# Assignment 2: SIP Speaker

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Modifications 2007 and 2008 by Erik Eliasson

# Outline

- What is "Voice over IP"?
- So what is this assignment about?
- Introduction to the protocols
  - SIP, SDP, RTP, RTCP
- Where to start

# What Is "Voice over IP"?

- Voice calls over IP networks
  - "IP-telephony", "Voice over packet", ...
- Integration
  - Voice and data in the same network
  - Applications may integrate different media
- "Free" market
- Anyone can be operator
  - Skype, iptel.org, pulver.com, ...

# This Assignment

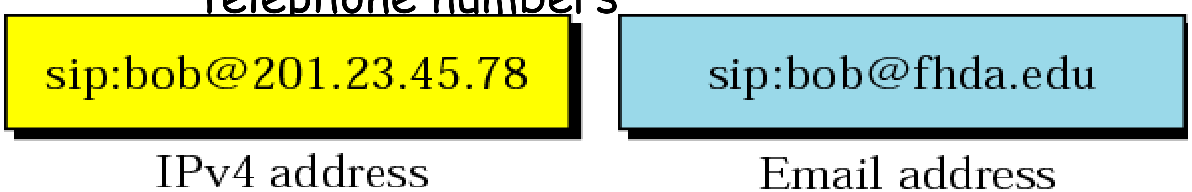
- VoIP answering service
  - One-way communication
  - Possible applications
    - Numerous: Dial-up service for news, weather, traffic conditions, …
- You will learn about
  - VoIP protocols
    - SIP, SDP, RTP
  - Multi-user state management with UDP
  - Multi-protocol servers

# VoIP Protocols

- Media
  - Transport voice calls in IP packets
  - Real-time Transport Protocol (RTP)
- Signaling
  - Keep track of users, call set up and tear-down, etc.
  - Session Initiation Protocol (SIP)
- Setting up voice channels
  - Session Description Protocol (SDP)
- Support protocols
  - Name-address translation, AAA (Authentication, Accounting, Authorization), Quality of service, ...
  - DNS, TRIP, COPS, Diameter, ...

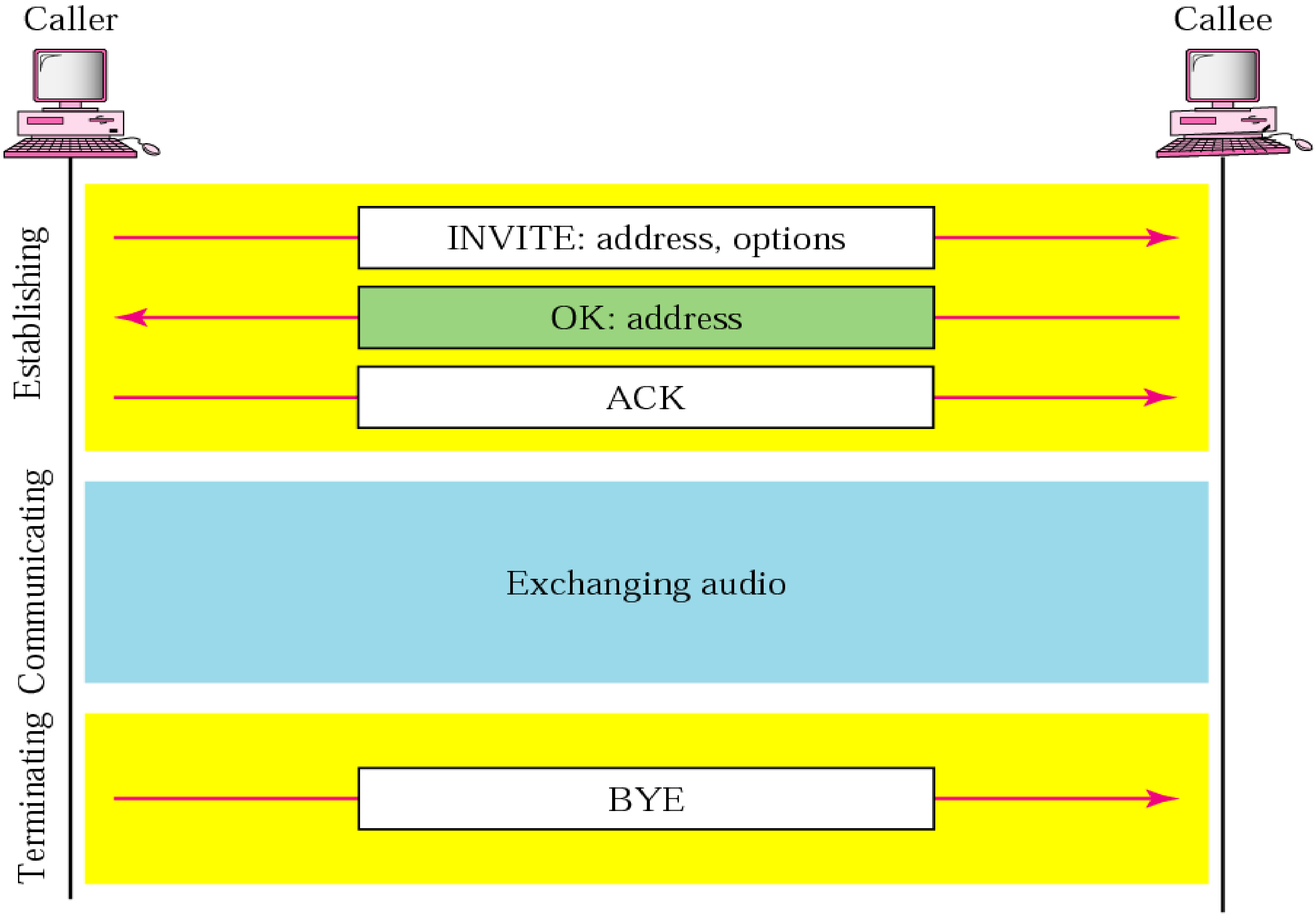
# Session Initiation Protocol (SIP)

