Introduction to SIP Telephony
Protocols and Services

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iptel.org

Builders of the SIP Express Router.
www.iptel.org/ser/

iptel.org Background

- Since its very origins, iptel.org has been a know-how organization, part of Fraunhofer.
- 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.
- Mostly used applications: VoIP, instant messaging and presence, voicemail2email.
- The infrastructure serves public subscribers as well as internal users with additional privileges.
- Increase in population size since introduction of Windows Messenger.
- Services powered by iptel.org's open-source SIP server, SER.

Outline

- Introduction
  - About VoIP
  - Current Industry Status
- Used Protocols
  - Protocol Zoo
  - SIP
- SIP Services
  - IN-like and Internet-integrated services
  - Service Programming
- Operational Issues
  - Routing policy, NAT traversal

Introduction

History

- Carrying voice on IP-based packet networks first identified by Cohen in 1977*
- Commercialization and standardization began in 1995; Vocaltec the first company to ship IP2PSTN gateways (proprietary)
- SIP standardization began in IETF in 1995
- Adoption of SIP for use in 3GPP in late nineties
- Motivation:
  - Cost saving through telco by-passing
  - Service Integration

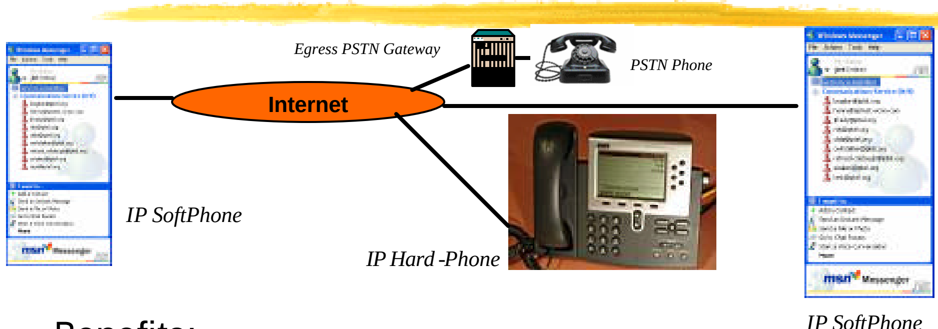
** D. Cohen, "Issues in transnet packetized voice communications", In Proceedings of the 5th Data Communications Symposium*

IETF – Where SIP Was Born

- The IETF is a large open international community of network designers, operators, vendors, and researchers concerned with the evolution of the Internet architecture and the smooth operation of the Internet.
- Working Groups related to Internet telephony:

- SIP: core Session Initiation Protocol
- SIPPING: Future SIP extensions and related issues
- IPTEL: Internet Telephony
- AVT: Audio Video Transport
- MIDCOM: Firewall/NAT Traversal
- MMUSIC: Multiparty Multimedia Session Control
- QoS Related: DiffServ, IntServ, RSVP
- PSTN legacy: SigTran, Megaco
- SIMPLE: SIP for Instant Messaging and Presence Leveraging
- interaction of PSTN and IP services: PINT,SPIRITS

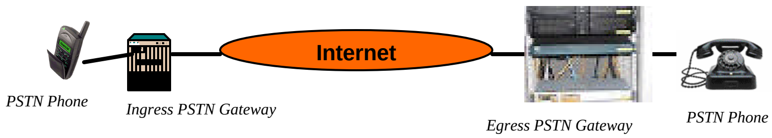
PC-to-phone/PC Scenario



Benefits:

- Cost-savings
- Service integration: Many today's installation serve as PBX replacement today.

Phone-2-phone Scenario



Benefit:

- Cost savings through telco bypass. Most of the voice path in the Internet which has no notion of distance and duration in its charging model.

Shift in Motivation Since Nineties

- Telco prices dropping, cost saving not as appealing as in the past:
  - Berlin-Prague before market liberalization in 1995: 1DM a minute
  - Berlin-Prague today*: 4.50 cents (factor of 11)
- Expectations set high on service integration
  - Voicemail2email, click-to-dial, conferencing, instant messaging and presence, etc.
- Shift in business mode: “be telco on your own” doable without large investments and operational costs. Seen as an opportunity for small businesses.

** Source: http://www.billigertelefonieren.de/*

Completed Work

- Since beginning, lot of focus has been paid to quality of service to assure “nice voice”:
  - Network provisioning
  - QoS Signaling (RSVP, diffserv)
  - Packet-loss concealment
  - Less concern today – backbones overprovisioned, end-devices improved and burned in hardware
- Signaling for IP Telephony Well Understood Today:
  - Session Initiation Protocol (RFC3261)
  - Gateways to PSTN

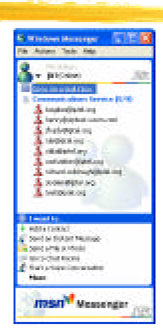
SIP Implementations Widely Available


SNOM


Pingtel


Allied Telesyn


Cisco


Microsoft


Intertex


Cisco


Siemens


Mitel

Free SIP server with capacity that can server VoIP signaling for Bay Area.

… and many more …http://www.iptel.org/info/products/

Status Update: Good News

- Basic VoIP services work, so do complementary integrated services such as instant messaging, voicemail, etc.
  - *Commercial deployments exist, mostly offering PSTN termination: Vonage, deltathree, denwa, Packet 8*
  - *Trial services: FWD, PCH, WCOM, SIP Center*
  - *Tens of intranet deployment of SER reported, probably many more unknown*
- Billing machinery works too: Accounting easy, though not standardized.
- Numbering plans easy to maintain and they complement domain names well.

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... Good News

- QoS mostly pleasant for broadband community:
  - *Links between iptel.org site and iptel.org user community have packet loss close to zero and RTT mostly bellow 150 ms, rarely above 200 ms.*
- SIP interoperability well established across mature implementations
- Interoperation with other technologies works too:
  - *Multiple products on the PSTN gateway market*
  - *Gateway to Jabber instant messaging up and running*
  - *Commercial H.323 gateways exist*

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Bad News

- ⌘Nightmare – NATs
- ⌘Why I keep my PSTN black phone in my room's corner: Reliability
- ⌘What Is It? Machines Do, Operators Don't ... Scalability and Manageability
- ⌘End-devices still expensive
- ⌘Future issues: spam, denial of service attacks

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Protocols

Refresher: IP Design Concepts

- **Distributed end-2-end design**
- Intelligence and states resides in end-devices
- Network maintains almost zero intelligence (except routing) and state (except routing tables).
- End-devices speak to each other using whatever applications they have. There is almost no logic in the network affecting this behavior.
- Result:
  - Flexibility. Introducing new applications is easy.
  - Failure recovery. No state, no problem on failure.
  - Scalability. No state, no memory scalability issues.

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What Problems Do Need to Be Solved for VoIP?

- Session management
  - Users may move from terminal to terminal with different capabilities and change their willingness to communicate
  - To set-up a communication session between two or more users, a **signaling protocol** is needed
  - **Session Initiation Protocol** (SIP) supports locating users, session negotiation (audio/video/instant messaging, etc.) and changing session state
- Media Transport
  - Getting packetized voice over lossy and congested network in real-time
  - **RTP** – protocol for transmitting real-time data such as audio, video and games
- End-to-end delivery: underlying IP connects the whole world
- Supporting protocols: DNS, IP , routing protocol, Authentication/Authorization/Accounting (AAA), gateway location, QoS, etc.
- IETF Practice: Decomposition Principle; Separate protocols are used for separate purposes. All of them on top of IP.

