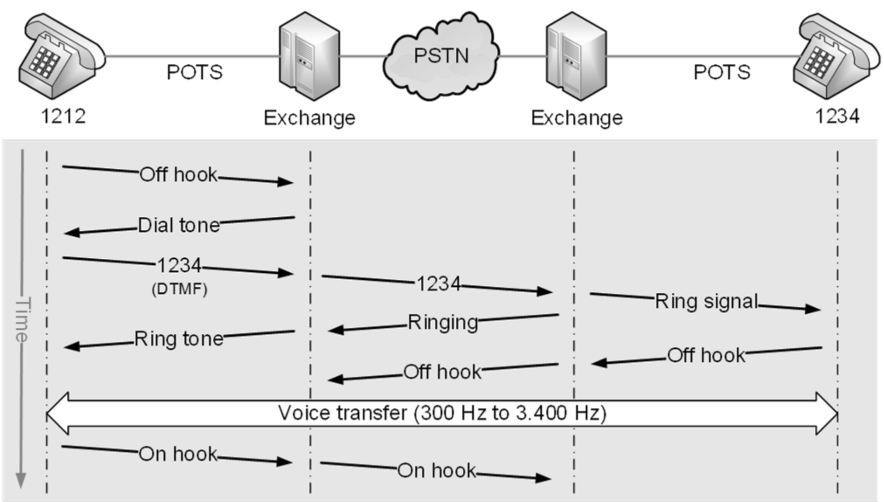
# DTMF Dialing

- Telephone sends dual tone when keypad pressed
- Exchange recognizes dual tones and interprets them as digits
- All tones are within the 300 to 3.400 Hz band
  - This is called in-band signalling
  - Voice and signalling carried in the same band

| Frequency | 1209 Hz | 1336 Hz | 1477 Hz | 1633 Hz |
|---|---|---|---|---|
| 697 Hz | 1 | 2 | 3 | A |
| 770 Hz | 4 | 5 | 6 | B |
| 852 Hz | 7 | 8 | 9 | C |
| 941 Hz | * | 0 | # | D |

# signalling and voice

## POTS call progress



## Circuit vs. packet switched

- Circuit switched network (PSTN)
  - A logical connection is made between the two endpoints. (Phones)
  - Bandwidth guaranteed ( 64 Kbps for voice)
  - Delay constant
- Packet switched network (IP network)
  - A logical connection use TCP/UDP between the two endpoints. (IP Phones)
  - No bandwidth guarantee
  - Delay <u>not</u> constant

## MOS
### Mean Opinion Score

- MOS used to validate the quality of telephone voice quality

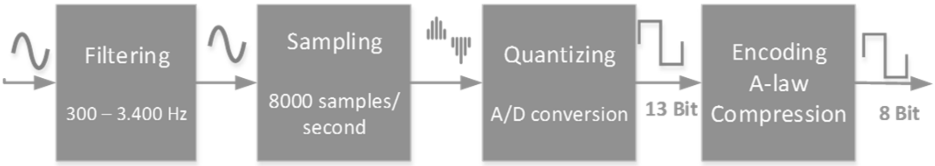
| MOS | Quality | impairment |
|---|---|---|
| 5 | Excellent | Imperceptible |
| 4 | Good | Perceptible but not annoying |
| 3 | Fair | Slightly annoying |
| 2 | Poor | Annoying |
| 1 | Bad | Very annoying |

| Codec | From | Bandwidth | Data rate | MOS |
|---|---|---|---|---|
| G.711 | 1972 | 300-3.400 Hz | 64 Kbps | 4,1 |
| G.722 | 1988 | 50-7.000 Hz | 64 Kbps | ~4,5 |
| G.726 | 1990 | 300-3.400 Hz | 32 Kbps | 3,85 |
| G.729 | 1996 | 300-3.400 Hz | 8 Kbps | 3,92 |
| GSM EFR | 1995 | 300-3.400 Hz | 12,2 Kbps | 3,8 |

# Codec G.711 A-law

- From analog audio signal to coded bit stream
  - Advanced mathematics used in encoding to preserve voice quality

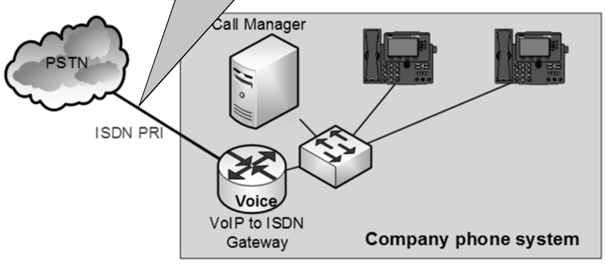
| Stage | Signal | Comment |
|---|---|---|
| Filtering | Analog | Input from microphone. High bandwidth |
| Sampling | Analog filtered | 8000 samples of input signal caught pr. Second |
| Quantizing | Analog samples | Analog to digital conversion. 13 bit precision |
| Encoding | Compression | 8000 compressions second to 8 bit = 64.000 bps |




# ISDN PRI

- Mainly used to connect Private telephone exchanges or IP phone system to PSTN
  - Call ange)
- For exam al 0 to get a public line
- Custome lls are routed through ISDN PRI

ISDN PRI up to 30 outgoing/incoming simultaneous calls in one ISDN PRI



## in/out of band signalling

- ISDN signalling
  - signalling carried in the D channel
  - Voice/data carried in B-channel
  - Signal and voice separated in two channels or bands
  - ISDN is an example of out-of-band signalling
- POTS
  - Voice carried in the 300 – 3.400 Hz band
  - signalling – DTMF – carried in the same band
  - POTS is an example of in-band-signalling

