# Creating Services for Internet Telephony using the Session Initiation Protocol

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# Overview

- What is SIP (and not)?
- What is it good for?
- Internet telephony architectures
- SIP for VoIP services
- SIP for instant messaging and presence
- hurdles for **Internet** telephony

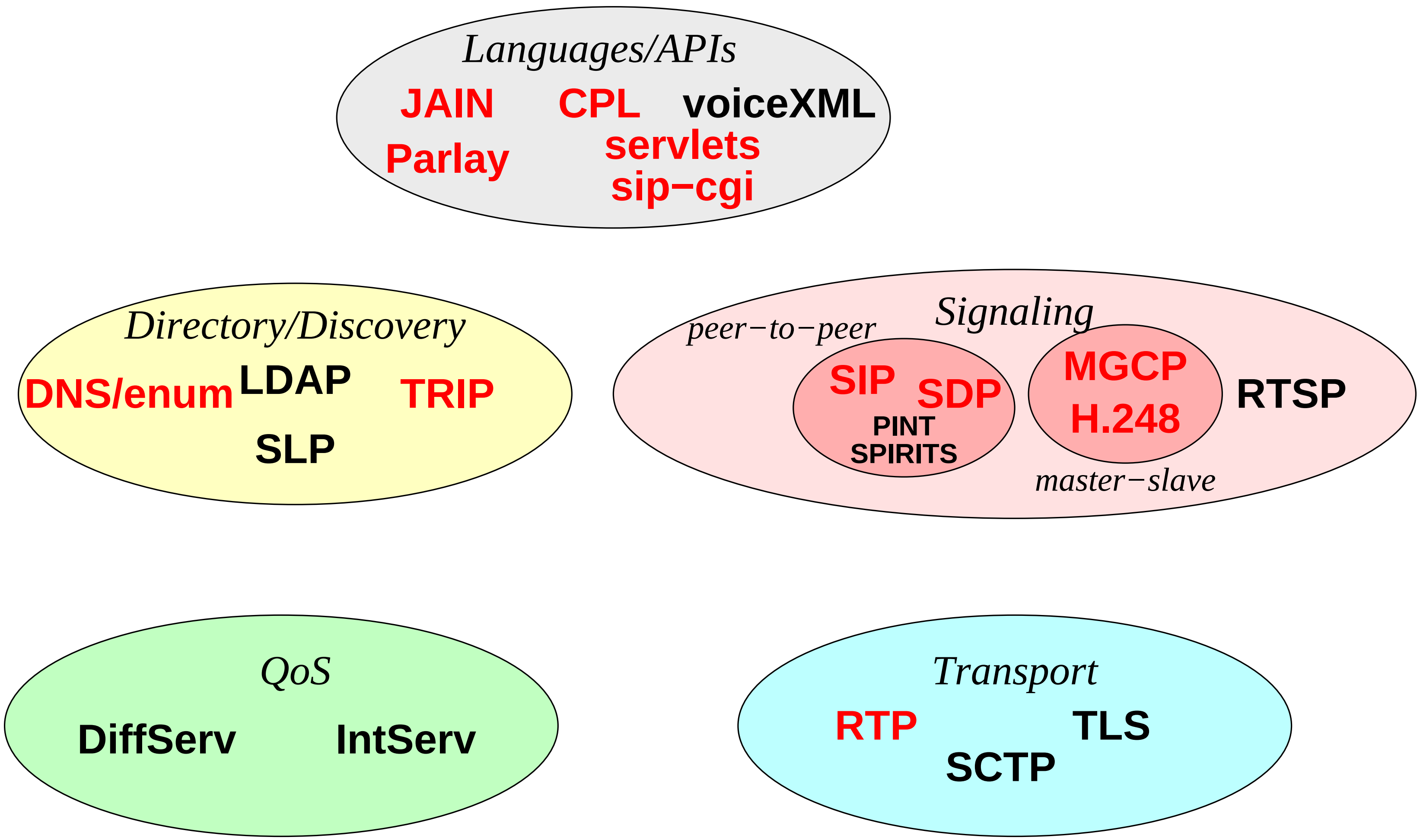
# What is SIP?

