VoIP - Basics and Protocols

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What is VoIP?

Transmission of **voice** and **signaling**

via data networks

using Internet Protocol (IP),

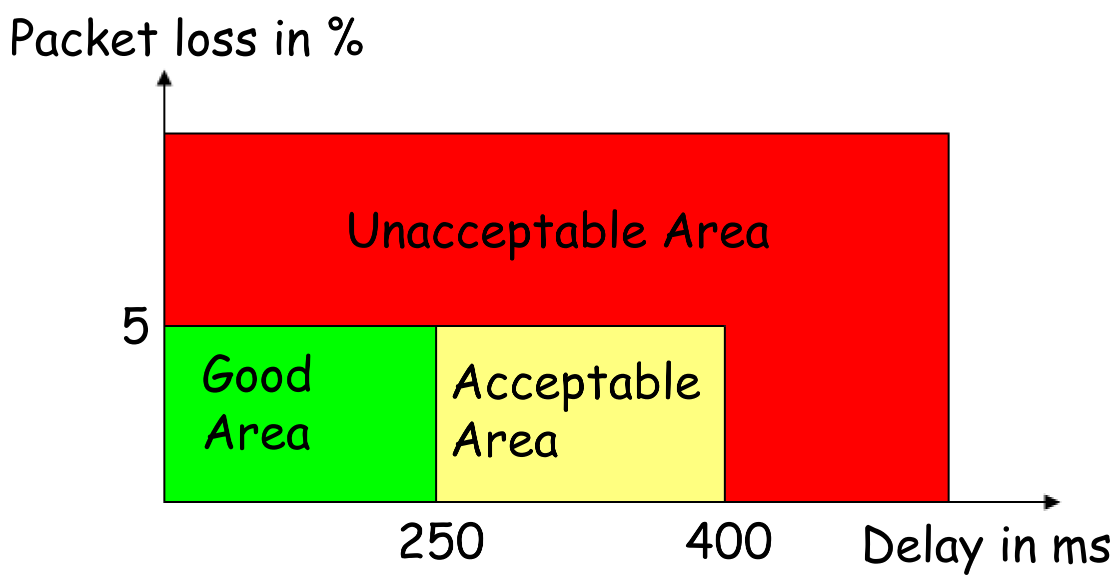
i.e. via so-called **IP netwoks** (Internet, Intranets)

Requirements for VoIP networks (1)

- Requirements from the user's point of view
  - Voice quality
  - Reachability
  - Handling
  - Eavesdropping Security
  - Availability
  - Performance Characteristics
  - Flexibility

Requirements for VoIP networks (2)

- Requirements from a technical point of view
  - Quality of Service (QoS)
  - Delay
  - Jitter
  - Echo
  - Packet losses
  - Distortions