- Establishing session
  - VoIP calls / video conferencing
  - Instant messaging
  - Presense
  - Event notification
- Locating users
- SIP URI:s
  - "telephone numbers"




*From B. A. Forouzan: Data Communications and Networking, 3rd ed, McGraw-Hill*

# Simple Session Directly Between Parties

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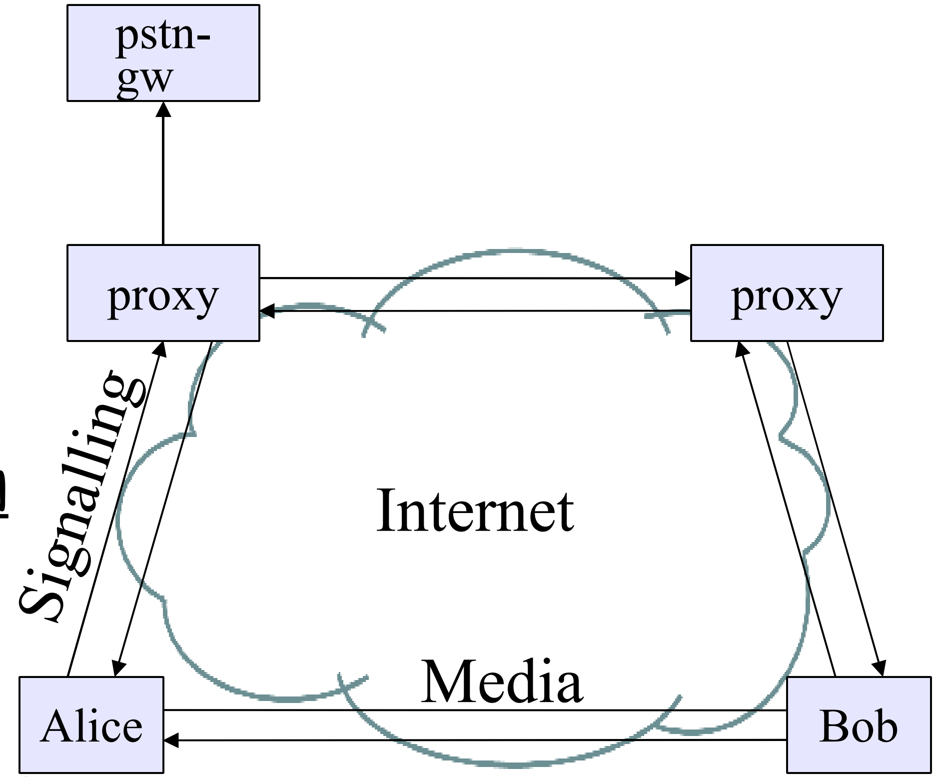
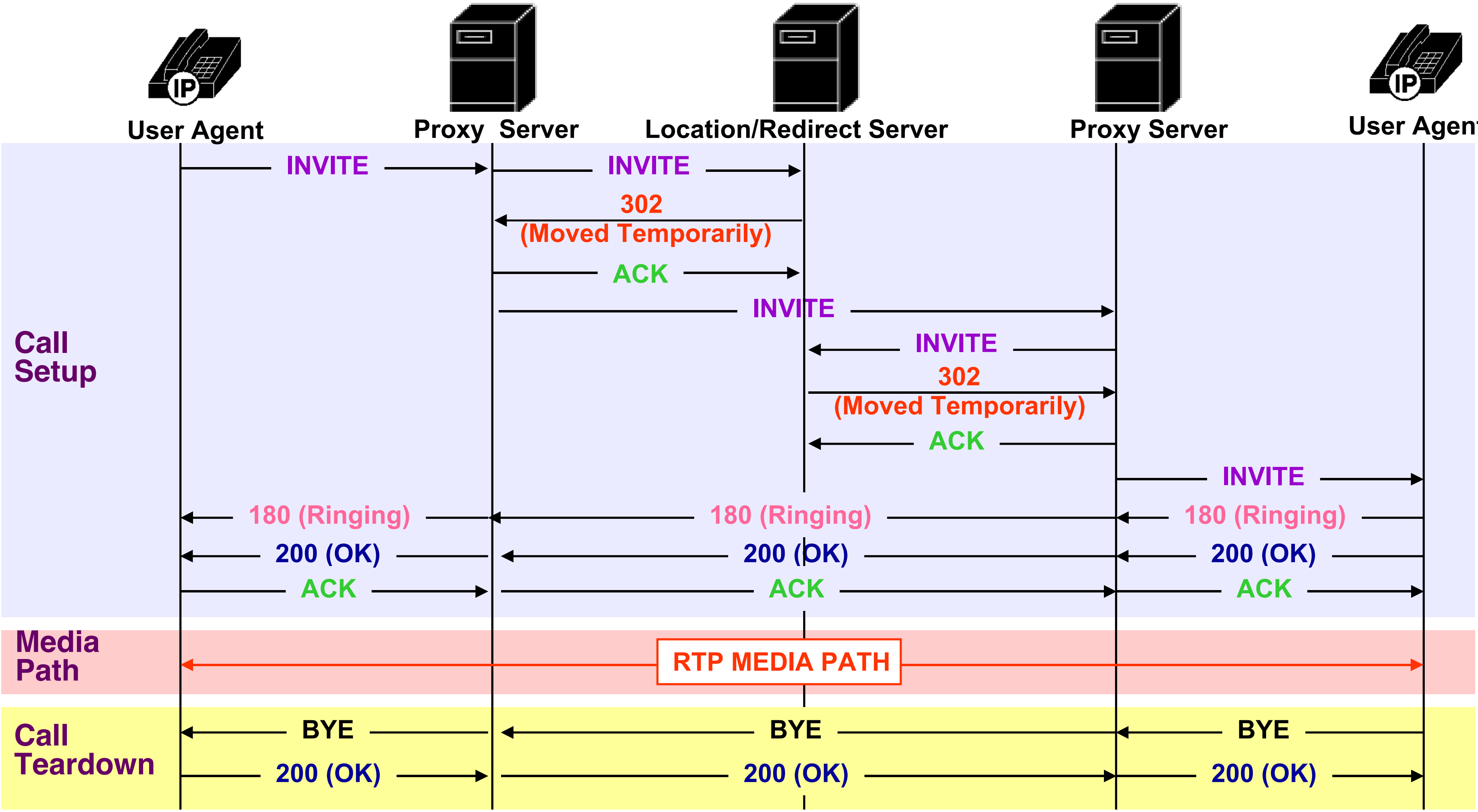
# SIP distributed architecture

