VoIP & SIP

Summary

- What is VoIP
- Real Time Protocol
- SIP
  - Servers, proxies, registrars
  - NAT traversal
  - mobility
  - Android support

Voice over IP (VoIP)

Sinnreich
Ch 18

- **Did you know that :**
  - **Most telephony today is transported with VoIP?**
  - **Most PBX-es installed use VoIP?**
  - **In Ethiopia, Oman VoIP use is a criminal offence?**
  - **Hangouts, Whatsapp, Skype, TeamSpeak, TeamViewer, Viber, Yahoo Mesg all use VoIP?**

Seven Myths About VoIP

1. VoIP is free
2. The only difference between VoIP and regular telephony is the price
3. Quality of service isn’t an issue nonadways, because there’s plenty of bandwidth in the network
4. VoIP can’t replace regular telephony, because it still can’t guarantee quality of service
5. VoIP is just another data application
6. VoIP isn’t secure
7. A Phone is a Phone is a Phone

What is VoIP?

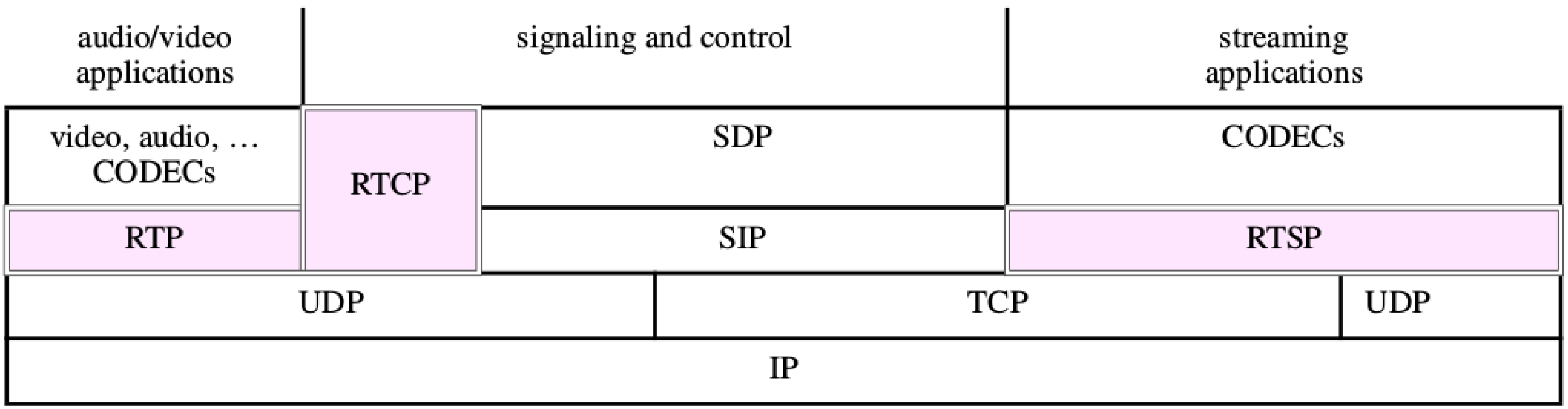
- VoIP is an end-to-end architecture
  - Voice transported in IP packets
- Comparison with PSTN
  - Circuit switch vs. Packet switch
  - Latency
  - Dataplane, control plane
  - Mobility
- VoIP headsets
  - Physical, Software
  - Built into Android

Packet Encapsulation

- **Encapsulate 10-20ms of speech in a packet**

IP	UDP	RTP	Voice
20	8	12	20-240

- **RTP, RTCP, RTSP**



Packet Encapsulation

RTP datagram

Version, flags & CC	Payload Type	Sequence Number	Timestamp	Synchronization Source ID	CSRC ID (if any)	Codec Data
1	1	2	4	4	0-60	0-1460

UDP datagram

Source Port Number	Destination Port Number	UDP length	UDP checksum	Data
2	2	2	2	0-1472