# VoIP signalling standards

VoIP – Voice over IP

## VoIP signalling standards

- A variety of IP based signalling standards to VoIP defined – and used
  - SIP
  - H.323
  - MGCP
  - ….
- SIP and H.323 included in this course
- The purpose of VoIP signalling
  - Establish, maintain and close voice calls
  - Eventually collect information for taxation

## signalling standards

- Two kinds of information necessary to phone
  - Call control (signalling phone to/from exchange)
  - Voice transfer (Between participating phones)
- In traditional telephony
  - Call Control (For example POTS or ISDN)
  - Voice transfer (Circuit switched channel)
- In VoIP – Voice over IP
  - Call control (For example SIP or H.323)
  - Voice transfer (Packet switched)

## SIP and H.323

- H.323
  - Defined by ITU in 1996
    - International Telecommunication Union
  - Uses many technologies from PSTN
  - A suite of different protocols incl. voice/video
- SIP – Session Initiation Protocol
  - Defined by IETF in 1996
    - Internet Engineering Task Force
  - Uses many technologies from the Internet
  - Is a signalling protocol for voice/video

## SIP SIGNALLING

Session initiation protocol

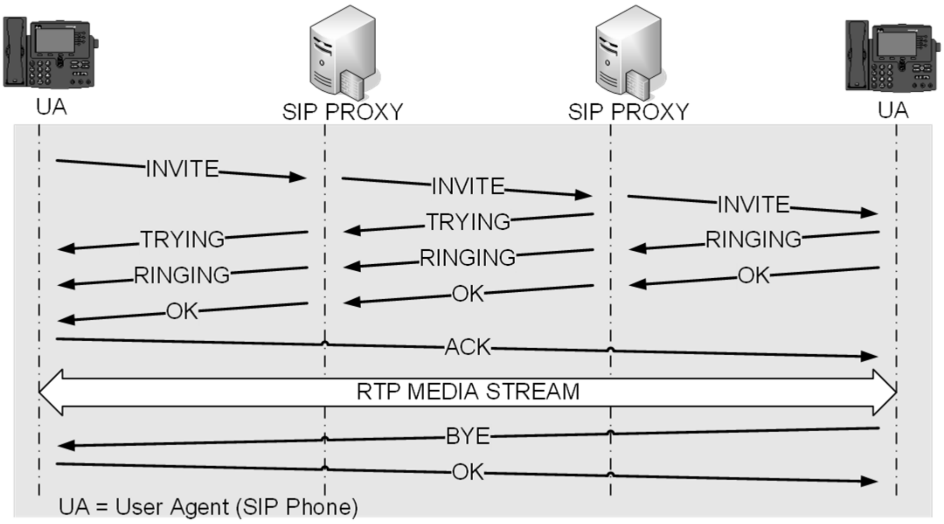
# SIP
## Session Initiation Protocol

- Defined by IETF in 1996
  - Internet Engineering Task Force
- Uses many technologies from the Internet
- Is a signalling protocol for voice/video
- SIP are widely used as VoIP signalling

# SIP Design

- Expandable
  - Easy integration of new facilities
- (Not) a substitute for PSTN
  - SIP are used by many service providers
- Diversity of end-devices
  - Intelligent end-devices
- Multimedia
  - Voice, video….

# SIP call flow example




# SIP

- Is a application layer protocol
- Is a text based signalling protocol
  - Reminding of the HTTP and SMTP protocols
- Are used to establish, maintain and terminate multimedia sessions
- Support unicast and multicast sessions
- Location independent
  - Suitable for mobile users

## SIP components

- There are two SIP components
- User Agents (UA) – SIP End-points (IP Phones) consisting of
  - User Agent Client (UAC): Initiates sessions
  - User Agent Server (UAS): Respond sessions
- SIP servers
  - Proxy server – Like a SIP IP-PBX
  - Registrar server – User Agent register
  - Redirect server – Redirect sessions

## SIP addressing

- SIP addresses are identified by SIP URI's
  - Uniform Resource Identifier
- Example

  sip:4103@mercantec.dk;transport=UDP
  - sip – The URI service
  - 4103 – The URI user part (extension/phone)
  - Mercantec.dk – The URI host part
  - ;transport=UDP – A URI parameter

## SIP addressing

- Different formats of URI’s
  - FQDN’s: sip:heth@mercantec.dk
  - E.164: sip:30539361@mercantec.dk
  - E.164: tel:30539361
  - Mixed: sip:heth@194.123.12.23
- E.164 numbers in DNS

## SIP messaging

- Messages contain header describing the communication in details
- Uses a text based syntax and header like HTTP
- Transport protocols TCP, UDP or SCTP
  - SCTP: Stream Control Transport Protocol
- Basically there are two kinds of messages
  - Requests: Sent by Clients
  - Response: Sent by servers

# SIP messaging

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- Request message header

| Method | Request URI | SIP version |
|---|---|---|

  - Method: INVITE, ACK, BYE ….
  - Request URI: The receiver of the request
  - SIP version: Used SIP version (2.0 in use)
- Response message header

| SIP version | Status code | Reason phrase |
|---|---|---|

  - SIP version: Used SIP version (2.0 in use)
  - Status code: An integer describing the answer
  - Reason Phrase: A text description of status code

