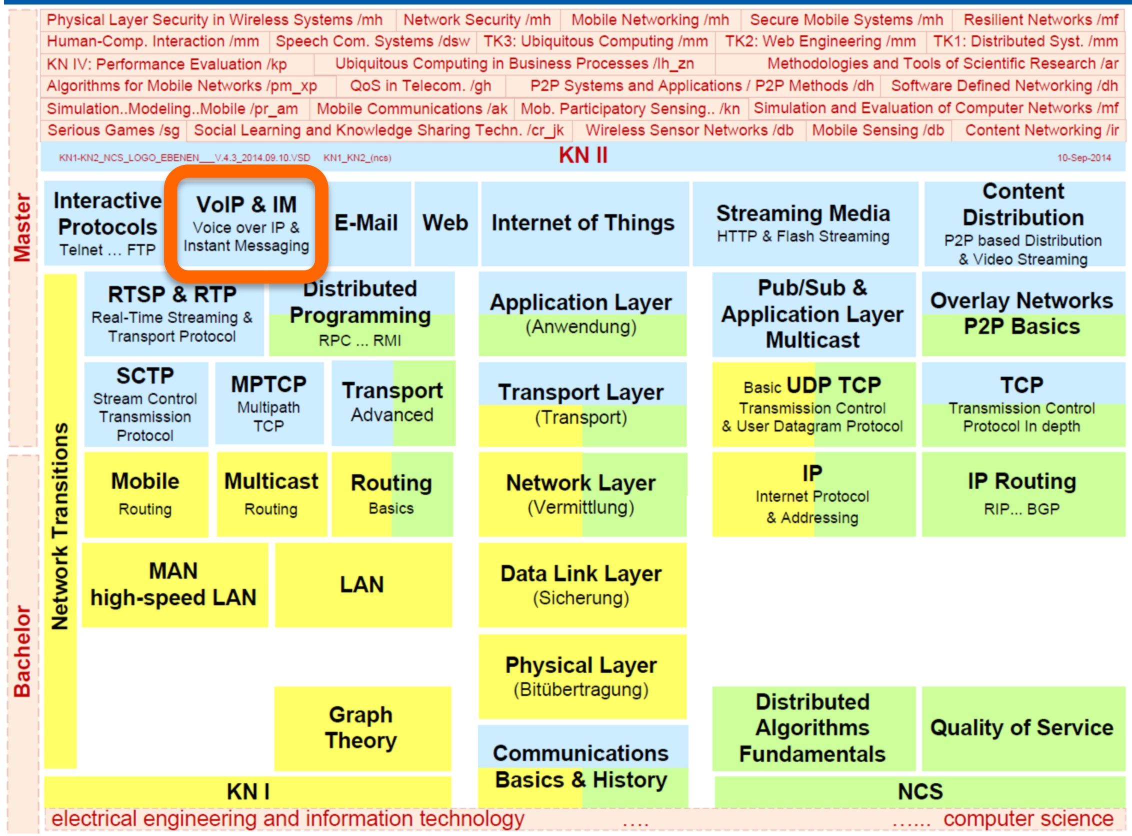
Communication Networks II

Real-time Interactive Multimedia
VoIP and Instant Messaging



Prof. Dr.-Ing. **Ralf Steinmetz**
KOM - Multimedia Communications Lab

Overview

Physical Layer Security in Wireless Systems /mh | Network Security /mh | Mobile Networking /mh | Secure Mobile Systems /mh | Resilient Networks /mf

Human-Comp. Interaction /mm | Speech Com. Systems /dsw | TK3: Ubiquitous Computing /mm | TK2: Web Engineering /mm | TK1: Distributed Syst. /mm

KN IV: Performance Evaluation /kp | Ubiquitous Computing in Business Processes /lh_zn | Methodologies and Tools of Scientific Research /ar

Algorithms for Mobile Networks /pm_xp | QoS in Telecom. /gh | P2P Systems and Applications / P2P Methods /dh | Software Defined Networking /dh

Simulation..Modeling..Mobile /pr_am | Mobile Communications /ak | Mob. Participatory Sensing.. /kn | Simulation and Evaluation of Computer Networks /mf

Serious Games /sg | Social Learning and Knowledge Sharing Techn. /cr_jk | Wireless Sensor Networks /db | Mobile Sensing /db | Content Networking /ir

KN1-KN2_NCS_LOGO_EBENEN___V.4.3_2014.09.10.VSD KN1_KN2_(ncs)

KN II

10-Sep-2014

Master

Bachelor

Network Transitions

Interactive Protocols Telnet … FTP

VoIP & IM Voice over IP & Instant Messaging

E-Mail

Web

Internet of Things

Streaming Media HTTP & Flash Streaming

Content Distribution P2P based Distribution & Video Streaming

RTSP & RTP Real-Time Streaming & Transport Protocol

Distributed Programming RPC … RMI

Application Layer (Anwendung)

Pub/Sub & Application Layer Multicast

Overlay Networks P2P Basics

SCTP Stream Control Transmission Protocol

MPTCP Multipath TCP

Transport Advanced

Transport Layer (Transport)

Basic **UDP TCP** Transmission Control & User Datagram Protocol

TCP Transmission Control Protocol In depth

Mobile Routing

Multicast Routing

Routing Basics

Network Layer (Vermittlung)

IP Internet Protocol & Addressing

IP Routing RIP… BGP

MAN high-speed LAN

LAN

Data Link Layer (Sicherung)

Physical Layer (Bitübertragung)

Graph Theory

Communications Basics & History

Distributed Algorithms Fundamentals

Quality of Service

KN I

NCS

electrical engineering and information technology …. …… computer science

Asynchronous communication:

- FAX
- E-Mail
- Forum
- Twitter
- …

Here we focus on Synchronous communication:

- Chat
- Instant Messaging (IM)
- Voice over IP (VoIP)
- Videoconference
- …

Synchronous Communications - Common Terms

VoIP (Voice over IP) / IP telephony

- Voice communication (telephone call) over
  - a packet-switched network using
    - the Internet Protocol
    - instead public telephone network
- Part of Internet Telephone services
  - like SMS, fax, … part of public telephone services
- Software applications or integrated into hardware devices

Videoconference

- Distributed meeting
  - two or more locations and two or more participants
  - with simultaneous voice and video transmission
- Additional document or desktop sharing

Chat

- Real-time communication over the Internet
- Text-Chat, Voice-Chat, Video-Chat

Instant Messaging (IM)

- Synchronous text-based communication
- Discussed in RFC 2778
  - A Model for Presence and Instant Messaging

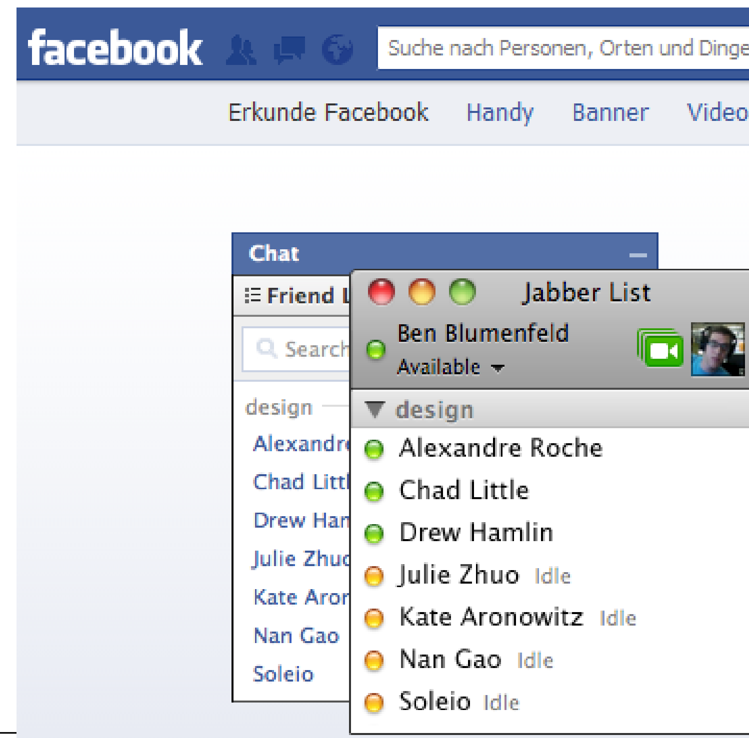
Common Applications and Tools

Tools for synchronous communication: IM, VoiP, Video Conference

- Whatsapp
- Viber
- iMessage (Apple)
- Facetime (Apple)
- Facebook chat
- Skype (Microsoft)
- Windows Live Messenger
  - successor of MSN Messenger
- Microsoft Lync
  - successor of Microsoft Office Communicator
  - Mainly used in corporate environments
- ICQ (since 1996 ICQ Instant Messenger)
- Yahoo! Messenger
- AOL Instant Messenger (AIM)
- …

Specialized Tools

- Voice Chats mainly used for gaming:
  - TeamSpeak, Ventrilo, Mumble, …
- Videoconference:
  - Microsoft NetMeeting, Adobe Connect, …



2 Messaging Protocols

History

1988 IRC - Internet Relay Chat
    - by Jarkko Oikarinen
- Inspired by BITNET Relay Chat
- RFC 1459

1996 H.323: Packet-based multimedia communication system (12/09)
- ITU-T recommendation

1996 SIP: Session Initiation Protocol
    - by Henning Schulzrinne and Mark Handley
- Standardized 1999
- RFC 3261-3265
- 2002 SIMPLE RFC 3428 and RFC 3856

1999 Jabber → XMPP - Extensible Messaging and Presence Protocol
    - by Jeremie Miller
- Standardized 2004 as Extensible Messaging and Presence Protocol (XMPP)
- RFC 6120-6122 and 3922-3923

1999 MSNP - Microsoft Notification Protocol
- Microsoft Instant Messenger Protocol (Mobile Status Notification Protocol)
- First used with MSN Messenger

Internet Relay Chat Protocol

- RFC 1459

Real-time internet chat – synchronous communication

Text messages / Chat Rooms (Channels) / Private Chat rooms

Mainly using TCP/IP and Client-Server

- Many client implementations,
  - e.g. mIRC, Trillian, Xchat, androIRC, Unreal Tournament build-in IRC
- Thousands of IRC networks worldwide,
  - e.g. EFnet, IRCnet, QuakeNet, Undernet

Security depending on server

No voice or video

Some properties of H.323 Packet-based multimedia communication system

- Current version v7(12/09)
- ITU-T recommendation
- Industry driven

- Initially defined for LANs (1996)
  - Visual telephone system and equipment for local area networks which provide a non-guaranteed quality of service
- Revised since version 2 for packet-switched networks in general

- Protocol properties
  - Stateful
  - Binary (ASN.1)

H.323 – Set of … and comprises further Recommendations

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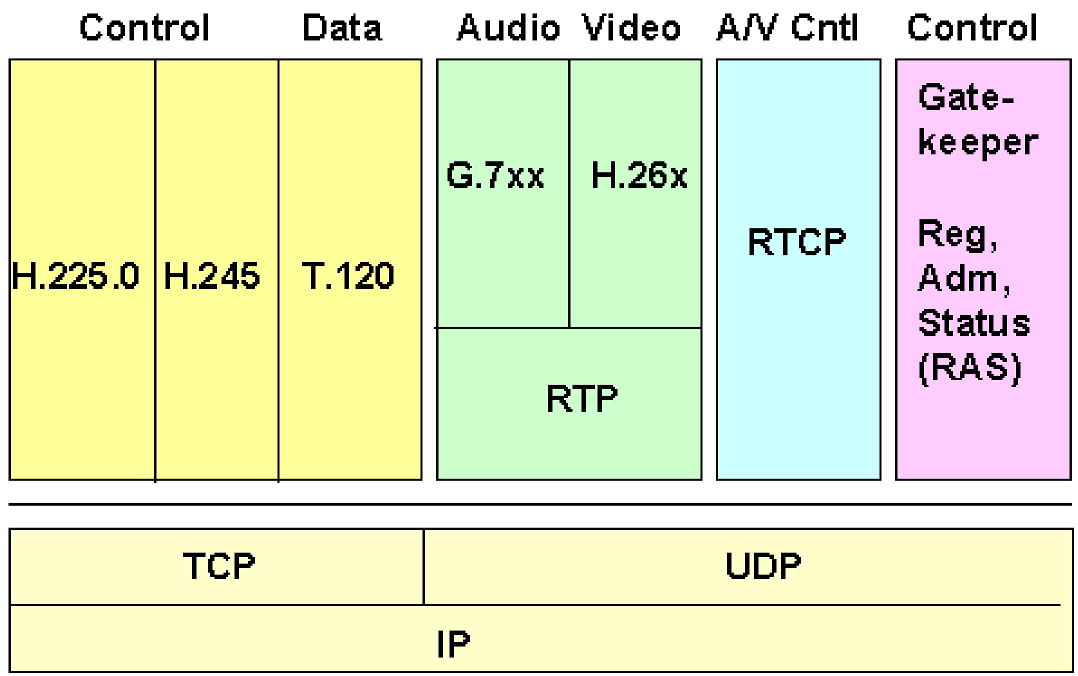
H.225.0: Call signaling protocols and media stream packet building for packet-based multimedia communication systems (2009) and Q.931

- Registration, authentication, status (RAS)
- Call control



H.245: Control protocol for multimedia communication (2009)

- Logical channel capabilities negotiation
- Dynamic port negotiation (problem with firewalls)

e.g. H.235.0: H.323 Security: Framework for security in H-series (H.323 and other H.245-based) multimedia systems (2005)

G.711: Pulse code modulation (PCM) of voice frequencies (1988)

Source of picture: http://users.skynet.be/spouseele/H323/NetMeeting_and_H323_Gatekeeper_as_HelpDesk_tool.htm

