SIP: 30” Tutorial

(“All you need to know” does not fit in 30” – send me an E-mail or grab me in bar for follow-up questions.)

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iptel.org Credit History

- iptel.org, a Fraunhofer organization, focuses on VoIP consultancy and manufacturing of SIP servers.
- iptel.org has been running SIP services on the public Internet since 2001. Users are able to pick an address username@iptel.org and a numerical alias. Increase in population size since introduction of Windows Messenger.
- Mostly used applications: VoIP, instant messaging and presence, voicemail2email.
- The infrastructure serves public subscribers as well as internal users with additional privileges.
- Services powered by iptel.org's open-source SIP server, SER. The server is freely available on the Internet and it powers numerous well-known SIP sites.

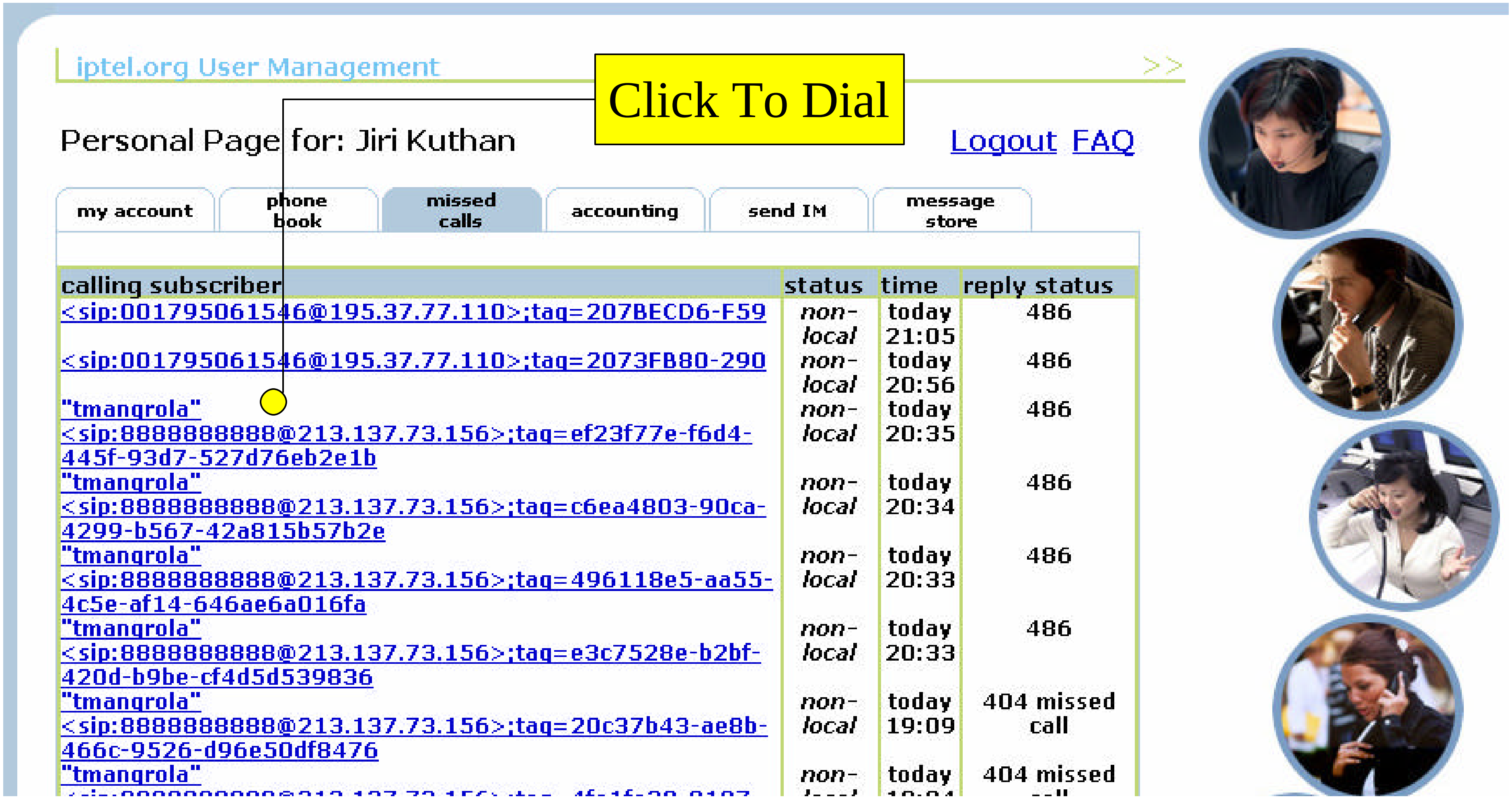
Outline

- Motivation
  - About Internet Telephony Application Space
  - Usage Scenarios for SIP
  - Feasibility Check: How Much Does It Cost?
- Technology:
  - SIP Refresher
  - Complementary Technologies
  - Concern Stack
- Conclusions

Convenience Applications

- What does make existing deployments use SIP? Applications and Cost effectiveness.
- The application driver is convenience.
- Applications demanded and deployed are mostly about service integration:
  - E-mail: replacement of IVR annoyance with voicemail-2-e-mail
  - Web: read list of missed calls from your webpage (both off-line and on-line) with click-to-dial.
  - Web: online phonebook.
  - Instant Messaging and Presence, Notification services (T-sturm alarm), SMS delivery
  - Telephony: conferencing