# SIP requests methods

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| Command | Function |
|---|---|
| **INVITE** | Used to establish a media session between user agents. Starting a call |
| **ACK** | Confirms reliable message exchanges |
| **BYE** | Terminates a established session between agents |
| **CANCEL** | Terminates a pending request |
| **OPTIONS** | Requests information about the capabilities of a caller, without setting up a call |
| **REGISTER** | Used by a user agent to register to registrar. The user agent informs the registrar of its IP address and URI. |

**NOTE:** There are defined more SIP request methods than indicated in the above table. The shown methods are from RFC 3261

## SIP response codes

With examples

| Respons class | Status code | Reason phrase |
|---|---|---|
| **Info** | **1xx** | **Provisional responses** |
| | 100 | Trying |
| | 180 | Ringning |
| **Succes** | **2xx** | **Succesful responses** |
| | 200 | OK |
| **Redirect** | **3xx** | **Redirection responses** |
| | 302 | Moved temporarily |
| **Client error** | **4xx** | **Client failure responses** |
| | 401 | Unauthorized |
| **Server error** | **5xx** | **Server failure responses** |
| | 503 | Service unavailable |
| **Global failure** | **6xx** | **Global failure responses** |
| | 603 | Decline |

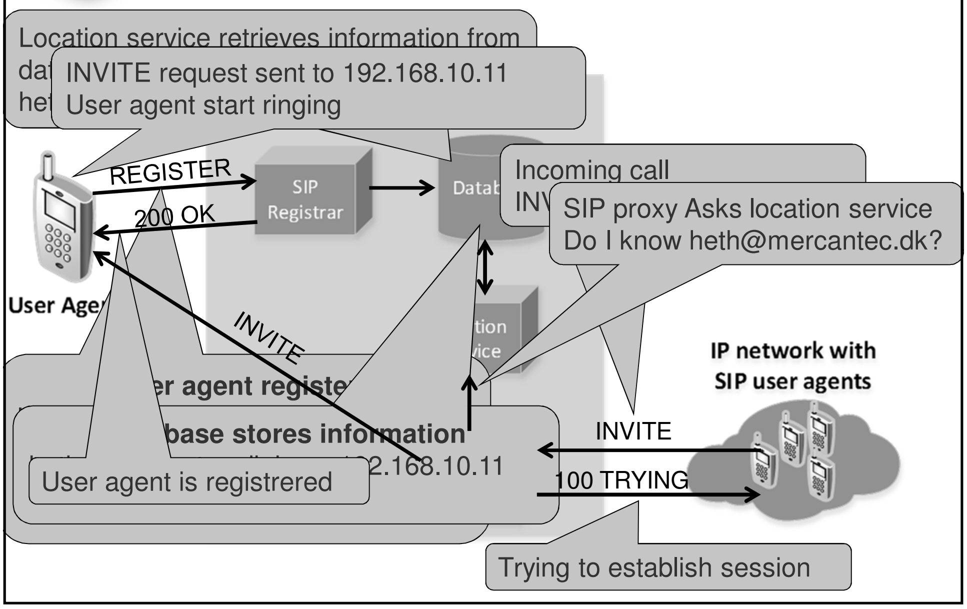
## SIP User Agent – UA

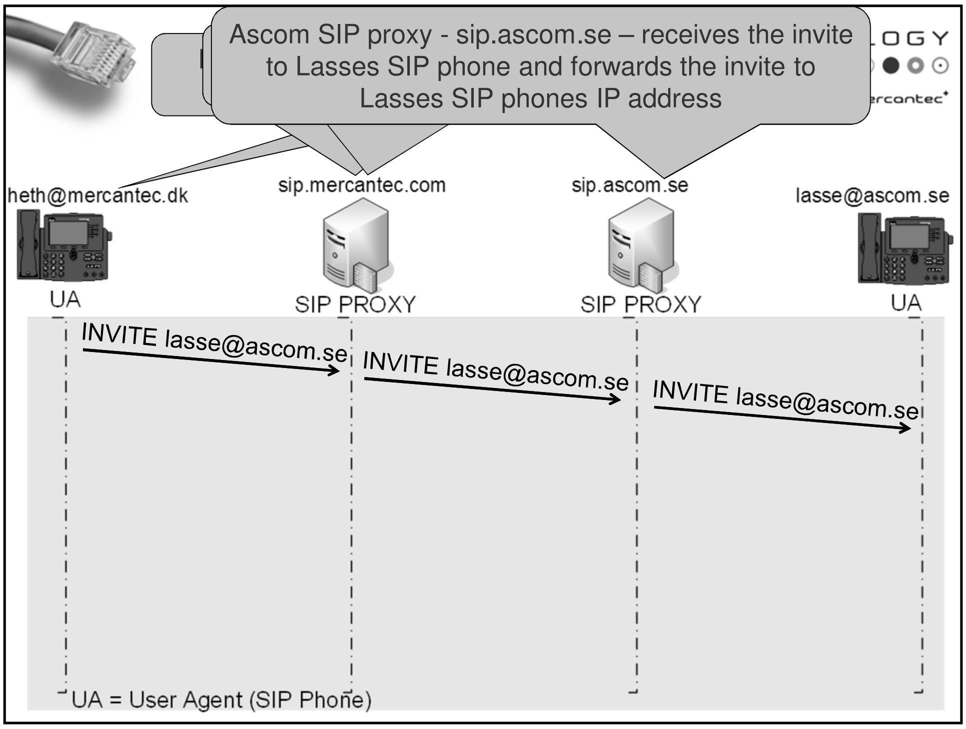
- A User Agent is a software component acting on behalf of a user
- The SIP phone soft or hard is a SIP UA
- User Agent Client – UAC
  - Initiates SIP sessions when calling
  - Send SIP requests
- User Agent Server – UAS
  - Accepts or rejects SIP session requests
  - Receives SIP requests and return SIP response