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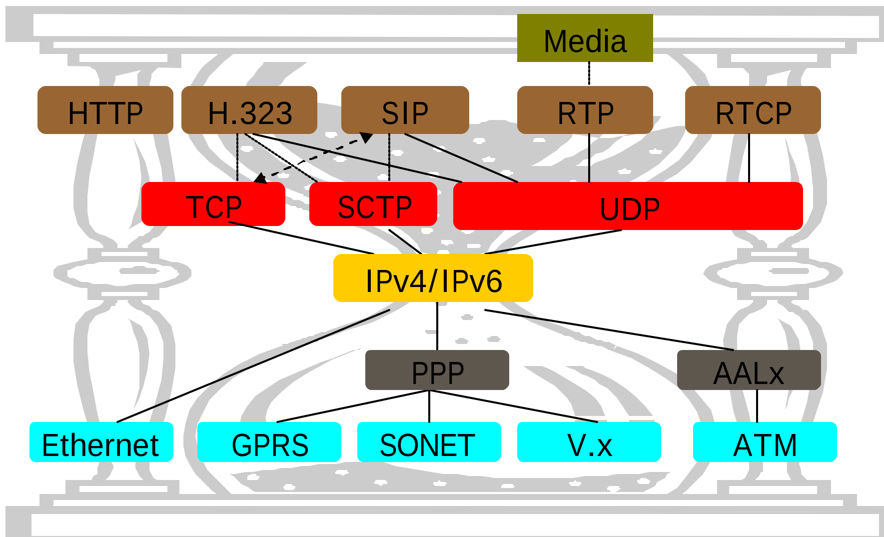
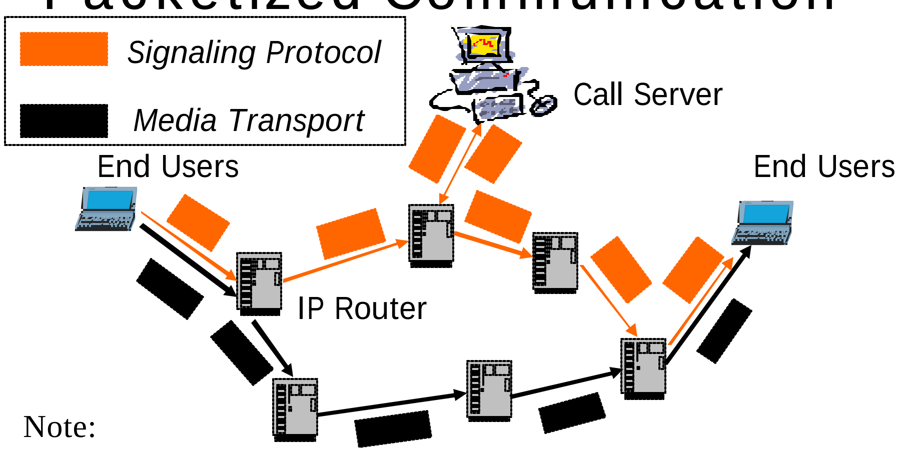
Protocol Zoo (Hourglass Model)



Packetized Communication



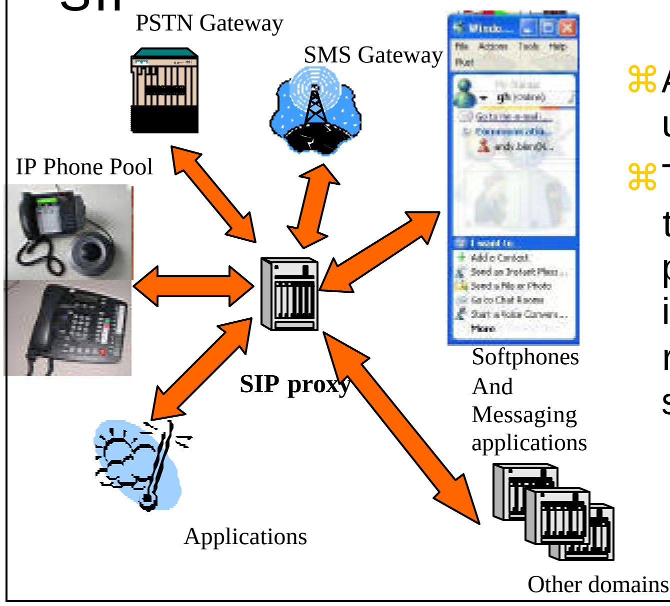
Note:

•Every packet may take a completely different path

•Signaling takes typically different path than media does

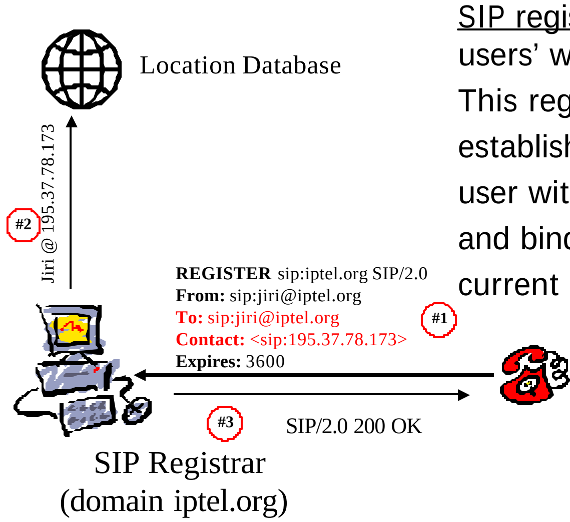
•Both signaling and media as well as other applications (FTP, web, email, …) look "alike" up to transport layer and share the same fate

Components Integration with SIP



- All components use SIP
- They are glued together using a proxy server that implements a routing policy for signaling

SIP Registrar

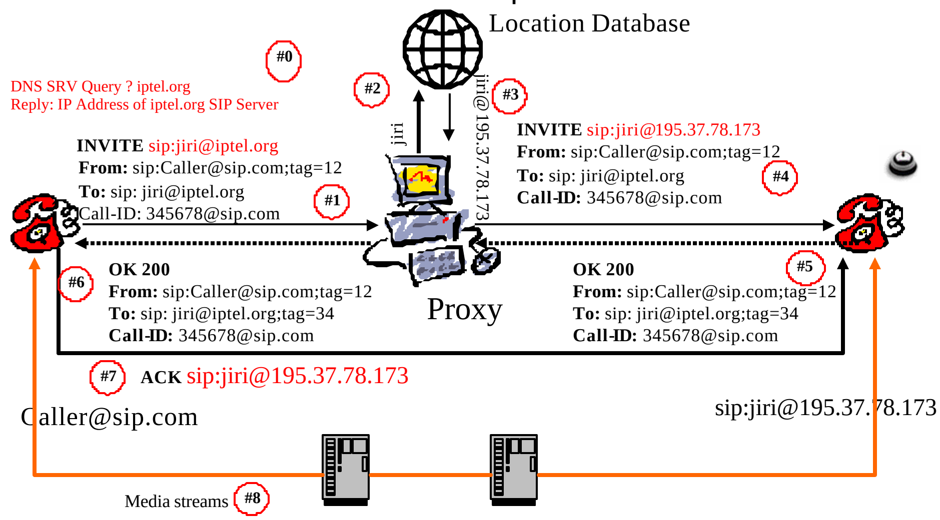


SIP registrar keeps track of users' whereabouts.

This registration example establishes presence of user with address jiri@iptel.org and binds this address to user's current location 195.37.78.173.

Basic SIP Call-Flow (Proxy Mode)

SIP Proxy looks up next hops for requests to served users in location database and forwards the requests there.



SIP (RFC3261) - General Purpose Presence Protocol

- SIP is not limited to Internet telephony
  - SIP establishes user presence
  - SIP messages can convey arbitrary signaling payload: session description, instant messages, JPEGs, any MIME types
- Suitable for applications having a notion of session
  - distributed virtual reality systems,
  - network games (Quake II/III implementations),
  - video conferencing, etc.
- Applications may leverage SIP infrastructure (Call Processing, User Location, Authentication)
  - Instant Messaging and Presence
  - SIP for Appliances

SIP Workhorses

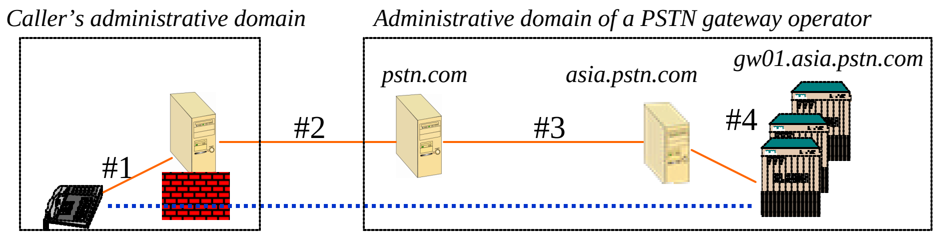
- SIP Proxy Server
  - relays call signaling, i.e. acts as both client and server
  - operates in a transactional manner, i.e., it keeps no session state
  - transparent to end-devices
  - does not generate messages on its own (except ACK and CANCEL)
  - Allows for additional services (call forwarding, AAA, forking, etc.)
- SIP Redirect Server
  - redirects callers to other servers
- SIP Registrar
  - accept registration requests from users
  - maintains user's whereabouts at a Location Server

  (like GSM HLR)

SIP End-devices

- User Agent (user application)
  - UA Client (originates calls)
  - UA Server (listens for incoming calls)
- Types of UAs:
  - Softphone and hardphones
  - Messaging clients
  - PSTN gateways
  - Media servers
  - Etc.