- sets up and tears down *sessions*
  - content-neutral: audio, video, shared applications, . . .
  - network-neutral: ATM, FR or IP, but mostly IP
- notifies users of *events*: “I’m online”, “person entered room”, “dishes are done”, . . .
- sends messages – instant (text) messages (“SMS”, “IM”),

# How does SIP work?

- protocol similar to HTTP
- uses either UDP or TCP
- uses URLs that identify *logical* destination, not IP address of end system

# SIP in the VoIP protocol ecosystem



# What is it not?

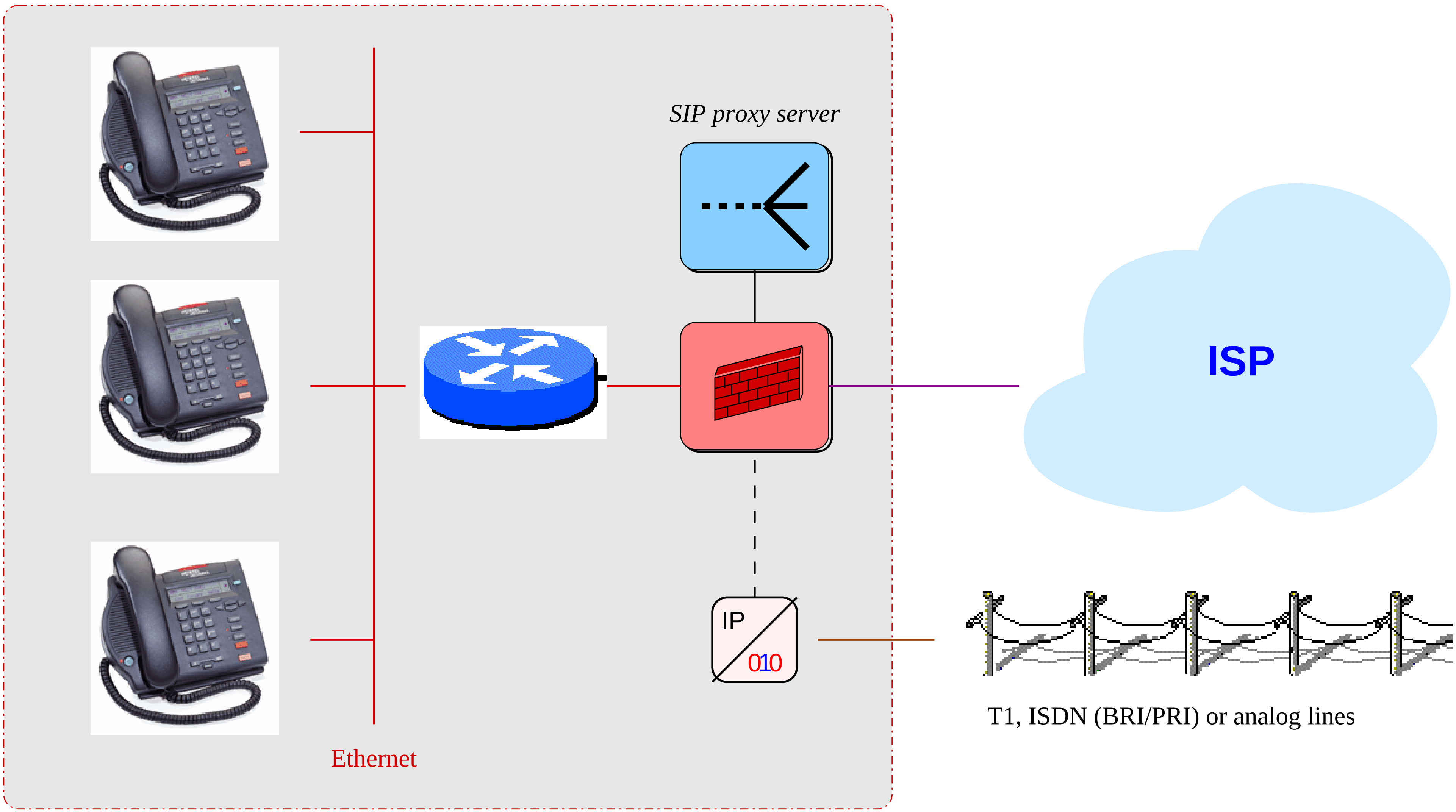
- replacement for the web (HTTP) or email
- media or data transport protocol ➠ RTP
- conference control protocol ➠ ?
- database access protocol ➠ LDAP, DNS