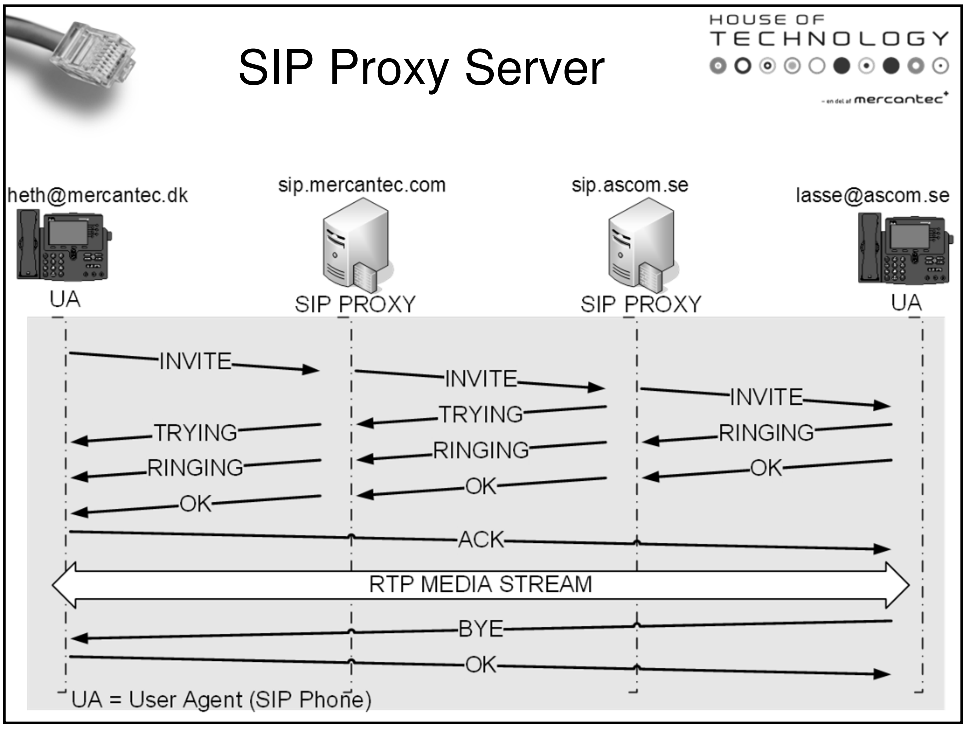
# SIP servers

- Proxy server
  - Receive and sends SIP request and responses on behalf of the User Agents. (UA)
- Registrar server
  - User Agents register to the registrar server
  - Keeps track on User Agents URI and IP addresses and acts as a location service
- Redirection Server
  - Returns “contact this address” responces
- The servers often live on the same “box”

# SIP servers




Ascom SIP proxy - sip.ascom.se – receives the invite to Lasses SIP phone and forwards the invite to Lasses SIP phones IP address
heth@mercantec.dk
sip.mercantec.com
sip.ascom.se
lasse@ascom.se
UA
SIP PROXY
SIP PROXY
UA
INVITE lasse@ascom.se
INVITE lasse@ascom.se
INVITE lasse@ascom.se
UA = User Agent (SIP Phone)


SIP Proxy Server
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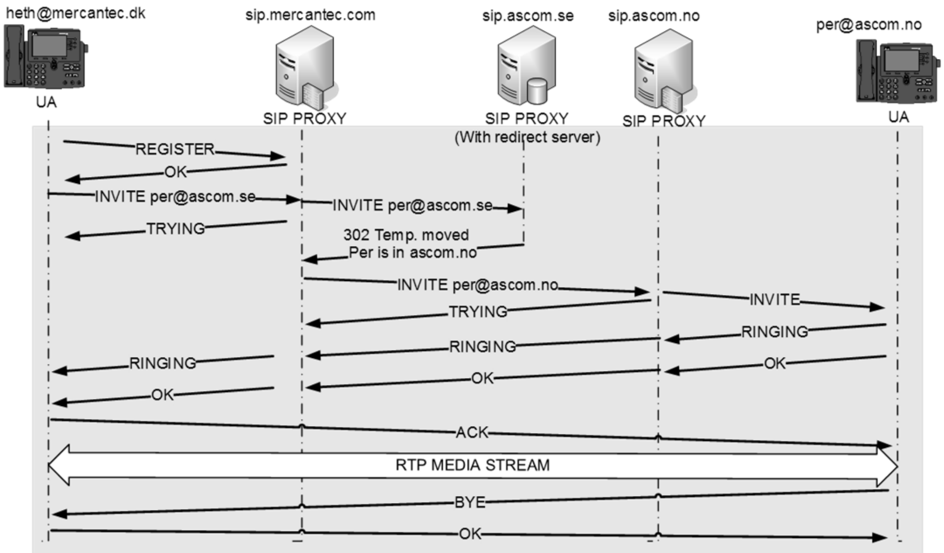
heth@mercantec.dk
sip.mercantec.com
sip.ascom.se
lasse@ascom.se
UA
SIP PROXY
SIP PROXY
UA
INVITE
INVITE
INVITE
TRYING
TRYING
RINGING
RINGING
RINGING
OK
OK
OK
ACK
RTP MEDIA STREAM
BYE
OK
UA = User Agent (SIP Phone)

