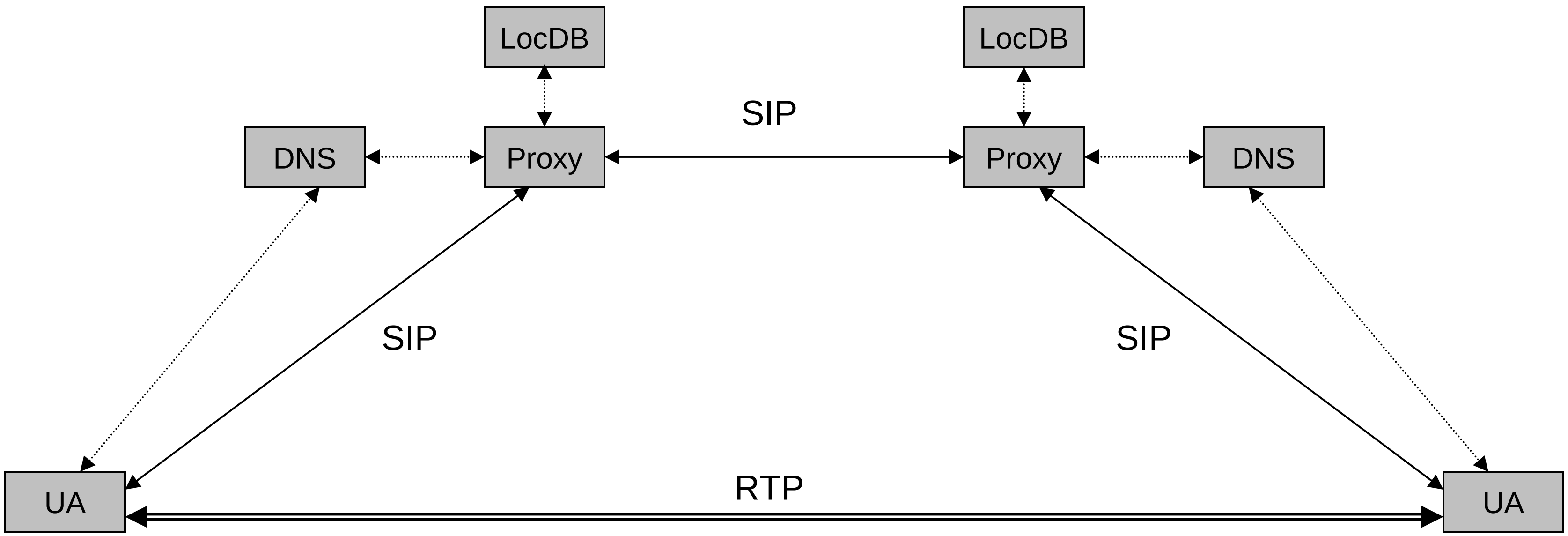
E2e with the help of a proxy server



Proxy does not keep call state information and does not stay in the signaling path starting with the ACK.

The SIP-RTP Trapezoid (RFC 3261)