Service composition: Added-value Server Chains



Caller's outbound proxy accomplishes firewall traversal.

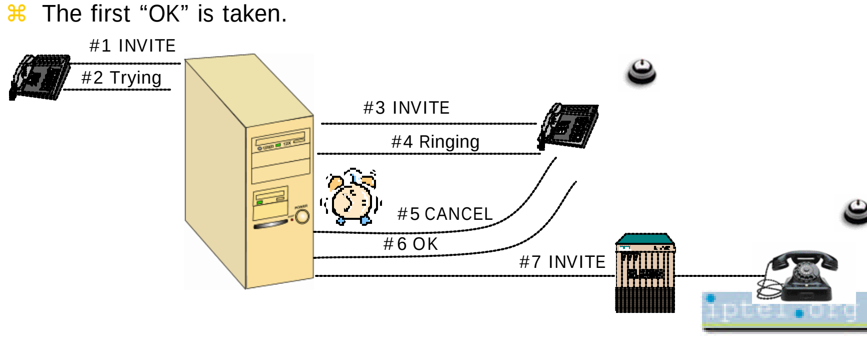
Destination's "first-hit proxy" identifies a proxy serving dialed area.

Proxy in the target area distributes load in a gateway farm.

Note: signaling (in red) may take a completely different path from media (in blue).

Ability to Try Multiple Destinations: Forking

- A proxy may fork a request to multiple destinations either in parallel ("reach me everywhere") or serially ("forward no reply").
- A proxy can cancel pending parallel searches after a successful response is received.
- A proxy can iterate through redirection responses ("recursive forking").
- The first "OK" is taken.



Stateful versus Stateless Proxy Operational Mode

- SIP Proxies may operate either in stateful or stateless mode; which of the modes is used depends on implementation or configuration.
- stateless mode:
  - **Usage:** good for heavy-load scenarios -- works well for example if they act as application-layer load distributors.
  - **Behavior:**
    - proxies just receive messages, perform routing logic, send messages out and forget anything they knew;
    - they should cache results of SIP routing logic as it is not able to distinguish between retransmissions and new requests -- and would result in new execution of SIP routing logic for every retransmission

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Stateful versus Stateless Proxy Operational Mode (cont.)

- stateful mode:
  - **Usage:** good for implementing some services (e.g., “forward on no reply”)
  - **Behavior:**
    - proxies maintain state during entire transaction; they remember outgoing requests as well as incoming requests that generated them until transaction is over; they do not keep state during the whole call
    - a forking proxy should be stateful
    - reduce retransmission time by acting on behalf of sender closer to destination

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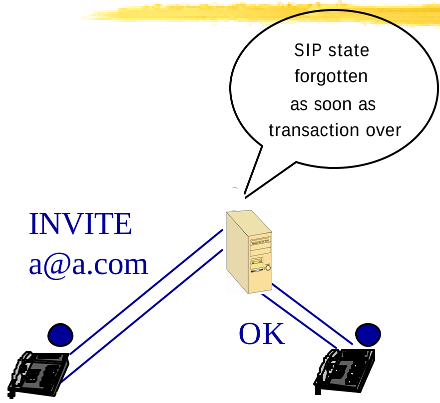
"Stateful" Proxy Refers to Transactions

Frequently Misunderstood Issue

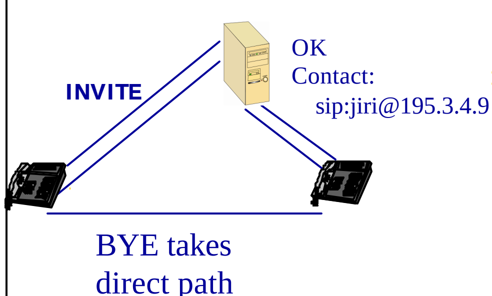


Legend
SIP signaling
SIP state
media

- SIP proxies deliver a "one-time rendezvous service" (as opposed to state storage service).
- Thus a stateful proxy just keeps state during a SIP "rendezvous transaction" and completely forgets it afterwards.
- A SIP proxy is not aware of existing calls. In case of failure, existing calls are NOT affected!
- Subsequent transactions may take a direct path!

Subsequent Transactions Bypass Proxy

Frequently Misunderstood Issue



- Unless route recording is used, subsequent transactions (e.g., BYE) may take a completely different path to destination indicated in **Contact**: header field.
- Generally, there may be session state as well. Unless there is a well defined use of it, it indicates unscalable implementation.

SIP Message Structure

Request

INVITE sip:UserB@there.com SIP/2.0

Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>;tag=123
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Subject: Happy Christmas
Contact: BigGuy <sip:UserA@here.com>
Content-Type: application/sdp
Content-Length: 147

Message Header Fields

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000

Payload

Response

SIP/2.0 200 OK

Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>;tag=123
To: LittleGuy <sip:UserB@there.com>;tag=65a35
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Subject: Happy Christmas
Contact: LittleGuy <sip:UserB@there.com>
Content-Type: application/sdp
Content-Length: 134

v=0
o=UserB 2890844527 2890844527 IN IP4 there.com
s=Session SDP
c=IN IP4 110.111.112.113
t=0 0
m=audio 3456 RTP/AVP 0
a=rtpmap:0 PCMU/8000

SDP (RFC2327): “receive RTP G.711-encoded audio at 100.101.102.103:49172”

SIP Addresses

- SIP gives you a globally reachable address.
  - Callees bind their temporary address to the global one using SIP REGISTER method.
  - Callers use this address to establish real-time communication with callees.
- URLs used as address data format; examples:
  - sip:jiri@iptel.org
  - sip:voicemail@iptel.org?subject=callme
  - sip:sales@hotel.xy; geo.position:=48.54_-123.84_120
- must include host, may include user name, port number, parameters (e.g., transport), etc.
- may be embedded in Webpages, email signatures, printed on your business card, etc.
- address space unlimited
- non-SIP URLs can be used as well (mailto:, http:, ...)

SIP RFC3261 Methods

- **INVITE** initiates sessions
  - session description included in message body
  - re-INVITEs used to change session state
- **ACK** confirms session establishment
  - can only be used with INVITE
- **CANCEL** cancels a pending INVITE
- **BYE** terminates sessions
- **REGISTER** binds a permanent address to current location; may convey user data (CPL scripts)
- **OPTIONS** capability inquiry

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SIP Extension Methods

- SUBSCRIBE/ NOTIFY/ MESSAGE — instant messaging and presence (RFC3265, RFC3428, draft-ietf-simple-*)
- REFER — call transfer (RFC3515)
- PRACK — provisional reliable responses acknowledgement (RFC3262)
- INFO — mid-call signaling (RFC 2976)

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SIP Response Codes

- Borrowed from HTTP: xyz explanatory text
- Receivers need to understand response class ("x")
- x80 and higher codes avoid conflicts with future HTTP response codes
- 1yz Informational
  - 100 Trying
  - 180 Ringing (ringing tone played locally)
  - 181 Call is Being Forwarded
- 2yz Success
  - 200 ok
- 3yz Redirection
  - 300 Multiple Choices
  - 301 Moved Permanently
  - 302 Moved Temporarily

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SIP Response Codes (cont.)