# SIP Redirect

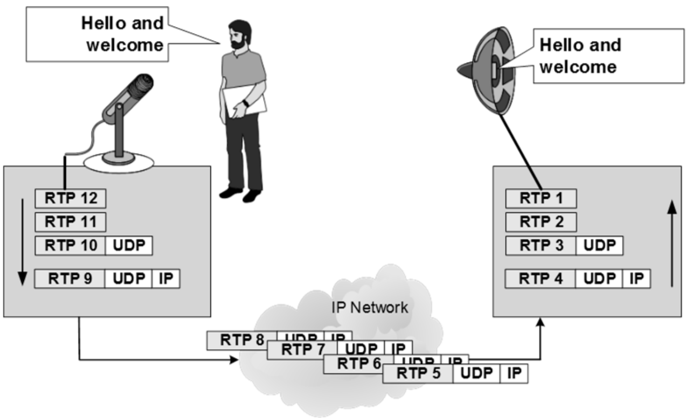



# RTP FAMILY

Real-time Transport Protocol

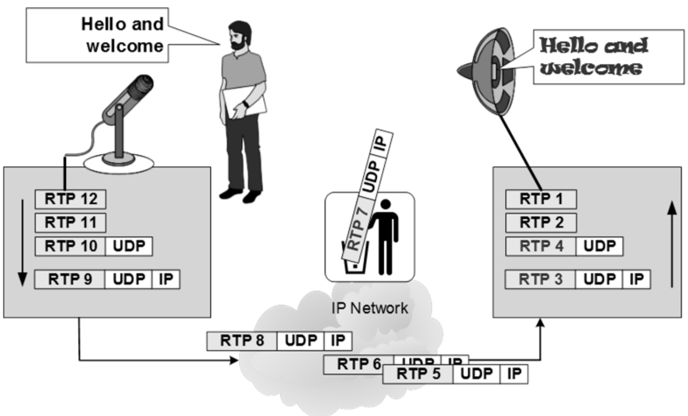
# Voice quality

- Low constant delay is important for a good voice quality with low delay




# Packet switched networks

– No guaranty of constant delay in packet !
– No guaranty of orderly delivery of packets !
– No guaranty packets are delivered !



# RTP

- Designed for end-to-end real-time transfer of data
- High lights
  - Primary standard for audio/video transport
  - Jitter Compensation
    - Variable delay between packets is called Jitter
  - Detection of out of sequence packets
  - Possibility for multicasting
  - Transport via UDP or TCP
  - Secure version available SRTP with encryption

# RTP

- Real-time multimedia streaming
  - Can tolerate small amounts of lost packets
    - Rule of thumb in VoIP is up to 1% packet loss
    - Receiver uses an error concealment algorithm
  - Receiver uses small jitter-buffer
    - Delays playback a little allowing a little jitter
    - 20 mS often used in VoIP
  - Requires an optional signalling protocol
    - Such as SIP or H.323 to establish a session

# RTP

- Most implementations of RTP uses UDP
  - If data is not present in the “magic” moment the data is useless
- Reliability and timely delivery is the responsibility of the IP network
  - QoS necessary for good voice quality

| Ethernet / frame header | IP header 20 bytes | UDP header 8 bytes | RTP header 12 bytes | Payload (Voice,video …) 20 -160 bytes |
|---|---|---|---|---|

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# RTP header

- The RTP header consist minimum of 12 bytes
  - 32 bits or 4 bytes in each line

Bit 0 2 3 4 8 9 16 31

| | V | P | X | CSRC | M | Payload Type | Sequence number |
|---|---|---|---|---|---|---|---|
| Minimum RTP header | Time stamp | | | | | | |
| | Synkronisation Source (SSRC) Identifier | | | | | | |
| | Contributing Source (CSRC) identifiers. (Optional) | | | | | | |
| | Extensions (Optional) | | | | | | |

# RTP header

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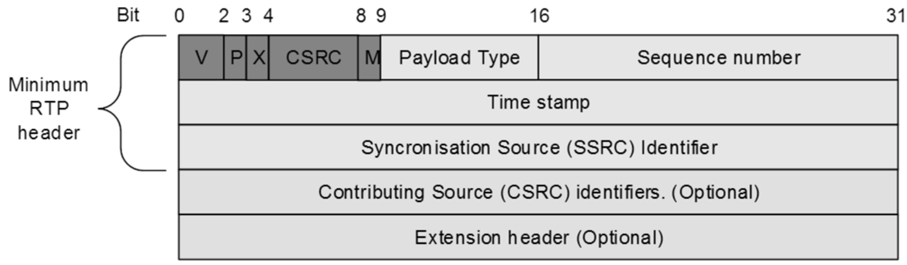
| Field | Bits | |
|---|---|---|
| V | 2 | Version. Current standard 2 |
| P | 1 | Padding – Indicating extra fill bytes to fill up a block to a certain size |
| X | 1 | Optional eXtension header present |
| CSRC | 4 | Optional CSRC present count (1 to 15) |
| M | 1 | Marker – Indicating this packet has special meaning.<br>For example can be used to synchronize clock rates between peers |