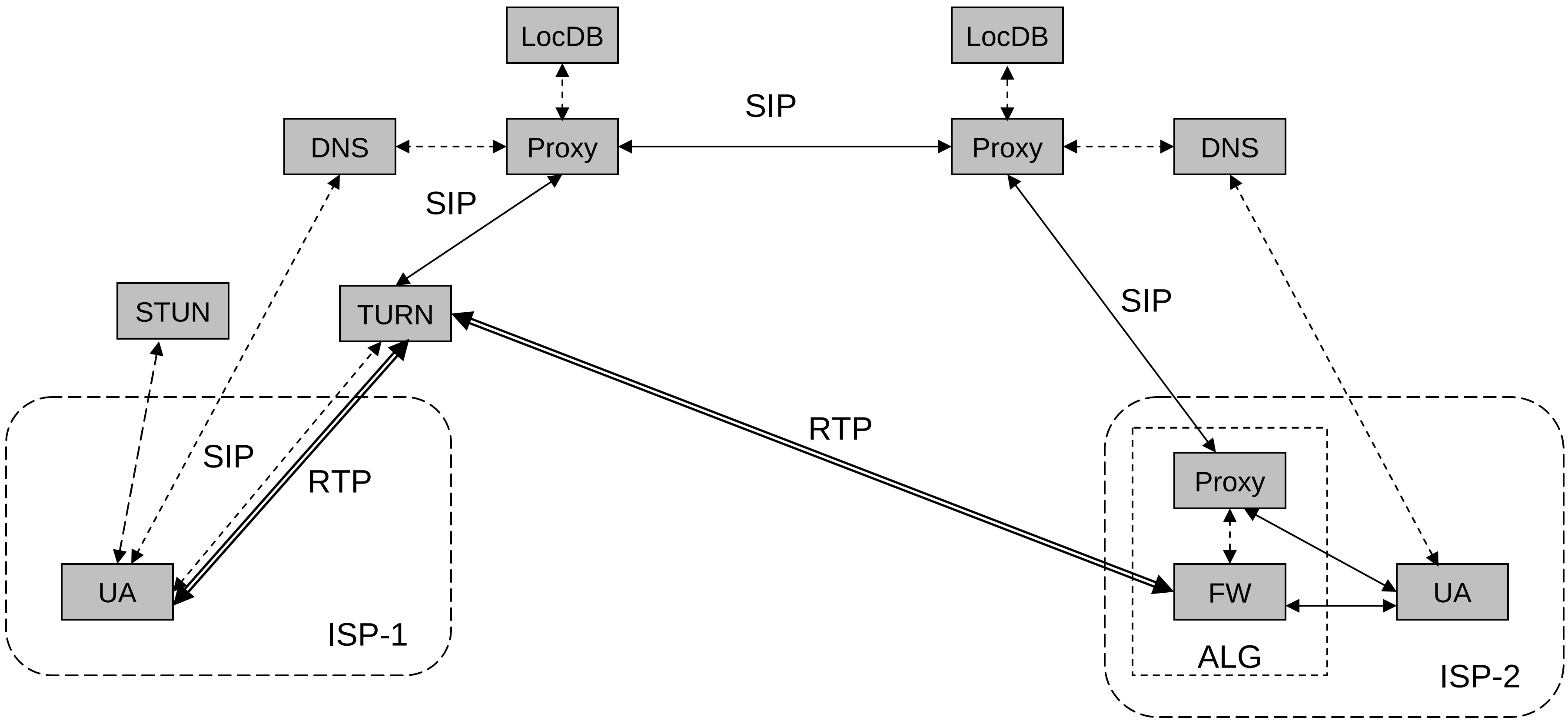
provides local control and service functions



Both proxies typically Record-Route in order to stay in the signaling path.

As long as Proxies obey RFC 3261 rules, SIP is still close to e2e (Proxies can be transaction stateful, not call stateful.)

Firewalls, NATs and local SIP proxies



Note: STUN and TURN servers are used for traversal of NAT in this ISP network

ALG is used for Firewall traversal in this ISP network.