2.3 SIP - Session Initiation Protocol

Session Initiation Protocol (SIP)

- RFC 3261

Application Layer control protocol

- to establish, modify, and terminate multimedia sessions (conferences),
- e.g. Internet telephony calls

SIP for Instant Messaging and Presence Leveraging Extensions (SIMPLE)

- Addition to SIP for
  - Instant Messaging (RFC 3428) and
  - presence (RFC 3856)

SIP - 5 facets of establishing and terminating multimedia communications:

- User location:
  - determination of the end system to be used for communication
- User availability:
  - determination of the willingness of the called party to engage in communications
- User capabilities:
  - determination of the media and media parameters to be used
- Session setup:
  - "ringing", establishment of session parameters at both called and calling party
- Session management:
  - including transfer and termination of sessions, modifying session parameters, and invoking services

Basics

- XMPP Extension Protocols by XMPP Standards Foundation
- RFC 6120 (core), RFC 6121 (IM and presence) and RFCs 6122, 3922, 3923
- Open XML protocol for messaging, presence and request-response services
- Open-source, further improved

Characteristics

- Client-Server Architecture
- Unique Identifier: Jabber ID *username@host*
- TCP/IP
- Transport
  - gateways to other messaging protocols, SMS or e-mail
- Security:
  - Simple Authentication and Security Layer SASL and Transport Layer Security TLS
- Jingle: Extension for P2P signaling as base for voice communication / video conferencing (draft standard XEP-0166 and others)

2.5 MSNP - Microsoft Notification Protocol

Also known as
the Mobile Status Notification Protocol
Microsoft IM-Protocol

- Used by .NET Messenger Service and its clients
  - E.g. Windows Live Messenger

Client-Server

MSN Messenger protocol consists of
a series of commands sent between the client and the server

- Notification server (NS) providing presence service
- Switchboard server (SB) providing instant messaging service
  - Acting as a proxy between clients (no directly connected conversations)
  - A switchboard session is not restricted in the number of participants

Session Initiation Protocol (SIP)

- Signaling-approach of IETF

Basic functions:

- Location of an end point
- Signal of a desire to communicate
- Negotiation of session parameters to establish the session
- Teardown of the session once established

Characteristics:

- Text-based signaling-message
- E-mail like addressing
  - sip:[user]@[domain]
- “fast in the core and smart at the edges”

```
e.g.
REGISTER sip:domain.com
 SIP/2.0
Via: SIP/2.0/UDP
 193.64.210.89
From: sip:bob@domain.com
To: sip:bob@domain.com
Expires: 3600
```

SIP Properties

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Advantages:

- Good interworking with other (orthogonal) internet protocols
  - DNS (Domain Name System)
  - STUN (Simple Traversal of UDP over NATs)
  - ENUM (E.164 Number and DNS)

Service provided by SIP servers:

- SIP registrar server
- SIP proxy server

SIP long-term vision:

- All telephone calls, video conference calls take place over Internet
- People are identified by names or e-mail addresses, rather than by phone numbers
- You can reach callee, no matter where
  - callee roams
  - what IP device callee is currently using

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Signaling protocol tasks

1. Pre-call
- User addressing (URL, number)
- Authentication / registration (security, billing)
- Locating (dynamic IP-address mapping on user ID)
- Heartbeat (client online)

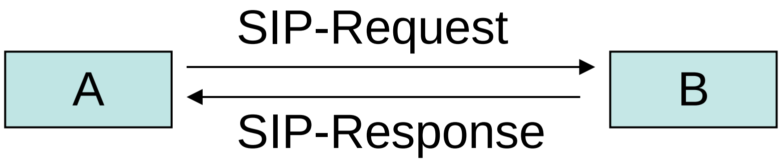
2. Call
- Invitation
- Parameter negotiation (media codec…)
- Management (redirect, forward, cancel,…)
- State (pick up, ongoing,…)

3. Post-call
- Release and reset of resources

Signaling sequence

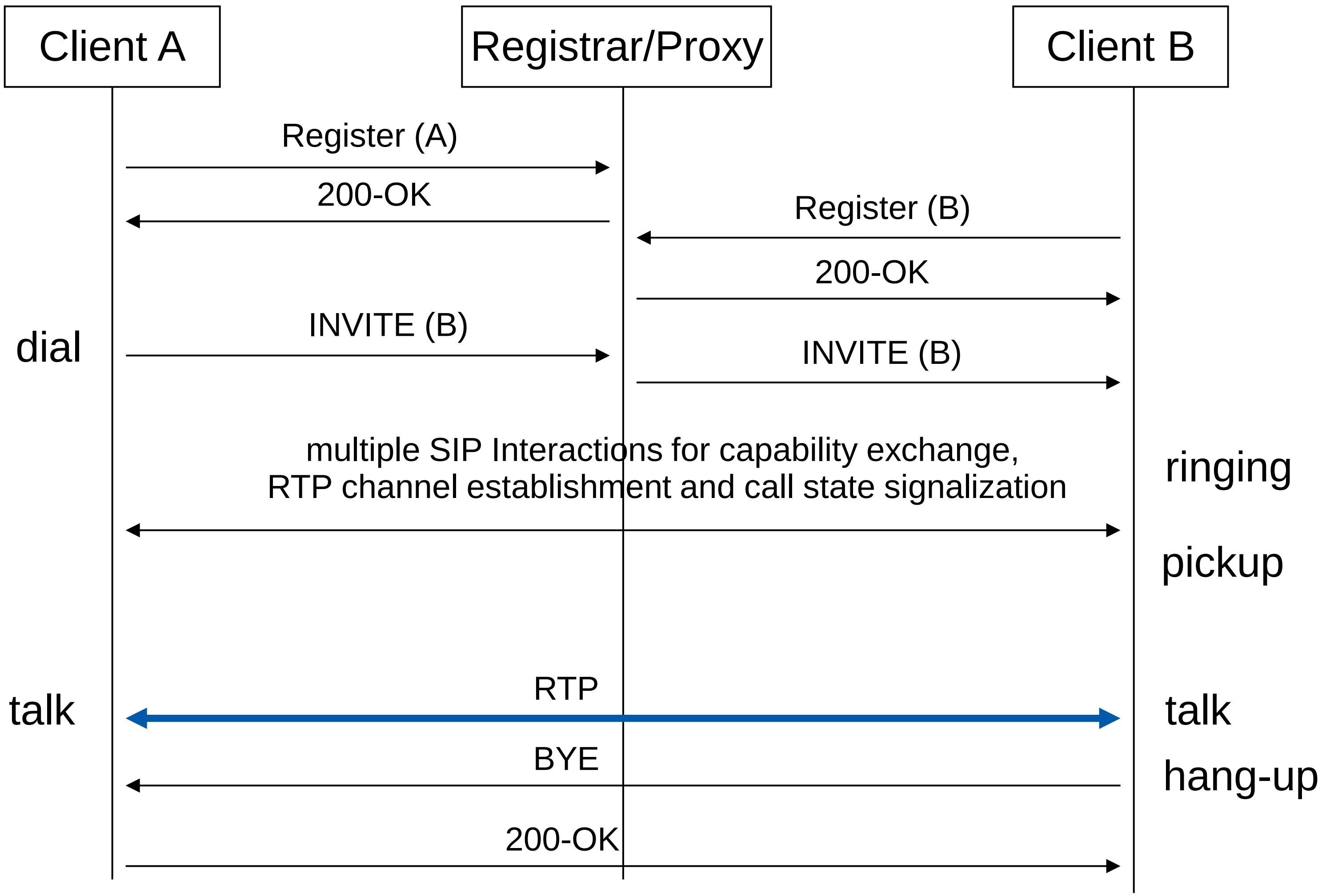
- similar to HTTP: request triggers response

2 message types:



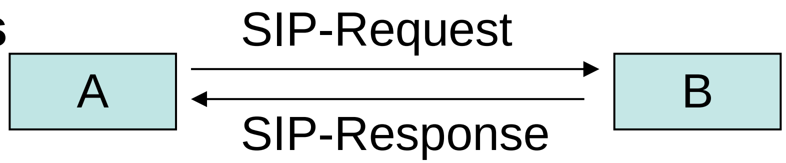
- SIP Requests
  - specific methods
- SIP Response
  - response codes from HTTP
  - additional explanatory text
  - extended with new numeric result codes

SIP – Signaling Procedure



3.2 SIP - Messages

SIP Requests



REGISTER
- binds id to the current IP
- Authentication

INVITE
- usually begins a session
- body contains session description (SDP)

OPTIONS
- returns client capability

ACK
- only used in session initiation
- end of 3-way handshake

CANCEL
- terminates an request

BYE
- terminates an open session

SIP Responses

1xx : Informational
- 100 Trying
- 180 Ringing

2xx : Success
- 200 OK

3xx : Redirect
- 302 Moved temporarily

4xx : Client error
- 400 Bad Request

5xx : Server error

6xx : Global failure

SIP – Message - Schema

First line

- method (SIP Request)

Header

- similar to E-Mail header
- request specific information
- mainly
  - URL: to / from / via
  - Call-ID (unique ID for call identification)
  - Body description (“Content-…”)

Body

- Session Description Protocol (SDP) format
- codec information
  - provided codecs, sampling rates
  - addresses and ports for media-data-receive
  - media destination, IP address and port number
- session name and purpose

SIP – Message - Example

First Line	**INVITE sip:00491266666@130.83.139.45:22400 SIP/2.0**
Header	**Via: SIP/2.0/UDP 130.83.139.206** **To: <sip:00491266666@130.83.139.45>** **From: <sip:00491266766@130.83.139.45>;tag=1c16296** **Call-ID: call-1036074256-8@130.83.139.206** **CSeq: 1 INVITE** **Contact: <sip:00491266766@130.83.139.206>** **Content-Type: application/sdp** **Content-Length: 310v=0**
Body	**o=Pingtel 5 5 IN IP4 130.83.139.206** **s=phone-call** **t=0 0** **m=audio 8766 RTP/AVP 96 97 0 8 18 98** **a=rtpmap:96 eg711u/8000/1** **a=rtpmap:97 eg711a/8000/1** **a=rtpmap:0 pcmu/8000/1** **a=rtpmap:8 pcma/8000/1** **a=rtpmap:18 g729/8000/1** **a=rtpmap:98 telephone-event/8000/1**

3.3 Setting up a Call to known IP Address

Alice's SIP invite message indicates

- her port number,
- IP address,
- encoding - she prefers to receive (PCM µlaw)

Bob's 200 OK message indicates

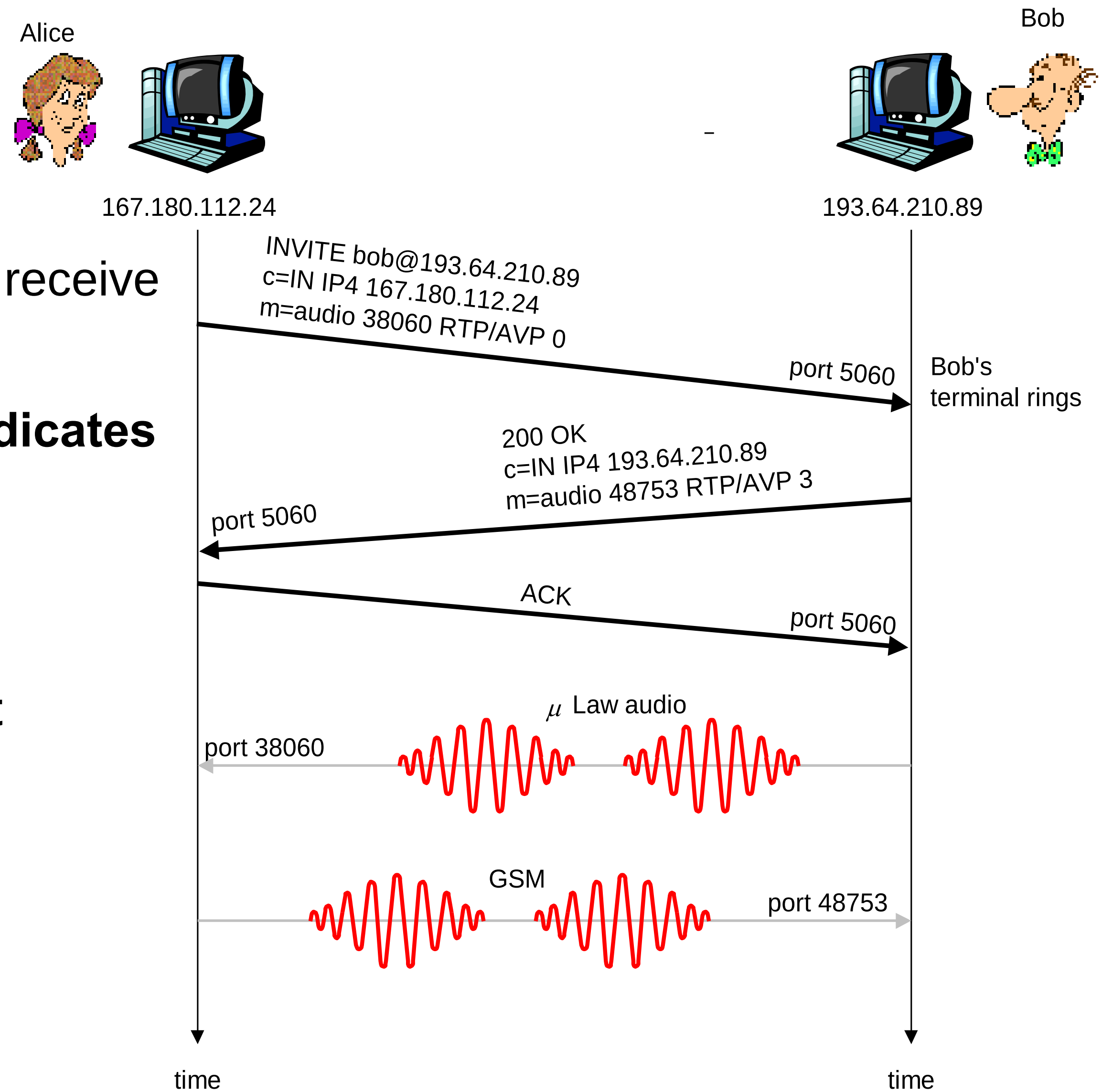
- his port number,
- IP address,
- preferred encoding