# RTP header

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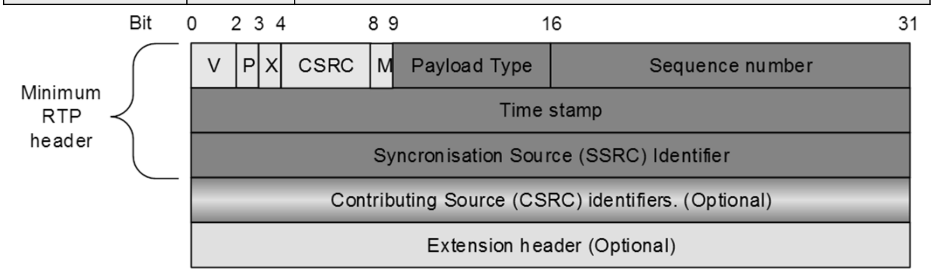
| Field | Bits | |
|---|---|---|
| Payload type | 7 | Which kind of data carried – Codec |
| Sequence nr. | 16 | Incremented in each packet. Used by the receiver to detect packet loss and restore packet order |
| Time stamp | 32 | Used by the receiver to start playback at appropriate time |
| SSRC | 32 | Uniquely identifies the source of a particular stream |
| CSRC | ´n X 32 | Optional field identifying if more contributing sources<br>(For example a voice conference) |



## RTP media stream example

- VoIP conversation profile
  - G.711 a-Law codec used
  - RTP time interval 20 mS = 0,02 second
- a-Law bit rate = 64.000 bps = 8.000 Bps
- RTP payload data pr. Packet
  - 8000 bytes/second * 0,02 second = 160 bytes
- Packet size: 20 + 8 + 12 + 160 = 200 Bytes
  - Without OSI layer 2 overhead (ethernet frame)

| **IP** header 20 bytes | **UDP** header 8 bytes | **RTP** header 12 bytes | Payload a-Law 160 bytes |
|---|---|---|---|

## RTP media stream example

- 0,02 second between packets
  - 1 / 0,02 = 50 packets pr. Second
- Each packets is 200 bytes
  - 50 packets * 200 byte * 8 = 80.000 bps
- VoIP streams can be bidirectional
  - Network load = 2 * 80.000 bps = 160.000 bps

| **IP** header 20 bytes | **UDP** header 8 bytes | **RTP** header 12 bytes | Payload a-Law 160 bytes |
|---|---|---|---|

## Silence suppression

- Voice conversations are often half duplex
  - One listen while the other talks
- No reason for sending RTP packets with background noise only
  - Silence suppression stops RTP transmission
    - Silence suppression is an option
  - Receiving phone makes comfort noise
    - Speaker has a feeling of the connection
- Saves bandwidth – important in trunks

## RTP family

- RTP family of protocols

| Protoc ol | Description |
|---|---|
| RTP | End-to-end transfer of real-time stream data (Unencrypted) |
| RTCP | Provides statistics and control information for an RTP media stream |
| SRTP | End-to-end transfer of real-time stream data (Encrypted) |
| ZRTP | Cryptographic key-exchange protocol between VoIP end-points |

| Protocol | Name | Date standardized |
|---|---|---|
| RTP | Real-time Transport Protocol | 1998 |
| RTCP | Real-time Transport Control Protocol | 1998 |
| SRTP | Secure Real-time Transport Protocol | 2004 |
| ZRTP | Zimmerman Real-time Transport Protocol | 2006 |

## RTCP

Real-time Transport Control Protocol

- Related to RTP
- Provides statistics and control information for an RTP media stream
- Sender report – SR
  - The RTP sender periodically sends SR reports including the sent and received RTP packets
- Receiver report – RR
  - Send by receive-only participants. Those who do not transmit RTP packets.

## SRTP

Secure Real-time Transport Protocol

- A version of RTP providing security
- Encryption
  - Only the intended receiver can read the data
- Authentication
  - Identity check of other party
- Integrity
  - Data cant be changed in transit
- Anti replay
  - Data cant be recorded and retransmitted

## SRTP

Secure Real-time Transport Protocol

- Uses standard security features
  - Defaults to AES for encryption
  - HMAC-SHA1 for authentication and integrity
- The security standards are out of scope in this course module
- A new version of RTCP providing security
  - SRTCP – Secure Real-time Control Protocol

## ZRTP

Zimmermann Real-time Transport Protocol

- Cryptographic key-agreement protocol
- Can be used without a PKI or certificates
  - Public Key Infrastructure
- Uses RTP as transport channel
- signalling protocol unaware of ZRTP
- Negotiate keys between end-points
  - Uses Diffie-Hellman key exchange