VoIP Signaling

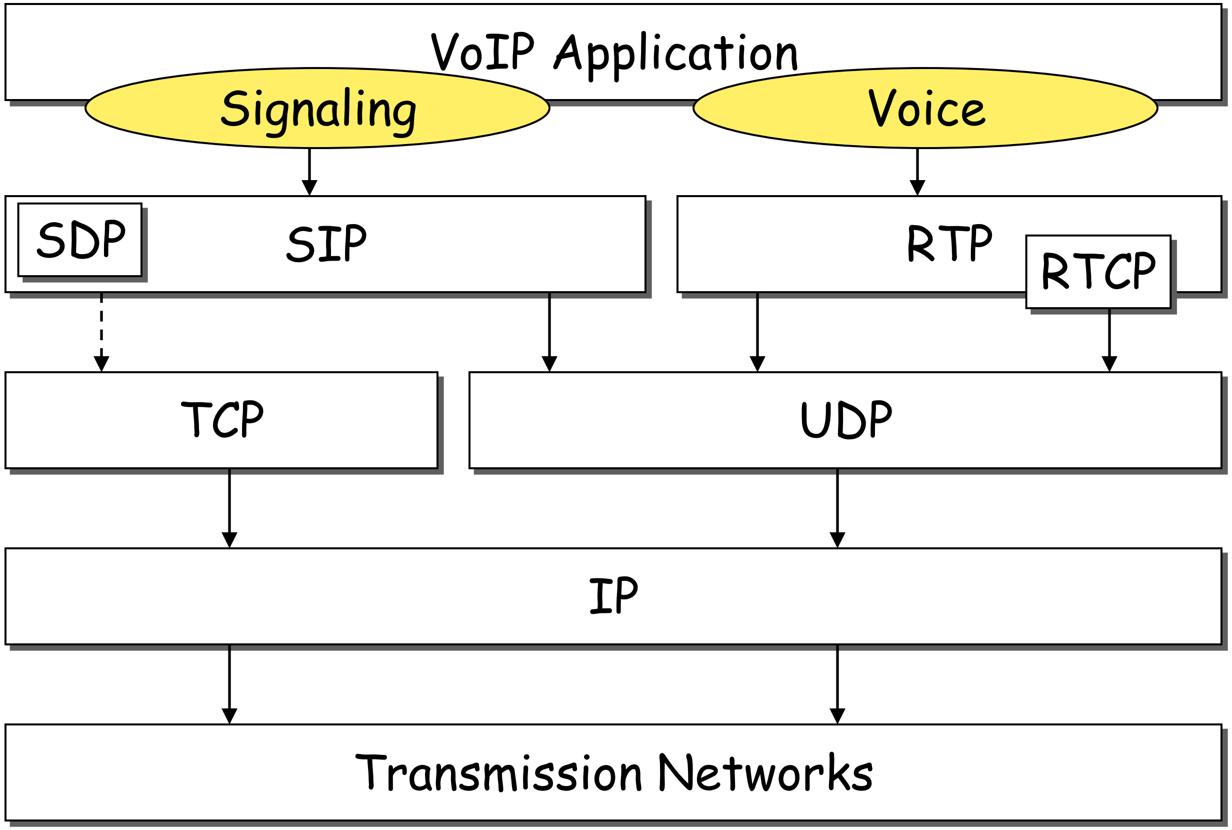
- Signaling
  - Session Initiation Protocol (SIP), [IETF]
  - H323, [ITU]
  - Scinny Client Control Protocol (SCCP), [Cisco]
  - Inter Asterisk Exchange Protocol (IAX™), [Digium]

- Protocols for the transmission of multimedia / voice data
  - Real Time Protocol (RTP) and
  - Real Time Control Protocol (RTCP), [IETF]

Session Initiation Protocol (SIP)

- Components
  - User Agent Client (UAC)
  - User Agent Server (UAS)
  - Proxy (stateless oder statefull)
  - Redirect Server
  - Registrar Server
  - Location Server

Session Initiation Protocol (SIP)