SIP messages can be sent

- over TCP or UDP;
- here sent over RTP/UDP

Default SIP port number

- is 5060



Setting up a Call to known IP Address

Codec negotiation:

- Suppose Bob doesn't have PCM µlaw encoder.
- Bob
  - will instead reply with 606 Not Acceptable Reply,
- Alice
  - listing his encoders can then
    send new INVITE message, advertising different encoder

Rejecting a call

- Bob can reject with replies
  - "busy," "gone," "payment required," "forbidden"

Media exchange

- Media can be sent over RTP or some other protocol

SIP Registrar

When Bob starts SIP client,
client sends SIP REGISTER message to Bob's registrar server

- (similar function needed by Instant Messaging)

Register Message:

```
REGISTER sip:domain.com SIP/2.0
Via: SIP/2.0/UDP 193.64.210.89
From: sip:bob@domain.com
To: sip:bob@domain.com
Expires: 3600
```

SIP Proxy

Alice sends invite message to her proxy server

- Contains address sip:bob@domain.com

Proxy responsible for routing SIP messages to callee

- Possibly through multiple proxies

Callee sends response back through the same set of proxies

Proxy returns SIP response message to Alice

- Contains Bob's IP address

Proxy analogous to local DNS server

3.4 Setting up a Call to unknown IP Address

Caller wants to call callee,
but only has callee's name or e-mail address

- Need to get IP address of callee's current host:
  - User moves around
  - DHCP protocol
  - User has different IP devices (PC, PDA, car device)

- Result can be based on:
  - Time of day (work, home)
  - Caller (don't want boss to call you at home)
  - Status of callee
    - (calls sent to voicemail when callee is already talking to someone)

i.e. Name Translation and User Location issue

Example of Setting up a Call to unknown IP Address

Alice doesn't know Bob's IP address

- Intermediate SIP servers needed
- Alice sends, receives SIP messages using SIP default port 5060
- Alice specifies in header that SIP client sends, receives SIP messages over UDP

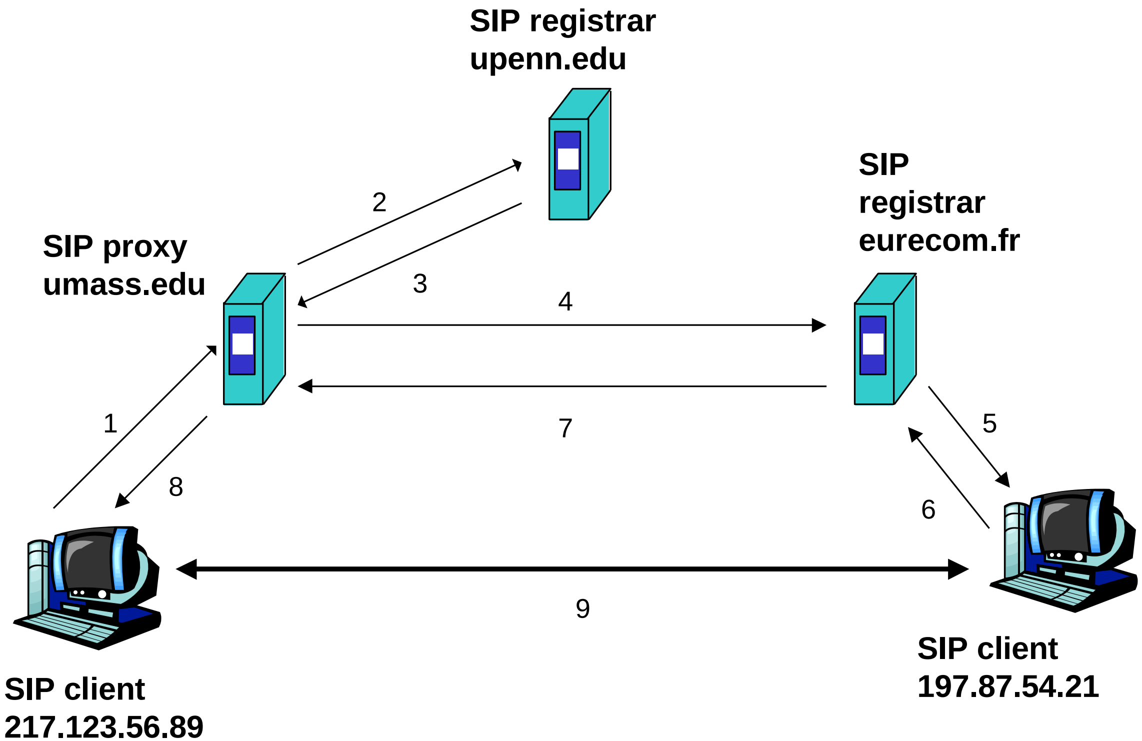
```
                     atlanta.com  . . . biloxi.com
                 .      proxy              proxy     .
               .                                       .
       Alice's  . . . . . . . . . . . . . . . . . . . .  Bob's
      softphone                                        SIP Phone
         |                |                |                |
         |    INVITE F1   |                |                |
         |--------------->|    INVITE F2   |                |
         |  100 Trying F3 |--------------->|    INVITE F4   |
         |<---------------|  100 Trying F5 |--------------->|
         |                |<-------------- | 180 Ringing F6 |
         |                | 180 Ringing F7 |<---------------|
         | 180 Ringing F8 |<---------------|     200 OK F9  |
         |<---------------|    200 OK F10  |<---------------|
         |    200 OK F11  |<---------------|                |
         |<---------------|                |                |
         |                       ACK F12                    |
         |------------------------------------------------->|
         |                   Media Session                  |
         |<================================================>|
         |                       BYE F13                    |
         |<-------------------------------------------------|
         |                     200 OK F14                   |
         |------------------------------------------------->|
         |                                                  |
```

Figure 1: SIP session setup example with SIP trapezoid

Example of Setting up a Call to unknown IP Address

Caller jim@umass.edu places a call to keith@upenn.edu

- (1) Jim sends INVITE message to umass SIP proxy
- (2) Proxy forwards request to upenn registrar server
- (3) upenn server returns redirect response, indicating that it should try keith@eurecom.fr
- (4) umass proxy sends INVITE to eurecom registrar



- (5) eurecom registrar forwards INVITE to 197.87.54.21, which is running keith's SIP client
- (6-8) SIP response sent back
- (9) media sent directly between clients
- Note: also a SIP ack message, which is not shown

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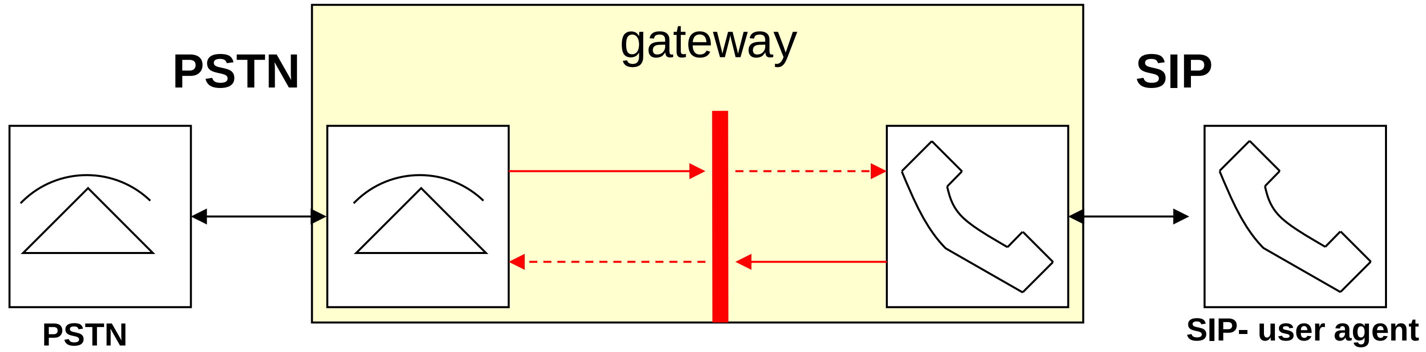
Gateways

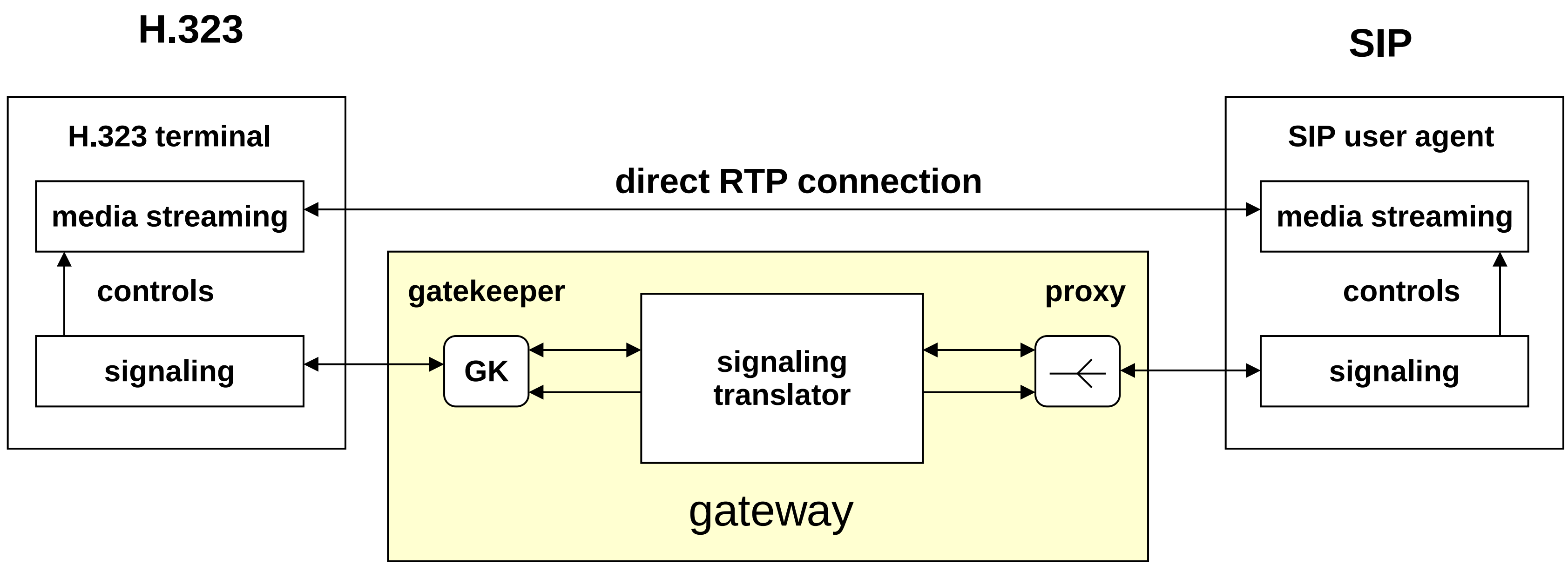
- Inter-connection to other communication systems
  - H.323, SIP, IAX2, PSTN, others…
- To extend the range of "reachable" users

Back-to-Back User Agent (B2BUA)

- Simulates an endpoint to each direction
- Media/signaling message transferred through gateway
- Example SIP-PSTN gateway:



Protocol-Interworking



Gateway with direct RTP connection

- Direct exchange of RTP media-data
  - Less data transfer through gateway
- Mapping of signaling-primitives
  - E.g. between
    - H.323 and SIP

3.6 Comparison SIP and H.323

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H.323

- Another signaling protocol for real-time, interactive communication

H.323 is a complete, vertically integrated suite of protocols for multimedia conferencing including

- Signaling, registration, admission control, transport, codecs

SIP is a single component.

- Works with RTP, but does not mandate it
- Can be combined with other protocols, services

H.323 comes from the ITU (telephony)

- H.323 has “telephony flavor”

SIP comes from IETF

- Borrows much of its concepts from HTTP
- SIP has “web flavor”

SIP uses the KISS principle: Keep it simple, stupid!