IP packet

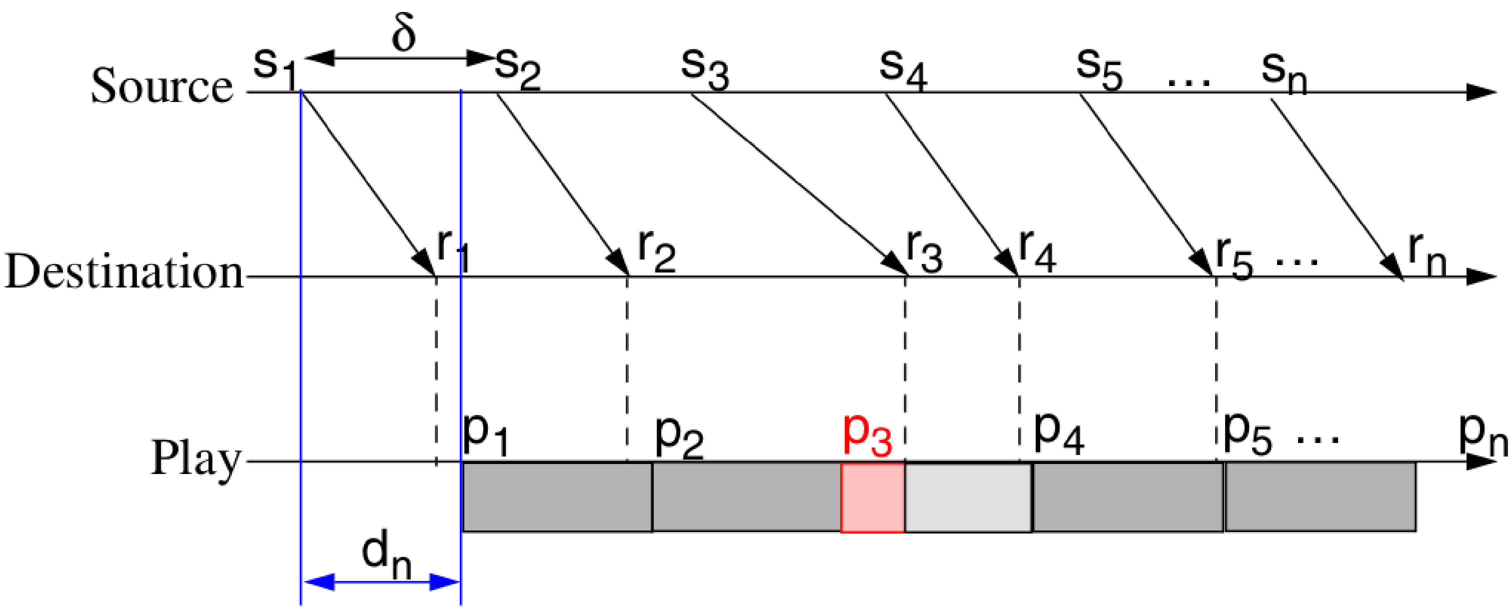
Version & header length	TOS	Total Length	Packet ID	Flags & Frag Offset	TTL	Protocol	Header Checksum	Source Address	Destination Address	Options (if any)	Data
1	1	2	2	2	1	1	2	4	4	0-40	0-1480

Ethernet Frame

Inter-frame gap	Preamble	Start of frame delimiter	Destination Address	Source Address	Length or Ethertype	Data	Pad	Checksum
12	7	1	6	6	2	0-1500	0-46	4