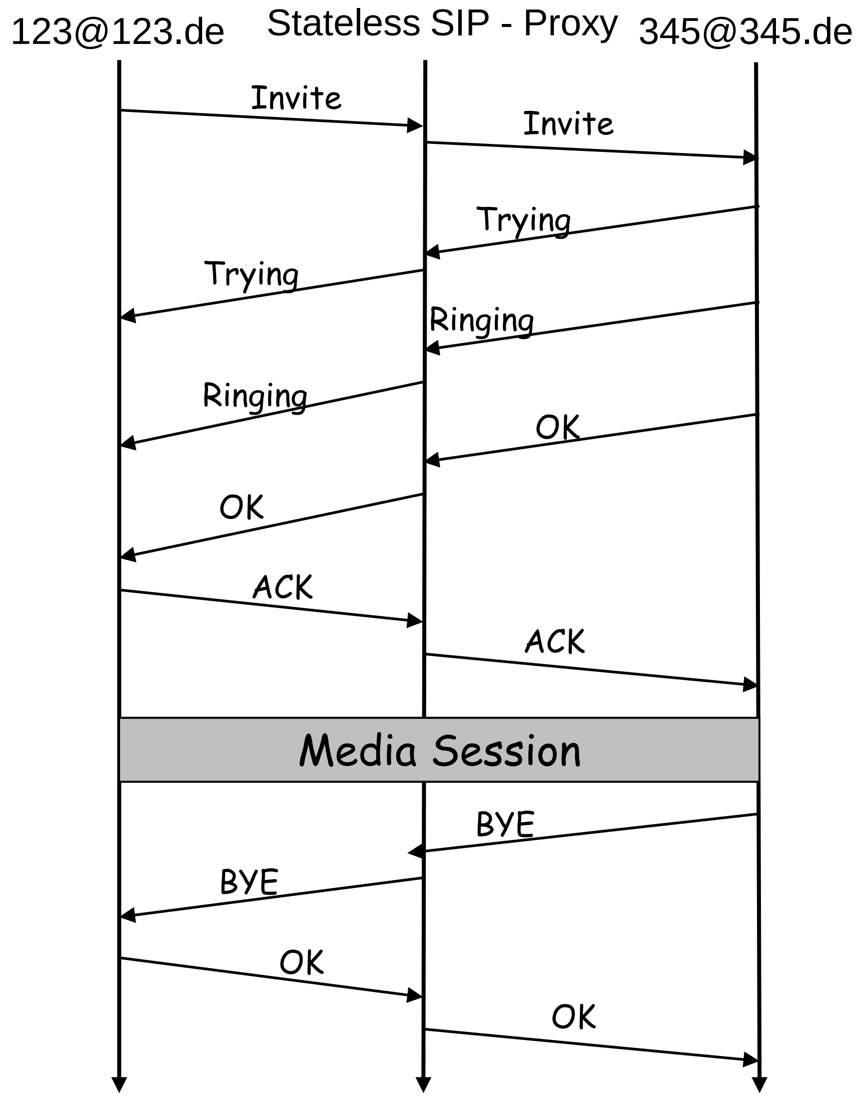
Classification of SIP and RTP

SIP: Session Initiation Protocol
SDP: Session Description Protocol
RTP: Real Time Transport Protocol
RTCP: Real Time Control Protocol

SIP - Message

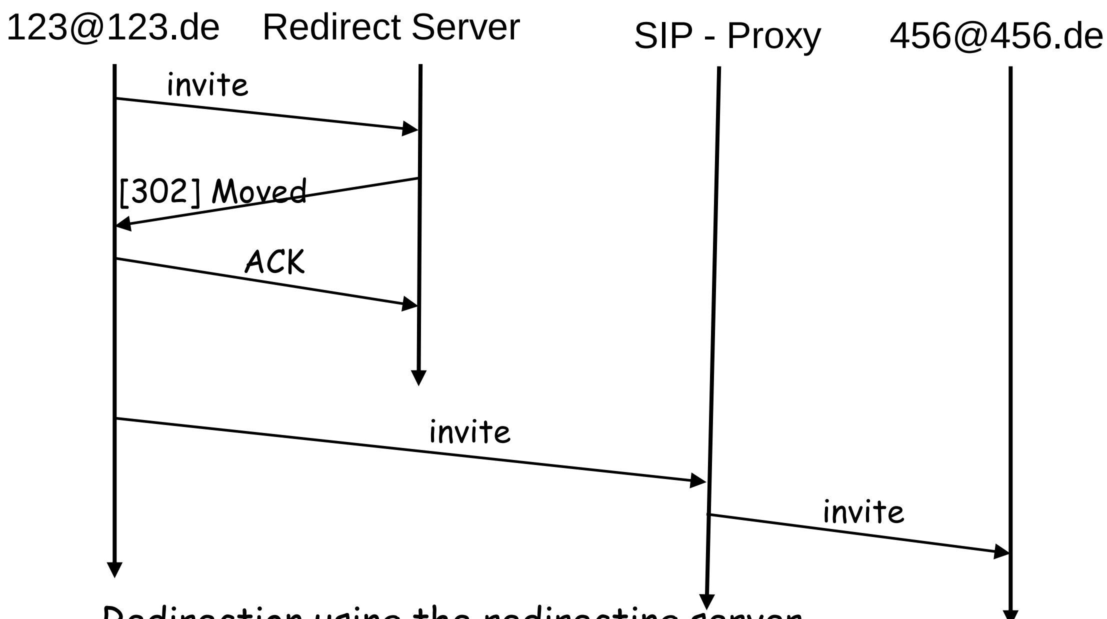
```
Session Initiation Protocol
    Request-Line: INVITE sip:10000@sipgate.de SIP/2.0
        Method: INVITE
        [Resent Packet: False]
    Message Header
        Via: SIP/2.0/UDP 192.168.178.102:5060;rport;branch=z9hG4bK60846
        Max-Forwards: 70
        To: <sip:10000@sipgate.de>
        From: <sip:user@sipgate.de>;tag=z9hG4bK66687806
        Call-ID: 841479524486@192.168.178.102
        CSeq: 1 INVITE
        Contact: <sip:user@192.168.178.102
        Expires: 3600
        User-Agent: mjsip stack 1.6
        Content-Length: 159
        Content-Type: application/sdp
    Message body
        Session Description Protocol
            Session Description Protocol Version (v): 0
            Owner/Creator, Session Id (o): <sip:user@sipgate.de> 0 0 IN IP4 192.168.178.102
            Session Name (s): Session SIP/SDP
            Connection Information (c): IN IP4 192.168.178.102
            Time Description, active time (t): 0 0
            Media Description, name and address (m): audio 21000 RTP/AVP 0
            Media Attribute (a): rtpmap:0 PCMU/8000
```