- Less complex approach compared with H.323
- With a good extensibility

Instant Messaging (IMS)

- Tries to feature real-time direct conversation
  - Like & Comparable
    - to SMS / pager services
    - To person to person chat
  - Using text messages
  - Different from email
- "instant" content sending and delivery
  - Immediate transport if possible
  - By "leaving" it for the receiver otherwise
- Usually coupled with presence mechanisms

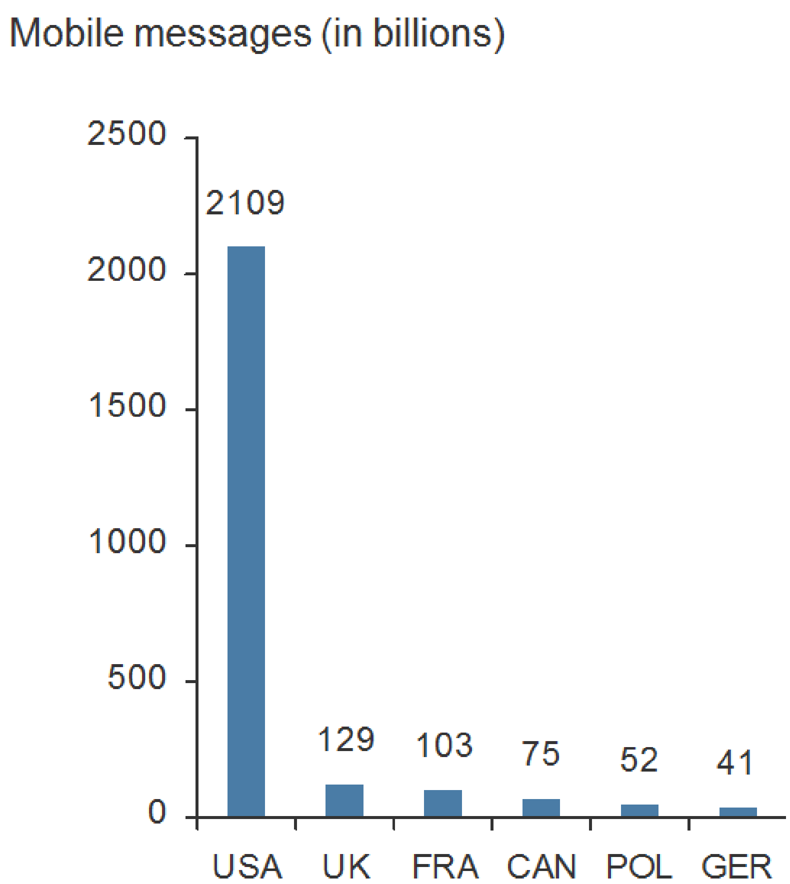
Frequently enhanced with additional services like

- Buddy lists
  - All users I want to communicate with
- Chat rooms (private/public)
- Notification of e.g. email
- News ticker
- Transfer of file, picture, video, location, …

Sources: www.pegasoft.cz/img_novinky/new_velka/icq.jpg
www.chitchat.org.uk

As part of the GSM – mobile telephone services

e.g. in 2010 per year



Source: Statista and Ofcom, International Communications Market Report 2011, page 262

Usage peak 2012 in Germany

- E.g. According to BITKOM, Nov. 2012
  - 700 per user per year
  - 58 billions of SMS in 2012
  - 3.1 billions EUR
    - More flat rates
    - More internet based alternatives

SMS – History

As part of the GSM – mobile telephone services

History

- Concept originates based on
  - Friedhelm Hillebrand (Deutschen Bundespost, today German Telekom)
- with contributions of
  - Bernard Ghillebaert (PTT, today France Télécom)
- 1984: well defined
- Feb.1985 incorporated into the GSM - Standard

„Merry Christmas"

- first SMS sent **3. Dez. 1992** in the UK, Vodaphone
- typed in at PC attached to mobile phone
  - Mobile terminal did not have any facility for characters to be typed in

SMS - Overall

Characteristics

- Usually 160 Characters
- At most 1120 bits

Enhancement

MMS – Mobile Multimedia Service

Enhancement

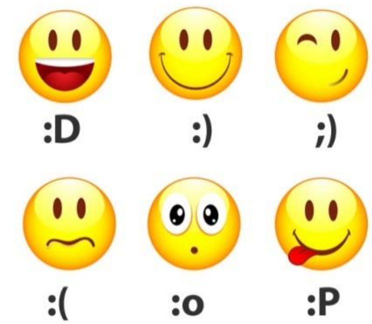
Future – Joyn

- Industry based initiative
- Based on Rich Communication Suite, Rich Communication Services

Social Impact

- Young generation
- politics

Smilies



Special abbreviations like

Words in full	Abbreviations or SMS language
As far as I remember	AFAIR
Thanks	THNX or THX
Today	2day
Before	B4
Have a nice day	HAND
See you	C U
At	@
Keep it simple, stupid	KISS

4.2 Joyn

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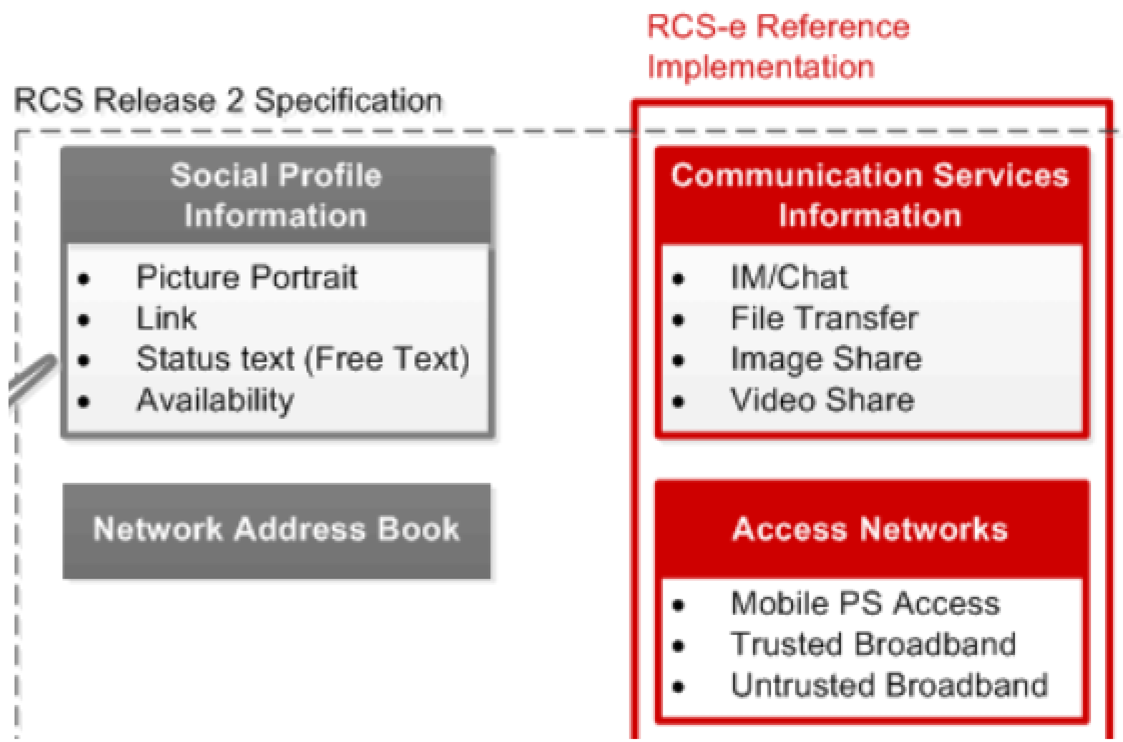
Overall

- Industry based initiative
- Based on Rich Communication Suite, Rich Communication Services
- Volume based tariff

Characteristics

- Enhanced Phonebook with
  - service capabilities
  - presence enhanced contacts information
- Enhanced Messaging
  - e.g. chat and messaging history
- Enriched Call
  - multimedia content sharing during a voice call



4.3 WhatsApp

Overall

- Large installed base of users
  - Jabber ID: [phone number]@s.whatsapp.net

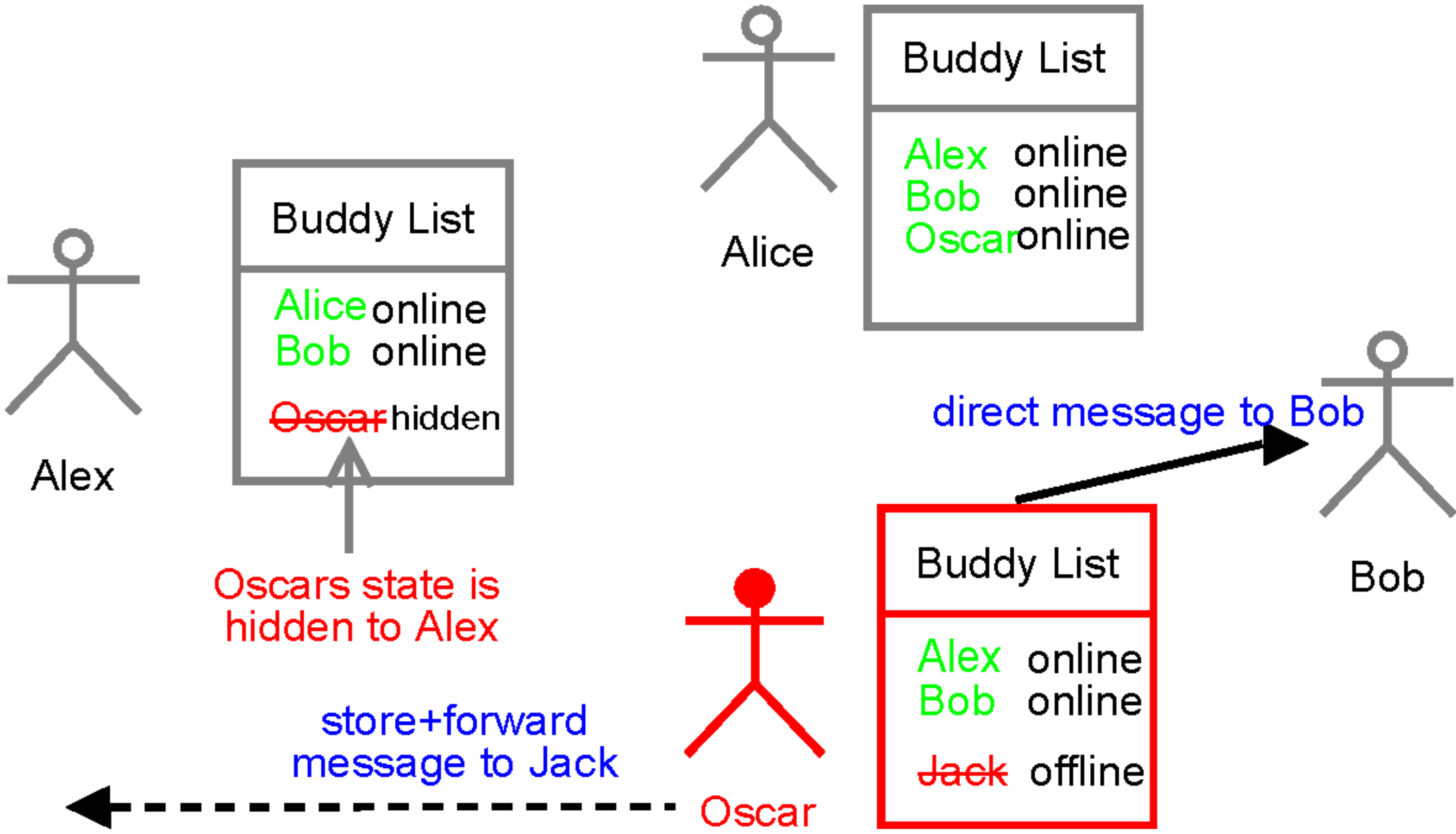
Characteristics

- uses customized version of
  XMPP - Extensible Messaging and Presence Protocol

- Upon installation it creates a user account
  - using one's phone number as username

- Some Privacy and Security concerns
  - Server knows all contacts from users
  - Identity theft possible

4.4 Instant Messaging - User View



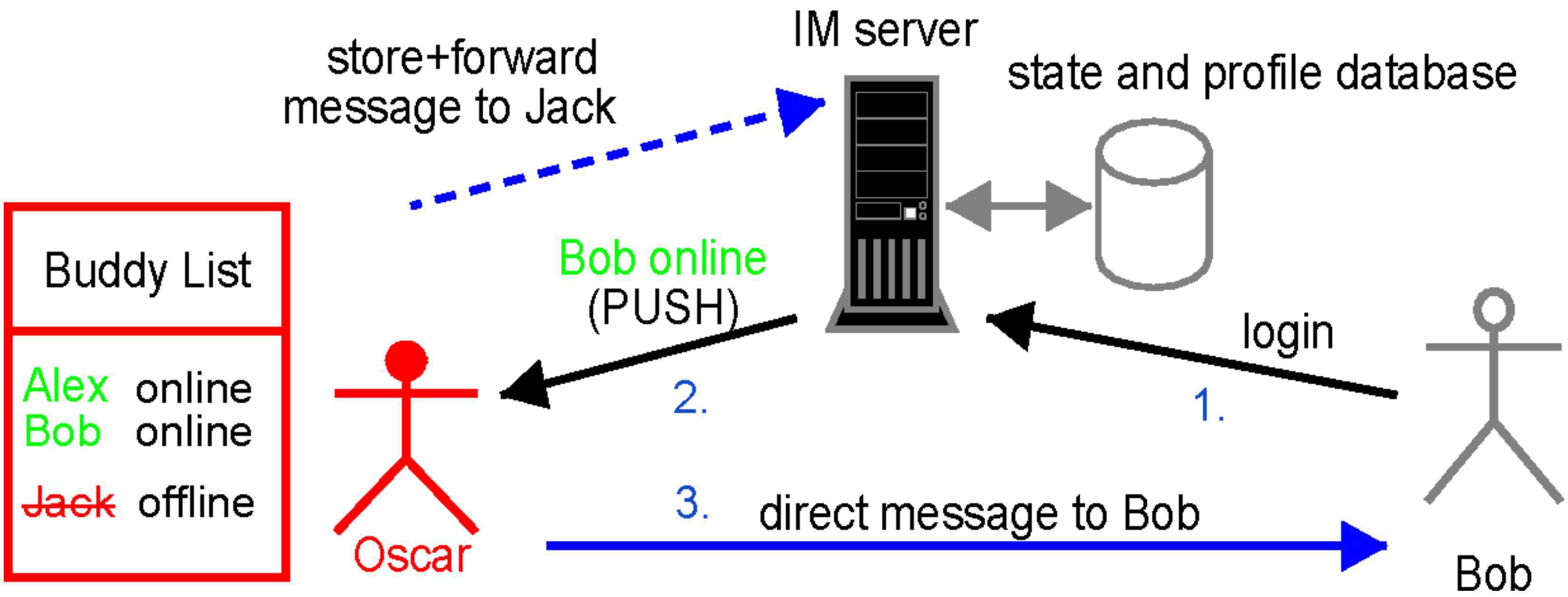
Let other people "see" my presence

- If I allow that

Let me see the presence of other people

Reaches available persons immediately or leaves them a message

4.5 Instant Messaging - System View



Message Transfer Alternatives

- Direct:
  - After learning partners point of presence from infrastructure
- Proxied:
  - Through infrastructure (makes coping with firewalls easier)
- Store and forward

Communication protocol

- Client / server model combined with server push mode
- Centralized databases allow to retrieve "profile / buddy list information"
  - From whatever client somebody is currently using (e.g. PC at work, home,..)

