Internet, Part 2

1) Session Initiating Protocol (SIP)

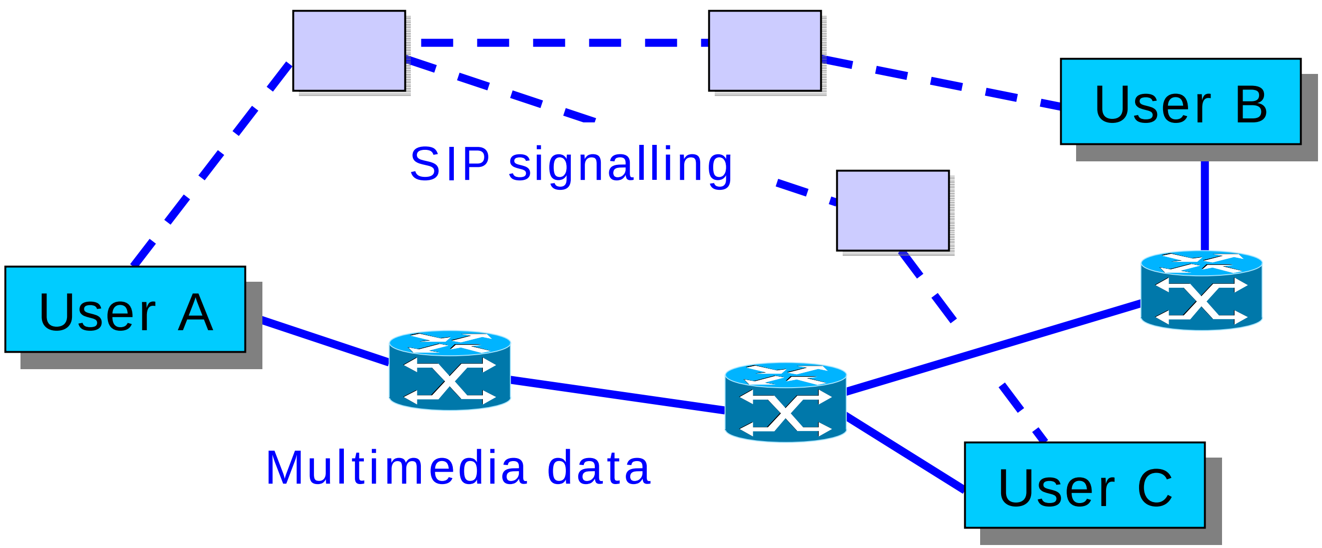
2) Quality of Service (QoS) support

3) Mobility aspects (terminal vs. personal mobility)

4) Mobile IP

Session Initiation Protocol (SIP)

SIP is a protocol for handling multiparty multimedia calls (sessions). The IP routes of the control plane (SIP signalling) and user plane (multimedia data) are separate.



http://www.ietf.org/rfc/rfc3261.txt

SIP vs. H.323

H.323 is a suite of protocols for managing multiparty multimedia calls in the PSTN (in other words using circuit switched technology).

Contrary to SIP, H.323 is used today (among others it forms the basis for Microsoft's NetMeeting application).

H.323 is more complex than SIP (this is the reason we will not go into more detail in this course).

SIP has been chosen for handling call control in the IMS (IP Multimedia Subsystem) specified in 3GPP Release 5.

H.323 standards (ITU-T) vs. RFC 3261 (good tutorial)

SIP offers the following features

Signalling for handling of multiparty calls

Call forking (several users are alerted at the same time)

Capability of multimedia calls (voice ,video, etc. at the same time) can be negotiated using Session Description Protocol (SDP) messages carried over SIP

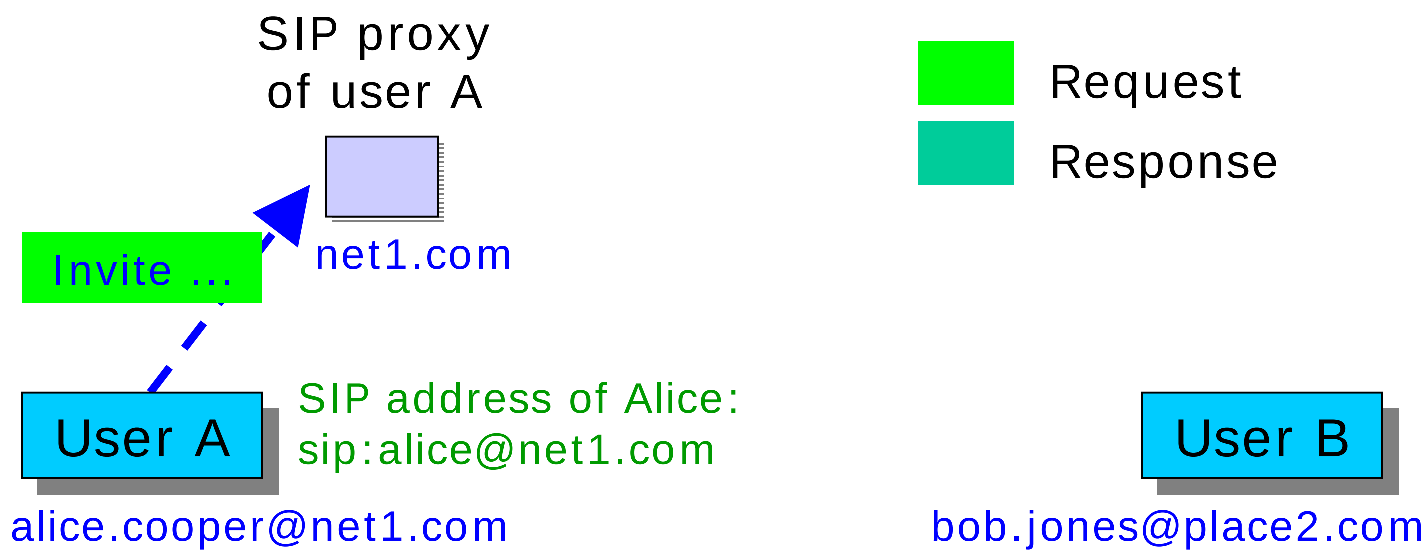
User-friendly addressing (sip:alice@net1.com)

Personal mobility (but not terminal mobility)

Good flexibility, scalability, extensibility

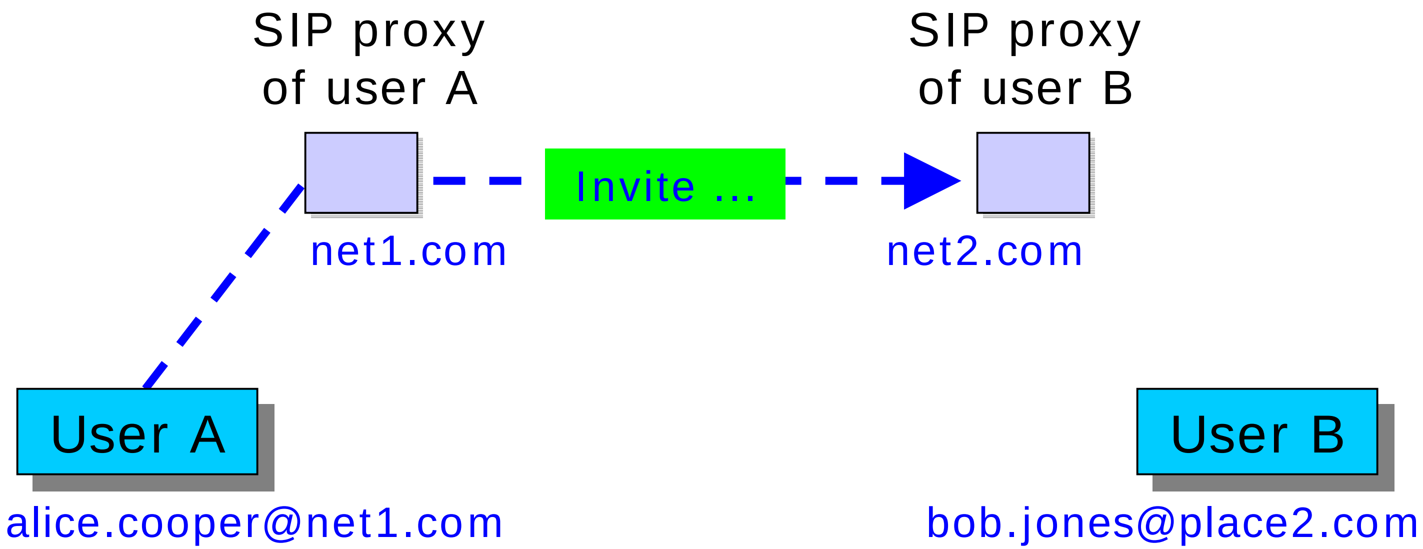
Interworking between SIP telephony and PSTN telephony (as well as between SIP addressing and E.164 addressing).

Basic (two-party) SIP call (1)



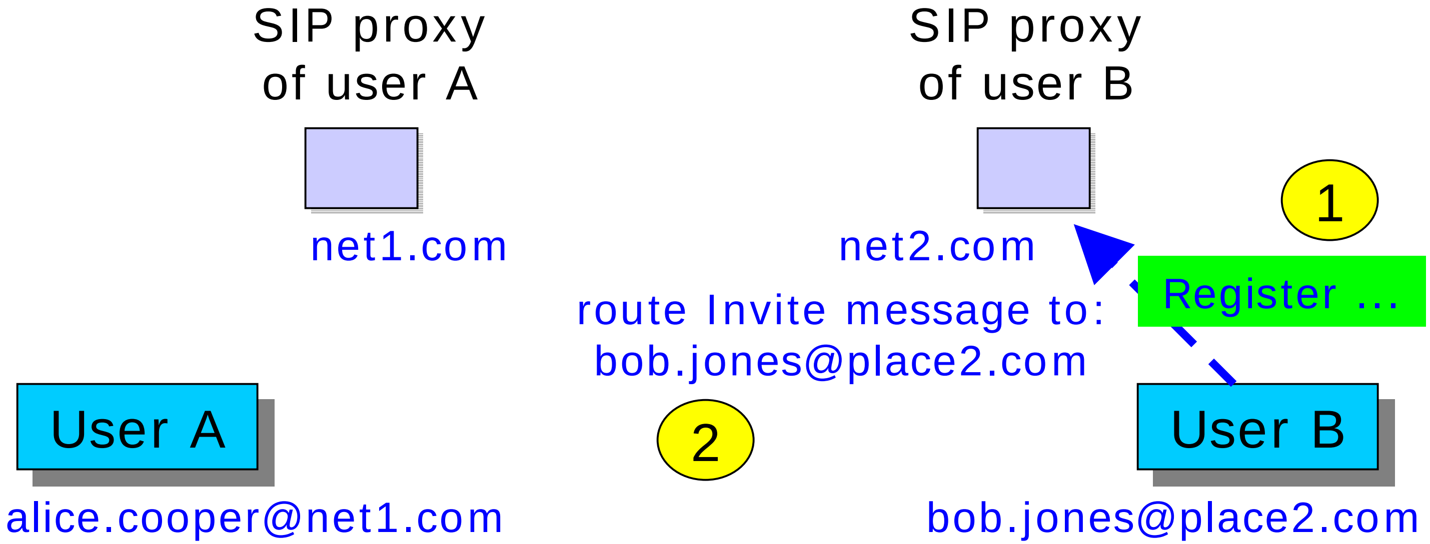
“Invite” message (corresponding to IAM message in ISUP) is sent to SIP proxy of user A. The message includes SIP address (sip:bob@net2.com) of user B.