SIP - Signalisierung



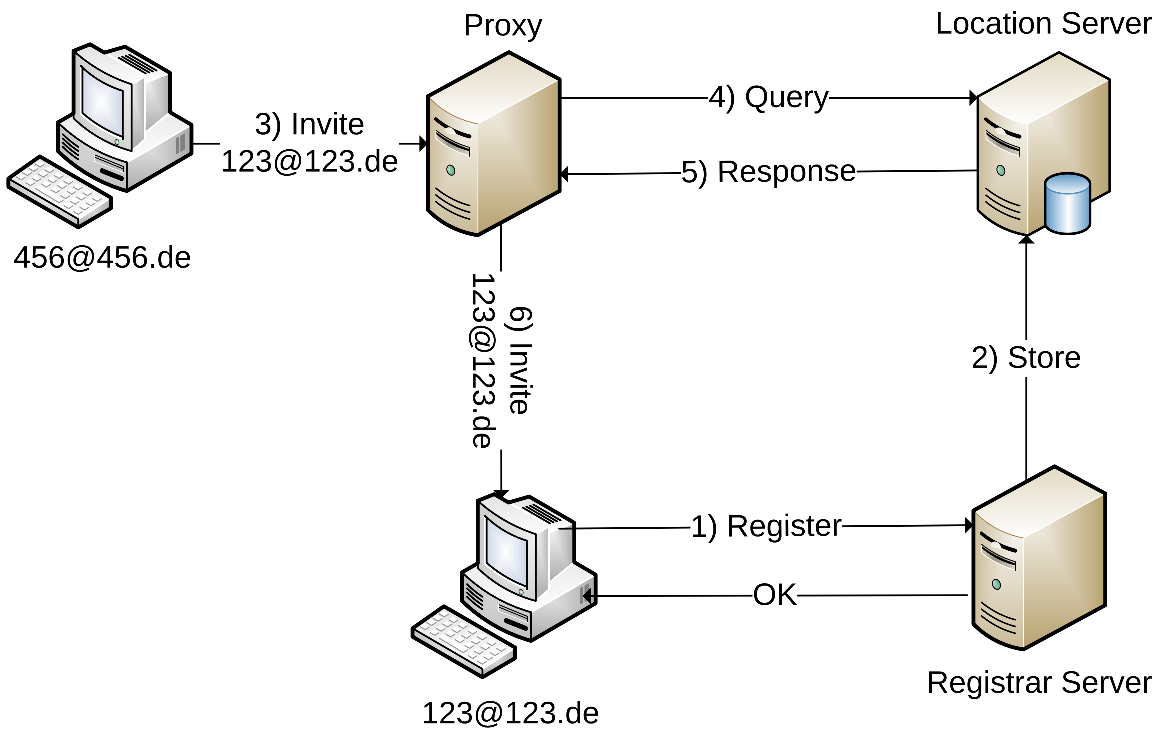
Exemplary process

SIP - Signalisierung (2)



Redirection using the redirecting server

SIP - Signalisierung (3)



Registration und Discovery

SIP – Free Software

- Free Software Clients
  - Ekiga (Unix, Windows)
  - Sipgate (Windows)
  - Kphone (Unix)

Real-Time Transport Protocol (RTP)

- Why not TCP?
  - TCP offers reliable, in-sequence data transfer
  - TCP embeds flow/error/congestion control
- Why not UDP?
  - Missing sequence control and not reliable!
- RTP: transport protocol for multimedia traffic
  - application level framing; integrated layer processing
  - usually RTP/UDP/IP, or RTP/IP
  - RTP does NOT guarantee real-time itself!