- 4yz Client error
  - 400 Bad Request
  - 401 Unauthorized
  - 482 Loop Detected
  - 486 Busy Here
- 5yz Server failure
  - 500 Server Internal Error
- 6yz Global Failure
  - 600 Busy Everywhere

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Summary of SIP Properties

- Textual (HTTP-like) client-server protocol
  - Easy to debug, extend and process with textual operating systems
- End-2-end
  - It puts most of intelligence into end-devices (“user agents”) – good for scalability and extensibility
  - The network infrastructure designed to be leight-weighted. Network functionality (registrar, proxy) are typically logical parts of a single server.
- Internet addressing using URIs
  - E.g., sip:jiri@iptel.org
  - Non-SIP URIs possible to (e.g., they may be used to redirect a caller to webpage)
  - Address space unlimited and may be used to create services (sip:sales@hotel.xy; geo.position:=48.54_-123.84_120)
- It delivers mobility: User can register from one or more locations with IP connectivity

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SIP Service Space

What's the Killer App?

- Q: Added-value services expected to be major source of revenues. So what is the killer app?
- A: If I saw raw gold on the street I would not tell you either.

- It is believed that the convenience of integrated services will be the killer.
- IN-like services reproducable, though with different mimics sometimes.
- Couple of examples follow...
- *(No, I really do not know which of them will be the best-seller.)*

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IN-like Services with SIP

- Most of IN services may be easily implemented with SIP in proxies/redirect servers or UAs:
  - (Un)conditional call forwarding
  - abbreviated dialing
  - Screening
  - distinctive ringing
  - call distribution
  - call transfer
  - etc.
- Sometimes, implementation logic may completely differ.
  - Televoting and IVRs likely to be replaced by Web in the long run.
  - Call-waiting is end-device implementation issue with no protocol support.
  - Music-on-hold may be played localy.

The real benefit is those services beyond IN: straight-forward integration with web, email, instant messaging, etc.

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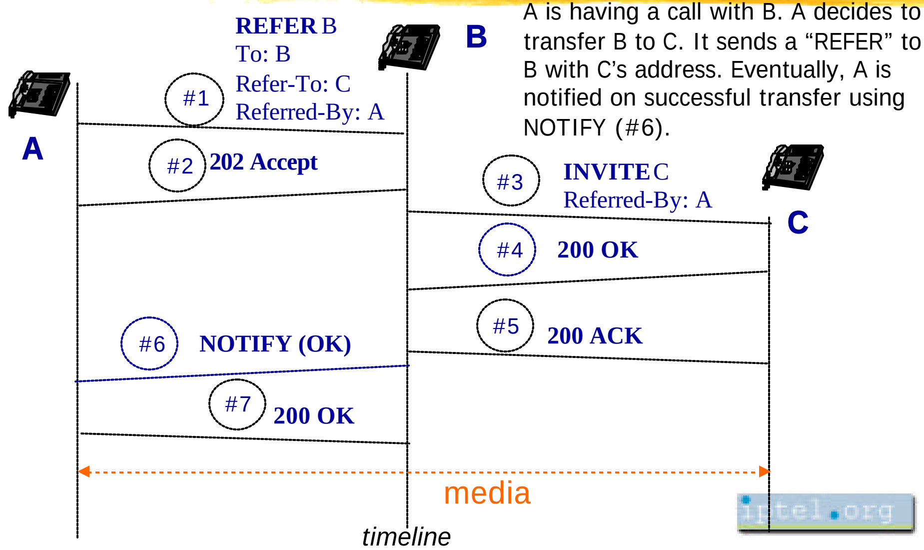
Example: Call Transfer Call Flow

A is having a call with B. A decides to transfer B to C. It sends a “REFER” to B with C’s address. Eventually, A is notified on successful transfer using NOTIFY (#6).



draft-ietf-sip-cc-transfer, ***RFC3515***

Call Transfer/REFER

- Accomplished using the REFER method.
- The REFER method indicates that the recipient (identified by the Request-URI) should contact a third party using the contact information provided in the method.
- New header fields: Refer-To, Refer-By.
- NOTIFY method used to report on result of referral.
- Note: No changes to proxy behavior required.
- Variants:
  - With Consultation Hold (SIP Hold and unattended transfer)
  - Attended Transfer, I.e., with a short conference
- Other REFER uses: Click-to-dial

Answering Machine

- Old-times behavior: set-up number of rings, plug-in, if you do not answer the machine will
- Easy to mimic with SIP: AM acts as a SIP UA; you need to set-up an answer timer, let the answering machine register using your credentials; when an invitation arrives it is forked both to your phone and your answering machine
- Added value examples:
  - **Unified messaging:** SIP answering machine can turn voice messages into email messages that follow you or comprehensive web-pages (cf. voice navigation)
  - **Programmability** allows to play variety of customized prompt messages:
    - **If** (caller $\in$ friends) **then** play ("You can reach me at Venice beach or leave a message") **else** play ("leave a message please");

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Instant Messaging and Presence

- Idea: Use the same signaling infrastructure for more services
- SIP already supports:
  - Notion of presence and user location mechanisms
  - Application-layer routing (incl. forking) and message processing (e.g., CPL)
  - Optimized for speed
  - Scalability by distributed design

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RFC3428

Instant Messaging

- Goal: deliver short messages rapidly
- SIP Extension: “MESSAGE” Method
  - Message body of any MIME type (including Common Profile for Instant Messaging, draft-ietf-impp-cpim)
  - im type URLs used

```
MESSAGE sip:user2@domain.com SIP/2.0
Via: SIP/2.0/UDP user1pc.domain.com
From: im:user1@domain.com
To: im:user2@domain.com
Contact: sip:user1@user1pc.domain.com
Call-ID: asd88asd77a@1.2.3.4
CSeq: 1 MESSAGE
Content-Type: text/plain
Content-Length: 18

Watson, come here.
```

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RFC3265

Subscribe-Notify

- Goal: ability to be notified when a condition occurs
- Applications:
  - User presence and related applications
  - Call-back (notify when the other party becomes available)
  - VoiceMail Notification (notify when a voicemail message is stored) *[draft-ietf-sipping-mwi]*
  - Traffic Alerts (notify on traffic jam)
- Extensions: “SUBSRIBE” and “NOTIFY” methods, “Event” and “Allow-Events” headers, “489 Bad Event” Response Code
- Subscription subject to expiration similarly to how REGISTER is

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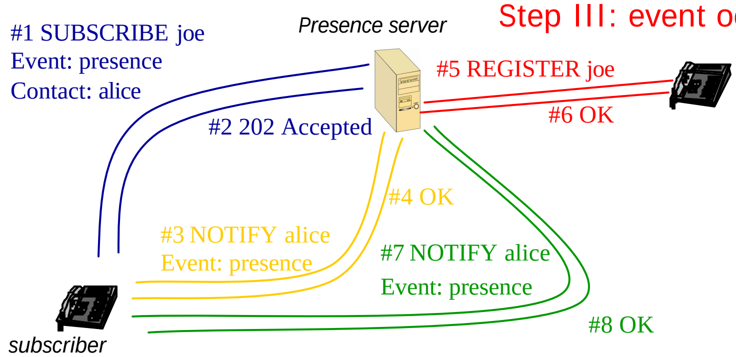
Subscribe-Notify For Presence Services

draft-ietf-simple-presence

Step I: subscription to a condition



Step IV: subscriber is notified whenever condition changes

Step II: subscriber is immediately notified on current condition

Service Programming

Programming SIP Logic

- Services examples
  - “discard all calls from Monica during my business hours”
  - “redirect authenticated friends to my cell phone, anyone else to my secretary”
- Programming SIP services
  - is not easy (our SIP Proxy server has 100k lines of code!) – lot of timers, dynamic allocation, parsing and other inconveniences
  - Some companies and standardization bodies have been seeking to standardize APIs (JTAPI, CTI, JAIN, PARLAY) – however, they APIs still feature lot of programming difficulties and are tightly coupled to specific programming environments such as Java
  - IETF approach: follow the textual interface tradition used in HTTP
    - CGI
    - Call Processing Language (CPL)