# Internet telephony service models

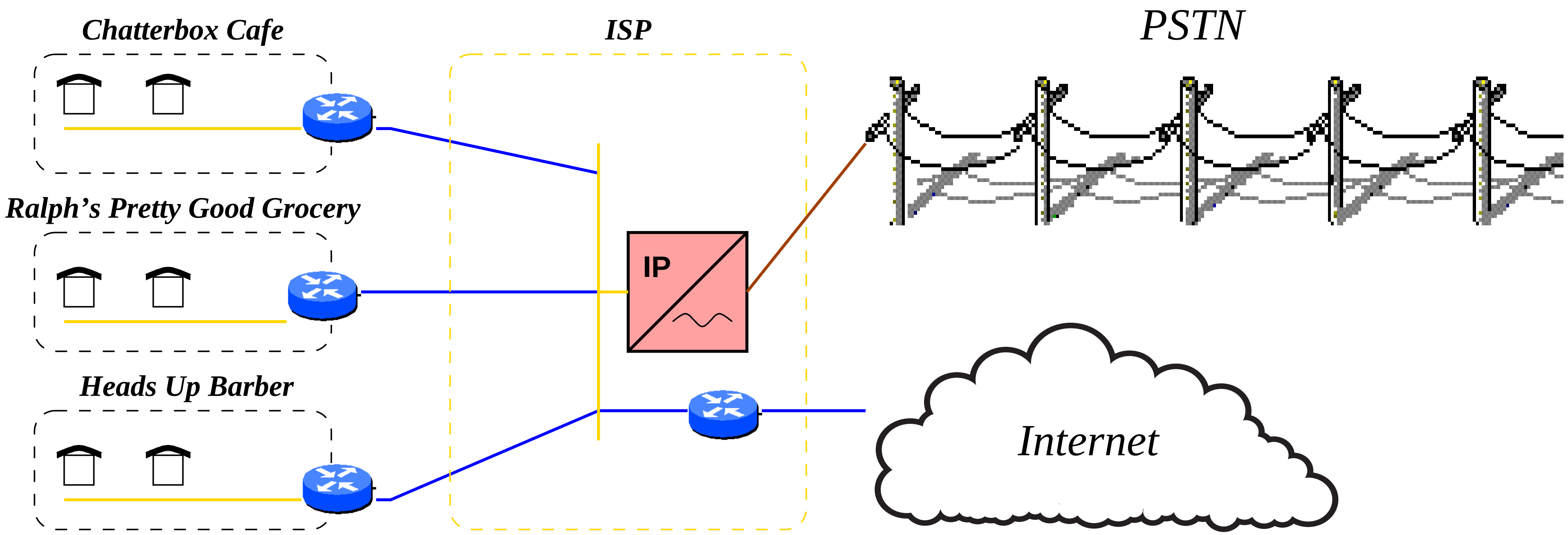
- Internet “PBX”
- Internet Centrex
- Internet Carrier