Real-Time Transport Protocol (RTP) (2)

- <u>RTP Data Transfer Protocol</u>: It carries real-time data.

- <u>RTP Control Protocol (RTCP)</u>: It monitors the quality of service and conveys information about the participants.

RTP Applications (1)

- Simple Multicast Audio Conference
  - The audio-conferencing application used by each conference participant sends audio data in small chunks of, say, 20 ms duration.
  - Each chunk of audio data is preceded by an RTP header; RTP header and data are in turn contained in a UDP packet.
  - The RTP header indicates what type of audio encoding (such as PCM, ADPCM or LPC) is contained in each packet.

RTP Applications (2)

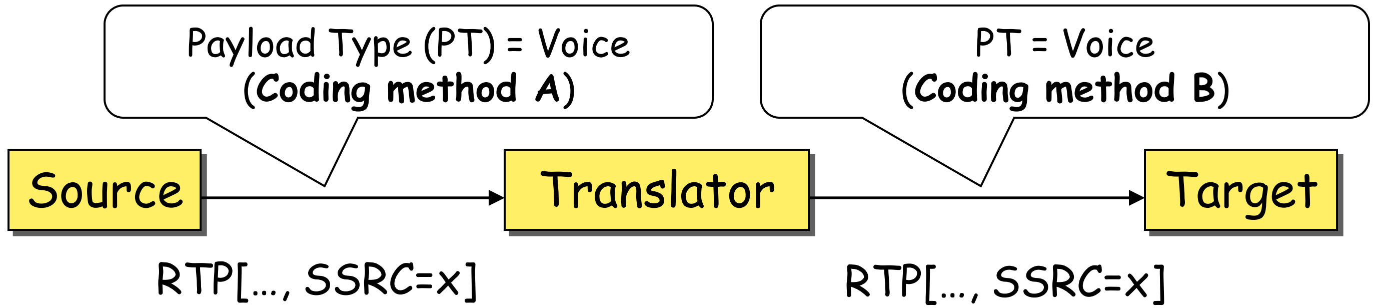
- RTP header contains timing information and a sequence number that allow the receivers to reconstruct the timing produced by the source.
- The sequence number can also be used by the receiver to estimate how many packets are being lost.
- the audio application in the conference periodically multicasts a reception report plus the name of its user on the RTCP port. The reception report indicates how well the current speaker is being received.
- A site sends the RTCP BYE packet when it leaves the conference.

RTP Translators and Mixers

- *A* mixer combines several media stream into a one new stream (with possible new encoding)
  - reduced bandwidth networks (video or telephone conference)
  - appears as new source, with own identifier
- *A* translator changes encoding from one format to another single media stream
  - may convert encoding