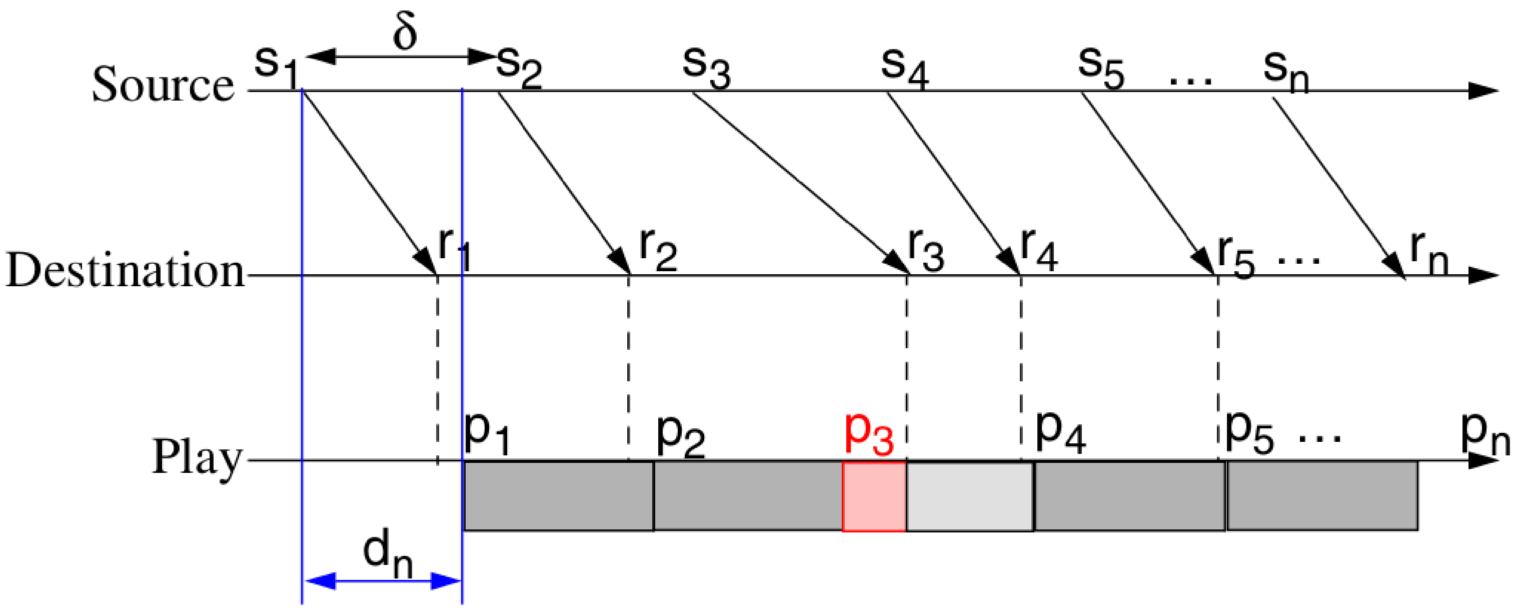
Realtime delivery

- Realtime app = maintain time relationship at receiver
  - Play in same order as original (sequence number)
  - Play time to reproduce original (time stamp)
  - Once decided p1, <u>all packets have deadlines</u>!