➠ same basic equipment, but size of gateway varies

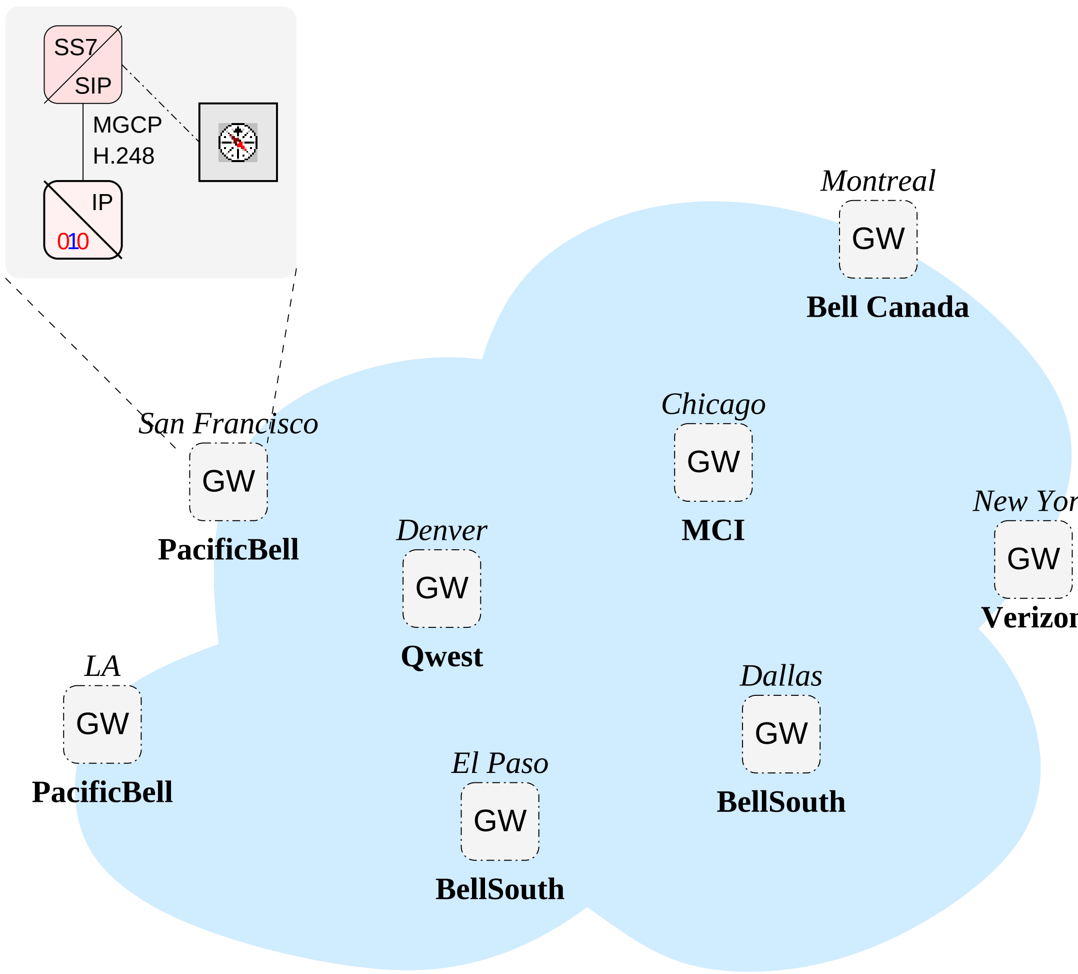
# Internet PBX



# IP Centrex


Chatterbox Cafe
ISP
PSTN
Ralph's Pretty Good Grocery
IP
Heads Up Barber
Internet

# IP Carrier


SS7
SIP
MGCP
H.248
IP
010
Montreal
GW
Bell Canada
San Francisco
GW
PacificBell
Chicago
GW
MCI
New York
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Verizon
Denver
GW
Qwest
LA
GW
PacificBell
Dallas
GW
BellSouth
El Paso
GW
BellSouth

# What is SIP good for?

- replicate functionality of traditional PSTN services:
  - caller id
  - call forwarding
  - call transfer
  - 800/900# services
  - find me/follow me
  - conference calls
- create new services:
  - Internet integration
  - programmable services
  - multi-destination routing
  - multimedia
  - event notification

# New SIP services: Internet integration

- typically, SIP URL $\equiv$ email address, e.g., `sip:joe@net2phone.com` or `tel:+1201-555-1212`
- URLs everywhere:
  - forward calls to email
  - forward calls to web page
  - forward calls to recordings
  - pager, cell phone numbers
  - IM addresses
- SIP messages can contain HTML and other web objects:
  - menu pops up when calling restaurant
  - error messages: "not here, but please choose from ..."
  - visual caller id – photos of callee

# New SIP services: programmable services

- three sources of services:

  **Vendor:** program into software ➠ efficient, robust, but long cycles, inflexible

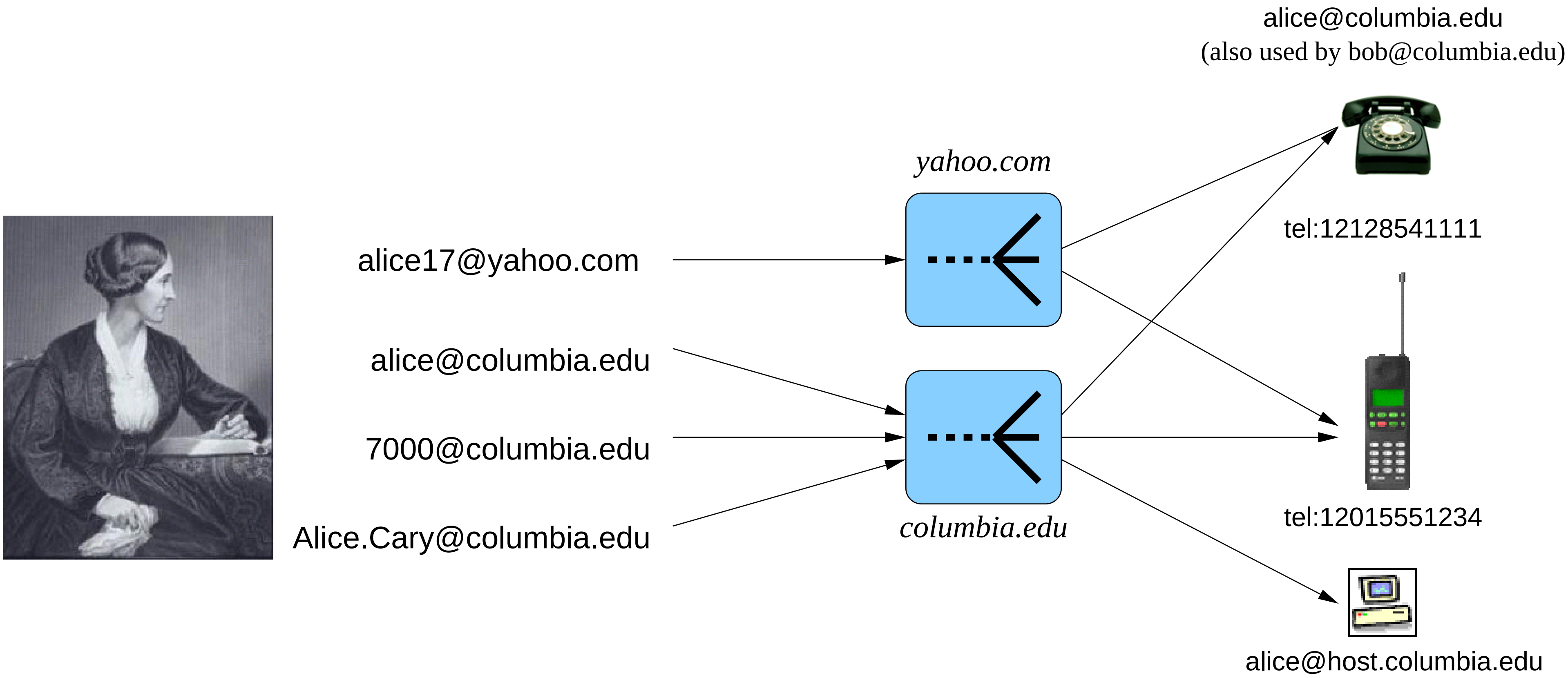
  **Service provider:** differentiation, vertical markets, but limited set