Basic (two-party) SIP call (2)



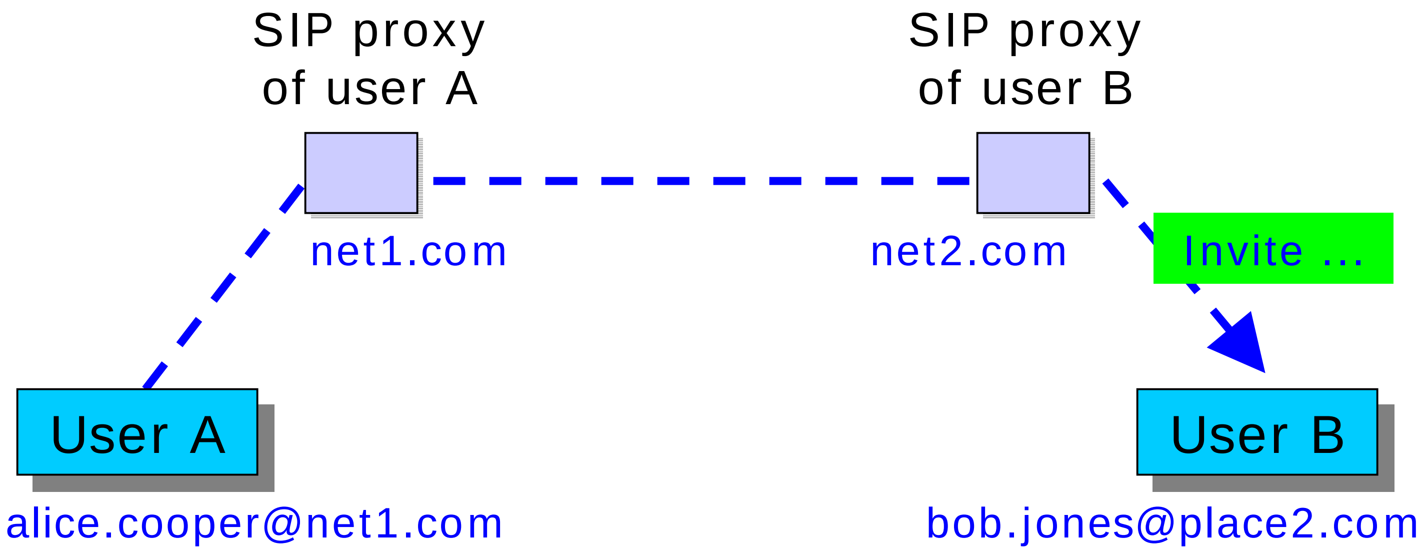
“Invite” message is routed to SIP proxy of user B (Bob). How does SIP proxy of Bob know where Bob is at this moment? (At home, at work, at computer Z, ...?)

SIP registration



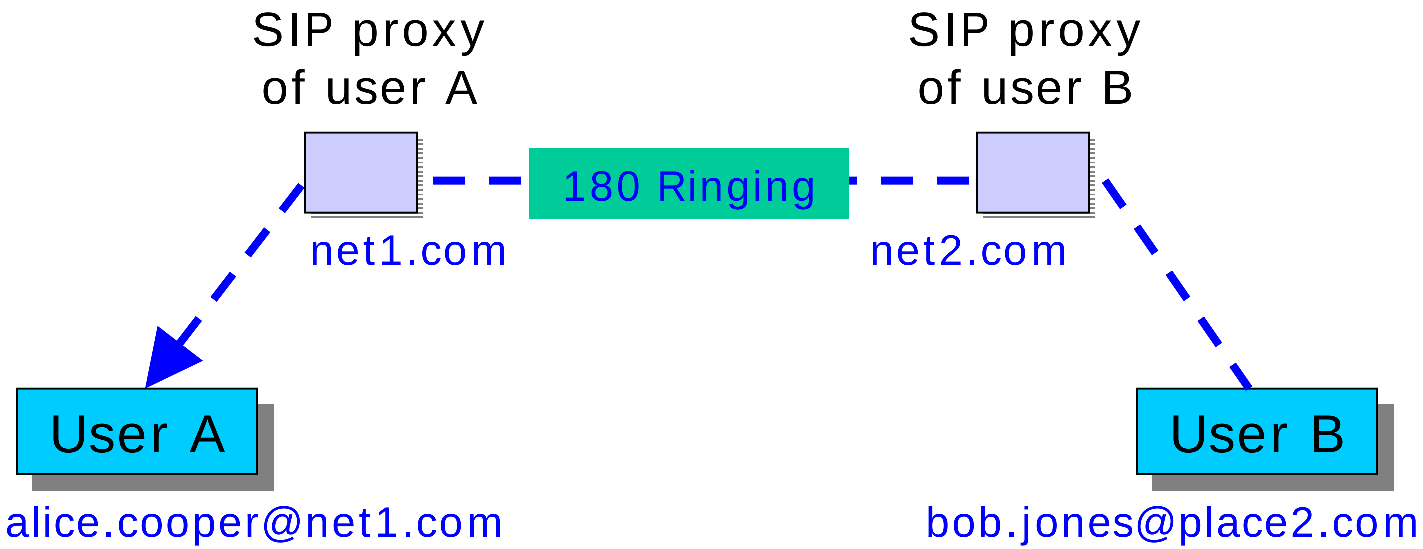
The answer is: the terminal of Bob has performed SIP registration. “Register” messages including the URL of Bob’s current terminal are sent initially (and at regular intervals) to the SIP proxy.

Basic (two-party) SIP call (3)



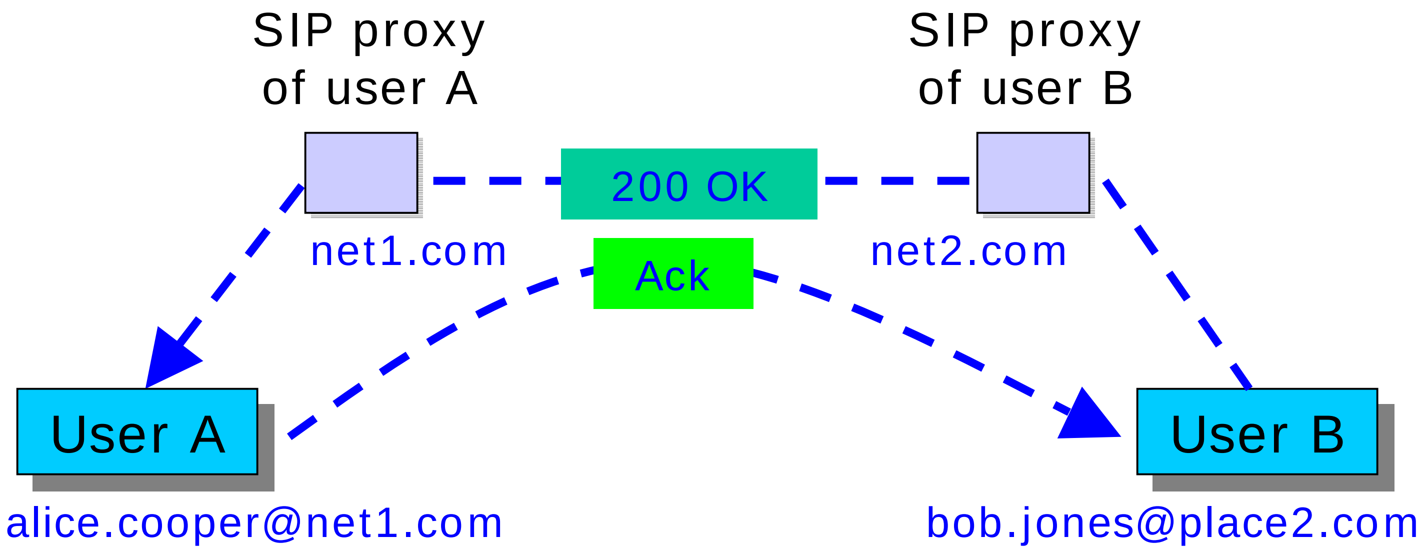
“Invite” message is routed to Bob’s terminal using Bob’s URL provided via SIP registration. Alice’s URL (alice.cooper@net1.com) is included in the message.