4.6 Instant Messaging - Enhancements

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Privacy and data security

- Encryption of data
- Ensure that data will not be monitored / sold
- Granting, managing and retracting permissions

Interoperable clients

- Increasing number of polyglot clients caused by
  - Competition, reverse engineering
  - Ongoing modifications, enhancements

Access from multiple devices

- PC, wireless devices, PDA, IP phones

Scalability

- Hundreds of
  - Millions of users
  - Billions of messages/notifications per day
- Possible speed limits for the propagation of states in distributed systems

Telephony integration

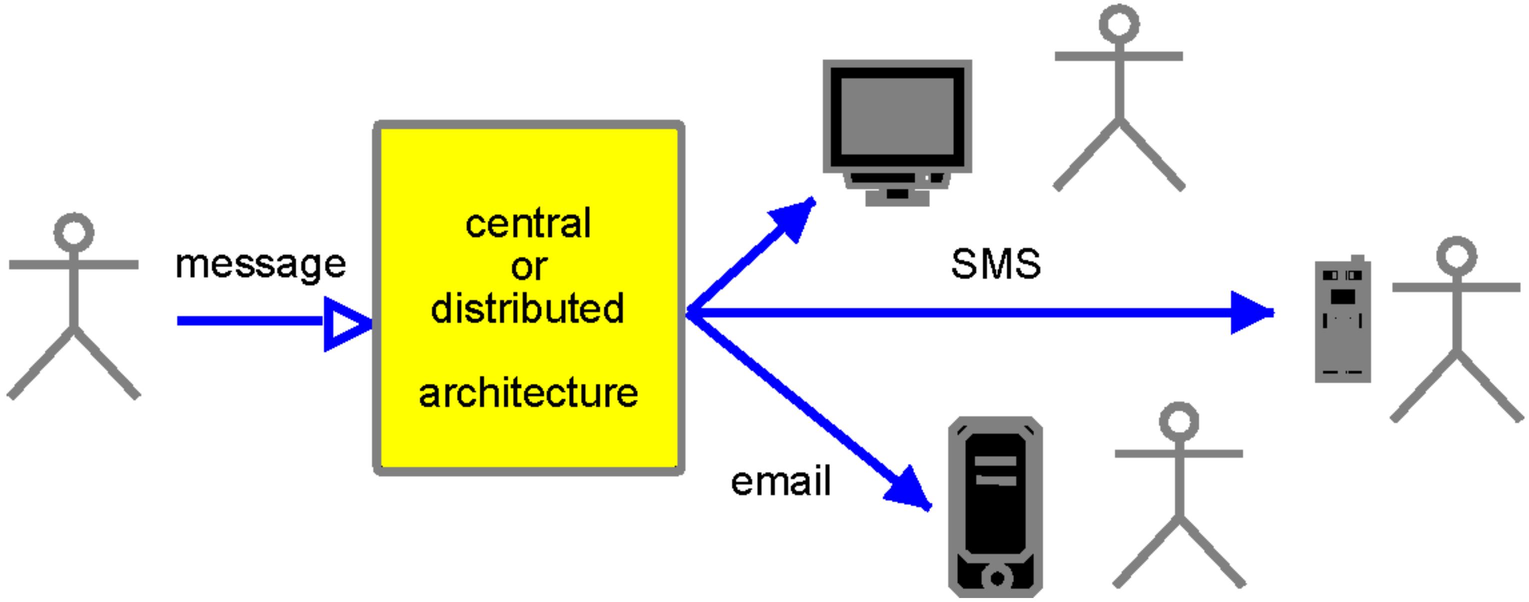
- Many services require voice integration industry mandates

Context awareness

Future Enhancements - Context Integration

New Approaches

- Include location and context awareness
  - Where is the user
    - at the moment
    - in near future
  - What is the user doing
  - Which are the available and most suitable communication devices and media



4.7 Instant Messaging with SIP

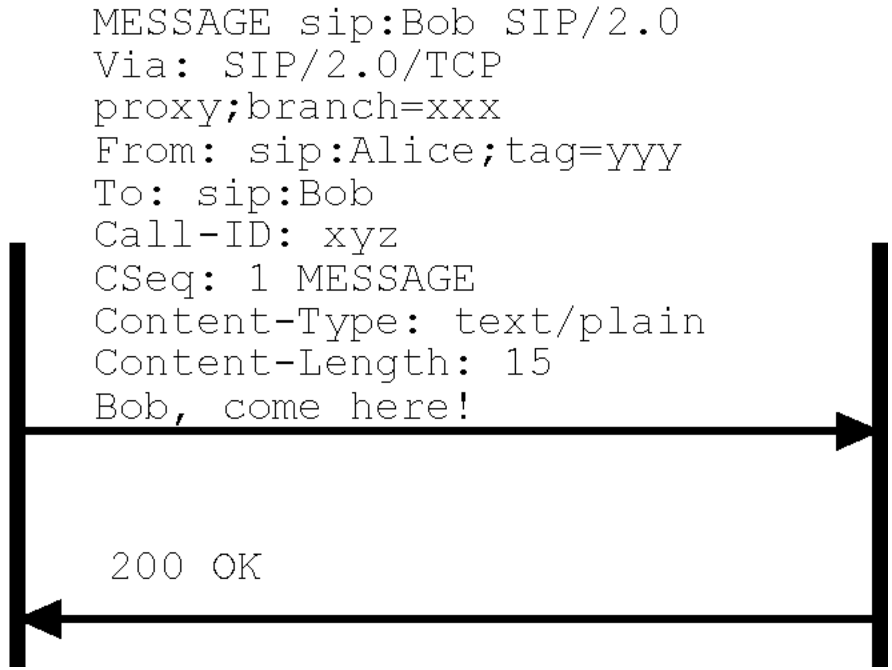
Use SIP infrastructure for Instant Messaging

- Provides signaling framework
- SIP
  - Servers route messages
  - Registrars provide means for addressing and locating users
  - Covers already some inherent security aspects

New aspects to SIP

- Instant Messaging
  - Problem:
    - possible congestion on SIP signaling path
      through large amount of SIP messages carrying user payload
  - Solution:
    - Peer-to-Peer technology
    - i.e. to define new MESSAGE method which is sent directly to the target and not through SIP server
- Presence
  - use SIP event concept
    - SUBSCRIBE / NOTIFY
  - new event type “presence”

Instant Messaging with SIP



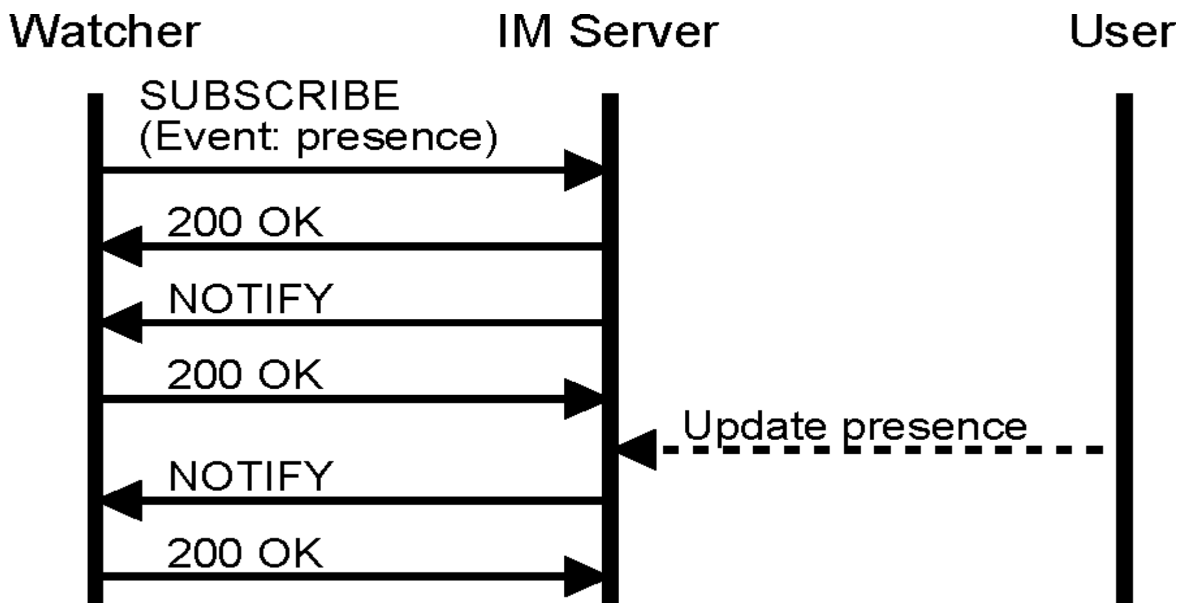
Payload transport

- Must support at least content type text/plain
- May support MIME type content including text/cpim (Common Presence and Instant Message Format)

SIP MESSAGE method for Instant Messaging

- Addressing uses SIP or IMS URLs
- 200 OK
  - Only indicates the message was accepted by the User Agent (UA)

Instant Messaging with SIP



SUBSCRIBE

- To subscribe to specific event source "presence"

NOTIFY

- PUSH message about status changes from the SIP server
- Presence information in message body with some non-SIP means
  - E.g. Content-Type: application/cpim-pidf+xm

Encoding of speech

- Codec
  - Transforming speech into binary data
- Defines:
  - Quality of encoding
  - Bandwidth
  - Robustness

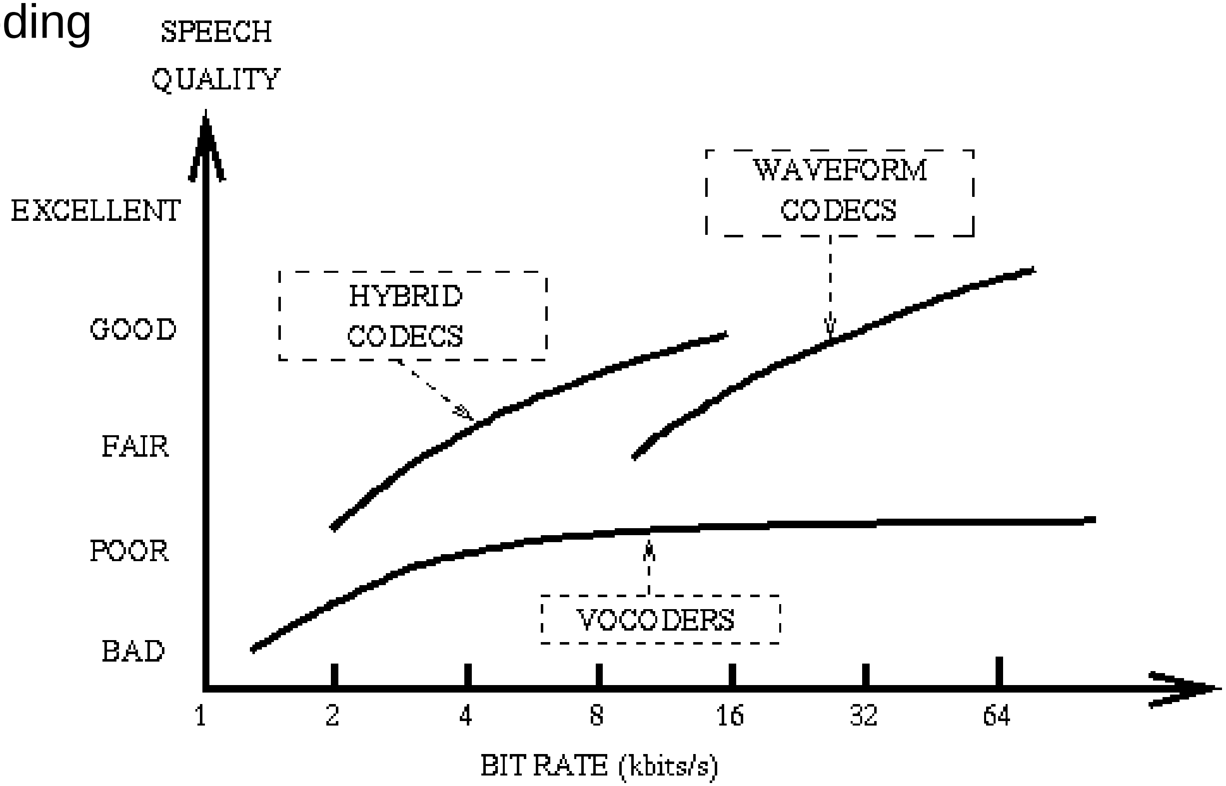
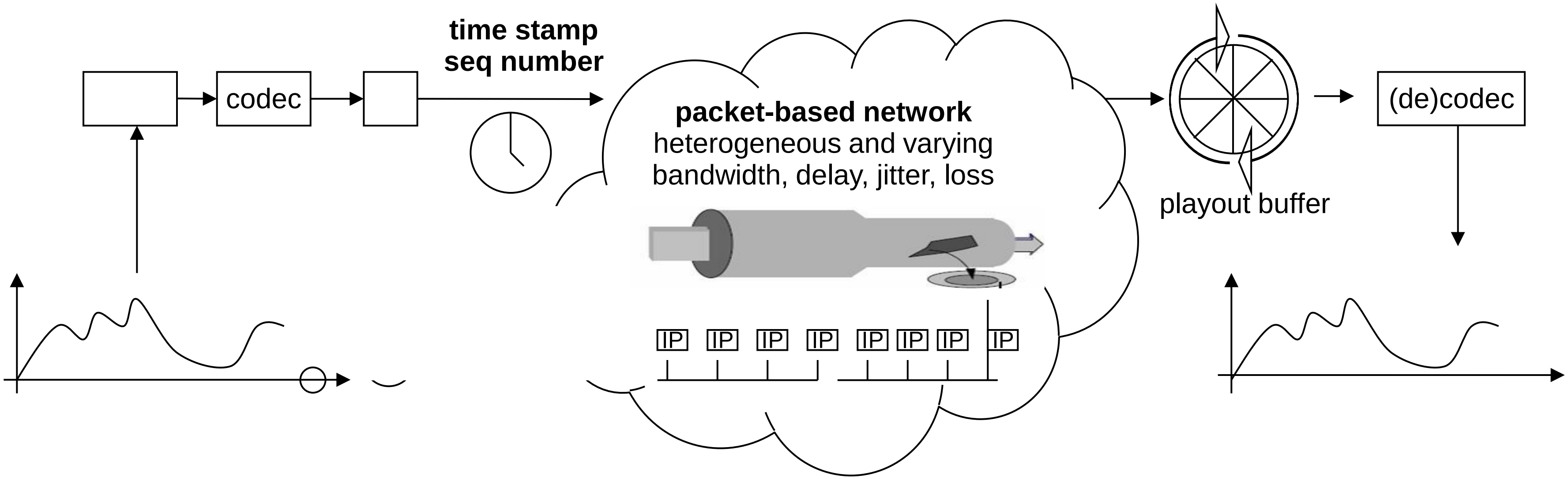