Options for Firewall Traversal

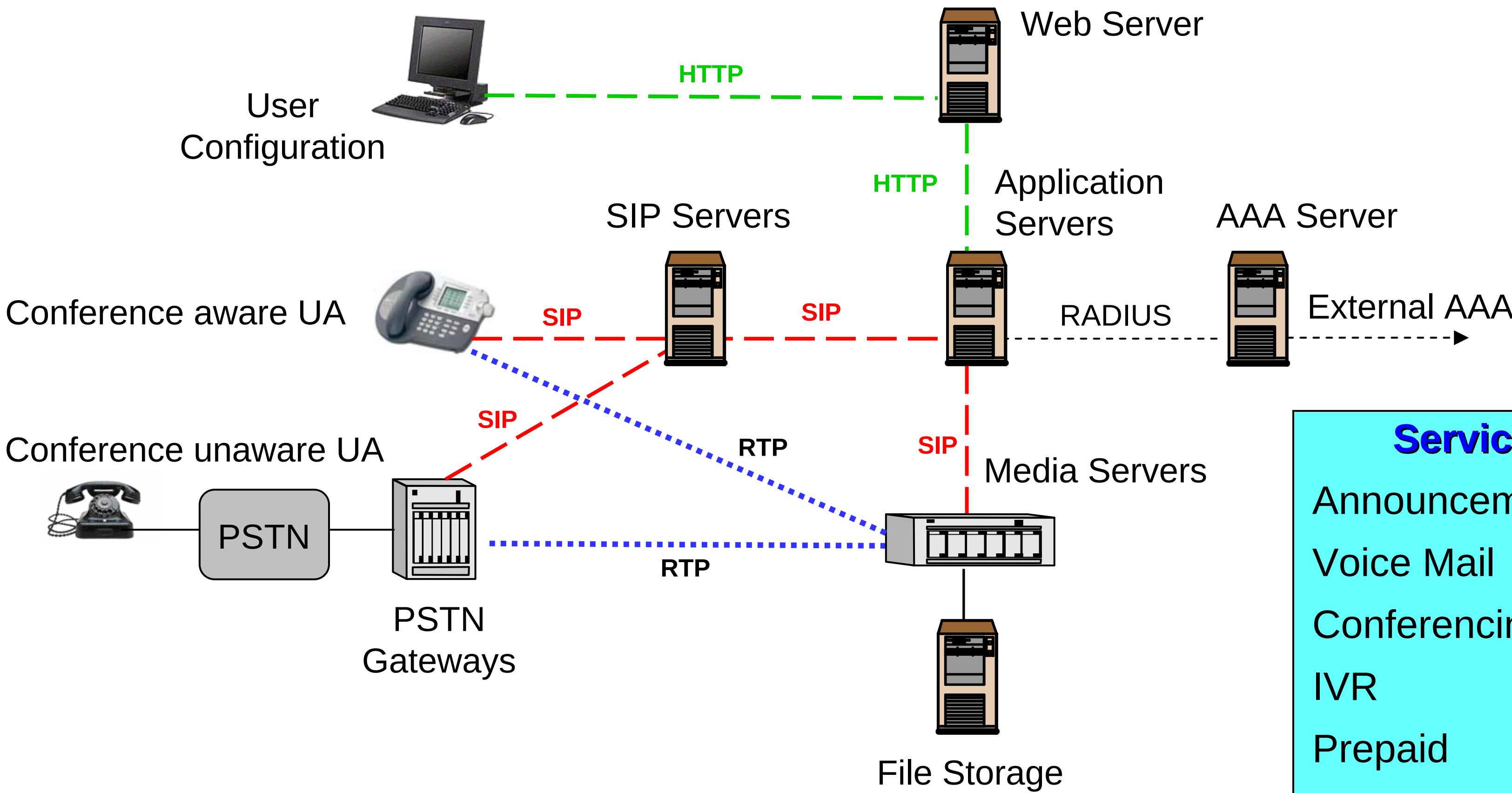
ALG (B2BUA)

- Breaks e2e
- ALG terminates SIP session and re-originates the dialog
- Can be separate from firewall.

SIP enabled firewall proxy

- Is close to e2e while still preserving security
- Proxy authenticates and selectively opens “pin” holes for RTP media.
- Needs MIDCOM protocol to separate from firewall.