Basic (two-party) SIP call (4)



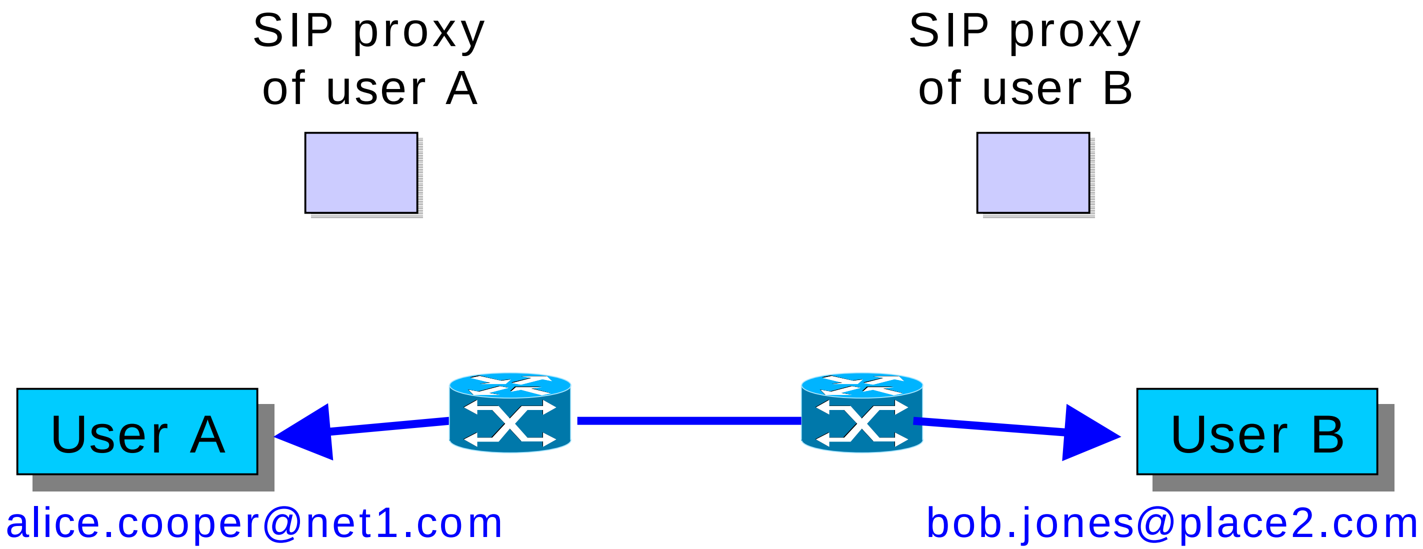
Bob’s terminal is ringing. An (optional) “180 Ringing” message is routed back to user A (Alice) and an audio ringing tone is generated in Alice’s terminal.

Basic (two-party) SIP call (5)



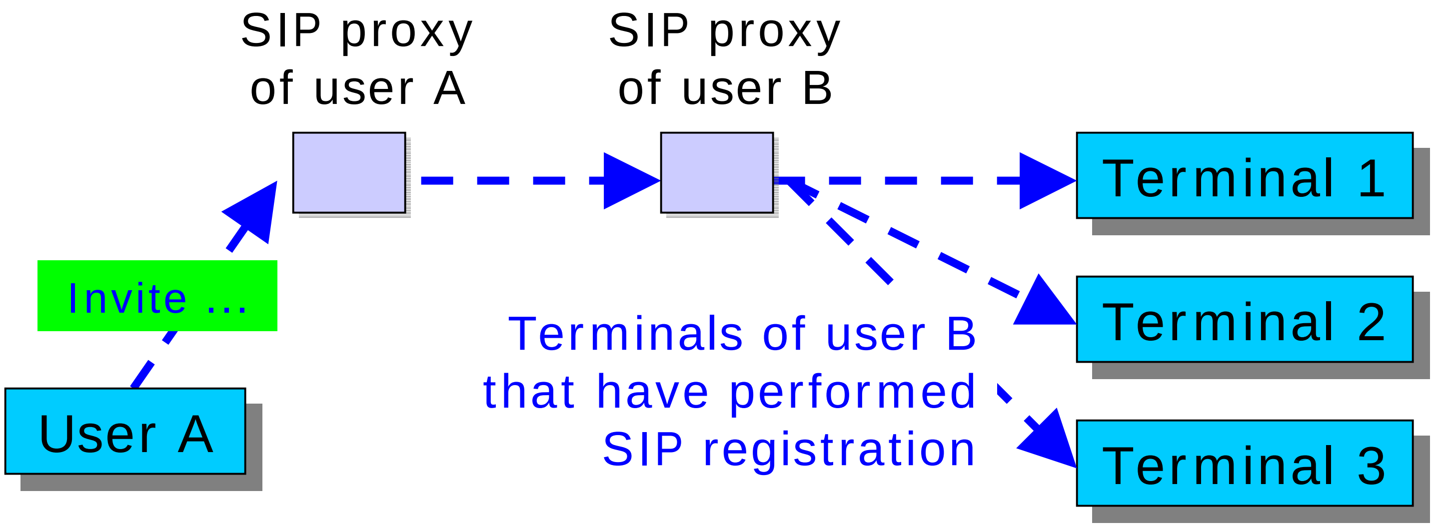
Bob answers the call. A “200 OK” message is routed back to Alice. Alice sends an “Ack” message to Bob.

Basic (two-party) SIP call (6)



After successful session establishment, the user plane data (e.g. coded speech carried on top of RTP) is carried between the terminals directly through the Internet without involving SIP proxies.

SIP forking example



Forking: Different terminals of user B are alerted at the same time. The one that answers first returns the 200 OK message ...

Three types of addresses

E.164 address → Address points directly to called user in the PSTN

358 9 1234567

MSISDN → Address points to HLR in GSM home network of called user

050 1234567

HLR knows where to route call

SIP address → Address points to SIP proxy of called user

sip:user@network.com

SIP proxy knows where to route "Invite" SIP message

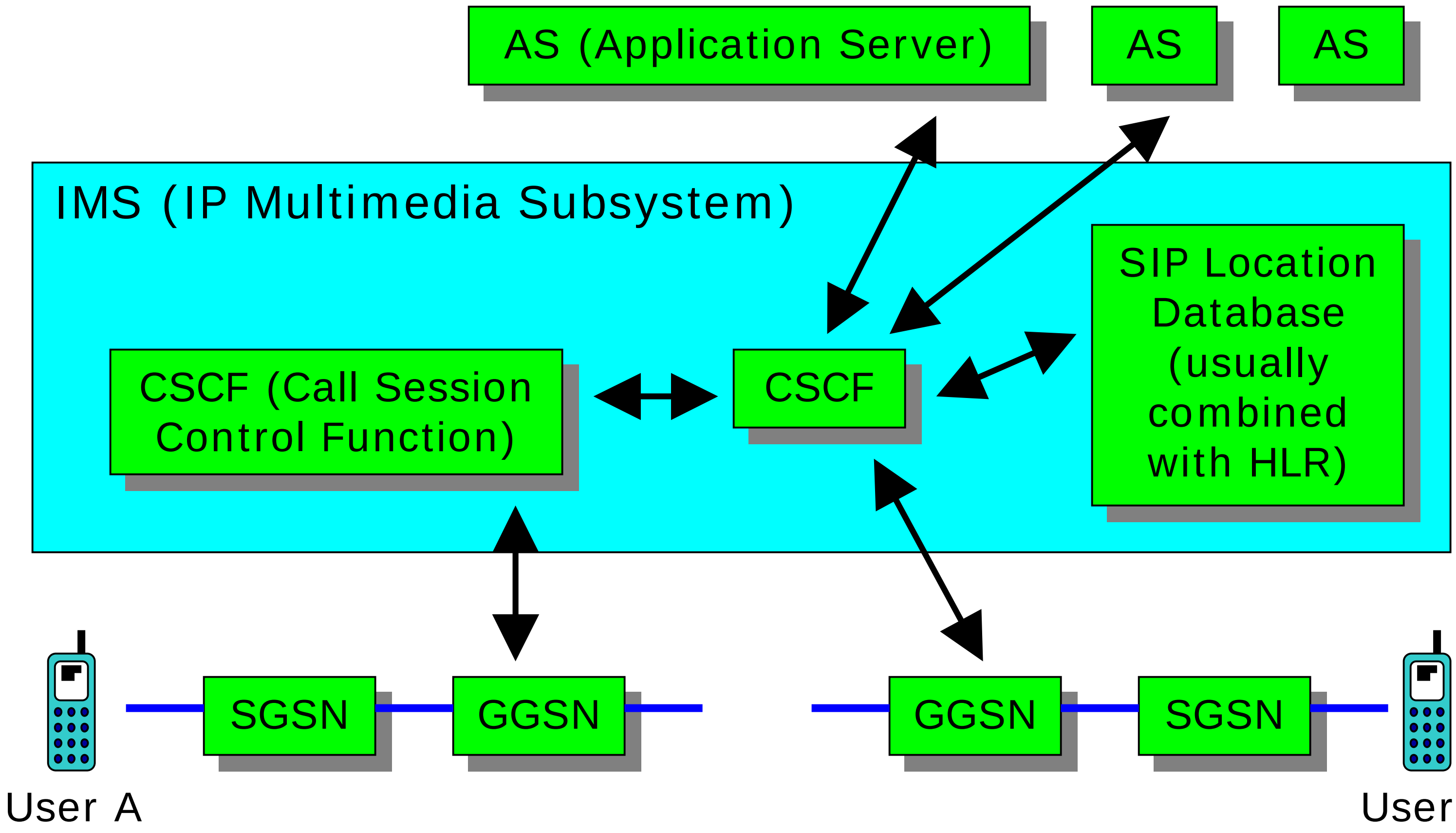
What can SIP do?

The most important task of SIP is to find out URLs of terminals to be included in the multimedia session (see previous example).

For negotiation of multimedia capabilities, SIP can carry SDP messages between end users (in “Invite” and “200 OK” SIP messages).

Unfortunately, SIP cannot influence the transport in the user plane (support of QoS and security features, inclusion of PCM/EFR transcoding equipment, etc.).

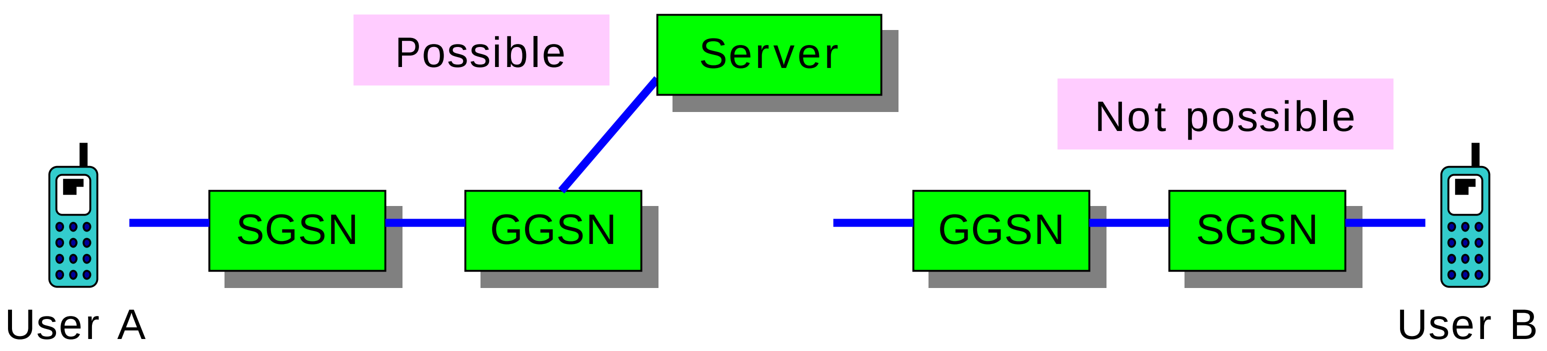
IP Multimedia Subsystem (IMS)



Why IMS?