Example: Web Integration, Missed Calls/Click-to-Dial

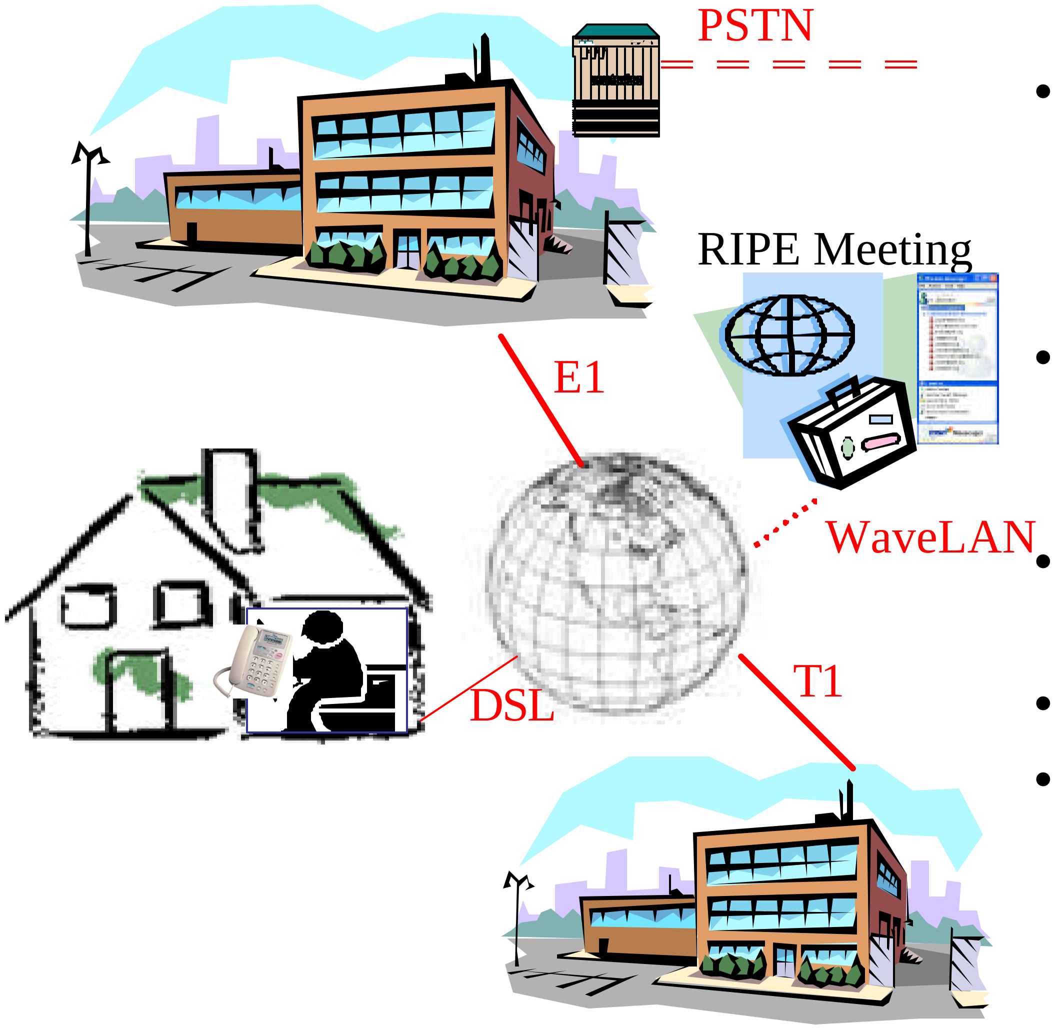


calling subscriber	status	time	reply status
<sip:001795061546@195.37.77.110>;tag=207BECD6-F59	non-local	today 21:05	486
<sip:001795061546@195.37.77.110>;tag=2073FB80-290	non-local	today 20:56	486
"tmangrola" <sip:8888888888@213.137.73.156>;tag=ef23f77e-f6d4-445f-93d7-527d76eb2e1b	non-local	today 20:35	486
"tmangrola" <sip:8888888888@213.137.73.156>;tag=c6ea4803-90ca-4299-b567-42a815b57b2e	non-local	today 20:34	486
"tmangrola" <sip:8888888888@213.137.73.156>;tag=496118e5-aa55-4c5e-af14-646ae6a016fa	non-local	today 20:33	486
"tmangrola" <sip:8888888888@213.137.73.156>;tag=e3c7528e-b2bf-420d-b9be-cf4d5d539836	non-local	today 20:33	486
"tmangrola" <sip:8888888888@213.137.73.156>;tag=20c37b43-ae8b-466c-9526-d96e50df8476	non-local	today 19:09	404 missed call
"tmangrola"	non-	today	404 missed

Scenario: Internet Telephony Providers

- Borderless customer base: Services available anywhere on the public Internet to subscribers very much like E-mail.
- Low CAPEX and OPEX.
- PSTN connectivity typically offered as an extra option; (example: deltathree charges <$.1 per US2UK minute and $11 a month for a US 800 number)
- Freebies: FWD, PCH, iptel SipPhone.
- PSTN-termination: deltathree, packet8, Vonage

Scenario: Use In Enterprises

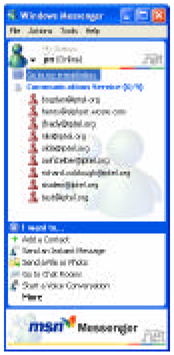


- Services available to all company’s users, on-site, off-site and multi-site – toll bypass.
- No telephone line required for home-workers and remote offices.
- Single infrastructure for data and voice.
- Effectiveness tools.
- Service operation can be outsourced in a Centrex-like manner (MCI Advantage). Like with web/email, single server may host multiple domains.

How Much in 2003?

- Very little! With IP infrastructure, a host and a skilled administrator already in place, PC-to-PC telephony is free:
  - Softphones Free (Windows Messenger, X-Lite)
  - Servers available freely (SIP Express Router)

- $0



- Your grandma does not want to talk through a PC? Buy her a hardphone. A freebie SIP site (sipphone.com) ships a pair for $129.99.

- N x $65

- Gateway for PSTN connectivity? Commercial T1/E1 gateways begin at $2500, software for experimental PC-based gateways available on the Internet for free.

- $2500

How Much Effort?