Many service components support 'e2e' calls



Services

Announcements

Voice Mail

Conferencing

IVR

Prepaid

Autoattendant

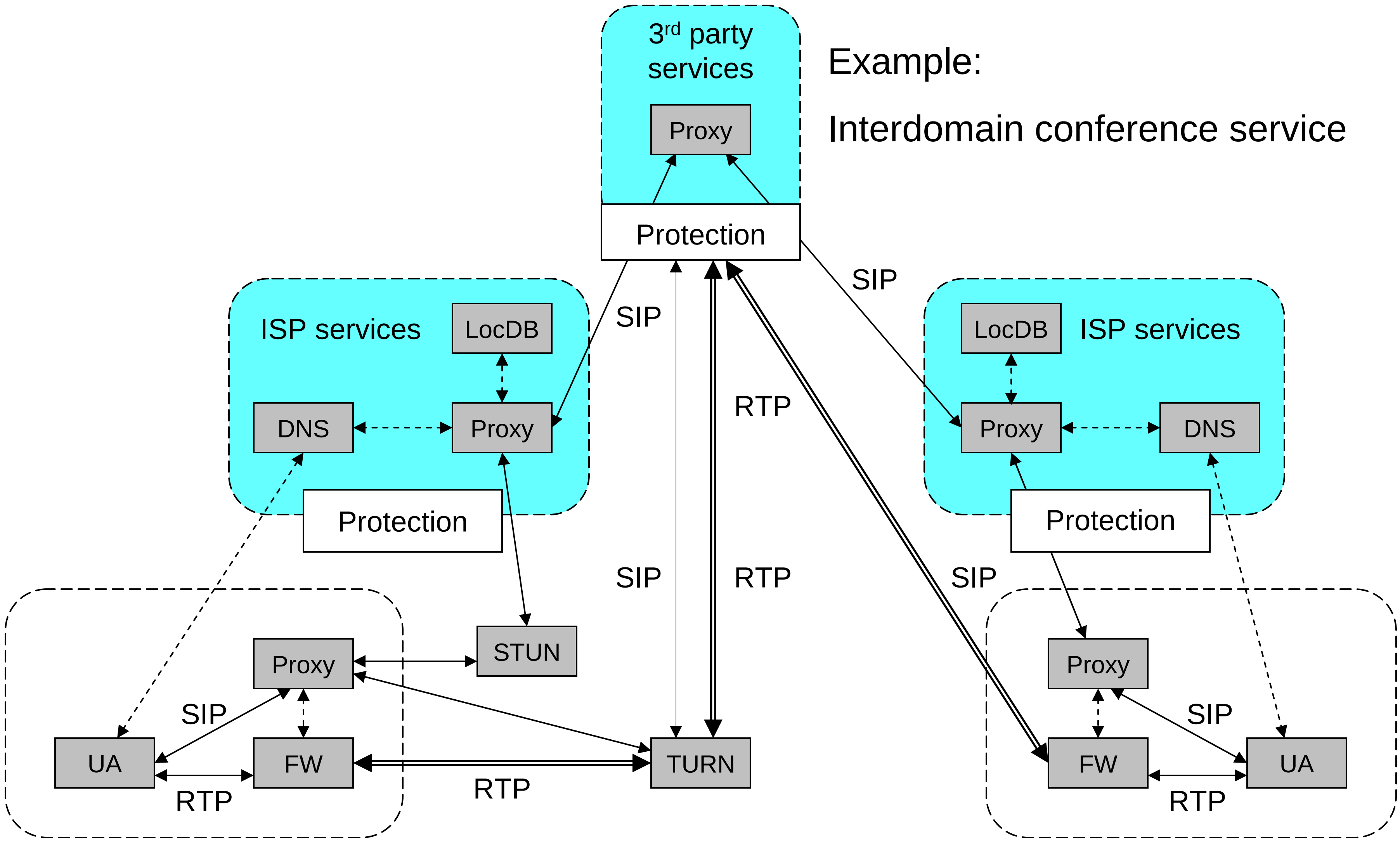
Centrex

Interworking of all network elements is a complex undertaking

Strict adherence to standards makes the interworking manageable

New services and new network elements should require minimal regression testing

ISP and 3rd party services



Example:

Interdomain conference service

How to preserve the goodness of e2e

Why is e2e valuable?

- Flexibility at the edge
- Enables innovation
- Scalable
- Enables integration with local IT and personal apps. This can be done only at the edge of the network
- Prevents spam and telemarketing…

Design principles

- User has choice
- User has control

or

- Inform the user
- Get user consent
- Components, not closed bundles

These guidelines are valid for any type of Web/IP service and have been applied to all IETF SIP standards.

They characterize the difference between Internet communications and proprietary or H.323 or master-slave MEGACO/H.248 VoIP protocols.

Reference

"The Rise of the Middle and the Future of End to End: Reflections on the Evolution of the Internet Architecture" by James Kempf and Rob Austein. IAB, March 2003, work in progress.

<draft-iab-e2e-futures-01.txt>

Dilemma for ISPs: B2BUA AKA Session Controllers

Pros (especially the <u>underlined</u>)

Many useful functions
- <u>Simplest FW/NAT traversal</u>
- Centrex
  - <u>Call park</u>
- SIP-SIP peering
- SIP-H.323
- IP PBX peering
- Metering
- Policy enforcement
  - Routing optimization
  - Access control
- QoS
- Dial plans
- CALEA
- Anonymity
- Topology hiding
- BW compression
- QoS monitoring
- …etc.,…

Lowest *initial* cost for all ISP business!

Cons

May block new service development
May not handle Presence, IM, video, etc.

If inside is compromised
- Telemarketing calls
- SPAM
- Theft of service
- Customer traffic data
- Customer voice (B2BUAM)
- Private IP addresses

Requires highest security environment

B2BUAWM requires double BW for ISP

Standards instead of B2BUA: Complexity

Function	IETF standards compliant approach
DHCP traversal	Dynamic DNS
NAT traversal	STUN, TURN servers, ICE, UPnP
Firewall traversal	SIP enabled firewall, UPnP
Centrex	draft-ietf-sipping-service-examples-05
SIP-SIP peering	SIP
SIP-H.323 peering	SIP-H.323 signaling gateway
IP PBX peering	SIP trunks
Metering	SIP session counting
Route optimization	SIP proxy
Access control	SIP proxy policy control
BW compression	RFC 2508, VAD in codecs
QoS	DiffServ on access link
Dial Plans	SIP proxy
CALEA	draft-baker-slem-architecture-02.txt
Anonymity	TURN, draft-dcsgroup-sipping-arch
Topology hiding	RFC 2543 Hide header field

B2BUA: Open Edge Pluggable Services WG

Inform: Services provided in the OPES framework should be traceable by the application endpoints of an OPES-involved transaction, thus helping both service providers and end-users detect and respond to inappropriate behavior by OPES components.