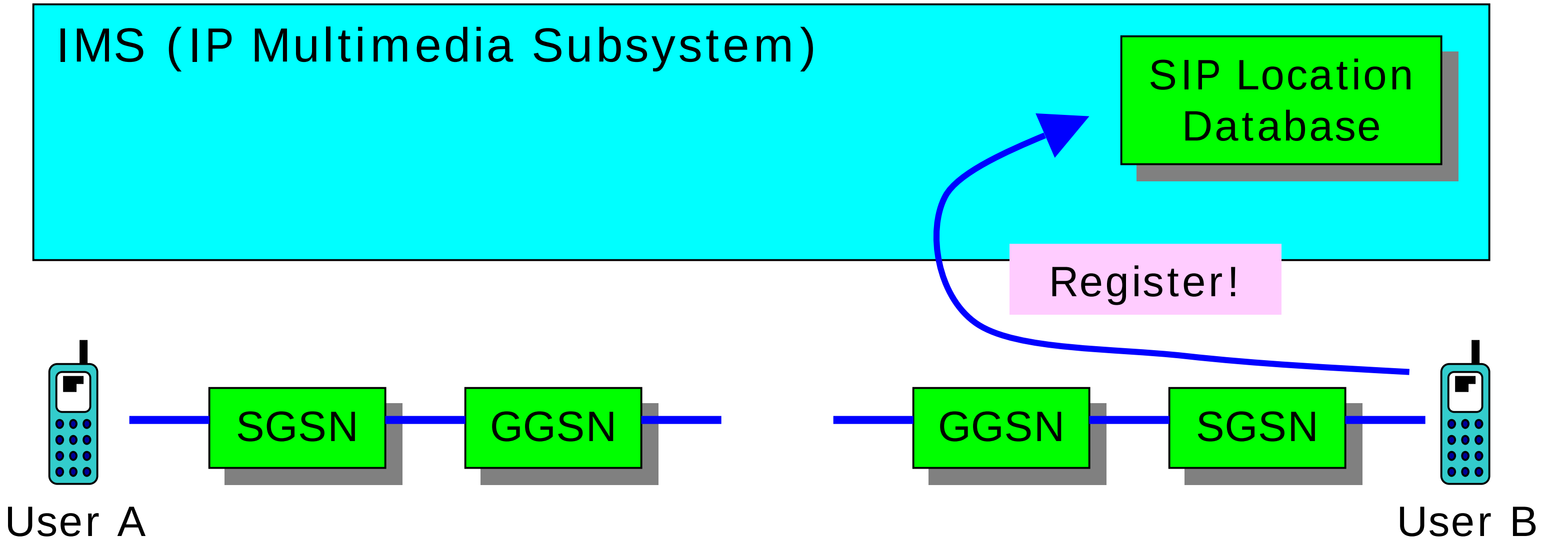
In a “conventional” GPRS-based IP network, only Client-Server types of applications are possible.

Client-Client (or peer-to-peer) types of applications are not possible since the dynamic IP address of the user B terminal is not known in the network.



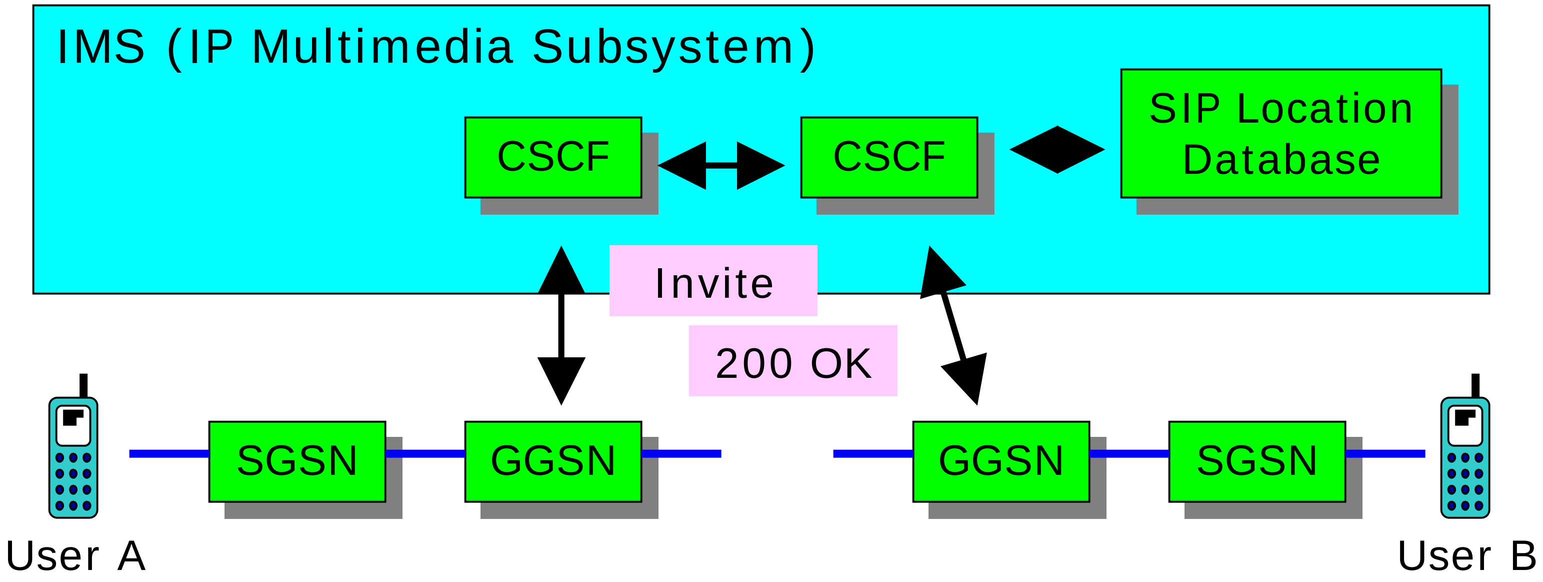
IMS operation (1)

User B can be reached only after registering in the IMS, which means binding her/his SIP address with the dynamic IP address that was allocated when the PDP context of the GPRS session was established.



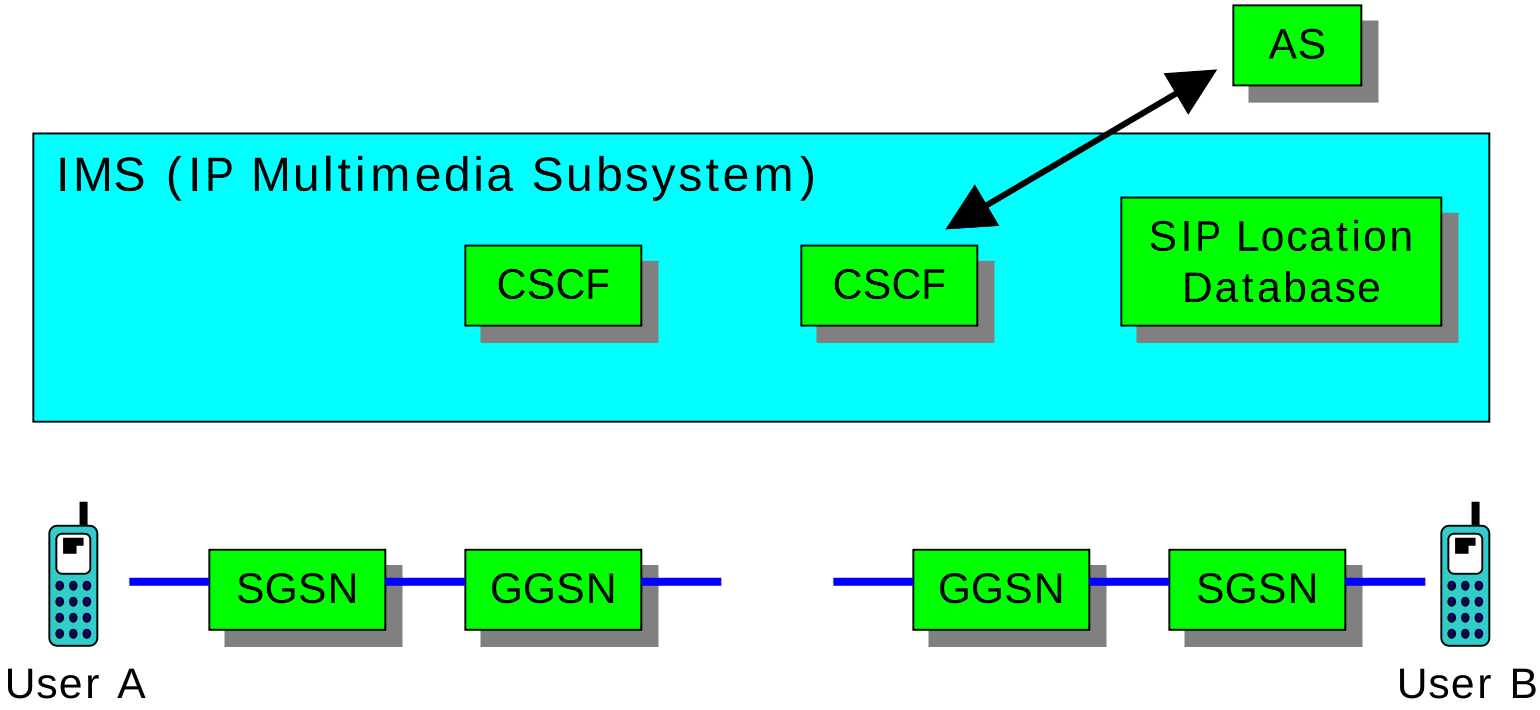
IMS operation (2)

Session (or call) control involves SIP signalling as well as network functions provided by the IMS (for instance, CSCF offers SIP proxy functionality).