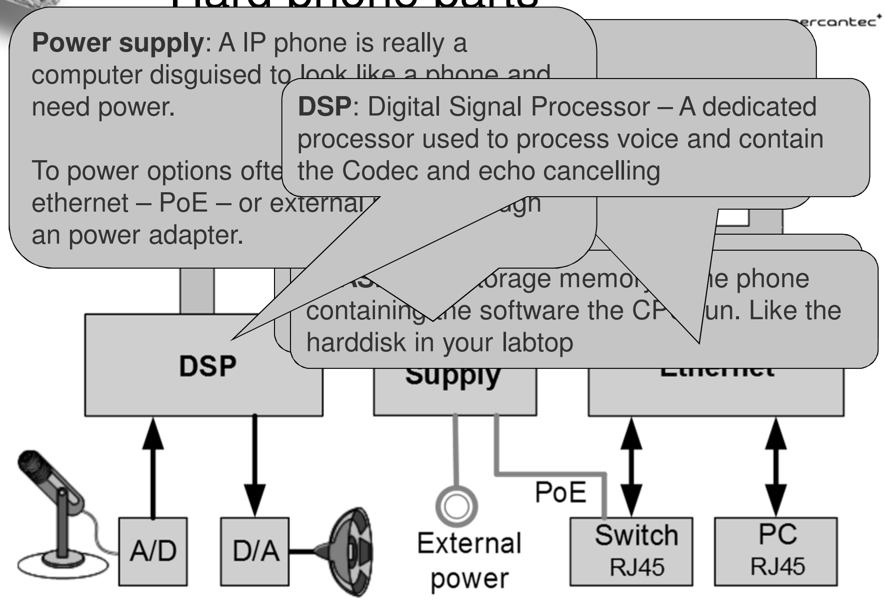
# THE IP PHONE

Hard and soft phones

## IP phones

- Soft, hard, wired and wireless
- Standardized signalling protocols
  - For example H.323 and SIP
- Vendors add on functionality
  - Not 100% comparability between vendors
- Configuration of IP phones
  - Manually
  - Automatically – Vendor specific configuration

# Hard phone parts

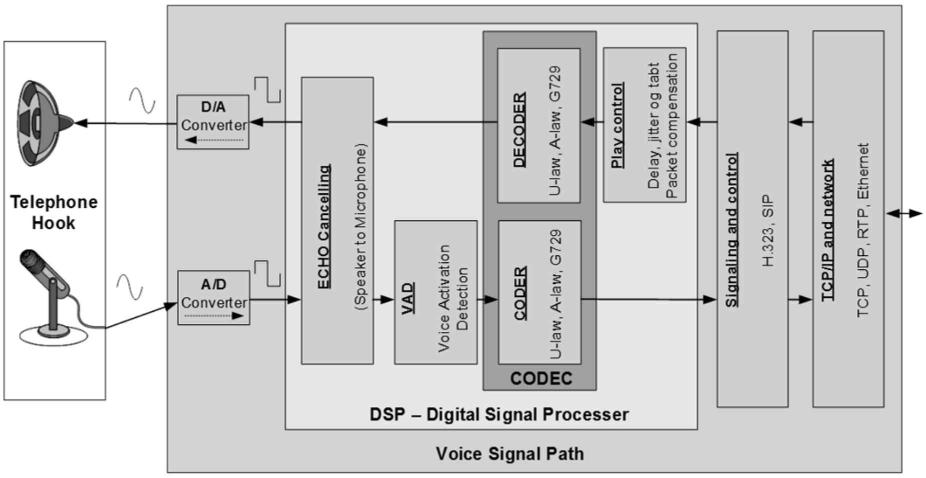



# DSP

## Digital Signal Processor

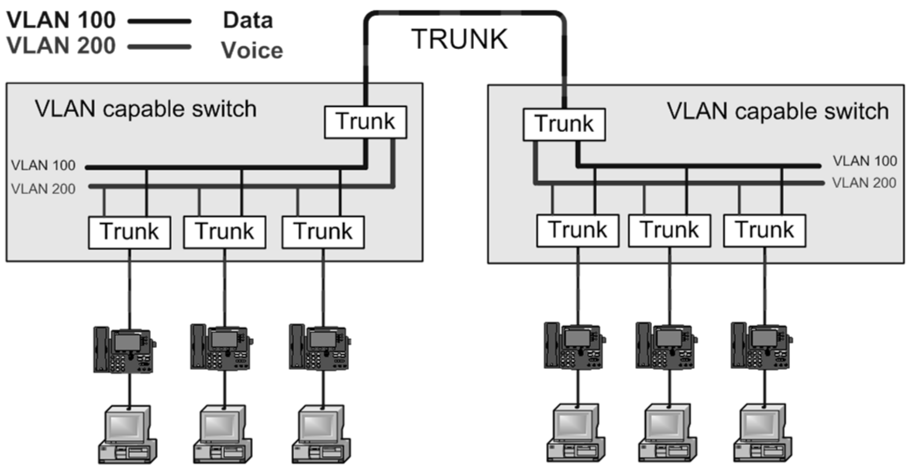
- A DSP is a microprocessor designed to process analog signal such as voice and video
- The analog signal has to be converted to digital by a A/D converter. (Analog to Digital converter)
- The DSP offload the CPU for working with the signal processing.
- A normal CPU can be used for signal processing
  - CPU's are not optimized for signal processing
  - All running programs share the same CPU. (Jitter)
- CPU's in PC's use Intels MMX instruction set
  - Special instructions for multimedia processing
- A dedicated DSP guaranties processing

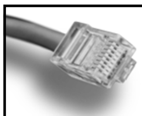
## DSP – Voice signal Path




## Voice VLAN

– Data traffic carried in VLAN 100
– Voice traffic carried in VLAN 200



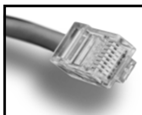


# PoE
## Power over Ethernet

- Most IP phones are powered by 48 Vdc
- Many IP phones can get power from
  - External power supply connected to mains
  - From switches capable of delivering power
    - PoE or Power over Ethernet
    - Picture below is a partial printout from a PoE capable switch