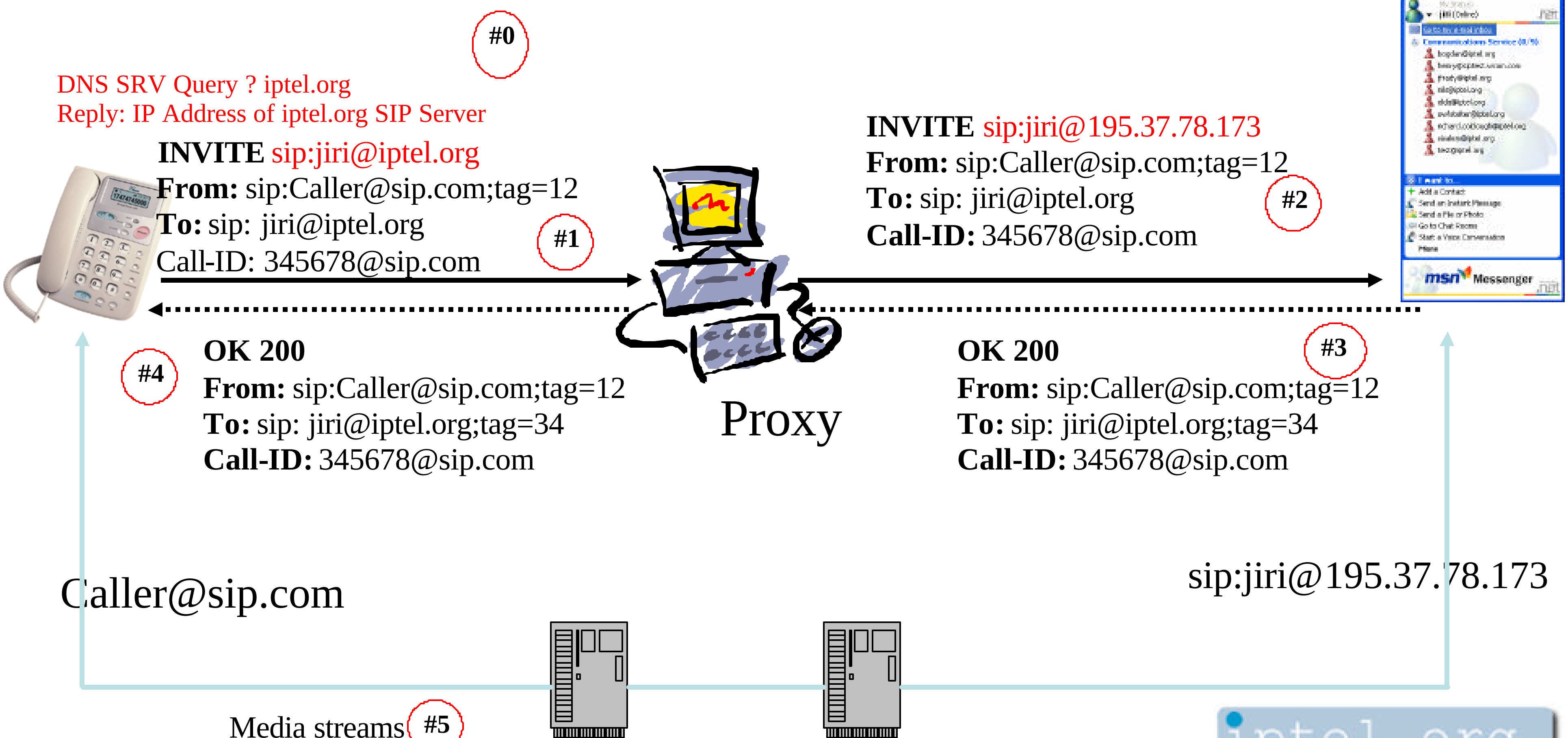
- Becoming an IP-Telephony operator takes complexity comparable to setting up E-mail server:
- Configuration Checklist:
  - Configure DNS
  - Download and configure a SIP proxy server
  - Configure supporting services: web provisioning, database back-end typically.
  - Configure PSTN gateway for use with your proxy server,

Does SIP Do All of It Today?