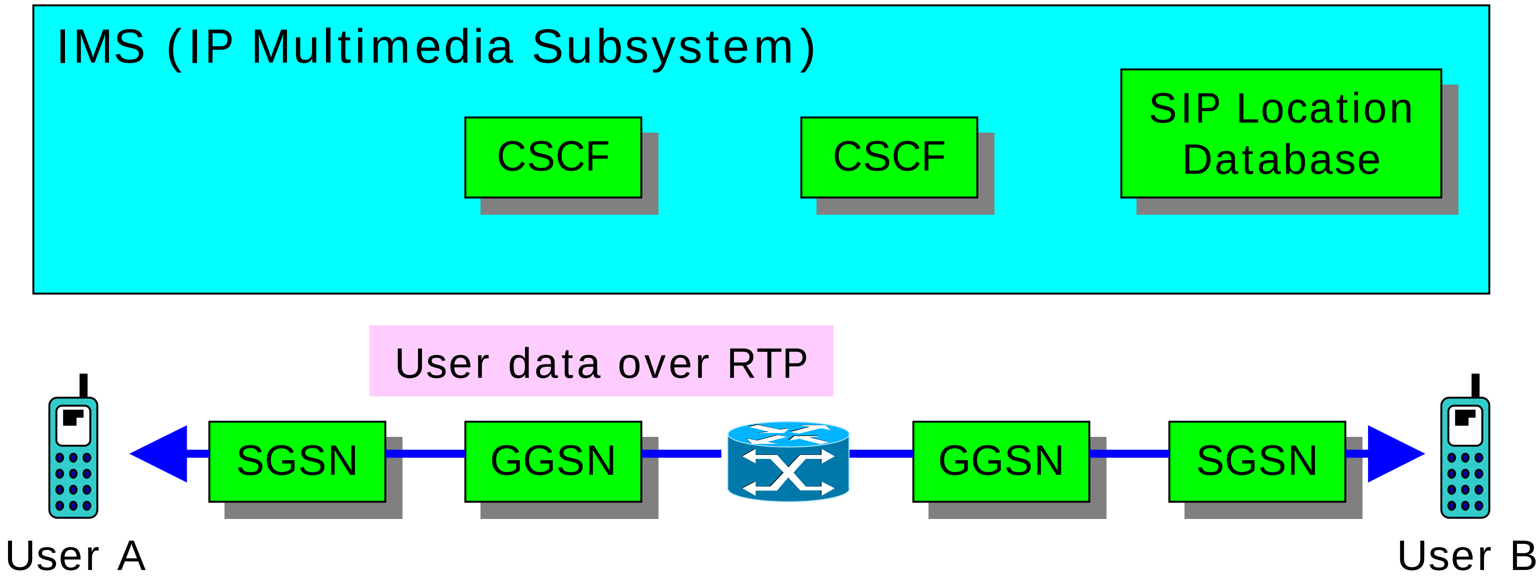
IMS operation (3)

Also, external application servers (e.g. presence server) can be employed during session control.



IMS operation (4)

After successful session establishment, the IMS is not involved in the transfer of user data (e.g., encoded speech) between the user A and B terminals.



QoS support in IP networks

“Best effort” service <=> no Quality of Service support

Some alternatives for introducing QoS in IP backbone applications (situation year 2004):

Alternative 1: RSVP (Resource ReSerVation Protocol)

Alternative 2: DiffServ (Differentiated Services)

Alternative 3: MPLS (MultiProtocol Label Switching)

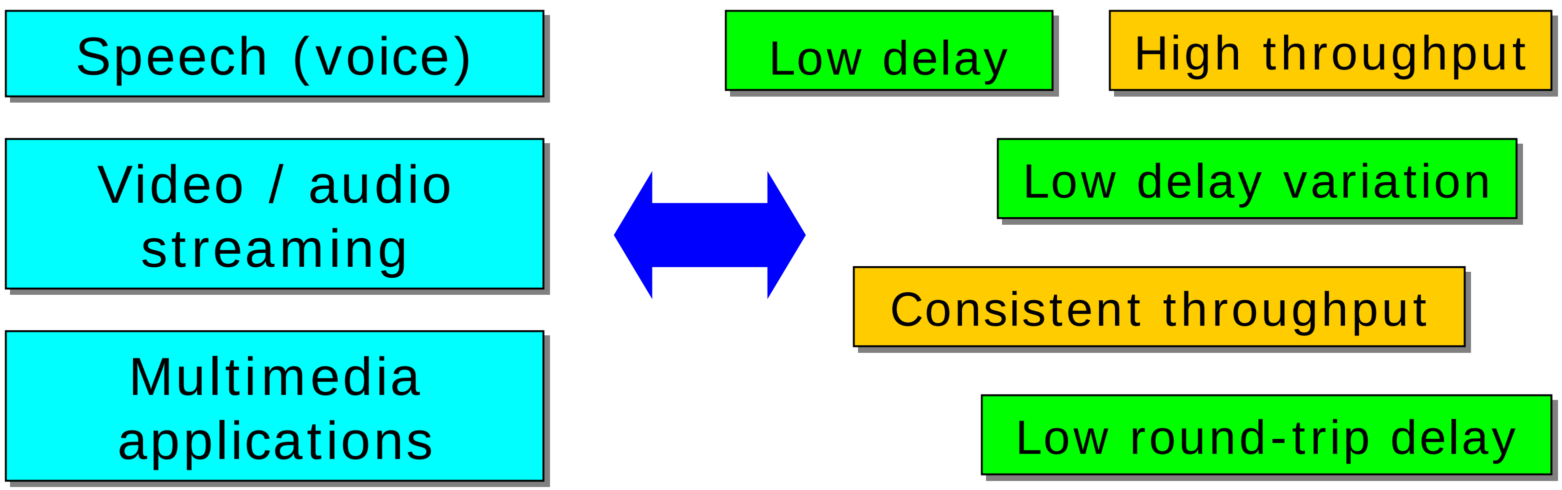
Alternative 4: IP tunneling over ATM

IETF terminology: Traffic engineering

Problems with “Best effort” IP transport

"Best effort" service is sufficient for traditional Internet applications like web browsing, e-mail, and file transfer.

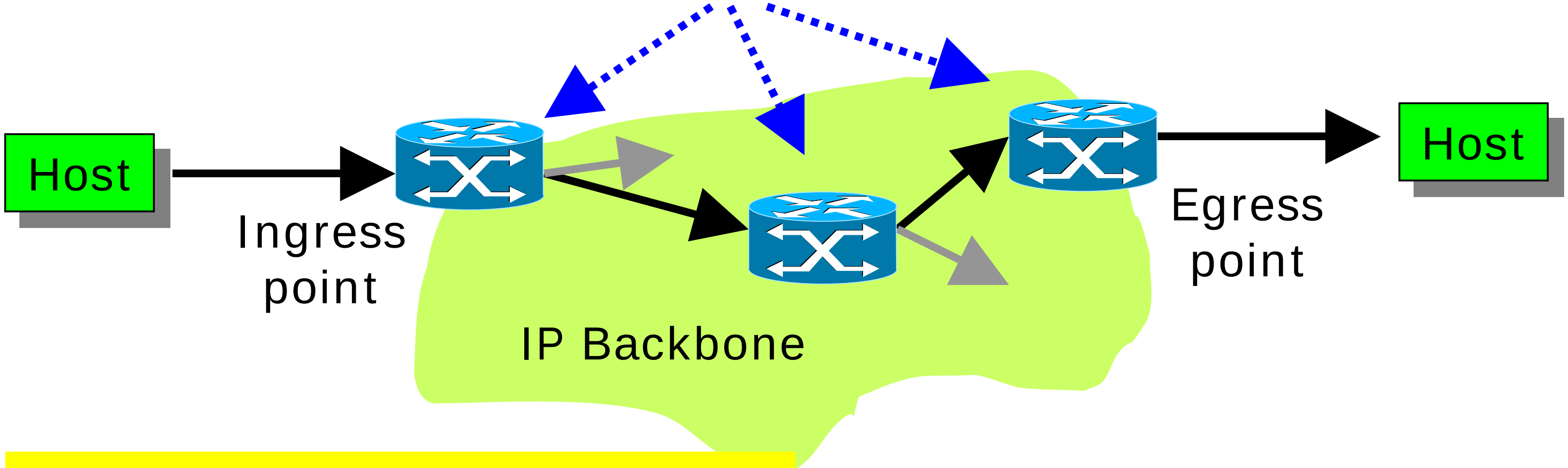
"Best effort" is not sufficient for real-time applications:



QoS support mechanisms (1)

RSVP (Resource ReSerVation Protocol) IETF RFC 2205

Resources are reserved beforehand (or at certain intervals)



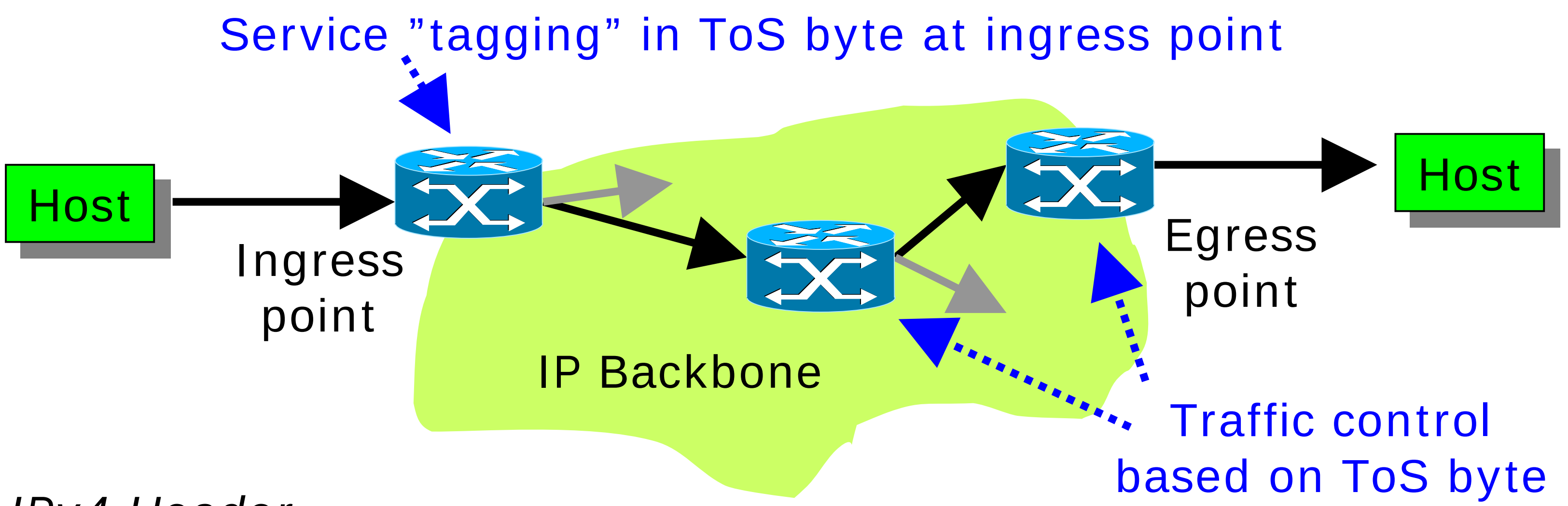
http://www.ietf.org/rfc/rfc2205.txt

RSVP can be considered an example of the *integrated services* concept (compare with *differentiated services*).