  **User:** customized and personalized

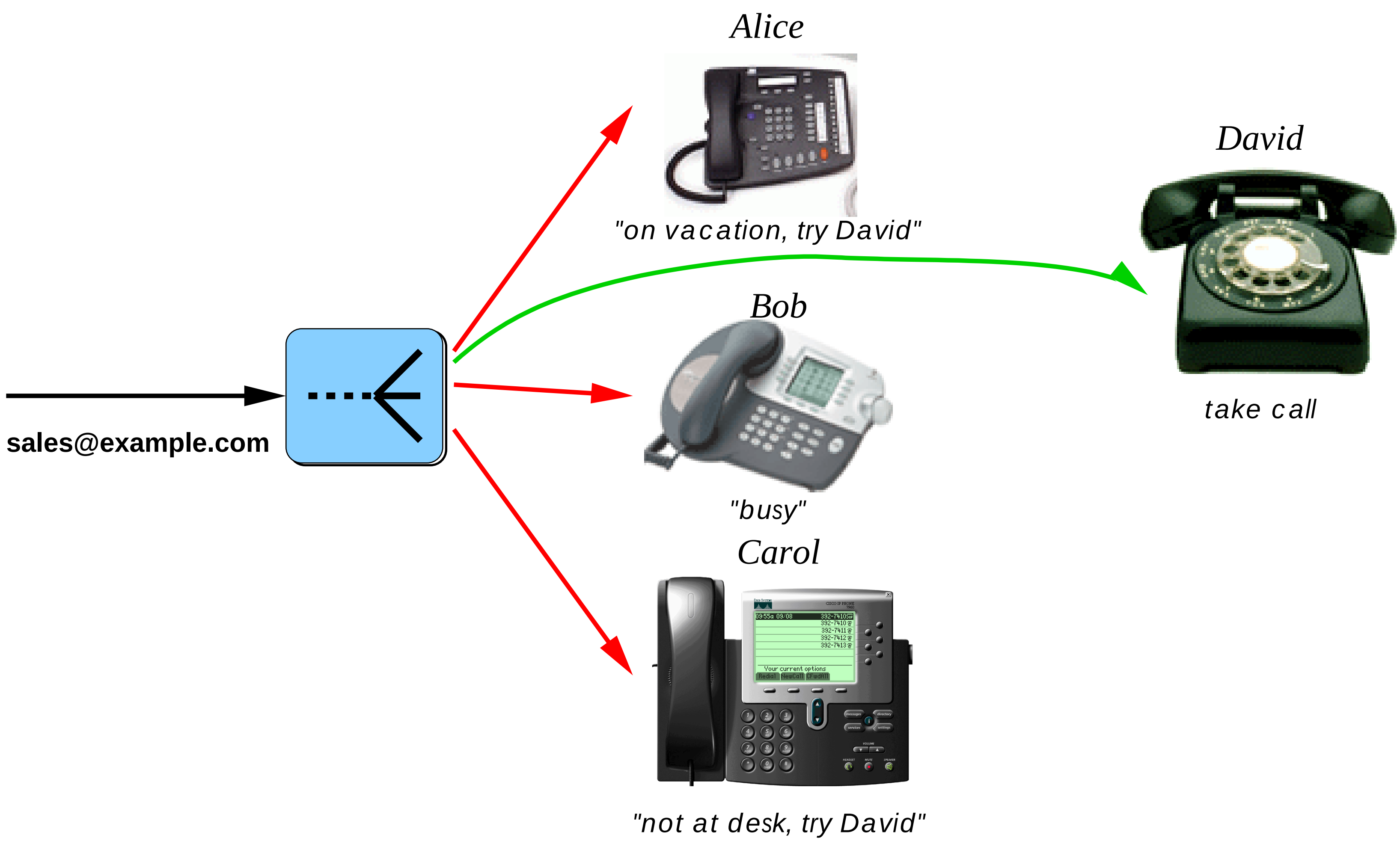
# SIP mobility

| | |
|---|---|
| terminal | cross-provider |
| personal | different terminals, same address |
| service | different terminals, same services |
| session | move sessions across terminals |