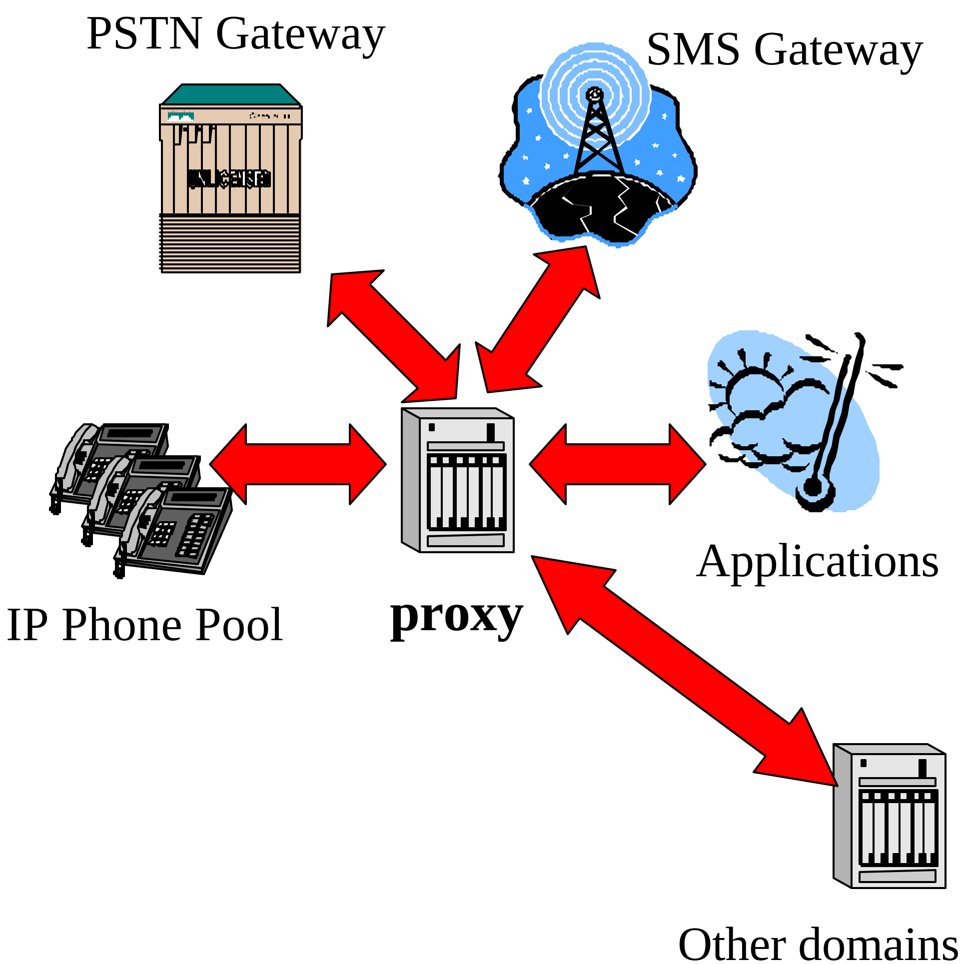
- ✓ Session Initiation Protocol (SIP) is an IETF signaling protocol (RFC 3261) that helps to:
  - ✓ Keep track of users.
  - ✓ Set up and maintain voice, video and other sessions between them
- ✓ Industry acceptance: SIP devices shipped by both established vendors (Cisco, Microsoft, Lucent, Lucent, …) as well as start-ups (Pingtel, Grandstream, Intertex, …)
  - ✓ See www.iptel.org/info/products/
- ✓ Interoperability: Good!
  - ✓ We returned from SIPIT last week: even advanced features such as IPv6 and TLS worked together!
- ✓ Future: Use of SIP for mobile networks standardized in 3GPP

Basic SIP Call-Flow

- SIP is HTTP-like, textual, client-server protocol, using email-like addresses
- So-called “Proxy” server takes care of setting up sessions between users
- Signaling independent on media – both take different path



Basic Server Element: SIP Proxy



- Proxy servers maintain central role in SIP networks:
- They glue SIP components such as phones, gateways, applications and other domains
- They provide place for service implementation (missed calls, forwarding, screening, etc.) and service access control
- SER: www.iptel.org/ser/

What Is SIP Good In?