RSVP is typically used together with other mechanism(s).

QoS support mechanisms (2)

DiffServ (Differentiated Services) IETF RFC 2475



IPv4 Header

Version	IHL	Type of Service	Total length
Identification		Flags	
Time-to-live	Protocol	Heade	

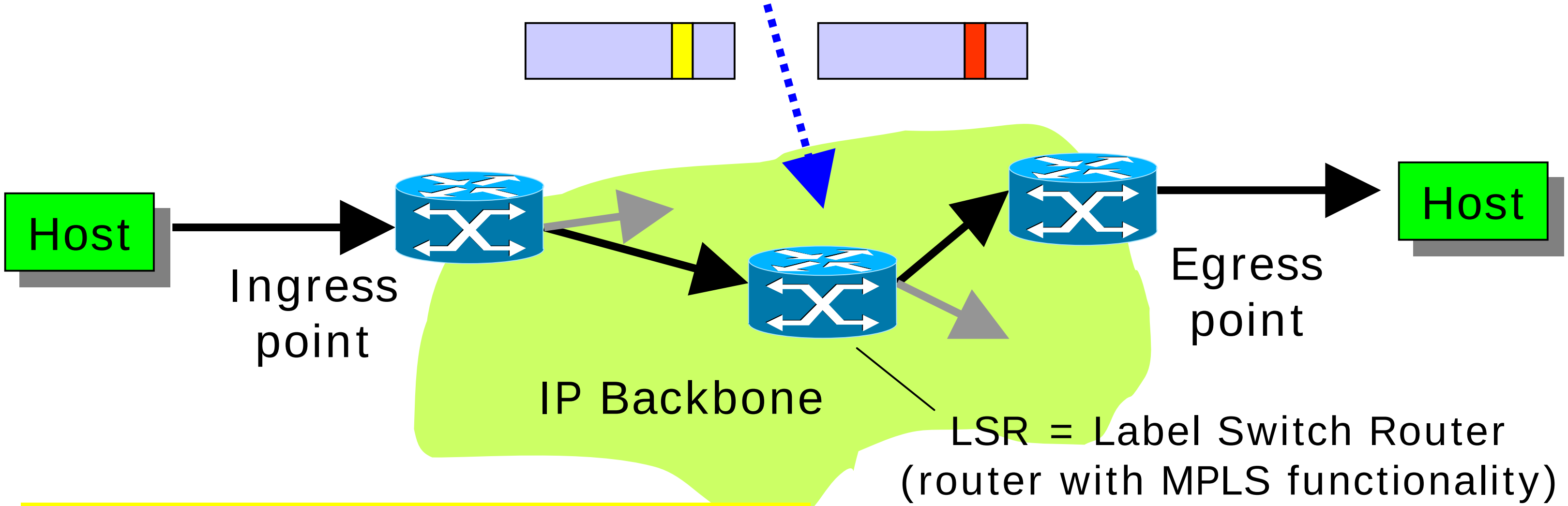
ToS byte = 8 bits (2^8 = 256 priority levels could be used)

http://www.ietf.org/rfc/rfc2475.txt

QoS support mechanisms (3)

MPLS (Multi-Protocol Label Switching) IETF RFC 2702

Label switching in all routers along the path



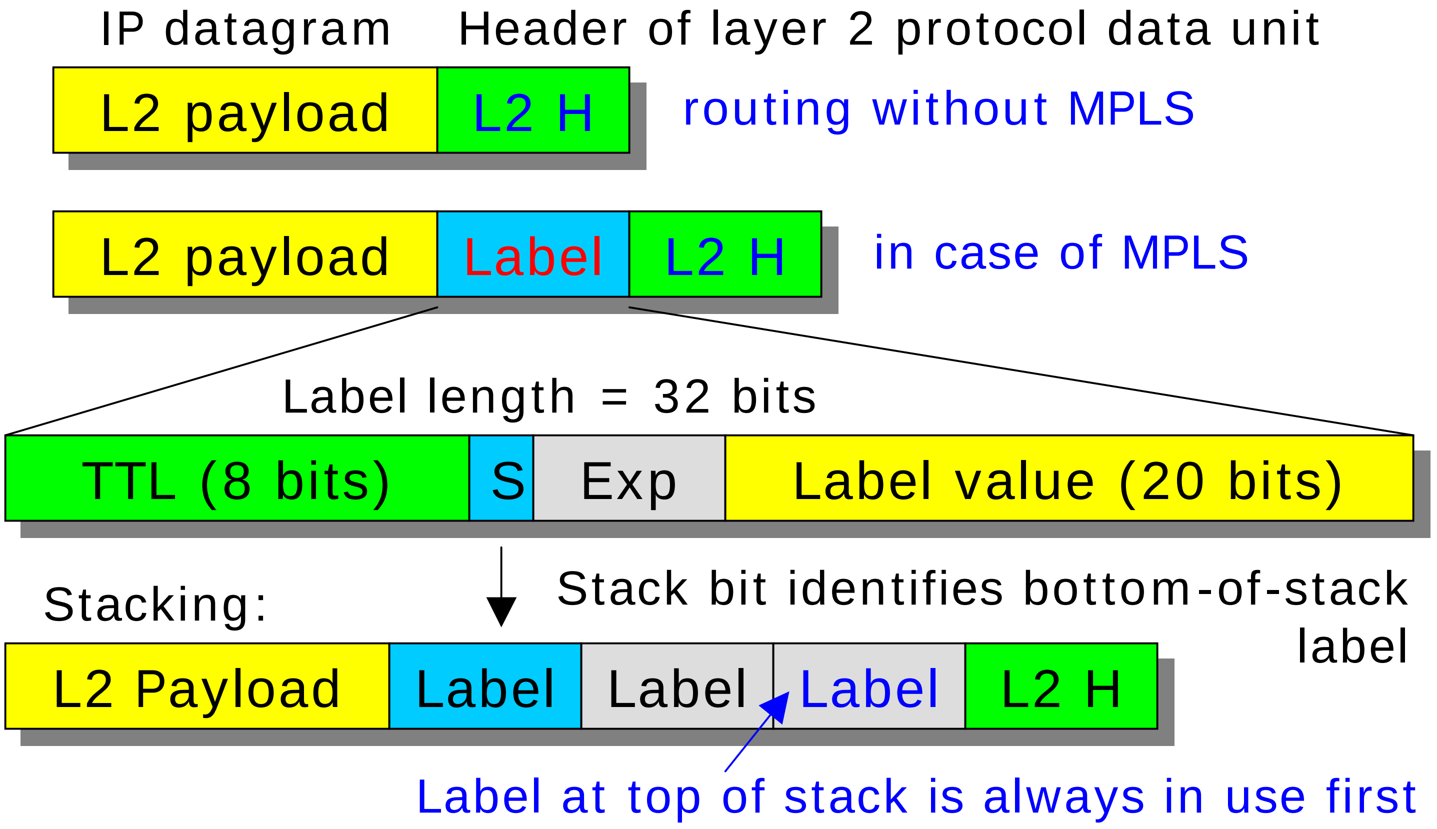
http://www.ietf.org/rfc/rfc2702.txt

Virtual connection must be established first (using e.g. RSVP).

IP datagrams are encapsulated in MPLS frames and relayed through label switch routers (only label is used for routing).

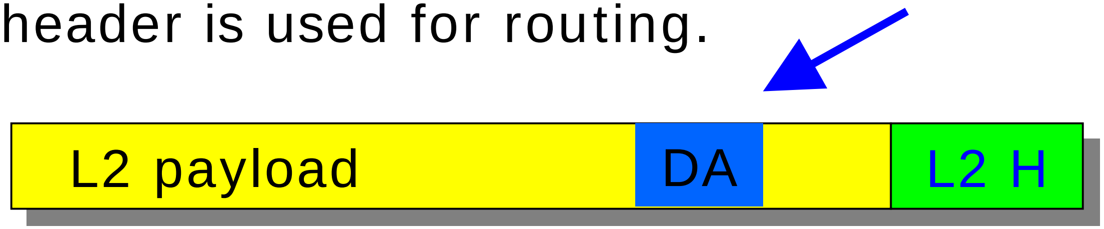
QoS support mechanisms (3 cont.)

MPLS label structure:

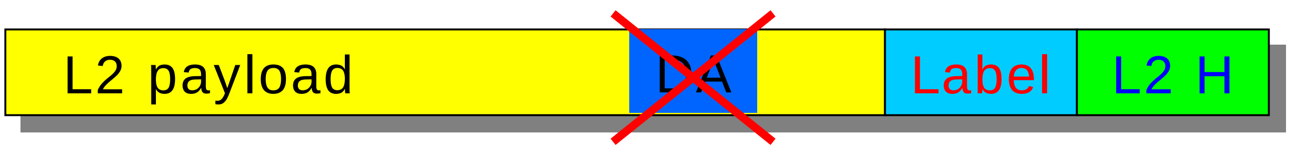


QoS support mechanisms (3 cont.)

Routing without MPLS: destination IP address in IP header is used for routing.