Figure 5 : Speech Quality Versus Bit Rate For Common Classes of Codecs

Image source: http://www-mobile.ecs.soton.ac.uk/speech_codecs/common_classes.html

Basics of Voice Data Transport and Voice Coding



Media-transport as general assignment:

- In many systems usage of
  - Real-Time Transport Protocol (RTP)
  - Real-Time Transport Control Protocol (RTCP)
- UDP-based transport
  - Qualified as non-network adaptive (like TCP)
  - Specification of media-endpoint-parameter
    - IP-addresses and ports
- Real-time requirements
  - Sensitivity to delay and jitter

Common Codec G.711

Recommendation ITU-T G.711 (1988), Pulse code modulation (PCM) of voice frequencies

- ITU (International Telecommunication Union)
- Waveform Codec, no Compression
- Used for ISDN, suited for VoIP
- Needs much bandwidth 64 kbit/s (plain) ~ 80 kbit/s (with headers)
- Frame size not fixed, normally 5ms – 20ms

Common Codec Speex

- Open Source Codec (Xiph.org)
- Hybrid Codec
- Flexible quality: sample rates 8kHz (narrowband), 16 kHz (wideband), 32kHz (ultra-wideband)
- Bandwidth 2,15 kbit/s – 44kbit/s
- Frame size 20ms

5.1 Speech Quality

Important network parameters:

- Delay
- Throughput
- Jitter
- Packet loss

Delay =

- Codec Delay +
- Serialization Delay +
- Queuing Delay +
- Propagation Delay

Throughput: depending on coding schemes and framing

- G.711 Codec (PCM) requires 64Kbps
- G.721 Codec (ADPCM) requires 32Kbps

Example of encoding / decoding delay for speex (frame size 20ms)

- Sample rate 8kHz -> delay = 30ms
- Sample rate 16kHz -> delay = 34ms

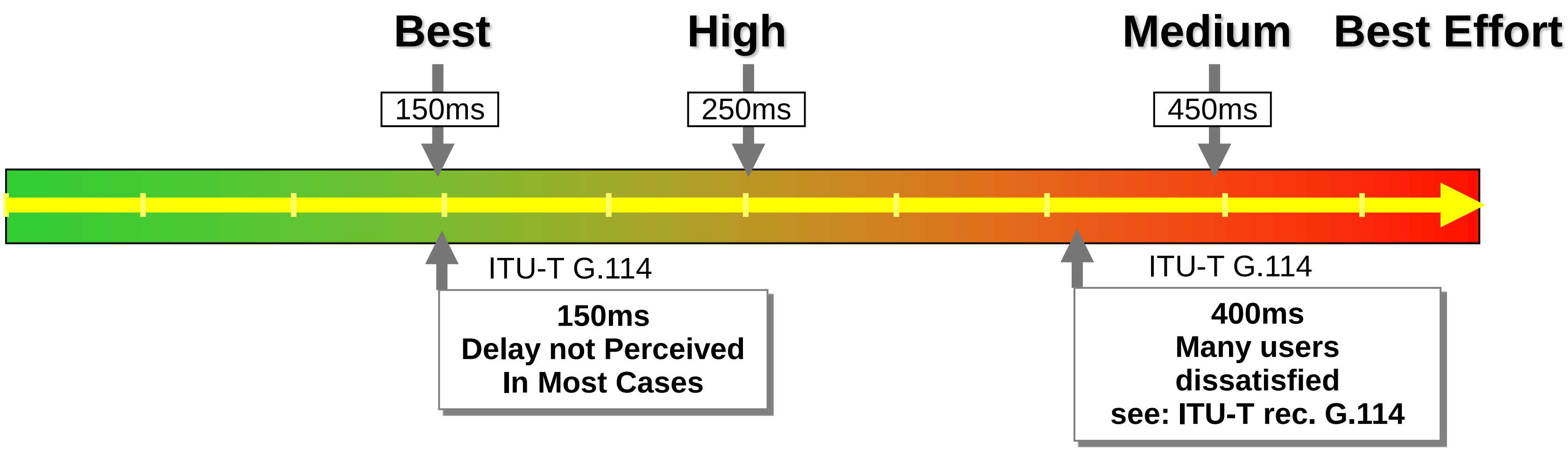
End-to-end delay requirements

< 150 msec – good

< 400 msec – ok

- Includes application-level and network delays

End-to-end Delay bounds:



MOS (Mean Opinion Score)

- ITU 1996
- Subjective measurement, many people needed
- Examples:
  - G.711 with 56-64 kbit/s -> MOS = 4,1 - 4,4
  - Speex with 2-44 kbit/s -> MOS = 2,3 – 4,0

MOS	Quality	Effort
5	Excellent	Complete relaxation possible; no effort required
4	Good	Attention necessary; no appreciable effort required
3	Fair	Moderate effort required
2	Poor	Considerable effort required
1	Bad	No meaning understood with any feasible effort

PESQ – PAMS - … Measurement of Speech Quality

PESQ (Perceptual evaluation of speech quality)

- ITU 2001
- Difference between source and destination signal is calculated
- Considers transmission losses (e.g. delay, packet loss)
- Can be “transferred” into MOS-values:

$$y = 0,999 + \frac{4,99 - 0,99}{1 + e^{-1,4945 * x + 4,6607}}$$

PAMS (Perceptual Analysis Measurement System)

- Like PESQ but different Algorithms
- Defines „listening quality“ and „listening effort“

E-Model

- ITU 2005
- Calculates R-factor for quality
  - 0 – 120 (more is better)
- R = Ro – Is – Id – Ie,eff + A

Für R < 0 :	$MOS = 1$
Für 0 < R < 100 :	$MOS = 1 + 0,035R + R(R-60)(100-R)^7 * 10^{-6}$
Für 100 < R :	$MOS = -4,5$

Tabelle 2.4: Berechnung des MOS-Wert mit Hilfe des R-Wertes [IT05]

Effect of delay

- Regarding the use of the E-model for speech applications
- Transmission Rating (R) versus delay
- Also shown are the speech quality categories
  - E.g. $90 \leq R < 100$ → Users very satisfied

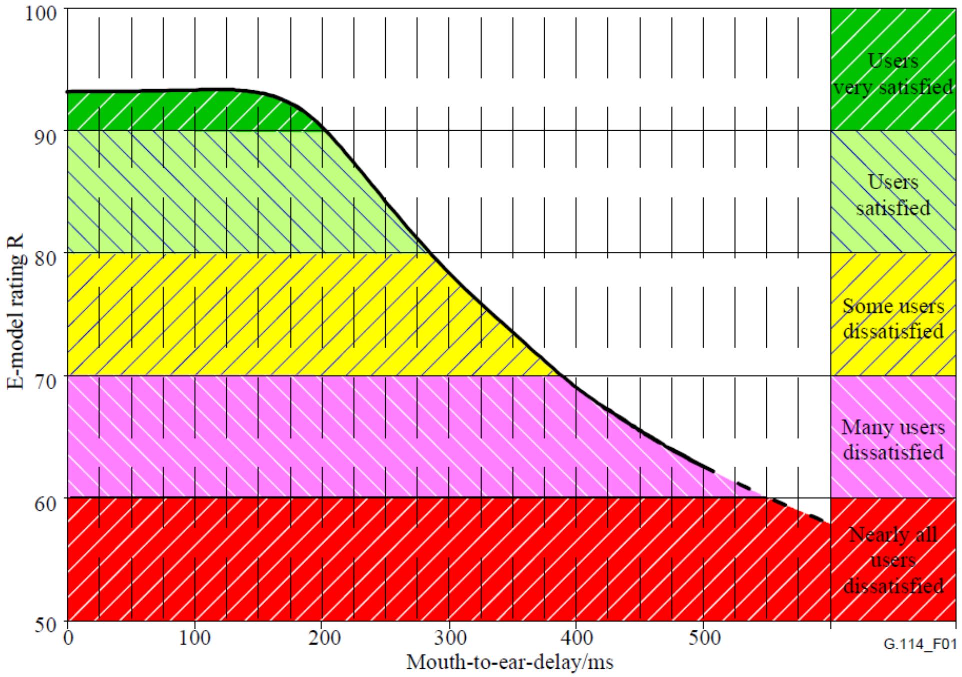


Figure 1/G.114 – Determination of the effects of absolute delay by the E-model

Measurement of Speech Quality

Consider all the way from speaker to listner including:

- Encoding / decoding
- Routing
- De-Jittering
- Playout

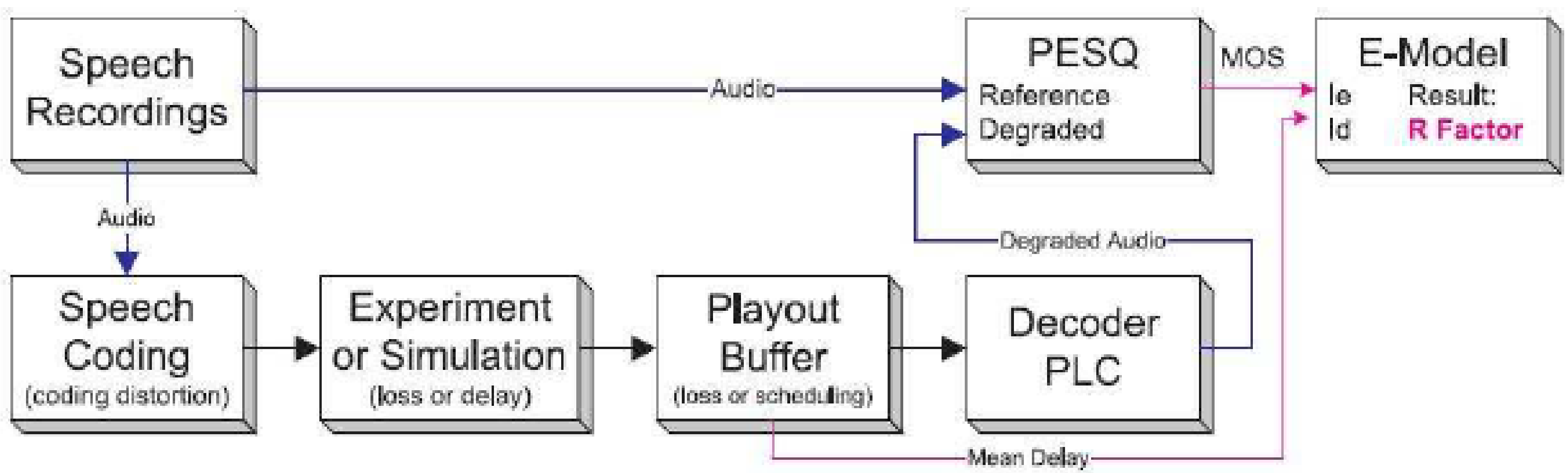


Abbildung 3.8: Aufbau des Qualitätsmodells von Hoehne [Hoe06]

Image source: C. Hoene. Internet Telephony over Wireless Links. PhD thesis, TU Berlin, 2006

VoIP / IP-Telephony / Internet Phone

- VoIP – “Voice Over IP Networks”
- Transmission of *signaling* and *media data* (voice streams)

Signaling approaches for VoIP:

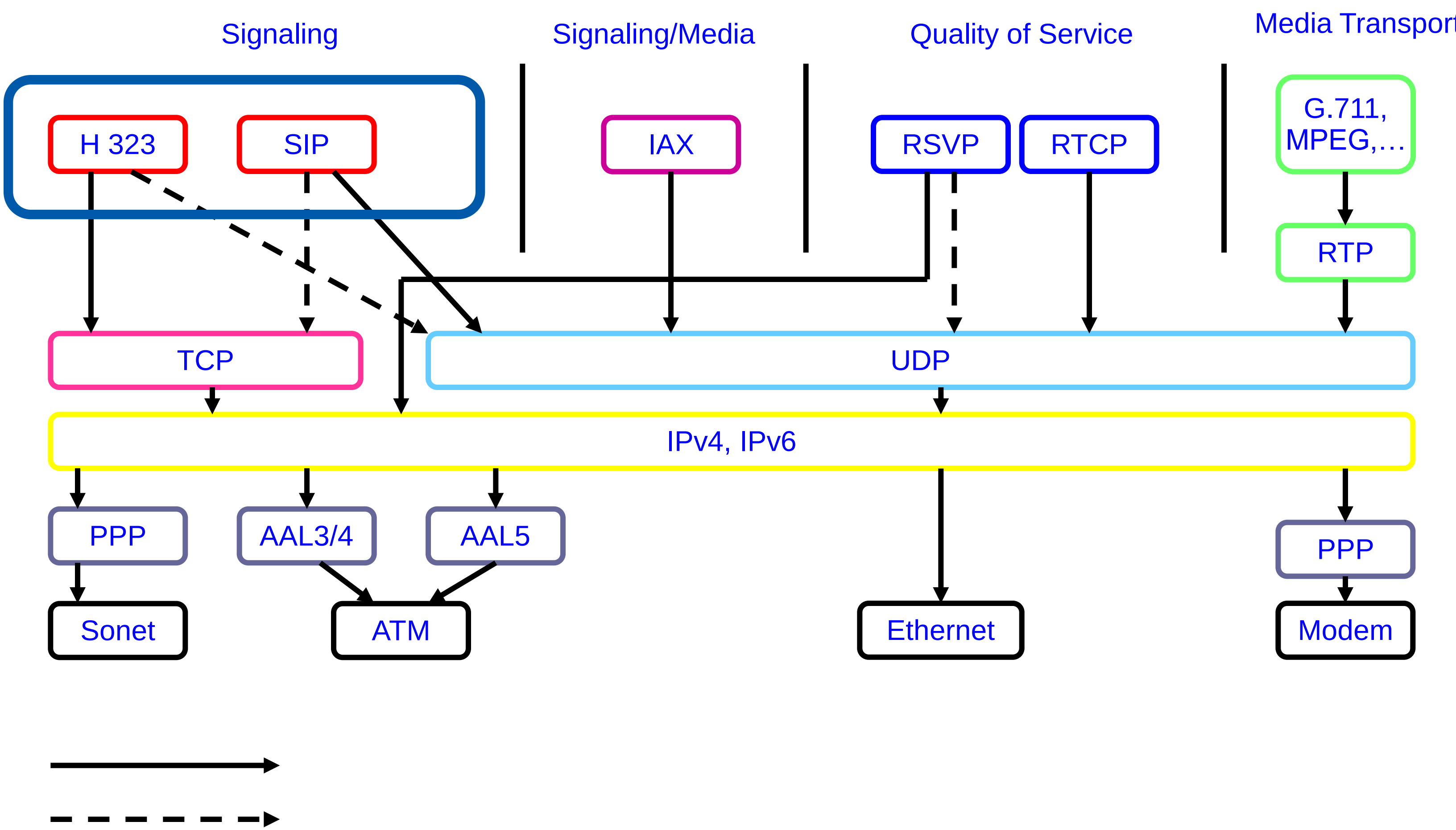
- 1. Proprietary / closed protocols
  - E.g. Skype

- 2. Open / standardized VoIP protocols
  - SIP signaling-approach from IETF
  - H.323 signaling-approach from ITU
  - IAX/IAX2 signaling and media-approach (IETF Draft)
  - MGCP (Media Gateway Control Protocol)

Focus here

- IP telephony systems
  - As contrast to individual “internet telephony”-approaches

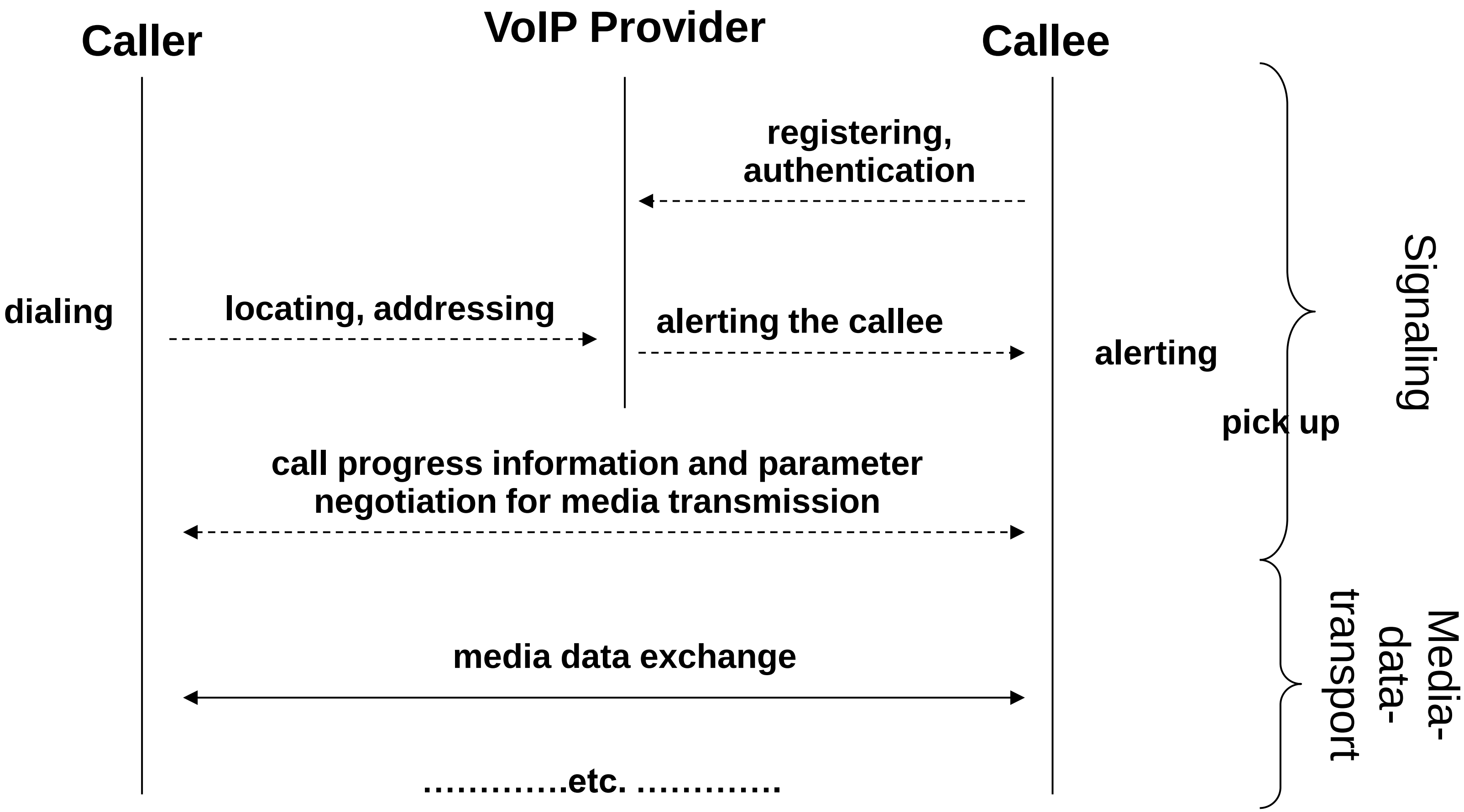
Based on Internet Real-time and Multimedia Protocols



Generic VoIP Call

VoIP as combination of

- Signaling
- Media-transport
- with quality of service



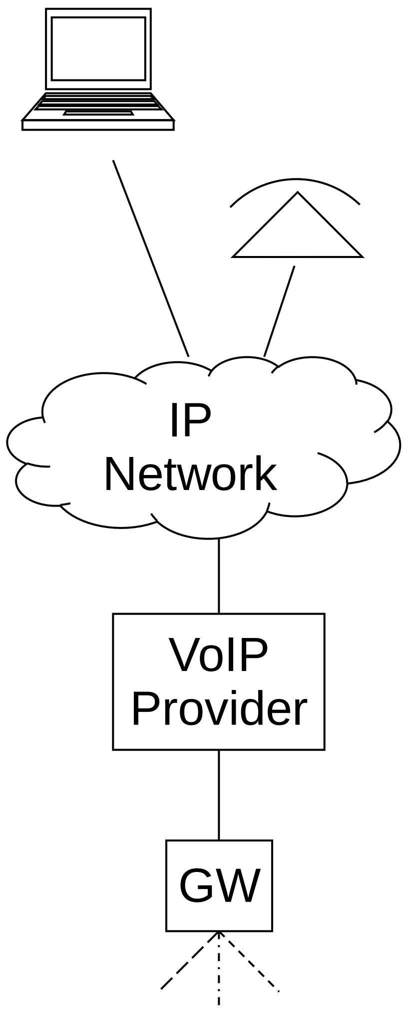
VoIP System Elements

Clients / Terminals

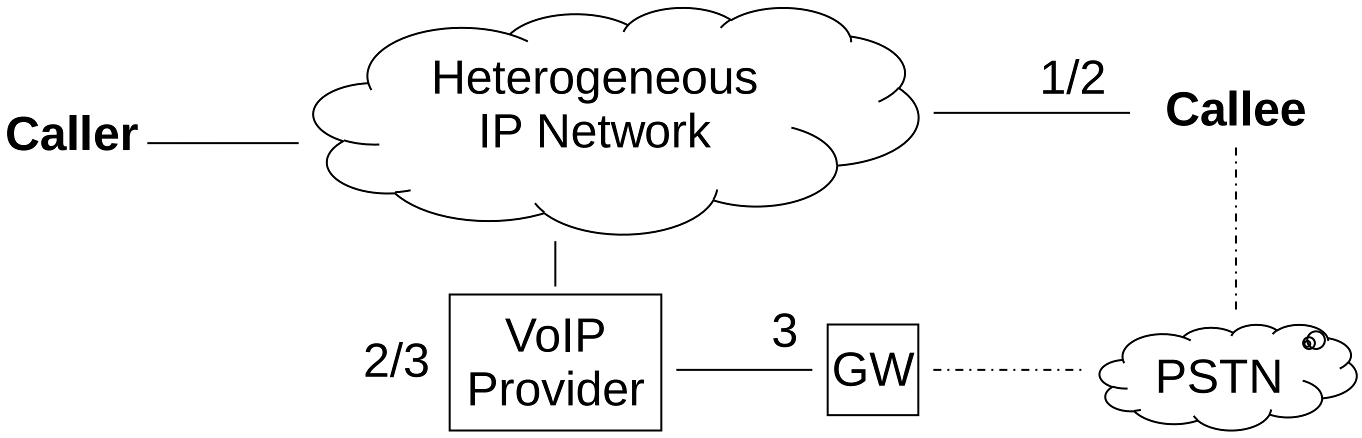
- Softphones
  - Software with VoIP functionality
- Hardphones
  - Device telephone with VoIP functionality

VoIP (Service) Provider

- Registrar (or gatekeeper)
  - For management of client IP addresses
  - For authentication
- Proxy or Redirect-Server
  - Forwarding of the signaling-messages
- Gateway
  - To other VoIP Systems or PSTN
  - Signaling and/or media conversion



VoIP Scenarios



Contact callee using

1. Direct IP-addressing
- Static IP-address needed

2. Directory service / registrar (VoIP Provider)
- Dynamic IP-address

3. PSTN gateway
- In order to reach phone partners on POTS

5.3 Example of Internet Phone

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Speaker's audio: alternating talk spurts, silent periods

- 64 kbps during talk spurt
- packets generated only during talk spurts
- 20 msec chunks at 8 Kbytes/sec: 160 bytes data

Application-layer header added to each chunk

- Chunk+header encapsulated into UDP segment
- Application sends UDP segment into socket every 20 msec during talkspurt

Example: Internet Phone

Packet Loss and Delay

Network loss:

- IP datagram lost due to network congestion (router buffer overflow)

Delay loss:

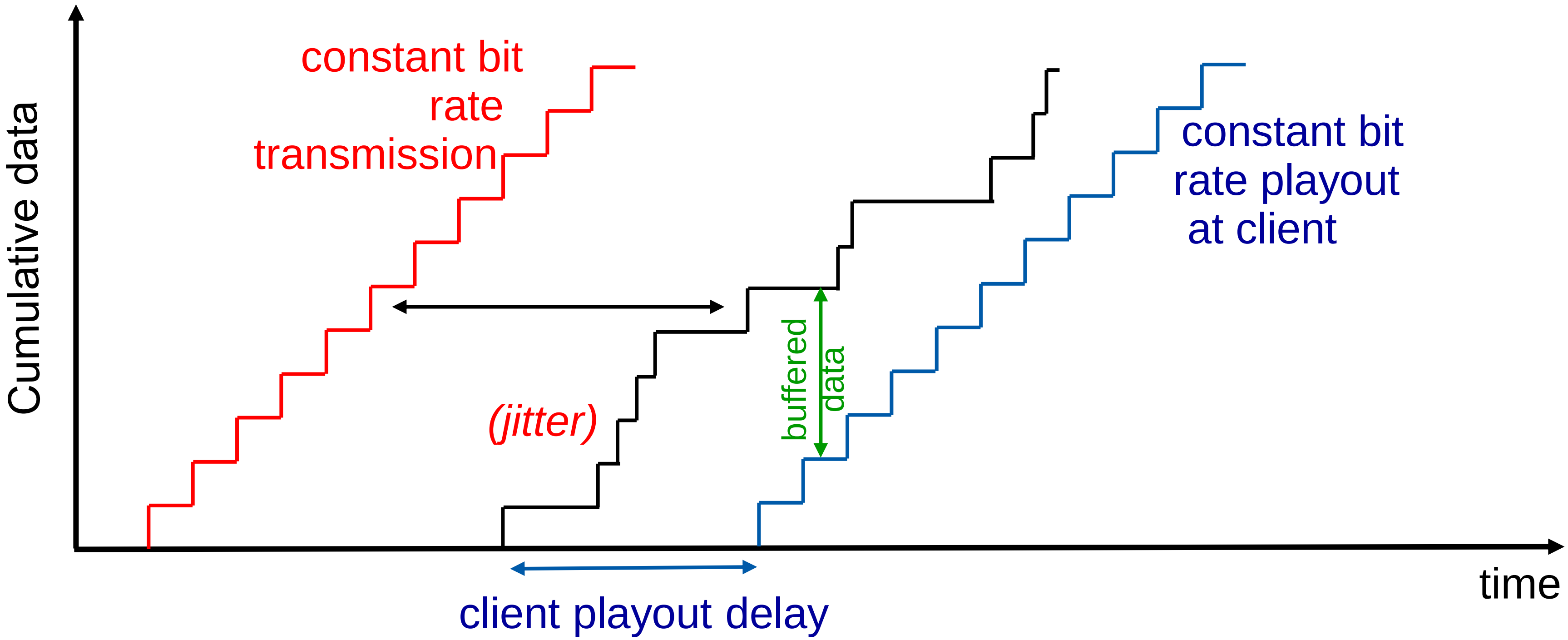
- IP datagram arrives too late for playout at receiver
- Delays: processing, queueing in network; end-system (sender, receiver) delays
  - Typical maximum tolerable delay: 400 ms

Loss tolerance:

- Depending on voice encoding, losses concealed
- Packet loss rates between 1% and 10% can be tolerated

Delay and Jitter

- Consider end-to-end delays of two consecutive packets: difference can be more or less than 20 msec (transmission time difference)



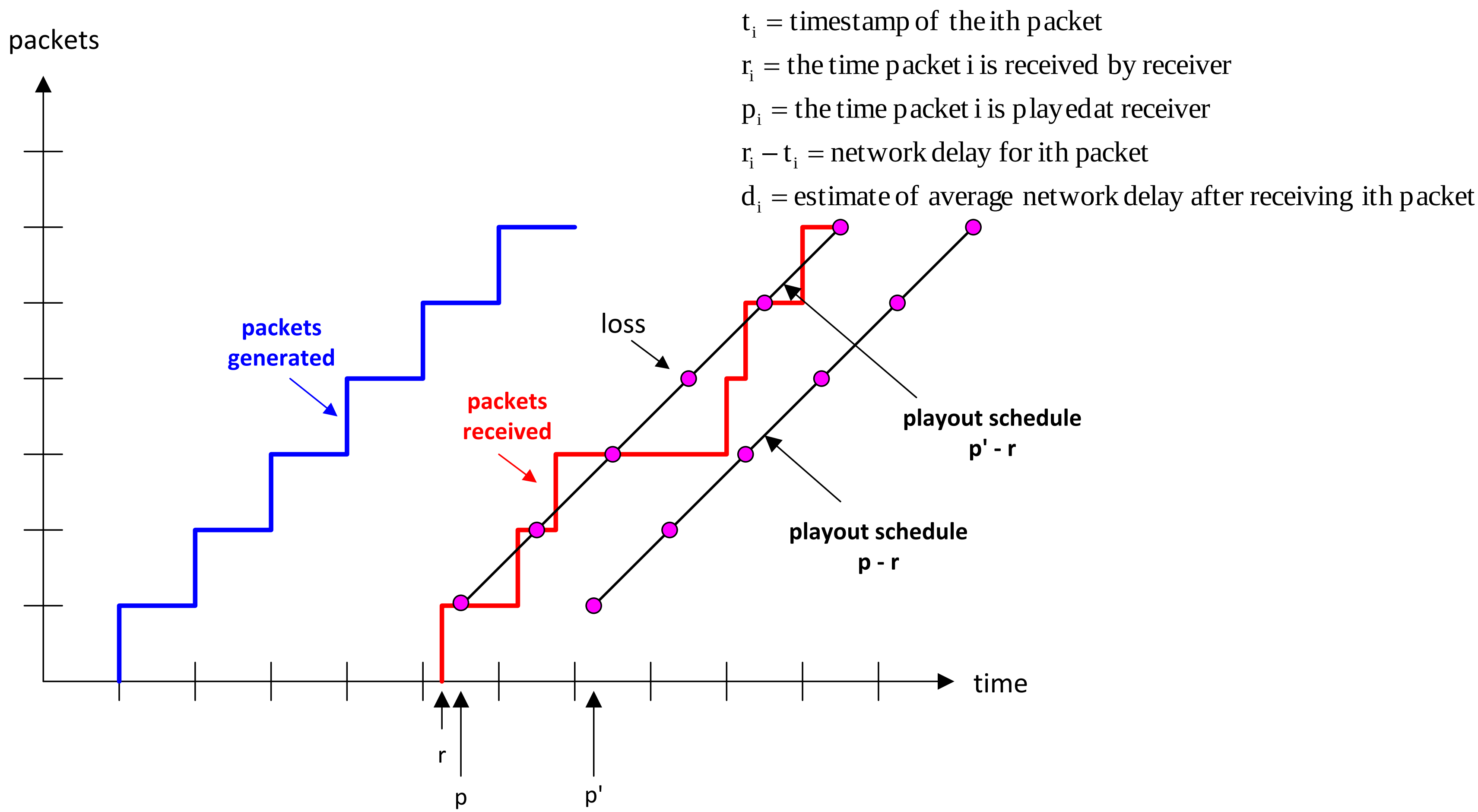
Fixed Playout Delay

- Receiver attempts to playout each chunk exactly q msecs after chunk was generated
  - chunk has time stamp t:
    - → play out chunk at t+q .
  - chunk arrives after t+q:
    - → data arrives too late for playout, data “lost”

Tradeoff in choosing q:

- large q: less packet loss
- small q: better interactive experience

Fixed Playout Delay



Sender generates packets every 20 msec during talk spurt

- First packet received at time r
- First playout schedule: begins at p
- Second playout schedule: begins at p'

Adaptive Playout Delay

Goal:

to minimize playout delay, keeping late loss rate low

Approach:

adaptive playout delay adjustment:

- to estimate network delay,
  to adjust playout delay at beginning of each talk spurt
- silent periods compressed and elongated
- chunks still played out every 20 msec during talk spurt.

$$t_i = \text{timestamp of the ith packet}$$

$$r_i = \text{the time packet i is received by receiver}$$

$$p_i = \text{the time packet i is played at receiver}$$

$$r_i - t_i = \text{network delay for ith packet}$$

$$d_i = \text{estimate of average network delay after receiving ith packet}$$

Dynamic estimate of average delay at receiver:

$$d_i = (1-u)d_{i-1} + u(r_i - t_i)$$

where u is a fixed constant (e.g., u = .01).

Adaptive Playout Delay

- also useful to estimate average deviation of delay, v_i :

$$v_i = (1-u)v_{i-1} + u\,|\,r_i - t_i - d_i\,|$$

- estimates d_i , v_i calculated for every received packet
  (but used only at start of talk spurt)

- for first packet in talk spurt, playout time is:

$$p_i = t_i + d_i + Kv_i$$

  - where K is positive constant

- remaining packets in talkspurt are played out periodically

Adaptive Playout Delay

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How does receiver determine whether packet is first in a talkspurt?

If no loss,
receiver looks at successive timestamps

- IF Difference of successive stamps > 20 msec
  → talk spurt begins.

With loss possible,
receiver must look at both time stamps and sequence numbers

- IF Difference of successive stamps > 20 msec
  AND sequence numbers without gaps
  → talk spurt begins

Forward Error Correction (FEC): simple scheme

- For every group of n chunks
  create redundant chunk by exclusive OR-ing n original chunks
- Send out n+1 chunks, increasing bandwidth by factor 1/n.
- Can reconstruct original n chunks if at most one lost chunk from n+1 chunks

Playout delay: enough time to receive all n+1 packets

- Tradeoff:
  - Increase n, less bandwidth waste
  - Increase n, longer playout delay
  - Increase n, higher probability that 2 or more chunks will be lost

Recovery from packet loss

2nd FEC scheme

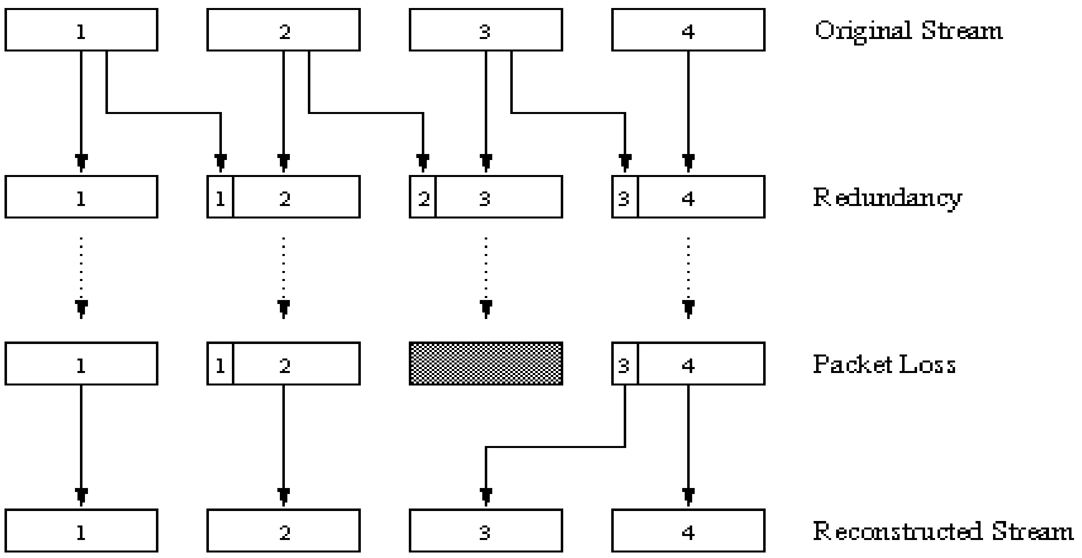
- “piggyback lower quality stream”
- Send lower resolution audio stream as redundant information
- E.g., nominal stream PCM at 64 kbps and redundant stream GSM at 13 kbps

Whenever there is non-consecutive loss,

- Receiver can conceal the loss
- Can also append (n-1)st and (n-2)nd low-bit rate chunk

2nd FEC scheme

- "piggyback lower quality stream"
- Send lower resolution audio stream as redundant information
- E.g., nominal stream PCM at 64 kbps and redundant stream GSM at 13 kbps



Whenever there is non-consecutive loss,

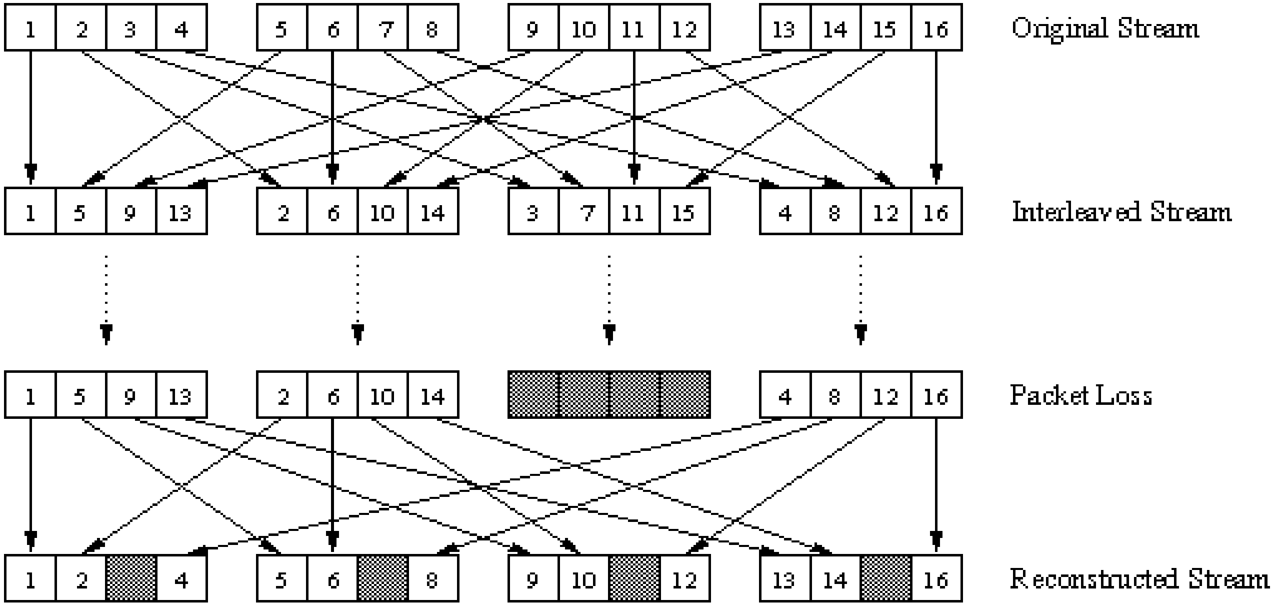
- Receiver can conceal the loss

Can also append (n-1)st and (n-2)nd low-bit rate chunk

Recovery from packet loss

Interleaving

- Chunks divided into smaller units
- For example, four 5 msec units per chunk
- Packet contains small units from different chunks



If packet lost, still have most of every chunk

- no redundancy overhead, but increases playout delay

5.5 Summary Voice Communication