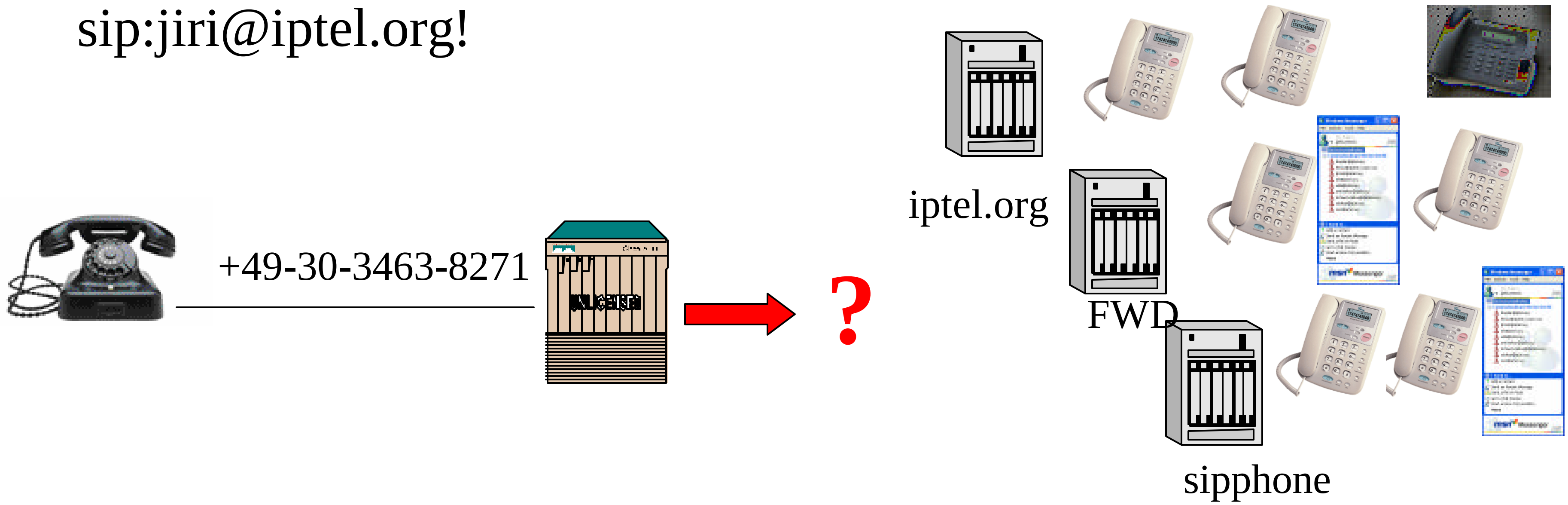
- Easy service integration: its design roots in SNMP and HTTP protocols; it integrates easily with applications built on top of them.
- Reusability, e.g., instant messaging and presence can be ran with the same protocol and infrastructure.
- High scalability: protocol maintains only transaction state in network. With SER, we achieve thousands of calls per second on a PC.
- Affordability: Free SIP servers and softphones exist.

SIP and Friends

- SIP alone is just a single piece in the protocol puzzle; this is the rest of the IETF puzzle. How do I …
- … transport voice in real-time? RTP!
- … find domain of called party? Like with email, use DNS to resolve address of server responsible for jiri@iptel.org/!
- … authenticate users generate Call Detail Records? De-facto RADIUS standard.
- … get over NATs? STUN.
- Other supporting protocols frequently used in SIP devices: NTP for clock setting, TFTP for image upgrade, HTTP for provisioning.

ENUM….

- That's all just fine but how do the 200 million PSTN callers find SIP callees? They really can't type in a SIP address like sip:jiri@iptel.org!



- Idea: provide a number-2-SIP-address mapping using DNS: "ENUM". E.g.: +49-30-3463-8271=> 8271@iptel.org.

✓ Things That Work

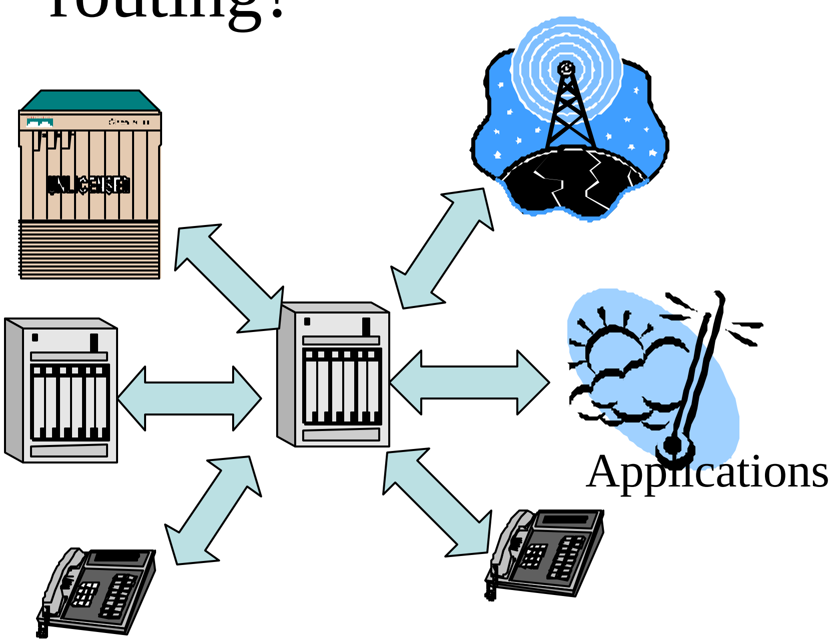
- ✓ Basic VoIP services work, so do complementary integrated services such as instant messaging, voicemail, etc.
- ✓ Numbering plans easy to maintain and they complement domain names well.
- ✓ QoS mostly pleasant. (Most broadband calls feature ~150 ms RTT and packet loss close to zero.)
- ✓ Solid SIP implementations interoperate fairly well.
- ✓ Billing machinery works too: Accounting easy, though not standardized. Gateways with accounting support exist today
- ✓ Interoperation with other technologies works too, PSTN gateway market established (single-vendor dominance too).