# SIP personal mobility



# New SIP services: multi-destination routing



# New SIP services: event notification

- many telecom services are really events:
  - voicemail notification
  - call supervision
  - automated call back
  - call waiting
- generalizes to
  - physical events: “water in basement”
  - communication events: “email has arrived”
  - network events: “print job is done”

# What kind of SIP products are emerging?

**SIP libraries:** for building end systems

**SIP “clients”:** also known as user agents; PC-applications

**SIP proxy servers:** call routing and applications

**SIP unified messaging servers:** record voice calls

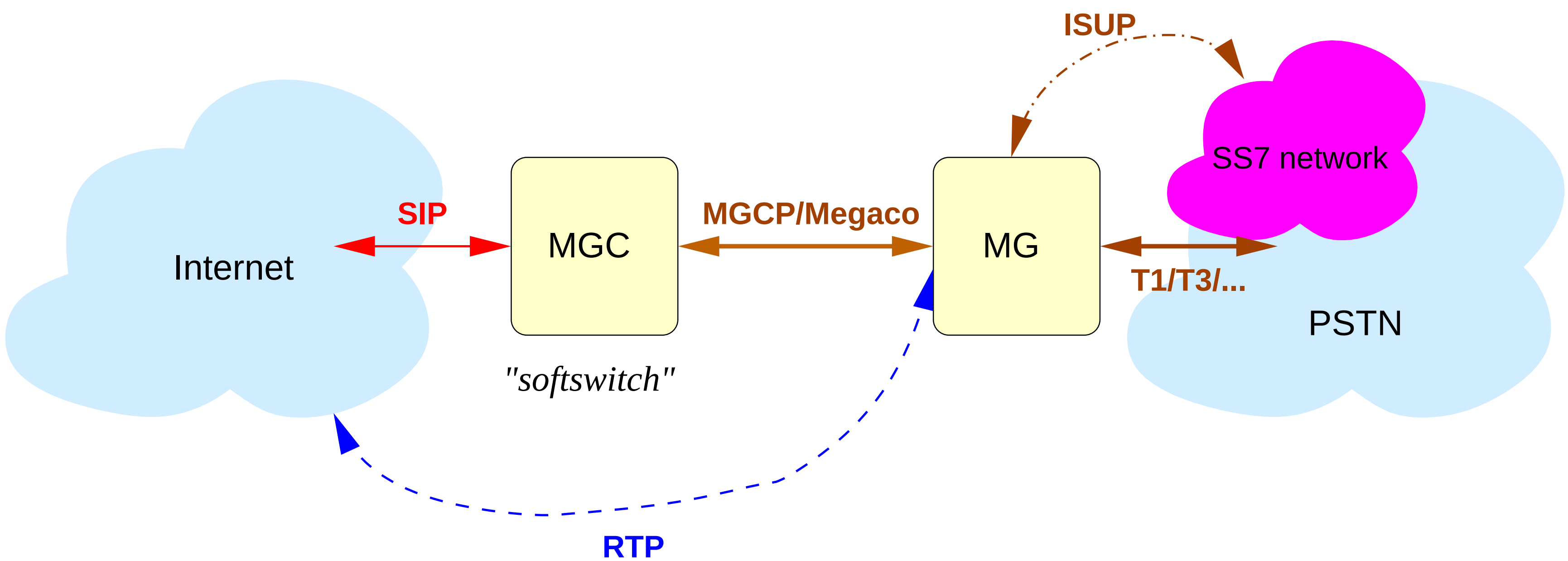
**SIP conference servers:** multipoint control units

**SIP testers:** debug applications, load testing

**SIP-enabled firewalls:** get voice through firewalls

# Marketing terms: softswitches

Commonly used in marketing, but pretty vague:




- Software version of class-4/class-5 switch?
- doesn't really "switch" voice

# Marketing term: application server

- supports Internet telephony applications
- typically, programmable:
  - APIs, such as JAIN and Parlay
  - Java servlets
  - Call Processing Language
- may be able to initiate calls or just route calls

# SIP-enabled networks

- Chunghwa Telecom, Taipei
- Level 3
- MCI Worldcom
- VONage

Others to follow soon.