Consent: …must include authorization as one if its steps, and this must be by at least one of the of the application-layer endpoints (i.e. either the content provider or the content consumer).

Reversible: In particular, services provided in the OPES framework should be reversible by mutual agreement of the application endpoints.

http://ietf.org/html.charters/opes-charter.html

Checklist for B2BUA's

Does it require application intelligence?

- For existing applications (example: Centrex and conferencing)
- For planned applications

Call flows compatible with the systems architecture

- Interoperability testing with SIP proxies, gateways, telephony devices

Is the behavior well defined and testable?

Security Considerations*

- Attack scenarios (DOS, silencing a client, stealing of identity, eavesdropping)
- Compromising a B2UA: Risk assessment
- Countermeasures

*draft-ietf-midcom-stun-04.txt

The Outlook for B2BUA’s

For practical reasons, ISP’s will deploy B2BUA’s

- **Do Networks Operations have the call flows, timers, etc. to run the B2BUA?**
- **Can new services be deployed without B2BUA upgrades? Non-voice?**
- **Other new e2e transparency based services?**

How can B2BUA’s support SIP mobility?

The effect of low cost SIP enabled IAD’s? SIP aware router/FW/NAT?

Intertex IX66 “SIP Switch”

Integrated Access Device

D-Link

CPE complexity that has not been foreseen

Integration of complex CPE

- IP router
- Firewall/NAT/DHCP
- UPnP
- Dynamic DNS client on WAN side
- WAN link voice/data QoS policy
- WAN link voice priority (DSCP)
- SLA monitor (RTCP extension reports)
- Local priority for voice
- Ethernet hub
- 802.11x wireless access points
- 802.1x port authenticator
- Local SIP proxy/registrar (FW/NAT ctrl and mini-PBX)
- Local gateways to PSTN (FXO ports)
- Local gateway for PBX/key system (FXS ports)
- Message waiting indicator (MWI)
- T.38 fax and interactive text support (FXS ports)
- Emergency (911) support

No single product has all these functions at present

* This is a far cry from the ATM based “multi-service” switch pursued for many years by the legacy telecom industry and is a showcase example of its failure to plan technology development.

IETF SIP and SIPPING Working Groups

Innovations that will change communications…

SIP System Architecture

- Multi-party call control with extensions
- Third party control BCP
- Content Indirection
- Globally Routable UA URIs (GRUU)

SIP Call Flows

- Basic
- With PSTN gateways
- Centrex/PBX style
- **Bridged appearances**

Caller Preferences Extensions with multiple use cases

Intermediaries

- NAT traversal: ICE based on STUN and TURN
- End-to-middle security using S/MIME
- SIP identity inserted by intermediaries

Event architecture – is IP specific and Internet-wide applicable

- Dialog event package
- Message waiting indication event package
- Limiting the rate of event notifications

IETF SIMPLE WG: Presence

Presence Events

- SIP extension for publishing event state
- Event package for SIP
- Event lists for resource lists
- Presence specific event notification filtering

Presence data format

- XML based format for watcher information
- Rich presence information data format

Policy

- Simple presence publication requirements
- Presence data manipulation requirements
- Filtering of watcher information
- XML configuration access protocol (XCAP)
- XCAP for setting presence authorization

Efficient delivery of presence information: Requirements and use cases (for 3GPP)

SIMPLE for Presence and IM

Short list of objectives

- Global-Internet wide standards based (no gateways)
- Presence is a generic event for all applications
- Same communication stack for all applications
- Same global routing infrastructure
- Same data sets and databases
- Same servers
- Same UAs as for other media
- Same authentication, message integrity and privacy
- E2E security, replay, DOS and other protections

Internet Conference Services

Integration of conferencing with calendaring and scheduling

Presence based conferencing

Change conference model and media ad-hoc

- Migrate from IM session to voice call
- Voice call to audio conference
- Voice conference to video conference
- A/V conference to collaboration through document sharing

All this without hanging up from the original call/session and while moving around between different end devices!