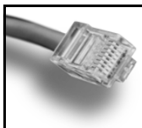
```
mars.tekkom.dk - PuTTY
Campus1#show power inline
Available:280.0(w)  Used:44.1(w)  Remaining:235.9(w)

Interface Admin  Oper       Power   Device              Class Max
                            (Watts)
--------- ------ ---------- ------- ------------------- ----- ----
Fa0/1     auto   on         6.3     IP Pnone 7940       2     15.4
Fa0/2     auto   on         6.3     IP Pnone 7940       2     15.4
Fa0/3     auto   off        0.0                         -     15.4
```



# PoE
## Power over Ethernet

- Two standards
  - 802.3af supplying device with up to 12,95 Watt
  - 802.3at supplying device with up to 25,50 Watt
- Can be used with 10,100 and 1000 Mbps ethernet
- Some vendors invented their own version of PoE before IEEE standards
  - IEEE 802.3af from 2003
  - IEEE 802.3at from 2009 also called PoE plus



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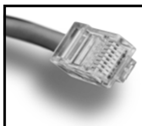
# PoE
## Power over Ethernet

| Pin RJ45 | T568A RJ45 | Pair | 10/100 Spare Mode B | 10/100 mixed Mode A | 1000 Mbps Mode B | 1000 Mbps Mode A |
|---|---|---|---|---|---|---|
| 1 | | 3 | Rx+ | RX+ DC+ | TxRx A+ | TxRx A+ DC+ |
| 2 | | 3 | Rx- | RX- DC+ | TxRx A- | TxRx A- DC+ |
| 3 | | 2 | Tx+ | TX+ DC- | TxRx B+ | TxRx B+ DC- |
| 4 | | 1 | DC+ | Unused | TxRx C+ DC+ | TxRx C+ |
| 5 | | 1 | DC+ | Unused | TxRx C- DC+ | TxRx C- |
| 6 | | 2 | Tx- | TX- DC- | TxRx B- | TxRx B- DC- |
| 7 | | 4 | DC- | Unused | TxRx D+ DC- | TxRx D+ |
| 8 | | 4 | DC- | Unused | TxRx D- DC- | TxRx D- |

| Abbr. | Explanation |
|---|---|
| Rx+, Rx- | Receive plus and minus signal (Differential on pair) |
| Tx+, Tx- | Transmit plus and minus signal (Differential on pair) |
| DC+, DC- | PoE DC delivered common mode in pair. Plus or minus in pair. |



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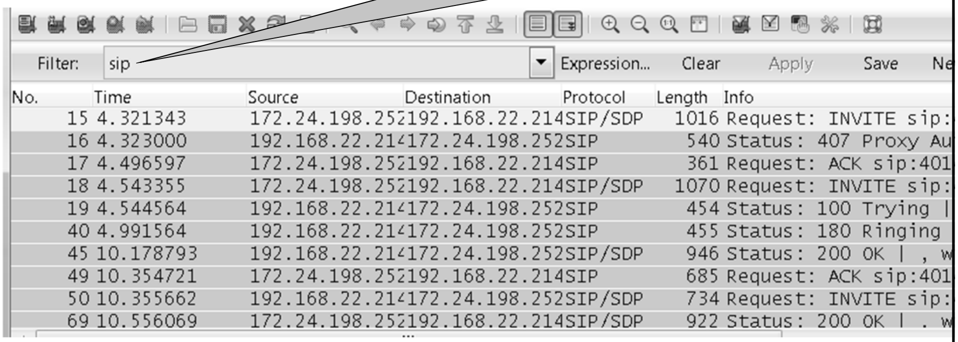
# WIRESHARK

WIRESHARK

Packet analyzer

# Packet filters

SIP packet filter –showing all SIP packets




| No. | Time | Source | Destination | Protocol | Length | Info |
|---|---|---|---|---|---|---|
| 15 | 4.321343 | 172.24.198.252 | 192.168.22.214 | SIP/SDP | 1016 | Request: INVITE sip: |
| 16 | 4.323000 | 192.168.22.214 | 172.24.198.252 | SIP | 540 | Status: 407 Proxy Au |
| 17 | 4.496597 | 172.24.198.252 | 192.168.22.214 | SIP | 361 | Request: ACK sip:401 |
| 18 | 4.543355 | 172.24.198.252 | 192.168.22.214 | SIP/SDP | 1070 | Request: INVITE sip: |
| 19 | 4.544564 | 192.168.22.214 | 172.24.198.252 | SIP | 454 | Status: 100 Trying \| |
| 40 | 4.991564 | 192.168.22.214 | 172.24.198.252 | SIP | 455 | Status: 180 Ringing |
| 45 | 10.178793 | 192.168.22.214 | 172.24.198.252 | SIP/SDP | 946 | Status: 200 OK \| , w |
| 49 | 10.354721 | 172.24.198.252 | 192.168.22.214 | SIP | 685 | Request: ACK sip:401 |
| 50 | 10.355662 | 192.168.22.214 | 172.24.198.252 | SIP/SDP | 734 | Request: INVITE sip: |
| 69 | 10.556069 | 172.24.198.252 | 192.168.22.214 | SIP/SDP | 922 | Status: 200 OK \| . w |

- Use wireshark to find interesting statistics about a SIP conversation

# Opgave

- Find minimum 7 default Feature Codes I FreePBX og beskriv hvad de bruges til i et Word/Writer dokument.
- De 2 af dem skal være *43 & *65
- Opret et hovednummer, og lav en announcement