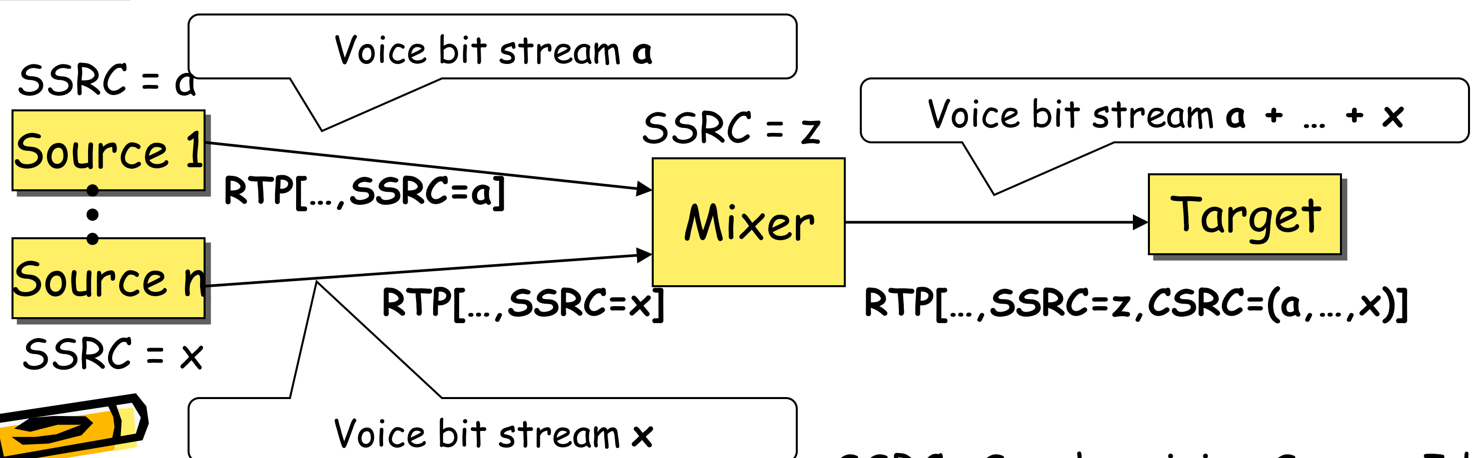
Translator und Mixer

Translator



A – Law
μ - Law

Mixer



SSRC= Synchronizing Source Identifier
CSRC= Contribution Source Identifier

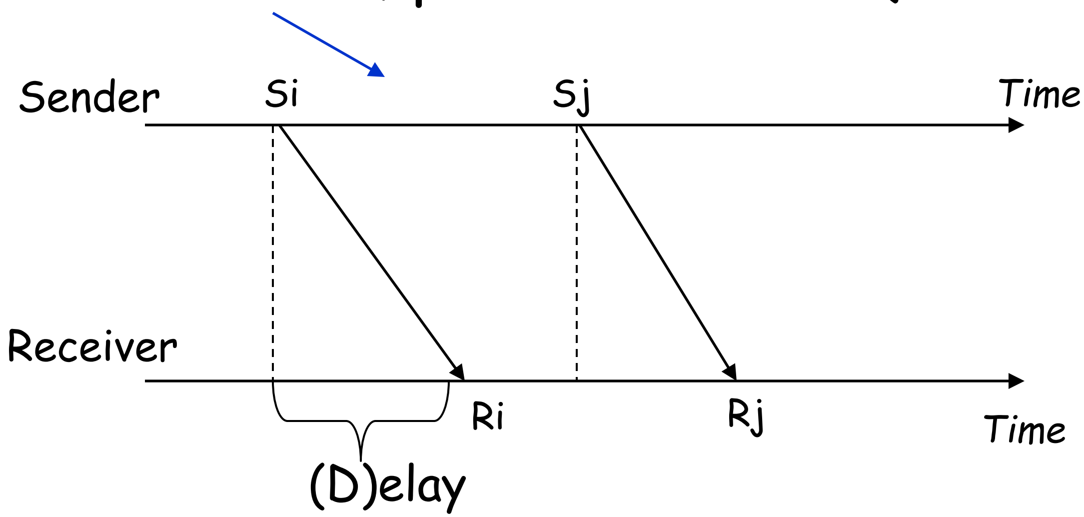
Real-Time Transport Protocol - Header

```
 0                   1                   2                   3
 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1 2 3 4 5 6 7 8 9 0 1
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|V=2|P|X|  CC   |M|     PT      |       sequence number         |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|                           timestamp                           |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
|           synchronization source (SSRC) identifier            |
+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+=+
|            contributing source (CSRC) identifiers             |
|                             ....                              |
+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+-+
```

V: Version
P: Padding
X: Extension
CC: Contribution Source Identifier Count
M: Marker

Real-Time Control Protocol (RTCP)

- An integral part of RTP
- Current properties of the RTP packet stream, e.g. Jitter, packet loss → QoS monitoring



$$D(i,j) = (Rj - Ri) - (Sj - Si) = (Rj - Sj) - (Ri - Si)$$

$$J(i) = J(i-1) + (|D(i-1,i)| - J(i-1))/16$$

$$= (15/16)\ J(i-1) + (1/16)\ |D(i-1,i)|$$

Typical VoIP Traffic Overhead

- To calculate the typical VoIP workload, it is necessary to understand what steps are taken to convert an analog signal to a digital one.
- The objective is that a VoIP terminal ultimately generates a digital signal from an analog sound signal.
- As input, the VoIP terminal receives a sound signal s(t), which is of an analog nature, and provides at the output a signal g(t), which is in binary form.