# Example: Pingtel SIP phone

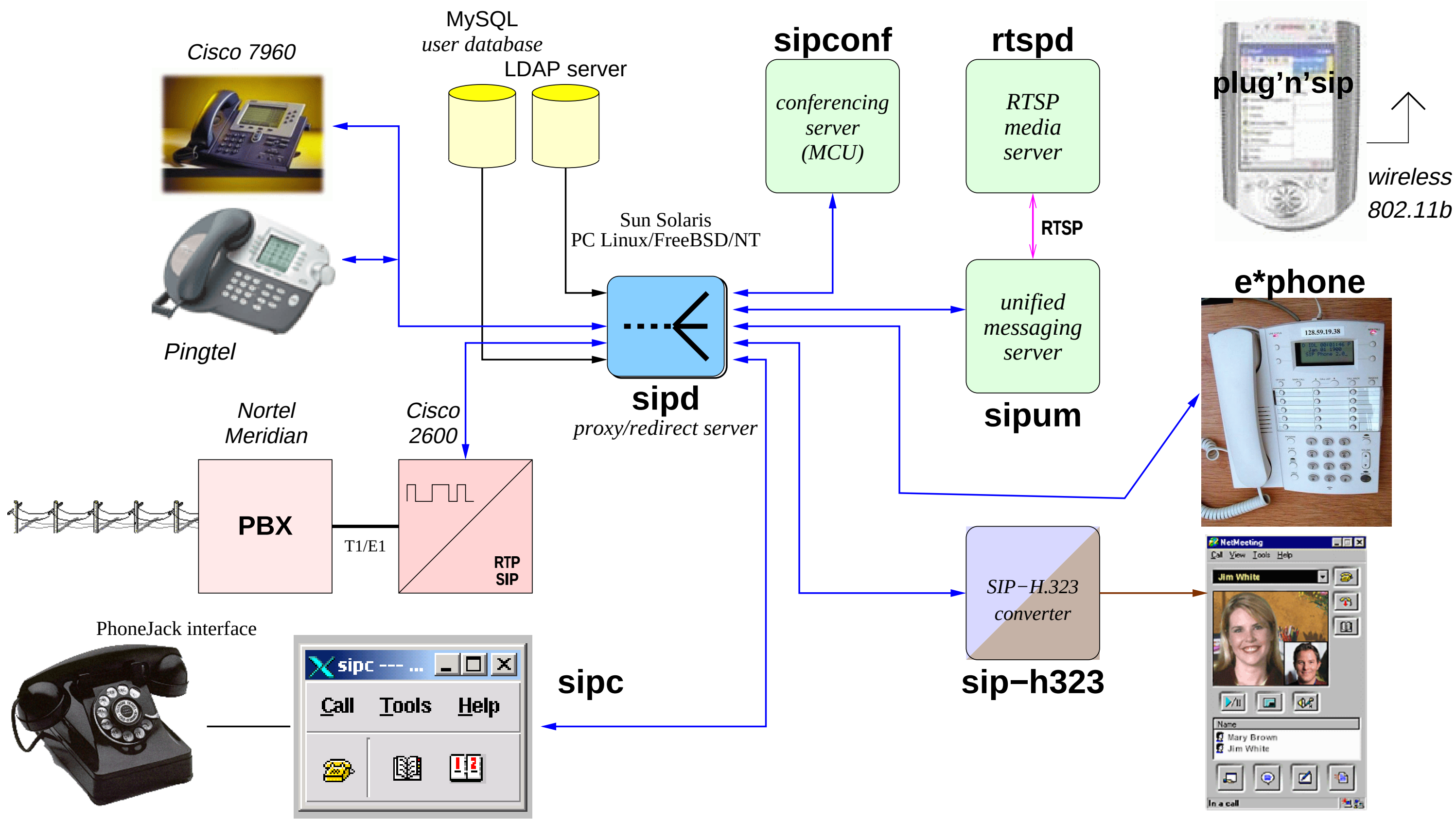
# Example: Cisco and 3Com SIP phones

Cisco

3Com ($395 list)

# Example: Columbia CS Phone System

Expand existing PBX via IP phones, with transparent connectivity



# Status of SIP in the market

- all major manufacturers of telephony-related equipment appear to be working on SIP
- generally, first-generation products
- not yet widely available as consumer products
- but largely interoperable
- use of SIP in Windows XP (MS Messenger) will accelerate uptake

# Differences: Internet Telephony $\leftrightarrow$ POTS

- separate control, transport (UDP) ➠ no triangle routing
- separate connectivity from resource availability
- separate services from bit transport
- datagram service ➠ less bootstrapping
- in-band signaling ➠ higher speed
- features “network” $\longrightarrow$ end system: distinctive ringing, caller id, speed dialing, number translation, . . . ➠ scaling
- features: intra-PBX = inter-LATA and general
- protocols: user-network = network-network signaling