- REGISTER to your proxy
- Proxies forwards requests
  - DNS SRV lookup to find servers
    - _sip._udp.domainb.com
- Gateways handles calls to PSTN
- Media is end-to-end
  - NATs are troublesome
- SIP uses either UDP, TCP or TLS



# SIP Call Setup




*From http://www.vovida.org/document/protocol.html*

# SIP Headers

```
INVITE sip:5120@192.168.36.180 SIP/2.0
Via: SIP/2.0/UDP 192.168.6.21:5060
From: sip:5121@192.168.6.21
To: <sip:5120@192.168.36.180>
Call-ID: c2943000-e0563-2a1ce-2e323931@192.168.6.21
CSeq: 100 INVITE
Max-Forwards: 70
User-Agent: Cisco IP Phone/ Rev. 1/ SIP enabled
Accept: application/sdp
Contact: sip:5121@192.168.6.21:5060
Content-Type: application/sdp
```

- SIP borrows much of the syntax and semantics from HTTP
- SIP message similar to HTTP message—message format, header and MIME support

*From http://www.vovida.org/document/protocol.html*

# SDP

- Session Description Protocol, RFC 2327
- Describes media parameters (CODEC, IP/port etc) for RTP initialization
- Attached to SIP INVITE method and 1xx/2xx responses

# SDP Example

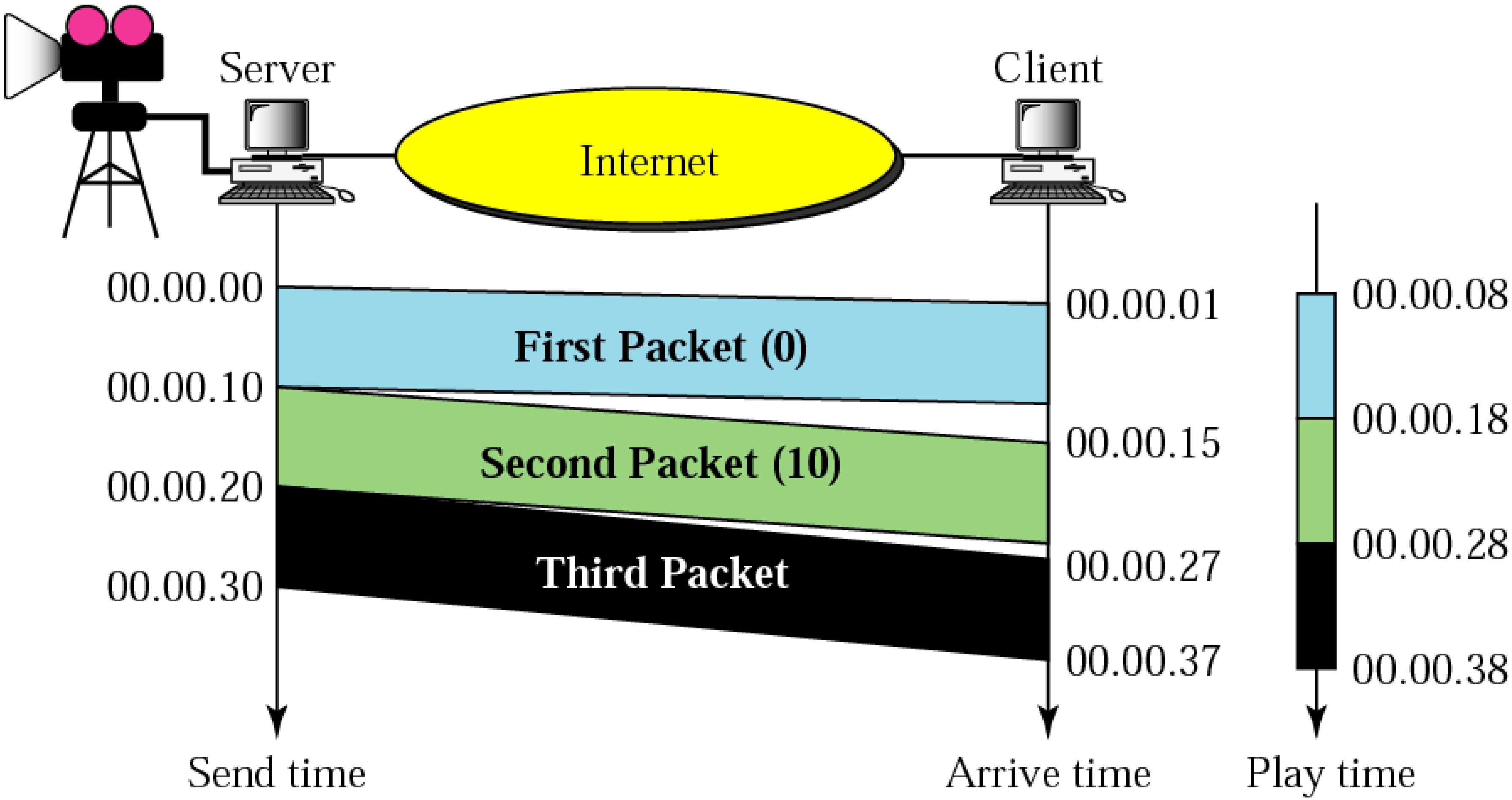
| | |
|---|---|
| v=0 | *Version* |
| o=- 12334 12334 IN IP4 172.31.212.23 | *Session origin and owner* |
| s=SIP Speaker | *Session name* |
| c=IN 172.31.212.23 | *Connect information* |
| t=31872345698 0 | *Session time* |
| m=audio 54434 RTP/AVP 0 | *Media,transport address, codec* |
| a=rtmap:0 PCMU/8000 | *Attribute (RTP codec type)* |
| a=ptime:20 | *Attribute (packet time)* |