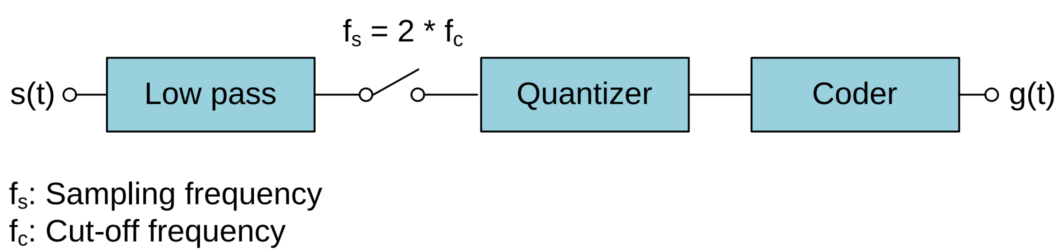


Figure 1. Conversion of an analog signal into a digital signal.

Typical VoIP Traffic Overhead (2)

- If an analog signal s(t) arrives, first a band limitation takes place. Particularly, the analog signal is narrowed in its frequency spectrum, provided that it can be reconstructed as clearly as possible afterward.

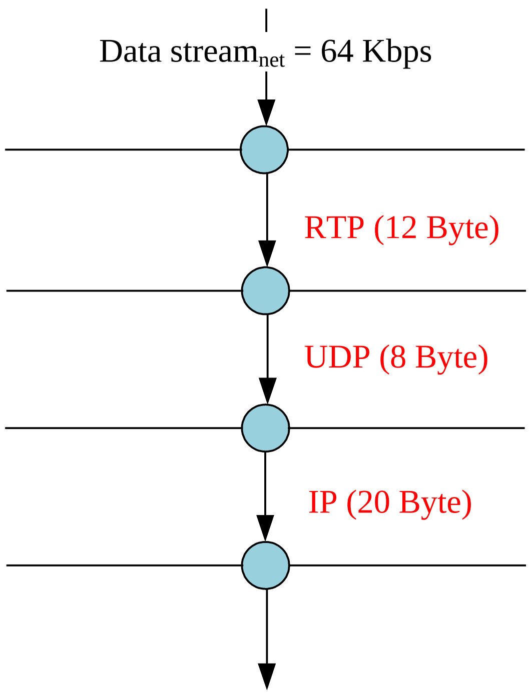
- This band limitation is caused by an element called a low pass. This maximum band-limited signal, also referred to as the cut-off frequency fc, is determined by speech intelligibility.

- The conventional telephony has a cutoff frequency of 4 kHZ.

Typical VoIP Traffic Overhead (3)

- The steps to be taken to convert this band-limited signal s(t) into a binary stream g(t) can be outlined as follows:
  - **Sampling:** The first step is to convert the signal from a time- and value-continuous form into a time-discrete form. For this purpose, the analog signal is sampled at twice the cutoff frequency fc.
  - **Quantization:** Thereafter, the sampled signal must be quantized to convert it from the time-discrete, but value-continuous form into a time- and value-discrete form.
  - **Coding and compression:** The time- and value-discrete signal must finally be coded and, if necessary, compressed.

Typical VoIP Traffic Overhead (4)



f_c = 4 kHz

f_s = 8 kHz

Coding: A-law à 8 bit per sample value

=> Data stream$_{net}$ = 64 kbps

=> Packetization time: 20 ms

=> Packet size = 160 Byte *(without header)*

=> Packet size = 160 Byte + 40 Byte = 200 Byte *(including header)*

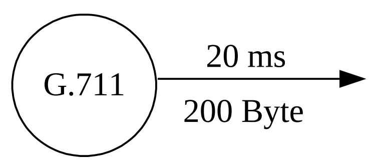


Figure 2. Additional overhead caused by TCP/IP protocol stack regarding VoIP

=> The overhead of 40 Byte in relation to 160 Byte, which one would only have to transfer in the classical telephony, shows the immense overhead caused by VoIP.

End of Presentation

Thank you for your attention!