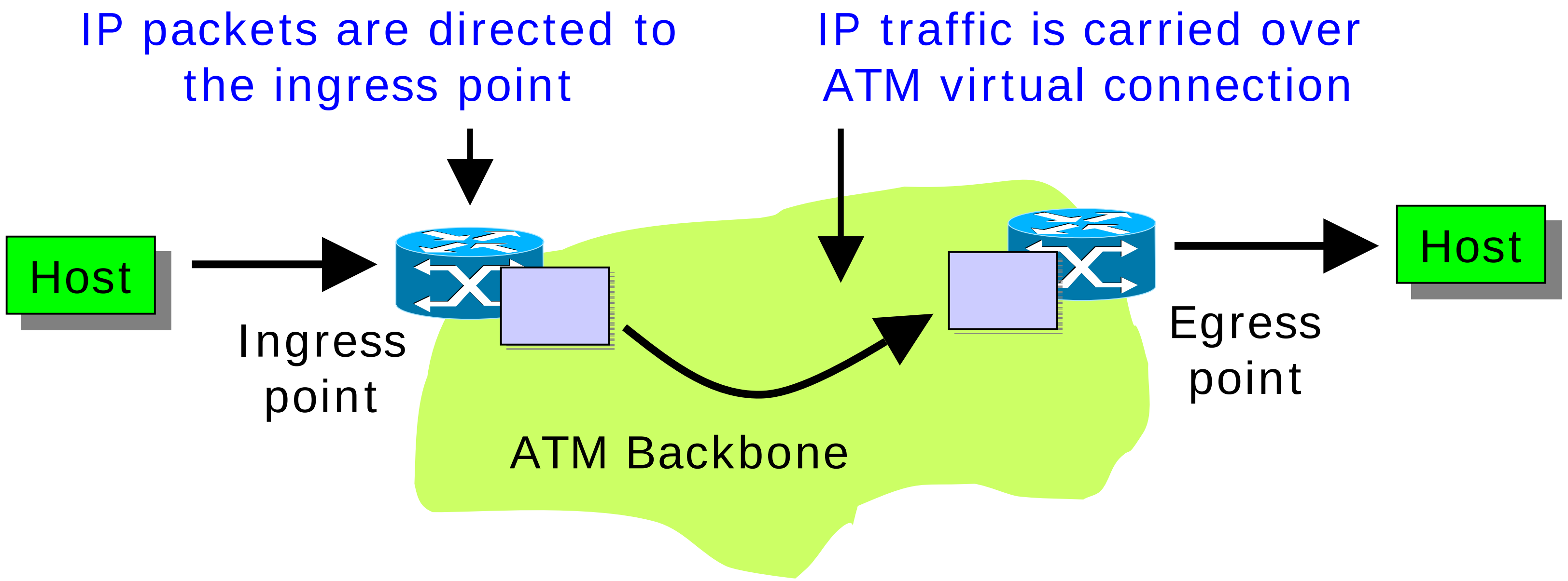


In case of MPLS: destination IP address is not used for routing along the virtual path between ingress and egress point. Routing is based on MPLS label instead.



QoS support mechanisms (4)

IP tunneling over ATM

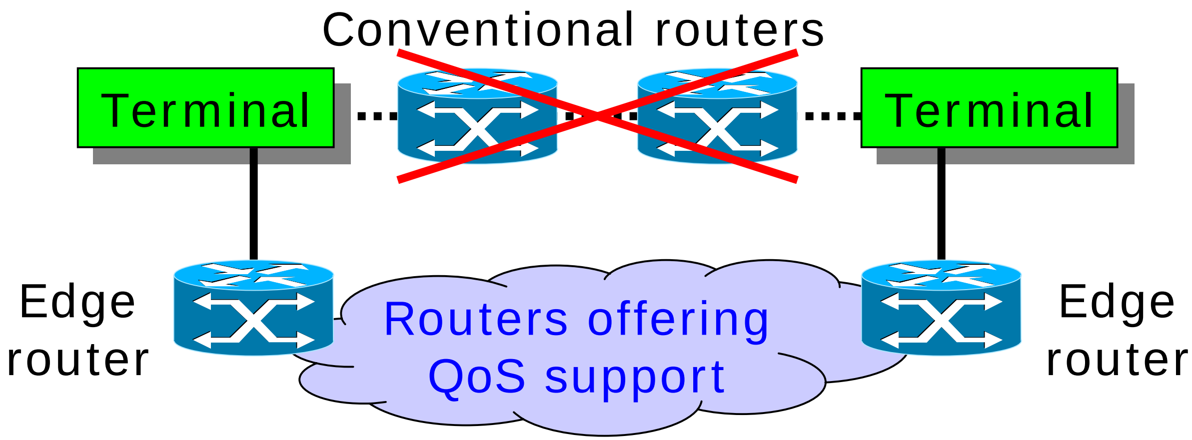


See lecture slides on ATM for protocol stacks involved

Problem with end-to-end QoS support

There are millions of Internet routers worldwide based on IPv4 and without any QoS support.

Efficient QoS support worldwide means that a large part of these routers must be updated.

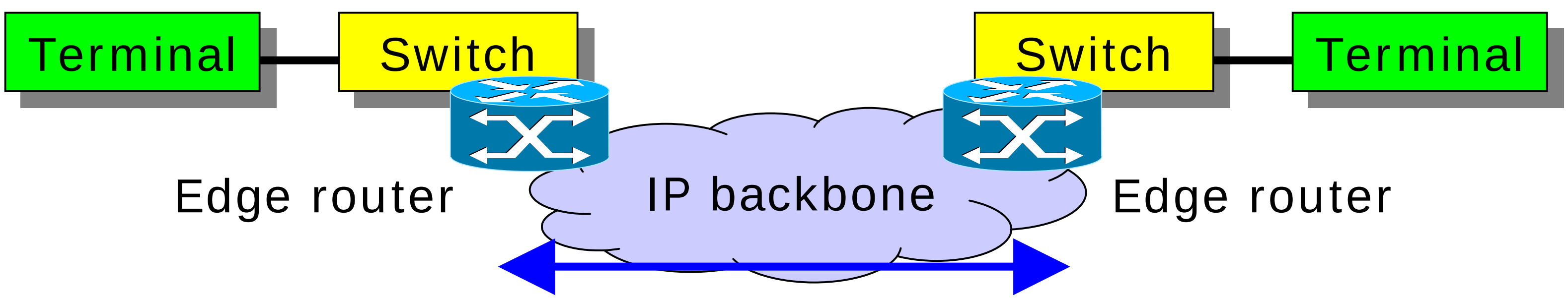


Situation is “easier” in this case

Traditional circuit switched network:

No store-and-forward nodes in network => predictable delay

Circuit switched network including IP bearer section:



QoS support: Small/predictable delay between edge routers

Mobility in IP networks

One can very generally define two types of mobility:

Personal mobility (e.g. offered by SIP)

Terminal mobility (e.g. offered by GPRS)

The concept “Mobile IP” tries to combine both, when implemented together with wireless LAN technology (see last slides of this lecture).

The IMS (IP Multimedia Subsystem) concept in 3GPP Release 5 also tries to combine both (using SIP and GPRS technology).

User mobility vs. terminal mobility

Personal mobility (e.g. offered by SIP):

User can move around in the network and use a new terminal after registration via the new terminal. The new terminal has the same address for incoming calls as the old terminal. However, terminal mobility is not supported.

Terminal mobility (e.g. offered by GPRS):

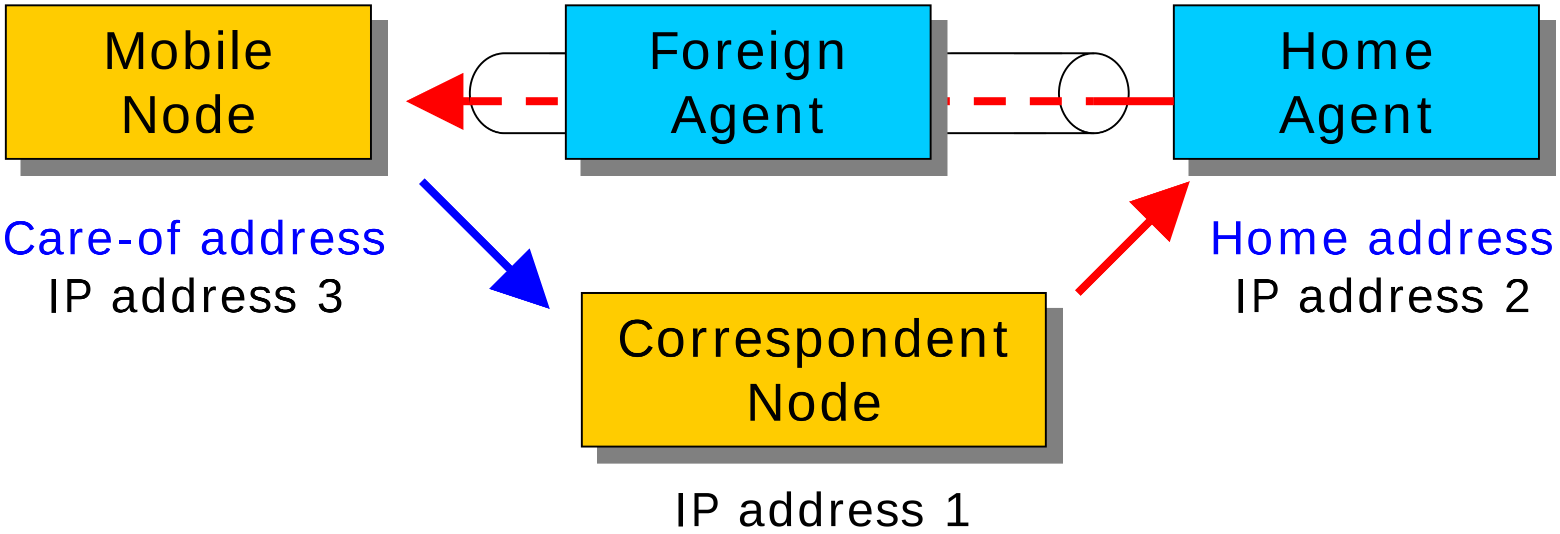
User can move around in the network and use the terminal at different locations => location updating. However, using different terminals means different addresses as far as incoming calls are concerned.

Mobile IP concept

- IETF solution for (e.g.) Wireless LAN –type applications
- Wide-scale deployment together with IPv6 ?