Jitter = $(r_i - r_{i-1}) - (s_i - s_{i-1})$

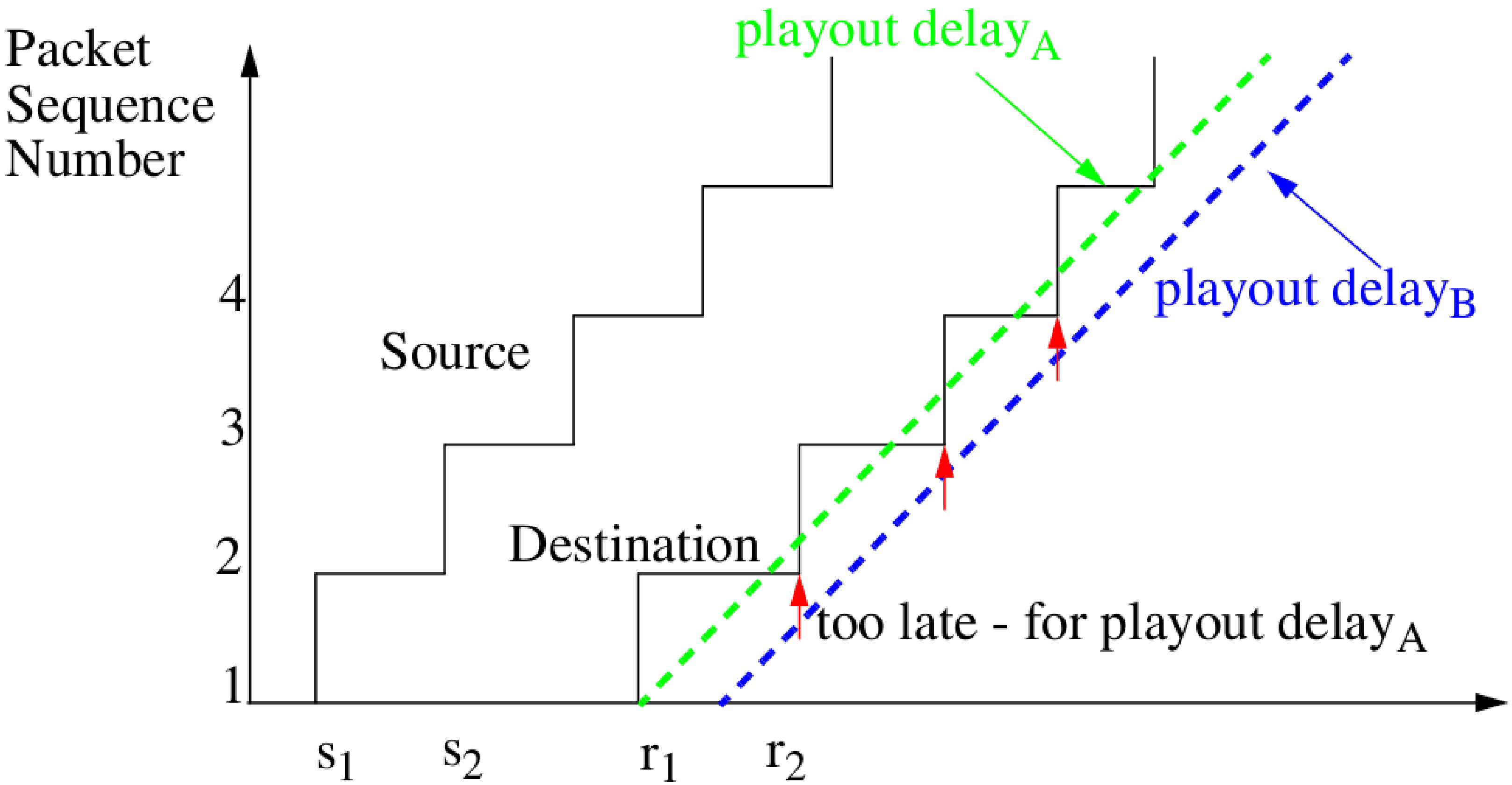
Realtime delivery



- What is jitter?
  - Packet delay variance $= (r_i - r_{i-1}) - (s_i - s_{i-1})$
  - Negative jitter: late packet
  - Positive jitter: early packet
- How to shield listener from jitter?
  - Playout buffer (extra delay)

Dealing with jitter

- **Playout buffer = delay at receiver to smooth jitter**



Delay and jitter

- Audio end-to-end delay components
  - encoding
  - packetization
  - transmission
  - propagation, switching/routing
  - receiving
  - decoding
  - dejittering
  - playing