# RTP

- Real time Transport Protocol, RFC 1889
- Runs over UDP
  - Established using SDP
- Each RTP packet contains:
  - Sequence number
  - Timestamp (time when sent)
  - Synchronization Source Identifier (SRC)
  - Payload – sound samples
    - Example: 8-bits per sample, sampling rate 8000 Hz, packet interval 20 msec  160 bytes per packet

Figure 28-6

# Timestamp

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# RTCP – RTP Control Protocol

- Reports packet losses and delay between RTP endpoints
- Sender Report (SR)
  - Sender counts (packets sent, bytes sent)
  - Timestamp
- Receiver Report (RR)
  - Receiver counts (packets received, lost)
  - Jitter (variations in inter-packet delay)
  - Last SR timestamp received
  - Delay since last SR (time between receiving SR and sending RR)
- Runs over UDP
  - Uses lowest odd-numbered port that follows the port number used for RTP session
    - RTP port plus one

# This Assignment

- VoIP answering service (SIP User Agent)
  - One-way communication, only speaks (ok to simply ignore audio sent from client)
  - waits for incoming calls
  - answers and play message
  - when message ends terminate the call
  - integrate a web server to change the current message
  - dynamically generate sound from text with Text To Speech library
- Java assignment. Recommended tools:
  - Eclipse IDE (development and debugging)
  - Ethereal (debugging)

# "Basic"

- Basic SIP signaling with SDP
- The SIP Speaker receives a call, establishes the call, waits and terminates the call and over again
- Only single user mode is needed
- Use UDP for SIP
- No existing libraries for SIP or SDP
- Support for command line interface and configuration file
  - java SIPSpeaker [-c config_file_name] [-user sip_uri] [-http http_bind_address]

# “Medium”

- As for grade “basic”, but also
  - support for one-way RTP stream to transmit to the caller
  - use an existing RTP and audio processing library  Java Media Framework
  - Use a static wav file as a message
  - Add multi user support to receive and handle several simultaneous incoming/ongoing calls

# “Advanced”

- As for “Medium”, but also
  - with a dynamically generated sound file
  - generated sound from text using an existing Text To Speech library FreeTTS
  - customize the Web Mail assignment as an interface to
    - submit, show and delete the current message
  - web server and SIP Speaker as one integrated application

# Deadline

- **April 25, 2008, 23:59** (midnight)
  - Submission via the course homepage
- April 30, Feedback on assignments
- **May 13, 2008, 23:59** (midnight)
  - Submission via the course homepage

- Supervision: April 18, 2008, 15:00-17:00
- Questions answered on discussion forum until April 22nd.

- Check course homepage for further details and updates

Good Luck!