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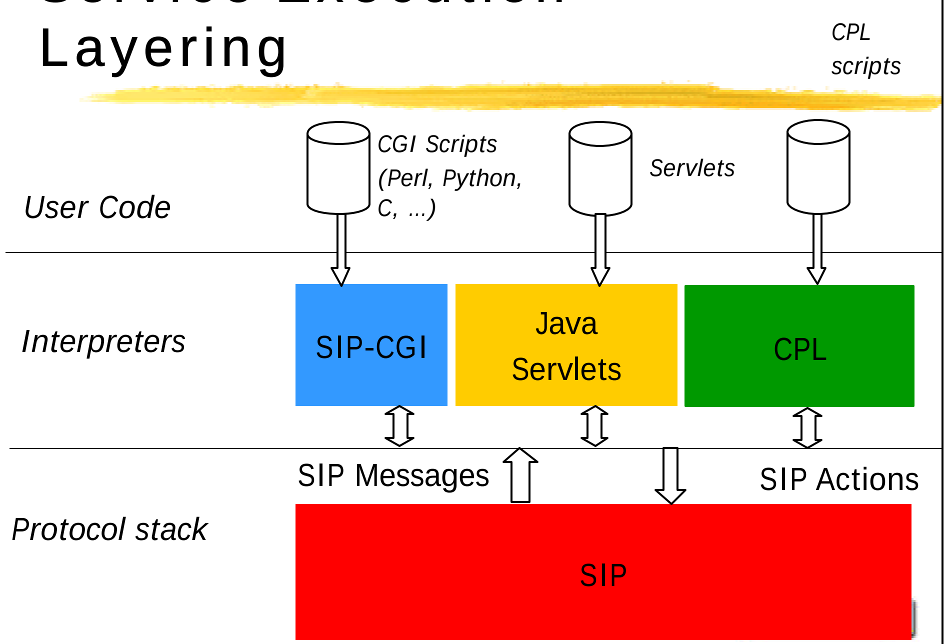
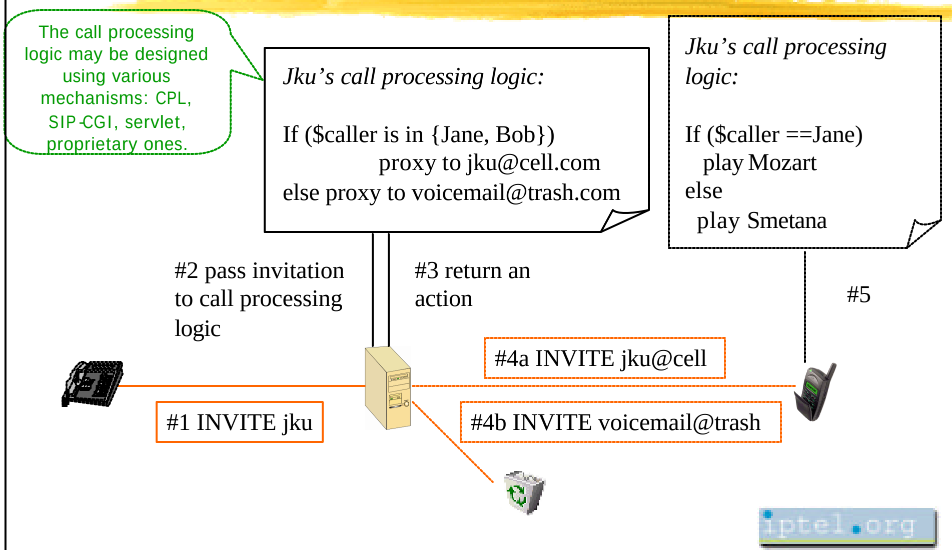
Service Execution Layering



Call Processing Logic Example



Where May Signaling Services Live?

- Some services have to live in the network:
  - call distribution
  - services for dial-up users without always-on IP connectivity
  - network servers may be located on users' premises (PBX-like) or operator's premises (Web-hosting-like, NetCentrex-like)
- Some services can be implemented in both places:
  - forward on busy
- Some services work best in end-devices:
  - distinctive ringing

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Service Location Examples

Feature	End-device	Proxy
Distinctive Ringing	Yes	Can assist
Visual call id	Yes	Can assist
Call Waiting	Yes	No
CF Busy	Yes	Yes
CF No Answer	Yes	Yes
CF No Device	No	Yes
Location hiding	No	Yes
Transfer	Yes	No
Conference Bridge	Yes	No
Gateway to PSTN	Yes	No
Firewall Control	No	No
Voicemail	Yes	No

Source: H. Schulzrinne: "Industrial Strength IP Telephony"

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Techniques for SIP Programming

- Stanardized SIP Technologies for programming follow web mechanisms:
  - SIP CGI-BIN
  - CPL
  - Java Servlets
- Non-standardized technologies may gain momentum too (think for example of PHP's penetration)
- They key is **programming efficiency**

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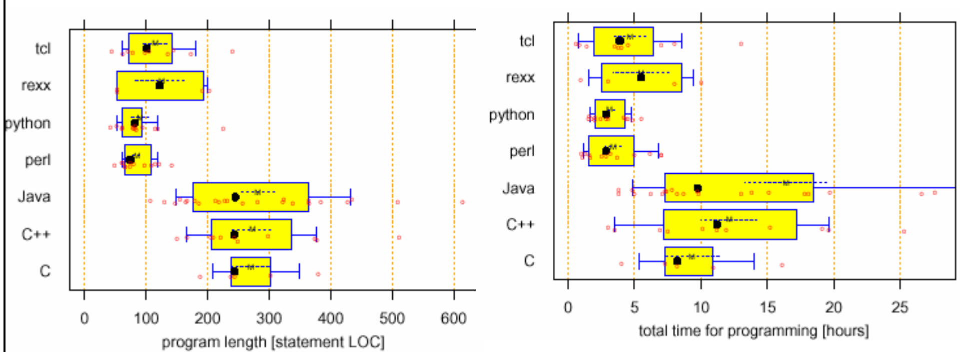
Scripting Languages Key To Efficiency

- Web lesson: variety of languages; PHP, Perl, Python, shell scripts....
- No dependency on a particular programming language – developers can use what they best understand, including scripting languages
- Use of scripting languages makes code shorter and takes less time (graphs from [*] demonstrate complexity for a specific problem)



(*) Source of both graphs: Lutz Prechelt: "An Empirical Comparison of C, C++, Java, Perl, Python,RXX, and Tcl", March 2000.

SIP Common Gateway Interface (CGI)

RFC 3050

- Follows Web-CGI. Unlike Web-CGI, SIP-CGI supports proxying and processes responses as well.
- Language-indpendent (Perl, C, ...)
- Communicates through input/output and environment variables.
- CGI programs unlimited in their power. Drawback: Buggy scripts may affect server behavior easily.
- Persistency token (cookie) is passed between SIP server and CGI to keep state across requests and related responses.

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SIP-CGI I/O

- Script input: environment variables (AUTH_TYPE, CONTENT_LENGTH, REQUEST_URI, etc.) and SIP message on stdin
- Script output: set of messages consisting of action lines, CGI header fields and SIP header fields on stdout
- Action lines:
  - Generating a response: status line
  - Proxying:
    - `CGI-PROXY-REQUEST <dest-url> <sip-version>`
    - Additional header fields may be followed – they will be merged with the original request.
  - Forward response: `CGI-FORWARD-RESPONSE <token> <sip-version>`
  - Set cookie for subsequent messages: `CGI-SET-COOKIE <token> <sip-version>`
  - Determine if the script should be called for the next message belonging to the same transaction: `CGI-AGAIN ("yes" | "no") <sip-version>`

draft-ietf-iptel-cpl

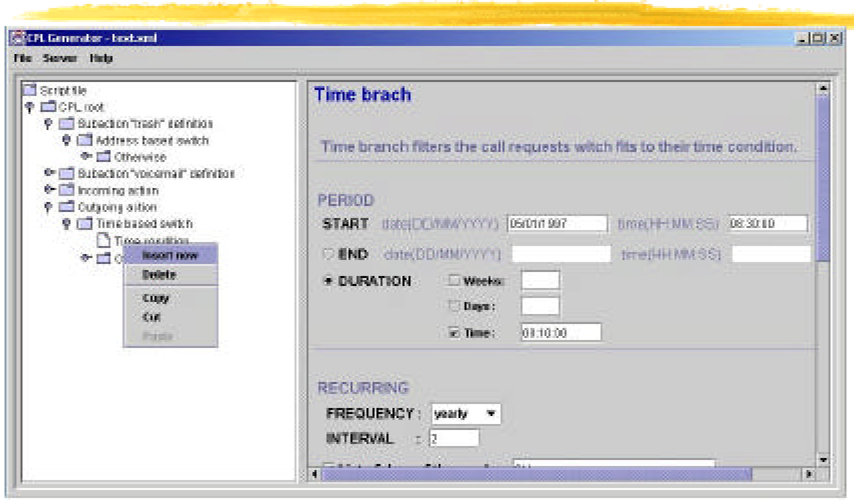
Call Processing Language

- Special-purpose call processing language.
- CPL scripts define a decision tree which may result in signaling (proxy, redirect, reject) or non-signaling (mail, log) action.
- CPL scripts triggered by SIP messages.
- May be used by both SIP and H.323 servers.
- Target scenario: users determine call processing logic executed at a server.
- Limited languages scope makes sure server's security will not get compromised.
- Portability allows users to move CPL scripts across servers.
- Scripts may be manually written, generated using convenient GUI tools, supplied by 3rd parties, ...