Distant learning – virtual classrooms

Advanced web call centers – multimedia with live agent

SIP for the hearing disabled is a special conference application

See XCON WG http://www.ietf.org/html.charters/xcon-charter.html

Telecom legacy errors

Are ‘softswitches’ and IP PBXs alternatives to SIP?

The proprietary IP PBX and softswitch are *Internet unaware*:

- Telephony-voice centric: PSTN & PBX emulations
- Services are unavailable outside of enterprise/ISP limits
- Central control
- Proprietary closed systems
- Ownership risk: There is no 2nd source for
  - phones
  - servers

  Traditionally designed to be not interoperable (some rare recent exceptions)
- Ownership cost: High for maintenance & custom development
- No standard presence
- No standard mobility

No integration with the web: Info, application, transactions

Single advantage: Turnkey systems

SIP Device Interoperability and Voice Quality

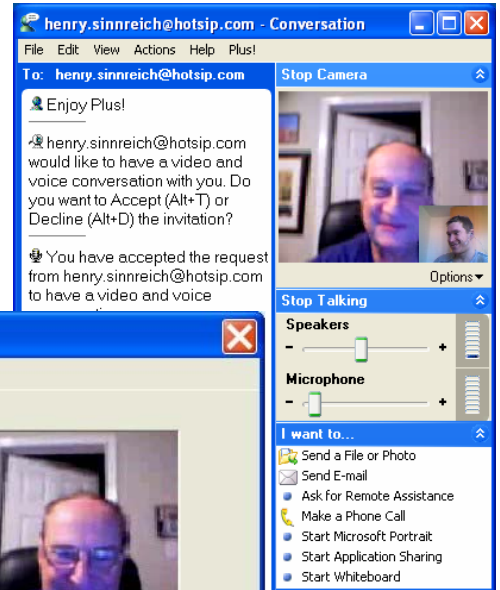
G.722 (and GIPS) 16 kHz sampling

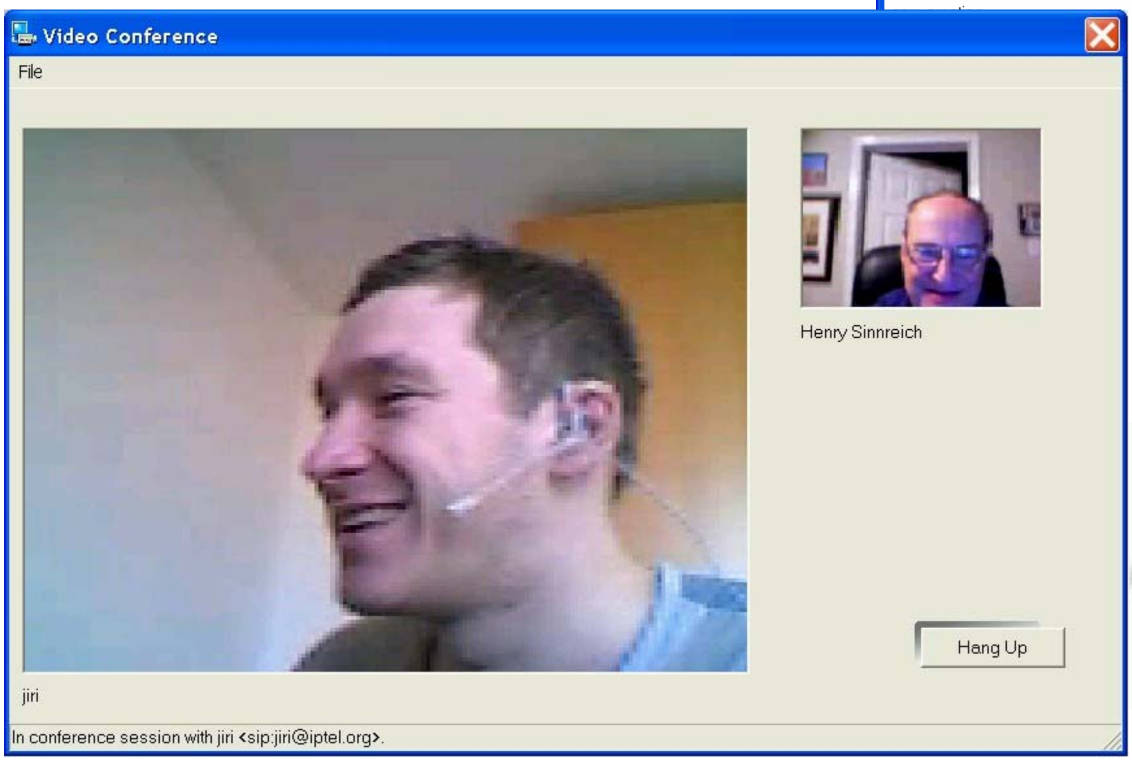
Messenger video

Seen in Berlin

Seen in Richardson

HotSIP large video





PSTN can be completely avoided

SIP Internet Voice Path: Dallas - Berlin

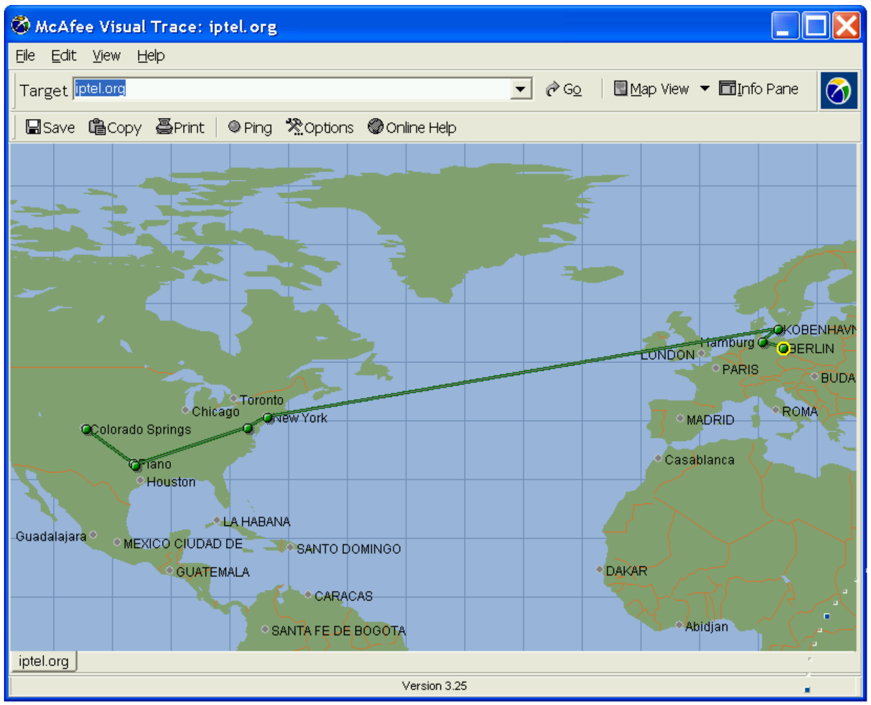
Better than PSTN voice on the Internet

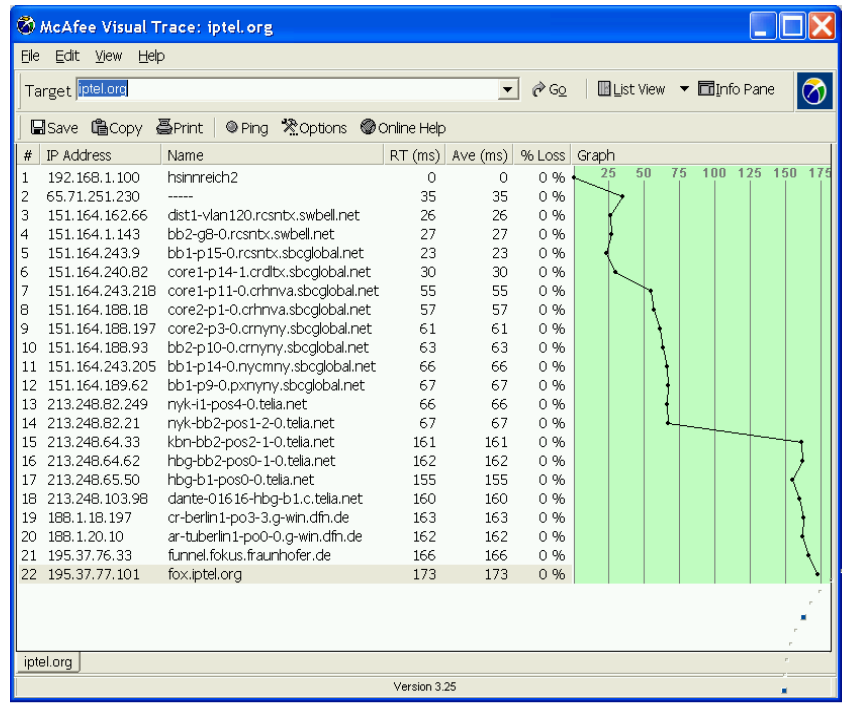
Path traverses 4 public networks and 22 IP router hops