# PSTN Legacies to Avoid

- E.164 numbers – might as well wear bar codes
- tones (e.g., failure indications)
- in-band signaling (DTMF)
- systems with user interface knowledge (12 keys, voice)
- voice-only orientation (e.g., MGCP/Megaco)
- integration of bit transport and services
- service-specific billing
- trigger model for service creation
- trusted networks without crypto authentication

$\longrightarrow$ confine PSTN knowledge to edge of network

# Replication of Existing Services

- "user is familiar with PSTN services"
- but how many users actually know how to use call transfer or directed pick-up?
- user interface is often just legacy of key systems or other ancient technology
- avoid binding of identifiers to devices – call person or group of people, regardless of location
- instead, model desired behavior
- single-server features don't need standardization
- find general mechanisms (e.g., REFER for three-party calls and various call transfers)

# Terminology Overload

Invasion of the meaningless technical-sounding terms, attempting to familiar mimic PSTN boxes:

- CO switch $\longrightarrow$ soft switches = gateway + SIP UA + ?
- SCP $\longrightarrow$ application servers = proxy? web server? media server?
- PBX $\longrightarrow$ Internet PBX = proxy? + gateway?
- …

Temptation: new name $\longrightarrow$ new protocols, APIs, … – the old box boundaries don't necessarily make sense!

# It's That Simple. . .

We really only have a few basic components:

- PSTN gateway, with some combination of FXO/FXS
- SIP proxy/redirect/registrar servers (or H.323 gatekeepers)
- SIP user agents (or H.323 terminals): PCs, phones
- media storage servers
- DNS, directory, web, email, news, . . . servers

# Invisible Internet Telephony

VoIP technology will appear in . . .

- Internet appliances
- home security cameras, web cams
- 3G mobile terminals
- fire alarms
- chat/IM tools
- interactive multiplayer games

# Programmable Internet Telephony

| | APIs | servlets | sip-cgi | CPL |
|---|---|---|---|---|
| Language-independent | no | Java only | yes | own |
| Secure | no | mostly | no, but can be | yes |
| End user service creation | no | yes | power users | yes |
| GUI tools w/portability | no | no | no | yes |
| Call creation | yes | no | no | no |
| Multimedia | some | yes | yes | yes |

Example: integration with iCal $\longrightarrow$ automatically export personal calendar to call handling

# Example cgi script: priority

```
#! /usr/bin/env perl -w

# Prioritize messages whose 'From:' matches 'sip:hgs@' by proxying them
# with 'Priority: urgent'.

# Translate the REGISTRATIONS env variable into a list of
# registration addresses, without name-addr forms or parameters.
sub get_regs {
    my($reg_str, @regs);

    if (!defined($ENV{REGISTRATIONS})) { return (); }

    $reg_str = $ENV{REGISTRATIONS};
    ...
    @regs = split(",", $reg_str);

    grep {
        # Eliminate parameters, then strip <> forms.
        s/;.*//;
        if (/\<(.*)\>/) { $_ = $1; }
    } @regs;

    return @regs;
}
```

```
if (defined $ENV{SIP_FROM} && $ENV{SIP_FROM} =~ /sip:hgs@/) {
    foreach $reg (get_regs()) {
        print "CGI-PROXY-REQUEST $reg SIP/2.0\n";
        print "Priority: urgent\n\n";
    }
}
```

# Example CPL script: lookup

```
<?xml version="1.0" ?>
<!DOCTYPE call SYSTEM "cpl.dtd">

<cpl>
  <incoming>
    <lookup source="http://www.example.com/cgi-bin/locate.cgi?user=jones"
            timeout="8">
      <success>
        <proxy />
      </success>
      <failure>
        <mail url="mailto:jones@example.com&Subject=lookup%20failed" />
      </failure>
    </lookup>
  </incoming>
</cpl>
```

# Potential stumbling blocks

**NATs:** VoIP violates new Internet “architecture” of TCP client-server ➠

- application-layer gateways
- avoid unnecessary NATs (get addresses)
- IPv6

**Firwewalls:** Out-of-band = control + data ➠ need modifications

**Walled-garden:** only carrier-approved services

- escalating attempts to try to prevent people from “bypassing” services
- probably violates common-carrier status
- WAP as warning

# Where is SIP being defined?

| | |
|---|---|
| IETF (Internet Engineering Task Force) | SIP core and extensions |
| 3GPP (3rd Generation Partnership) | mobile networks (SIP for signaling) |
| Softswitch Consortium | profiles for soft switches |
| PacketCable | profiles for cable modems |
| SIP Forum | evangelism, operational guidelines |

# For more information...

**SIP:** http://www.cs.columbia.edu/sip

**SDP:** http://www.cs.columbia.edu/˜hgs/internet/sdp.html

**RTP:** http://www.cs.columbia.edu/˜hgs/rtp

**Papers:** http://www.cs.columbia.edu/IRT