CPL Example

```
<incoming>
      <address-switch field="origin" subfield="host">
                <address subdomain-of="example.com">
                <location url="sip:jones@example.com">
                <proxy timeout="10">
                        <busy> <sub ref="voicemail" /> </busy>
                        <noanswer> <sub ref="voicemail" /> </noanswer>
                        <failure> <sub ref="voicemail" /> </failure>
                </proxy>
                </location>
        </address>
        <otherwise>
                <sub ref="voicemail" />
        </otherwise>
      </address-switch>
 </incoming>
```

Example: Creating CPL Scripts



iptel.org: CPL Composer

SIP Security

SIP Security Tools

- Most commonly use security protocol: digest
  - Based on private shared secret
  - Allows to establish user identity
  - Does not provide message integrity or privacy
- TLS – addresses shortcomings of digest but not widely deployed yet
  - It is based on a transitive trust model: upstream client trusts downstream proxy servers, which again trust their servers downstream from them
  - Servers "see" SIP in plain-text
- End-2-end security delivered with S/MIME
  - With e2e security, proxy servers in the middle do not see plain-text message bodies
- Alternate security protocols for 3GPP (AKA, RFC3310)

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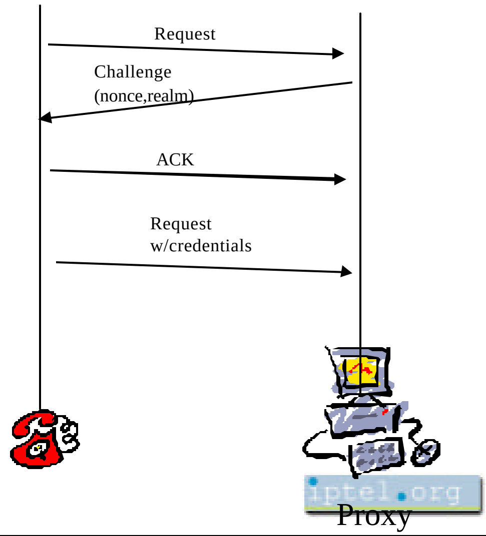
RFC 2617

SIP Digest Authentication

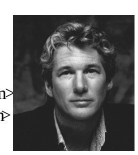
- Required for user identification and admission control for services.
- Protocol:
  - challenge-response using MD5
  - Based on secret shared between client and server
  - No message integrity provided



Disclaimer: Security Protocols Don't Implement Social Engineering

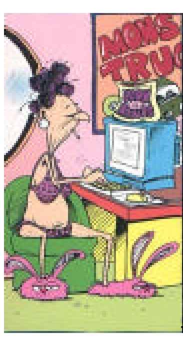
SIP INVITE w/JPEG

INVITE sip:UserB@there.comSIP/2.0
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
...



200 OK w/JPEG

SIP/2.0 200 OK
Via: SIP/2.0/UDP here.com:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345601@here.com...



Interworking with PSTN

PSTN Gateways

- Basic building block of PSTN interworking scenarios: gateways convert signaling and media
- The gateway can be split in media and signaling components and connected through MGCP or Megaco
- They need to be found on the Internet: problem similar to that of IP routing. Methods include:
  - Static configuration
  - TRIP routing protocol [RFC3219]
  - ENUM -- used to map digits into SIP URIs [RFC2916]

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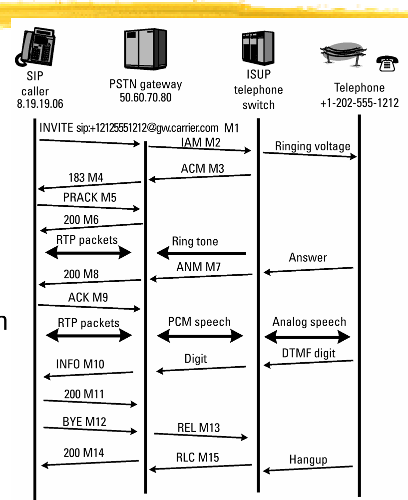
Call Flow SIP to PSTN

- Request-URI in the **INVITE** contains a Telephone Number which is sent to PSTN Gateway.
- The Gateway maps the **INVITE** to a SS7 ISUP IAM (Initial Address Message)
- **183 Session Progress** establishes early media session so caller hears Ring Tone.
- Two way Speech path is established after ANM (Answer Message) and **200 OK**



Slide courtesy of Alan Johnston, WorldCom. (See reference to Alan's SIP book.)

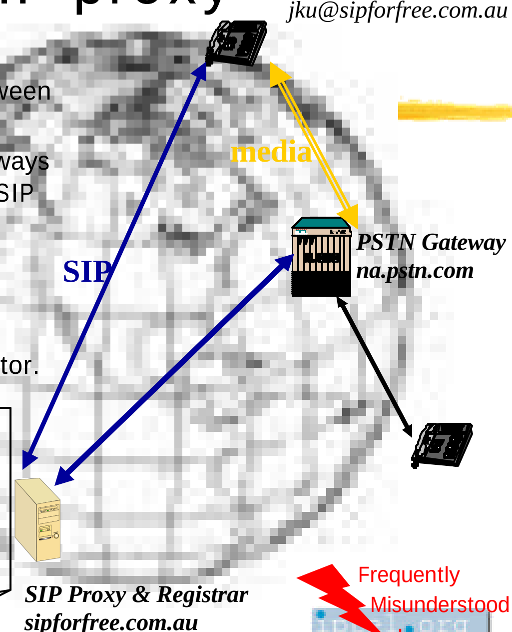
PSTN GW != SIP proxy

jku@sipforfree.com.au

- PSTN gateways are adapters between two different technologies.
- From SIP perspective, PSTN gateways are SIP termination devices, i.e., SIP User Agents just like IP phones.
- **PSTN gateway functionality separate from call processing logic residing at a proxy.**
- Gateway operator != proxy operator.

call processing logic:

If *($destination* ***in*** *PSTN)* ***then***
route_to_least_cost_gateway();
elseif *local("sipforfree.com.au")* ***then***
lookup_registry;
else *proxy_to_foreign_domain();*



More PSTN-Related Reads

- Mapping of of Integrated Services Digital Network (ISUP) Overlap Signalling to the Session Initiation Protocol [draft-ietf-sipping-overlap]
- Session Initiation Protocol PSTN Call Flows [draft-ietf-sipping-pstn-call-flows]
- Integrated Services Digital Network (ISDN) User Part (ISUP) to Session Initiation Protocol (SIP) Mapping [RFC 3398]
- Session Initiation Protocol for Telephones (SIP-T): (SIP-T): Context and Architectures [RFC3372]
- Interworking between SIP and QSIG [draft-elwell-sipping-qsig2sip]

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Advanced Topics

NAT Traversal, QoS Protocols, 3GPP, SIP Routing, Performance

DNS for Failure Recovery & Load Balancing *RFC 3263*

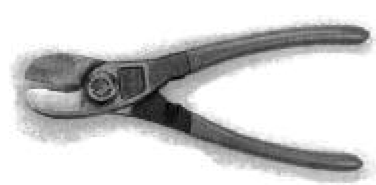
- Unavailable SIP servers can be dealt with using DNS in the same way as mail servers are:
  - DNS servers maintain multiple prioritized SRV entries
  - callers initiate calls to high-priority server; if unavailable, they proceed to lower-priority server
- Load balancing can be accomplished similarly
  - DNS servers maintain multiple SRV entries with equal priority
  - a random pick is chosen out of the server list
- Notes on DNS
  - Too few implementations have implemented DNS SRV properly (2003)
  - it's good do have multiple DNS servers for each zone of authority;
  - DNS may be a pain …

Firewall Traversal

Ultimately Secure Firewall

Installation Instructions: For best effect install the firewall between the CPU unit and the wall outlet. For Internet use install the frewall between the demarc of the T1 to the Internet. Place the jaws of the firewall across the T1 line lead, and bear down firmly. When your Internet service provider's network operations center calls to inform you that they have lost connectivity to your site, the firewall is correctly installed.