✓ Concern: Performance

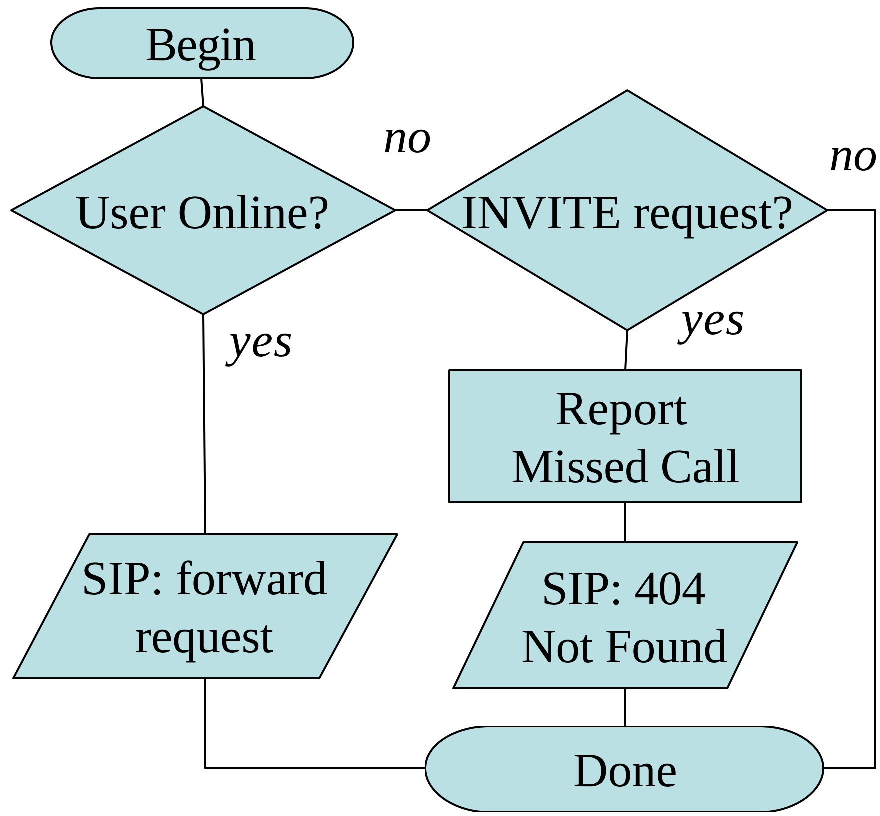
- Performance – are you really able to process all the crap messages you receive over the public Internet?
- iptel.org's operational observation: 80% of traffic is invalid messages caused by misconfigured or broken devices.
- Use of applications such as presence increase per-user load compared to VoIP roughly by factor of 100.
- Nevertheless we have the capacity today: our measurements indicate proxy transactional throughput of hundreds to thousands of calls per second. Sufficient to power large subscriber populations.

✓ Concern: SIP Routing

- Flexible signaling among a variety of components by proxy is good for service creation, but how do you define proper routing?



- Iptel's answer: <u>routing language</u> that allows precise definition of server behaviour.