sender

network

receiver

playout buffer ADDS delay

Voice quality metrics

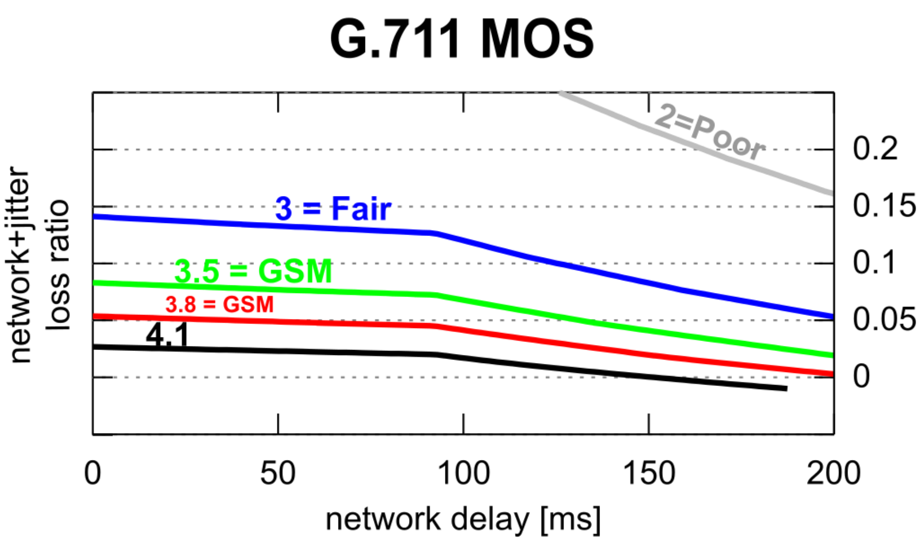
- Listening quality
- Conversational quality
- Network quality
  - Delay, loss, jitter
  - Delay limits
    - < 150ms acceptable
    - < 400ms tolerable
    - > 400ms unacceptable
- Mean Opinion Score (MOS)
  - Excellent = 5, Good = 4, Fair = 3, Poor = 2
  - Functions derived using human listeners to assign MOS to a given (loss,delay) conversation[1]

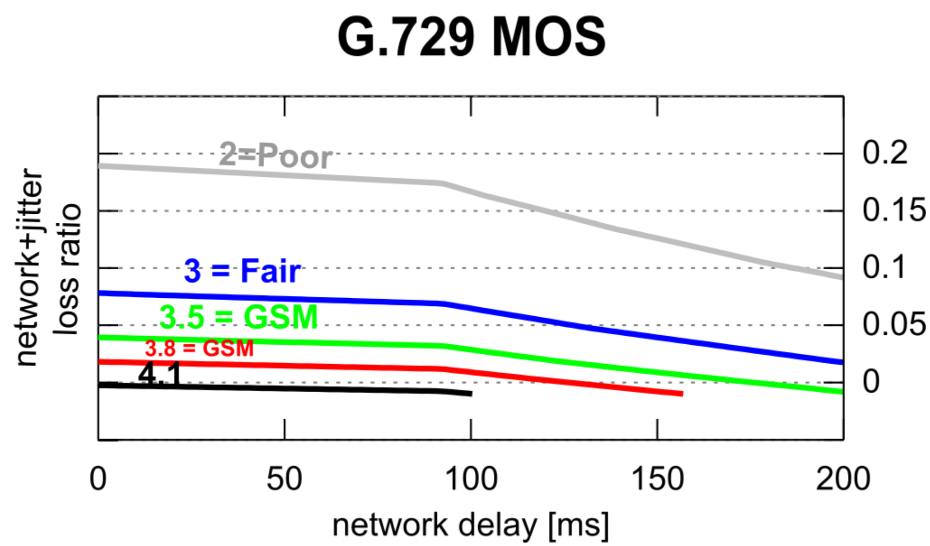
[1] Cole , Rosenbluth "Voice over IP Performance Monitoring", http://ccr.sigcomm.org/archive/2001/apr01/ccr-200104-cole.pdf

voice codecs

Codec	bitrate	Framesize [ms]	codec Delay[ms]	MOS ideal cond.
G.711	64kbps	10, 20,30	25	4.1
G.729	8kbps	10, 20,30	15, 25,35	3.92
GSM-FR	14kbps	22.5	20	3.5
SILK (skype)	6-40Kbps	20	?	5

MOS(delay, loss)





- Conditions: 25ms vocoder delay, 60ms playout buffer
- Used known MOS(delay,loss) functions to generate curves
- G.729 = high compression, less resilient to loss
- G.711 = needs more bandwidth, more loss resilient