(© *Marcus Ranum*)



Problems with Firewalls and NATs

- Firewalls
  - Interest to keep policy restrictive conflicts with dynamic nature of VoIP
  - Solutions space: ALGs, external ALGs (MidCom)
- NATs
  - Address translations conserves IP space but causes inconsistency between address in IP/transport headers and application payload
  - Solutions space: ALGs, external ALGs (MidCom), STUN
- Problem size: HUGE

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Where FWs/NATs affect SIP

```
INVITE sip:UserB@there.com SIP/2.0
Via: SIP/2.0/UDP 192.168.99.1:5060
From: BigGuy <sip:UserA@here.com>
To: LittleGuy <sip:UserB@there.com>
Call-ID: 12345600@here.com
CSeq: 1 INVITE
Subject: Happy Christmas
Contact: BigGuy <sip:UserA@192.168.99.1>
Content-Type: application/sdp
Content-Length: 147

v=0
o=UserA 2890844526 2890844526 IN IP4 here.com
s=Session SDP
c=IN IP4 100.101.102.103
t=0 0
m=audio 49172 RTP/AVP 0
a=rtpmap:0 PCMU/8000
```

- Contact, Route, Record-Route header fields
- Via header fields (received tag)
- SDP payload

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NAT Traversal

- NATs popular because they conserve IP address space and help residential users to save money charged for IP addresses.
- Problem: SIP does not work over NATs without extra effort. Peer-to-peer applications' signaling gets broken by NATs: Receiver addresses announced in signaling are invalid out of NATted networks.
- Straight-forward solution: IPv6 – unclear when deployed if ever.
- There are many scenarios for which no single solution exists (they primarily differ in design properties of NATs – symmetric, app-aware, etc.)

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Current NAT Traversal Practices ...

- Application Layer Gateways (ALGs) – built-in application awareness in NATs.
  - Requires ownership of specialized software/hardware and takes app-expertise from router vendors (Intertex, PIX).
- Geeks' choice: Manual configuration of NAT translations
  - Requires ability of NATs, phones, and humans to configure static NAT translation. (Some have it.) If a phone has no SIP/NAT configuration support, an address-translator can be used.
- UPnP: Automated NAT control
  - Requires ownership of UPnP-enabled NATs and phones. NATs available today, phones rarely (Snom).

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… Current NAT Traversal Practices

- STUN (RFC 3489): Alignment of phones to NATs
  - Requires NAT-probing ability (STUN support) in end-devices and a simple STUN server. Implementations exist (snom, kphone).
  - Does not work over NATs implemented as "symmetric".
  - Troubles if other party in other routing realm than STUN server.
  - + Works even if NAT device not under user's control.
- Relay: Each party maintains client-server communication
  - Introduces a single point of failure; media relay subject to serious scalability and reliability issues
  - + Works over most NATs

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NAT Practices: Overview

	ALG	STUN	UPnP	Manual	Relay
Works over ISP's NATs?	N/A	Ltd. (*)	N/A	N/A	Maybe
Symmetric NATs?	N/A	No	N/A	ok	Ltd.
Phone support needed?	No	Yes	Yes	Yes	Yes
NAT support needed?	Yes	Ltd. (*)	Yes	Ltd. (+)	No
Scalability	? (o)	Ok	Ok	Ok	poor ⊠
User Effort	Small	Small	Small	Big ⊠	Small

*… does not work for symmetric NATs
+ … port translation must be configurable
o … application-awareness affects scalability

NAT Traversal Scenarios

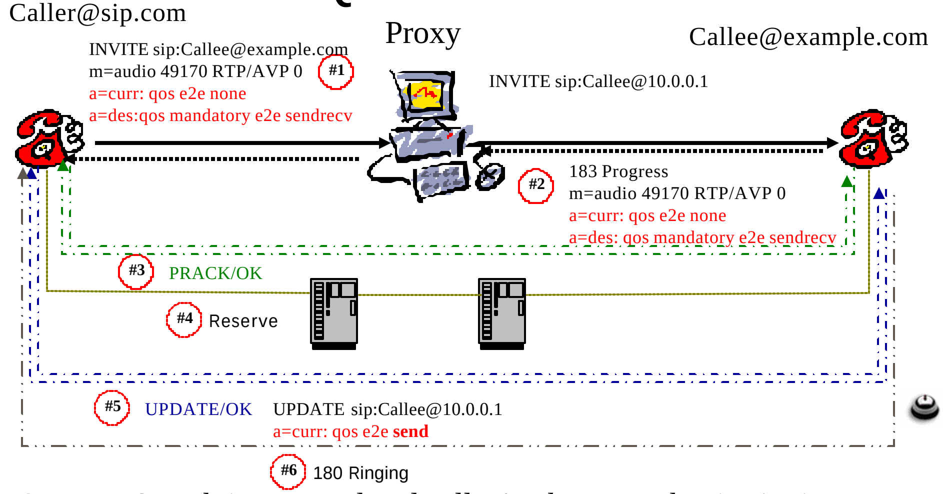
- There is no "one size fits it all" solution. All current practices suffer from many limitations.
- iptel.org observations for residential users behind NATs: Affordability wins: SIP-aware users relying on public SIP server use ALGs or STUN. First UPnP uses sighted.
- Our plan: hope for wider deployment of
  - STUN and STUN-friendly firewalls
  - ALGs
  - UPnP-enabled phones and NATs

RFC3312

QoS: SIP and QoS Control

- In many cases, you don't need complex QoS protocols: use Ethernet switches (as opposed to hubs), sufficient bandwidth, and DiffServ if needed.
- SIP DOES NOT provide QoS support: QoS protocols are kept separate from signaling.
- Deadlock:
  - QoS signaling cannot begin until I learn through signaling who is the other party.
  - SIP signaling cannot complete and alert callee until QoS is established
- Proposal: "QoS Preconditions": if QoS signaling is enabled, find the called party, ask it not to ring, carry out QoS reservation, and start ringing when QoS is ready (UPDATE)

SIP and QoS Control

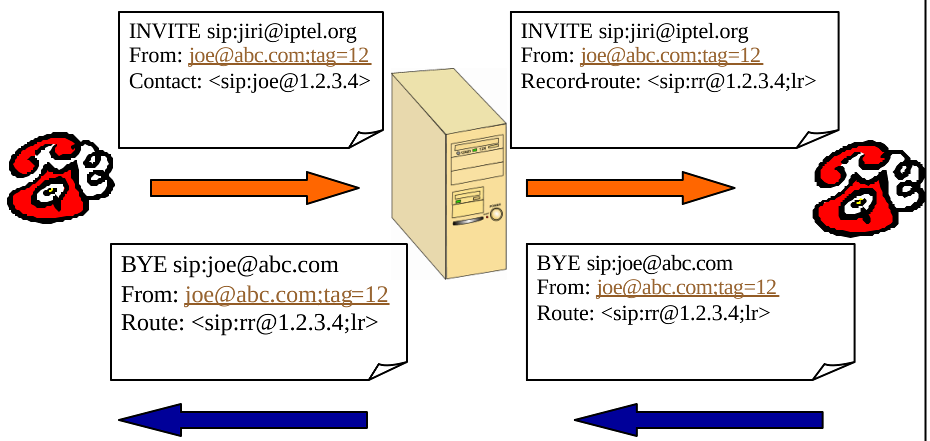


At step #6, path is reserved and callee's phone can begin ringing. Then, SIP completes as usual (180 confirmed by PRACK, 200 sent when callee answers, media exchange begins.)