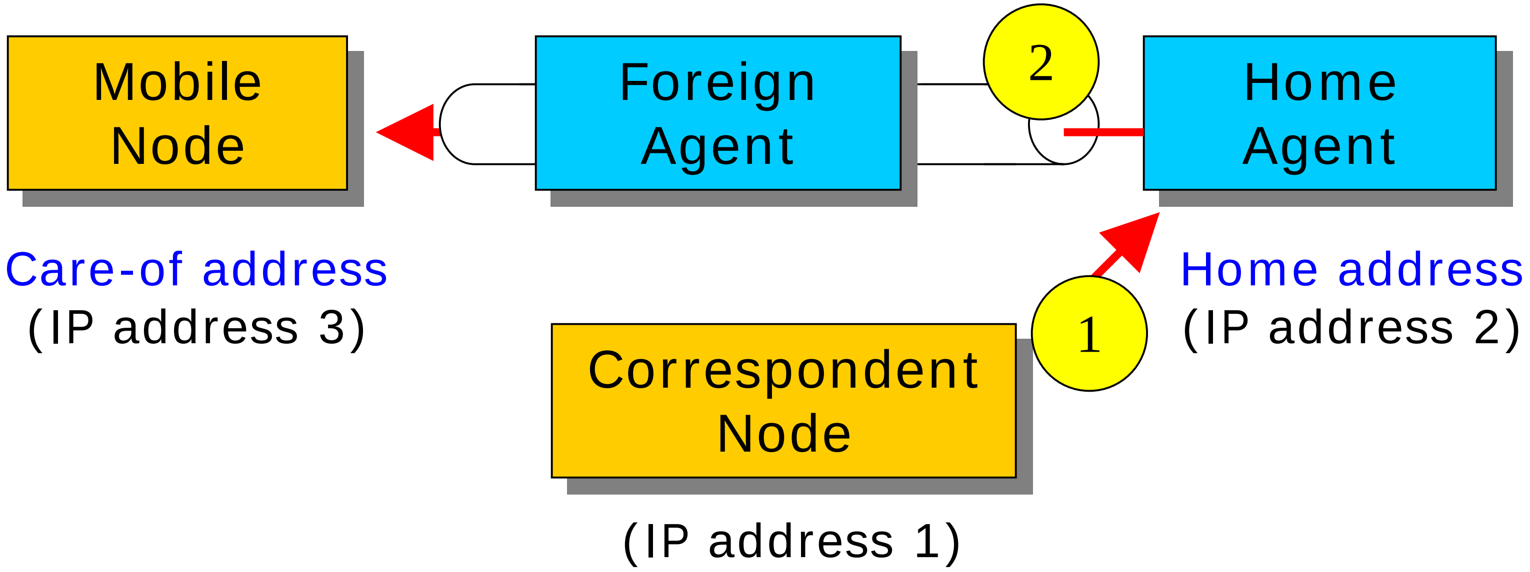
http://www.ietf.org/rfc/rfc2002.txt

Basic architecture:



Mobile IP (cont.)

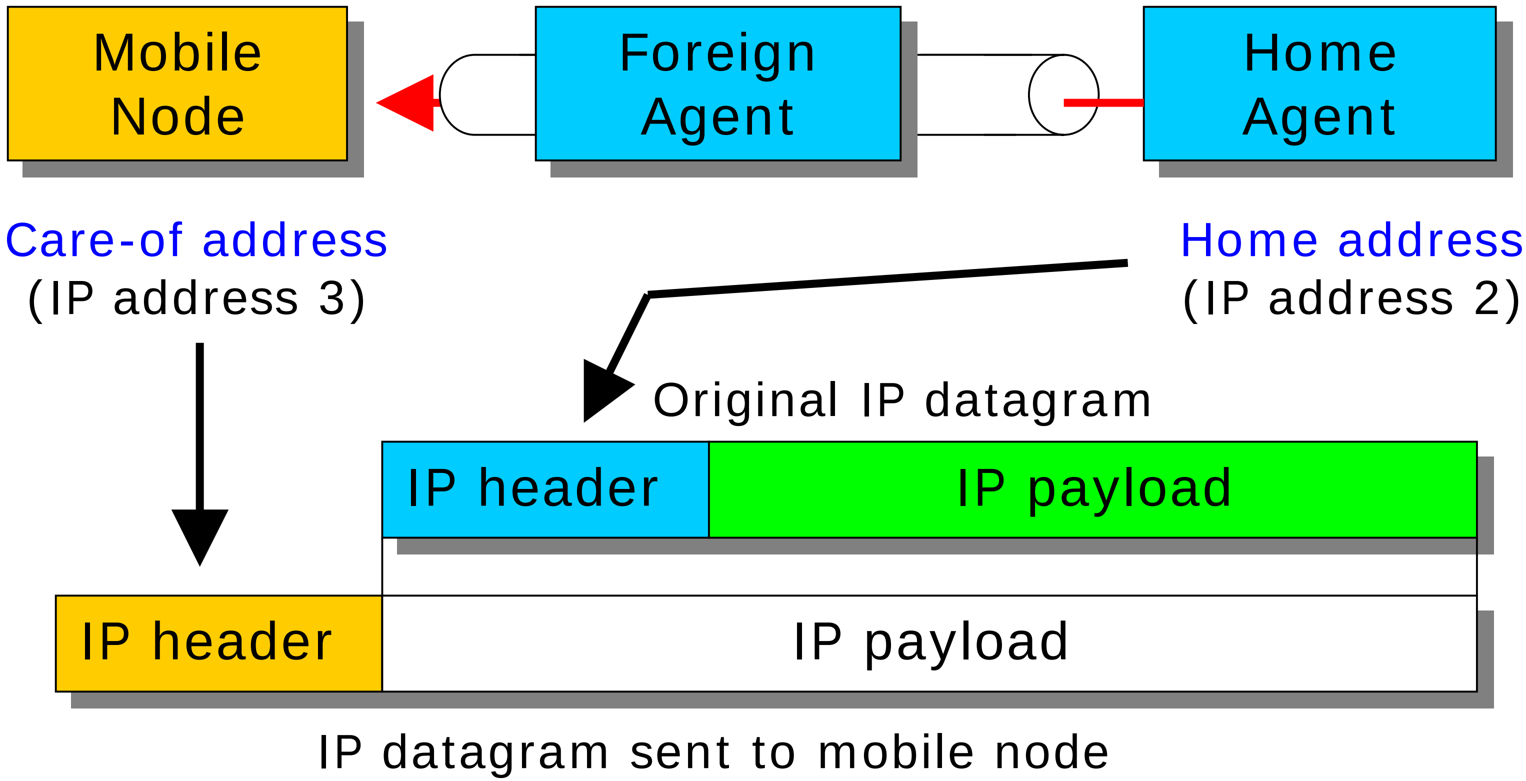
Mobile node -terminated IP transport:



1. Correspondent node sends IP packet to permanent home address (corresponding URL is known).
2. Home agent tunnels IP datagram to care-of address.

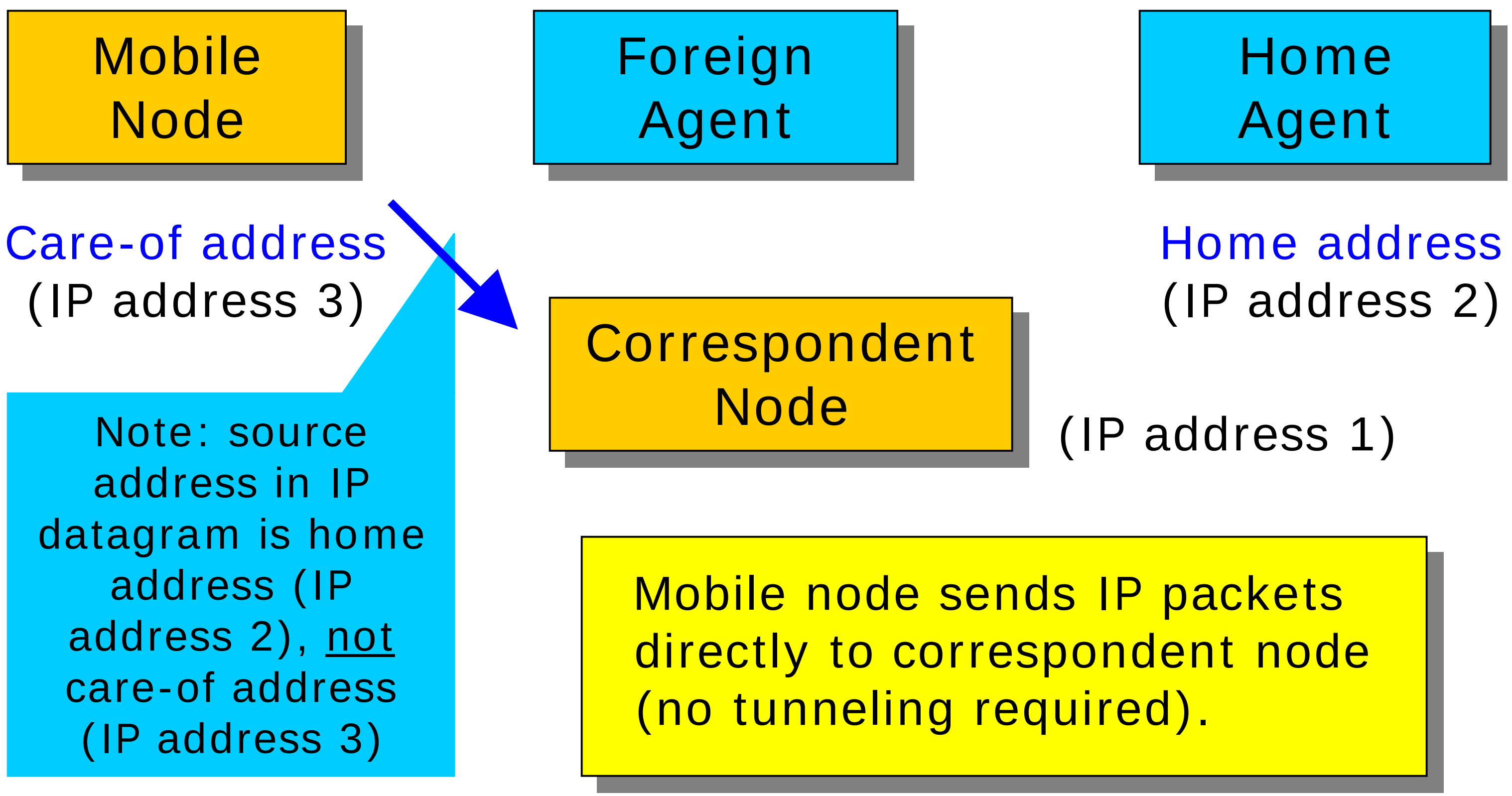
Mobile IP (cont.)

Tunneling in Mobile IP means encapsulation:



Mobile IP (cont.)

Mobile node -originated IP transport:



Mobile IP (cont.)

Mobility requires: (1) agent advertisements

1. “New” mobile node has no valid care-of address.
2. Foreign agents continuously broadcast (at ≈ 1 s intervals) lists of free care-of addresses within their area.
3. Mobile node selects a care-of address and informs the foreign agent.

Mobile IP (cont.)

Mobility requires: (2) registration

1. Mobile node informs home agent about new care-of address.
2. Home agent replies with ”OK”-message (or resolves the problem if situation is not OK).
3. From now on home agent can tunnel IP packets to mobile node (using care-of address).