✓ Application Programming

- Site administrator service request examples:
  - "Implement a welcome announcement for new subscribers".
  - "Show My on-line status on my web-page!"
- Problem: Do you really want to put your hands on 100k LOC server code with timers, locks, shared memory usages, etc.?
- Fortunately easy to handle: SIP's textual nature allows easy combination with UN*X and web applications known to be effective for programming.
- Example: FWD's online status; few lines of HTML/PHP

```
<a href="http://fwd.pulver.com/callme.php">
<img
   src=http://fwd.pulver.com/myicon.php?userid=nnnn
   n border=1 alt="FWD
Status">
</a>
```

😐 NAT Traversal

- NATs popular because they conserve IP address space and help residential users to save money charged for IP addresses.
- Problem: VoIP does not work over NATs without extra work.
- Straight-forward solution: replace NATs with IPv6 – unclear when deployed if ever.
- There are many scenarios for which no single solution exists. Solutions include: STUN, ALGs, symmetric communication, media relay, UPnP, …

😐 Other Troublemakers

- Phone makers: Some phone features still either in infancy or in chaos:
  - Few phone vendors support NAT traversal (STUN, symmetric signaling).
  - Very few SIP phone vendor support fail-over using DNS/SRV.
  - No standardized means of phone provisioning.
- Politicians: recently, state of Minnesota put unrealistic requirements on Vonage in response to telcos' attempt to rule out VoIP competition.

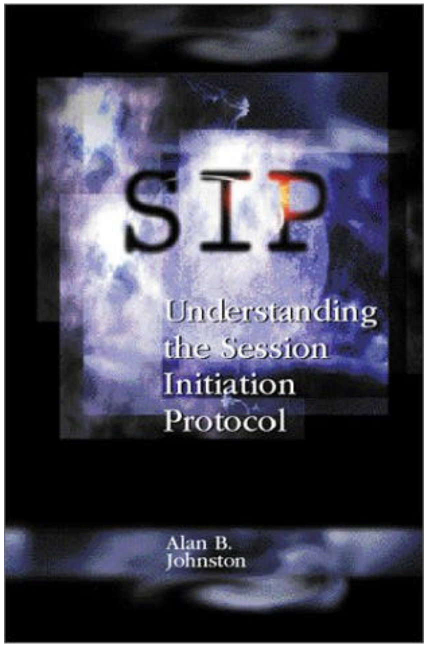
Concluding Observations

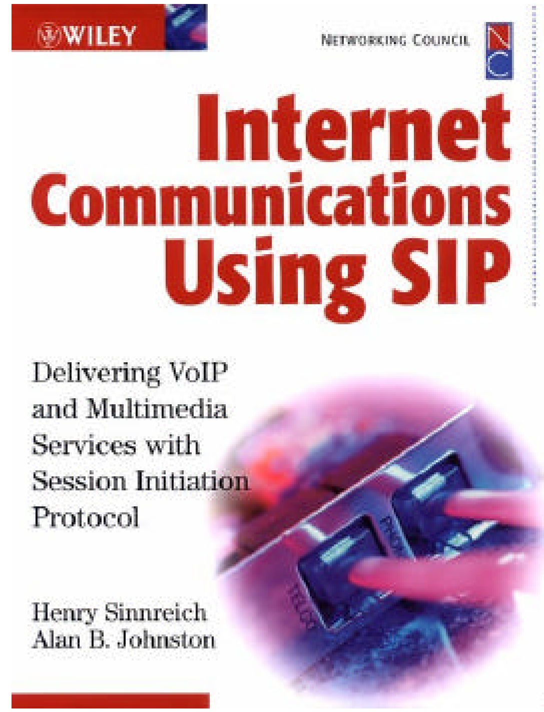
- Basic VoIP & complementary services up and running. Many problems of past years are gone: QoS, performance, SIP routing, etc.
- Infrastructure can be set up in an inexpensive way: Just download the software from the Internet and call “make install”.
- Still on the agenda: DoS, NAT traversal, solid fail-over (mostly phone vendor’s guilt).

Information Resources

- Email: jiri@iptel.org
- IP Telephony Information: http://www.iptel.org/info/
- SIP Services: http://www.iptel.org/user/
- SIP Express Router: http://www.iptel.org/ser/
- SIP Implementations: http://www.iptel.org/info/products/
- Full-size SIP Tutorial: http://www.iptel.org/sip/

There Are SIP Books!





- Alan B. Johnston: “SIP: Understanding the Session Initiation Protocol”
- Artech House 2001

- Henry Sinnreich, Alan Johnston: Internet Communications Using SIP: Delivering VoIP and Multimedia Services with Session Initiation Protocol
- John Wiley & Sons, 2001