Record-Routing

- Refresher: by default, only the initial request (INVITE) visits a proxy, subsequent requests (BYE) travel directly to offload servers
- Problems:
  - some applications need to see all signaling, accounting for example
  - UAs may live in different protocol realms (TCP vs UDP, IPv4 versus v6) and can communicate only through the proxy server
- Solution: record-routing: proxy servers append a hint to processed requests which advices phones to keep the servers in path for subsequent communication

Record-Routing Example



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Record-Routing Apps

- Record-Routing can be also use to piggy-back session-state in SIP messages to leave server state-less
- Example:
  - A RR-parameter can include timestamp for initial invite
  - When CDRs are generated on receipt of BYE, the call duration is calculated as “current_time()-rr_timestamp_parameter()”
  - Note: In security-sensitive application like above, it is necessary to introduce message integrity

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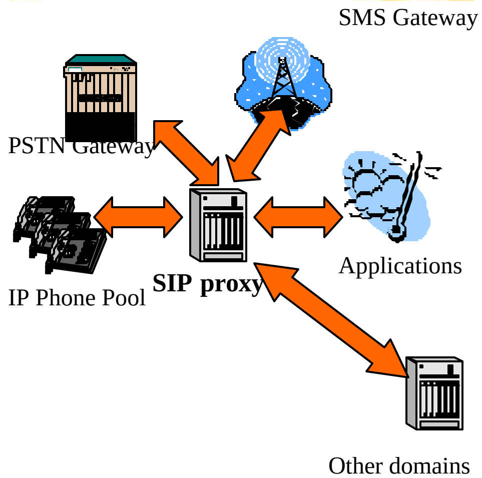
SIP Routing

- One of primary benefits of SIP: Ability to link various service components speaking SIP together.
- The "glue" are signaling servers. Their primary capability is routing requests to appropriate services.
- Issues:
  - Routing flexibility – how to determine right destination for a request
  - Troubleshooting when routing failures occur



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Routing Policy

- SIP request-routing decision can depend on a variety of factors. Iptel.org example:
  - address-based routing – requests to numeric destination are forwarded to PSTN gateway, whereas others to IP phones
  - Policy-based processing – calls to international PSTN requests require authentication and privileges
  - Method-based routing – requests to numerical destinations are split by method between SMS and PSTN gateway
  - Further factors include request's transport origin, address claimed in From header field, content of Contact, etc.
- **Operational observation: mighty tools for specification of routing policy are needed.**

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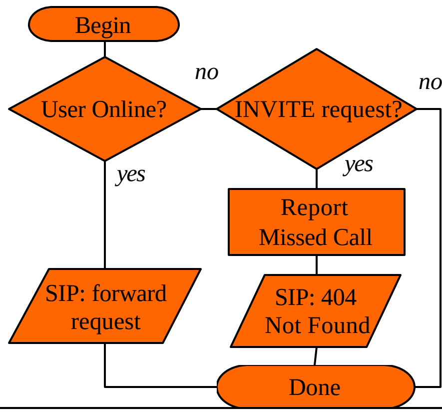
Routing Language

- ⌘ Request routing flexibility needed to link SIP components (voicemail, PSTN gateway, logging facility, etc.) together
- ⌘ Answer: request routing language (features conditions, URI-rewriting, request modification, replying, etc.)
- ⌘ Example: reporting missed calls



SER Routing Language

```
/* user online ? */
if (lookup("location")) {
     t_relay();
     break;
};
if (method=="INVITE") {
     /* report to syslog */
     log("ACC: missed call\n");
};
sl_send_reply("404","Not Found");
```

Performance Concerns

- ⌘ New applications, like presence, are very talkative
  - Presence status update frequent
  - Each update ventilated to multiple parties
- ⌘ Broken or misconfigured devices account for a fair part of load; few of many real-world observations:
  - Broken digest clients resend wrong credentials in an infinite loop → heavy flood
  - Mis-configured password: a phone attempted to re-register every ten minutes (factor 6) →2400 messages a day
  - Mis-configured Expires=30 (factor 120)
  - Keeping NAT bindings up – SIP request each 20 seconds
- ⌘ Replication, Boot avalanches

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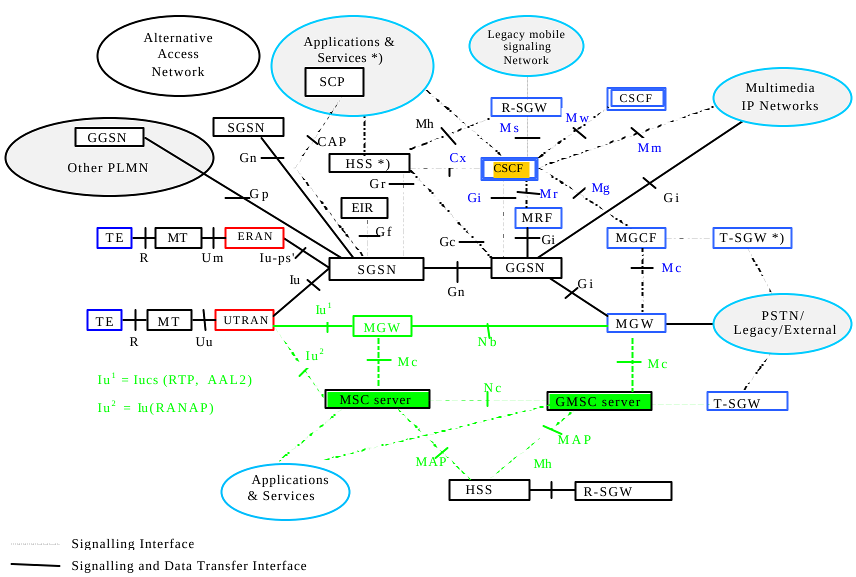
Achievable Scalability

- Good news: well-designed SIP servers can cope with load in terms of thousands of calls per second (CPS)
  - *Example: lab-tuned version of SIP Express Router able to process 5000 Calls Per Second to a static destination statefully on a dual-CPU PC – capacity needed by telephony signaling of Bay Area*
- Pending concern: denial of service attacks
  - *Example: hundreds of megabytes of RAM can be exhausted in tens of seconds with statefull processing*

3GPP: Architecture



Political Issues

- Telcos feel threatened and try hard to ban IP Telephony operation
- Governments of authoritative countries prevent free information exchange
- As a result, IP telephony is impeded in many countries, including:
  - Czech Republic
  - Pakistan
  - China
  - Panama
- US ILECs attacking VoIP industry (“numbering issues”)

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Information Resources

Information Resources

- Author: jiri@iptel.org
- Related IETF work: http://www.iptel.org/ietf/
- SIP Express Router: http://www.iptel.org/ser/
- SIP Tutorial: http://www.iptel.org/sip/
- SIP Site: http://www.cs.columbia.edu/sip/

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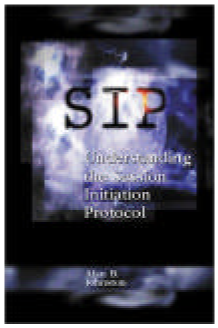
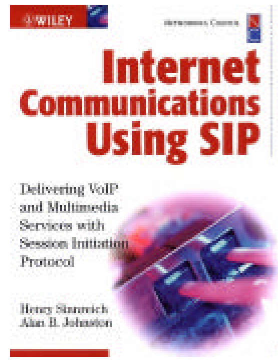
Glossary

- ALG Application-Level-Gateway
- CDR Call Detail Record
- CGI Common Gateway Interface
- CPL Call Processing Language
- DTMF Dual Tone Multi-Frequency
- ETSI European Telecommunications Standards Institute
- IETF Internet Engineering Task Force
- ITSP Internet Telephony Service Providers
- ITU International Telecommunication Union
- IVR Interactive Voice Reponse
- JAIN Java APIs for Integrated Network Framework
- LEC Local Exchange Carrier
- LNP Local Number Portability
- NAT Network Address Translation
- MGCP Media Gateway Control Protocol
- OSP Open Settlement Protocol
- PSTN Public Switched Telephone Network
- QoS Quality of Service
- RTCP RTP Control Protocol
- RTP Real-Time Transport Protocol
- RTSP Real-Time Streaming Protocol
- SDP Session Description Protocol
- SIP Session Initiation Protocol
- SS7 Signaling System Nr. 7
- TRIP Telephony Routing over IP
- VoIP Voice over IP

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

-The End –