CD quality sound with HotSIP softphone and GIPS codec

Consistent quality for over a year of observation

Yokohama-Dallas is of similar quality as experienced at the 54 IETF meeting

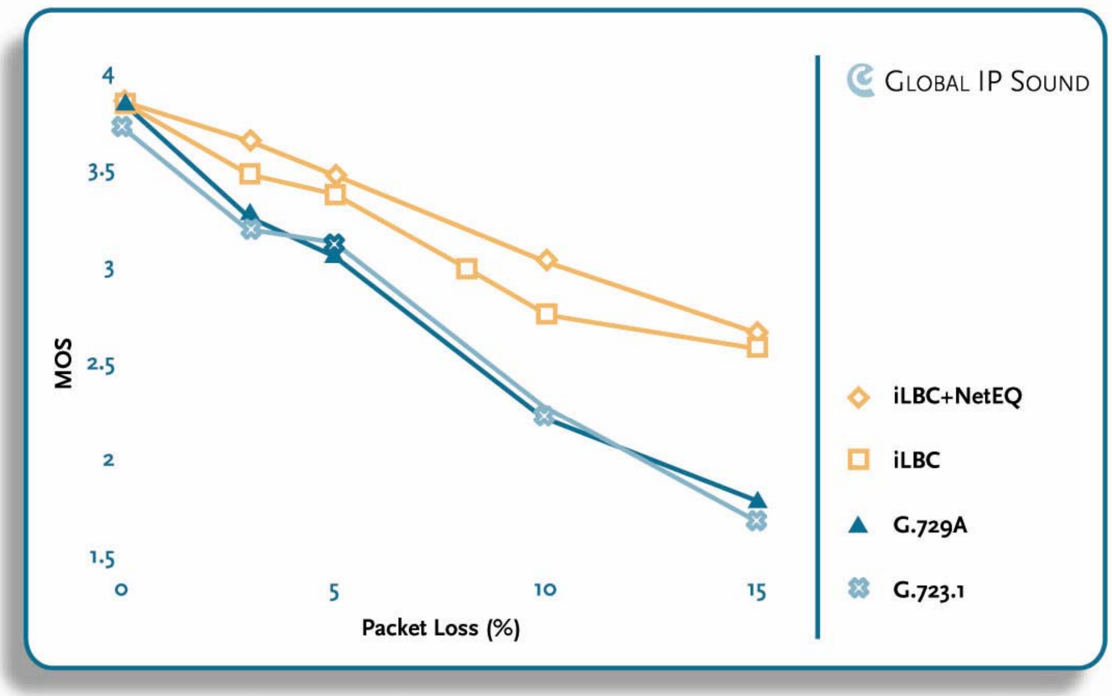




Conclusion: SIP services work well globally on the Internet ‘as is’

Single Internet Codec

(Internet standards are always better and license free)



The tests were performed by Dynstat, Inc., an independent test laboratory.
Score system range: 1 = bad, 2 = poor, 3 = fair, 4 = good, 5 = excellent

http://www.ietf.org/internet-drafts/draft-ietf-avt-ilbc-codec-00.txt

http://www.ietf.org/internet-drafts/draft-duric-rtp-ilbc-01.txt

Conclusion: Telecom disruption from SIP is far from over

Wireless surpasses wired telephony, 3G uses SIP, 4G is home for SIP

Largest carriers* (MCI, AT&T) consolidate all traffic on IP backbone

What happens to legacy networks (TDM, ATM, SONET) and telecom industry?

Regulation and taxation?

Mistakes: Rebuilding TDM over IP, who pays?

The impact of SIP has already started

The complexity of integrated SIP/IP communications, applications and transactions will fuel development for many years to come, see the early birds.

* References

http://www.nwfusion.com/news/2003/1201eslambolchi2.html

http://www.channelsupersearch.com/news/crn/41598.asp