Harvard University
CSCI-40, Communication Protocols and Internet Architectures

Reading Assignment for Lecture 12

We will discuss Voice Over IP (VoIP), SIP, WebRTC and QoS in this lecture.

- In the course textbook Internetworking with TCP/IP Volume One - 6th Edition
  * Read Chapter 26 on Voice and Video over IP. **(Required Reading)**
  Note that our focus in this course is on SIP and not H.323. The textbook describes H.323 and you should understand its historical importance, but we will not study it further.
- The following article provides background information on codecs and is a required reading.
  http://en.wikipedia.org/wiki/Codec
- Read RFC # 3550 on RTP, Read pages 1 – 15. Skim the section of Definitions.
- Read RFC # 3261, SIP: Session Initiation Protocol. **(Required Reading)**
  Read pages 1 – 31. Skim the section of Definitions.
- Required Readings on WebRTC:
  WebRTC is a suite of web-based protocols that enables real-time voice and video communication. It uses HTML5 as a foundation and its focus is on peer-to-peer communication between browsers. Read the introductory material on WebRTC in the following documents; focus on the concepts and topology, not the details of the APIs or the protocols. WebRTC is supported today on all major browsers and on mobile devices.
  https://en.wikipedia.org/wiki/WebRTC
  https://tools.ietf.org/html/draft-ietf-rtcweb-overview-19 (SECTIONS 1 and 2)

- The technical details of Quality of Service (QoS) and Differentiated Services are described in the following RFCs.
  RFC 2475, Sections 1 – 3. Skim the section of Terminology
  RFC 2474, Sections 1 – 5. Skim the section of Terminology.
  RFC 3260, pages 1 - 4

OPTIONAL: Readings on SIP and VoIP
http://datatracker.ietf.org/wg/sipcore/
http://datatracker.ietf.org/wg/clue/
https://tools.ietf.org/wg/rtcweb/
http://www.voip-info.org/
https://blog.opensips.org/
http://www.nojitter.com
http://www.webopedia.com/TERM/G/G_7xx.html (lists various codec's)
http://www.webrtc.org/
https://www.sipit.net/Main_Page

Some historical information on SIP and RTP by Prof. Schulzrinne, who was one of the developers of these protocols:

http://www.cs.columbia.edu/sip/
http://www.cs.columbia.edu/~hgs/rtp/ (Historical information on RTP.)

OPTIONAL: Readings on WebRTC

http://www.webrtc.org/
https://www.w3.org/TR/webrtc/
https://webrtchacks.com/
https://webrtchacks.com/guide-to-safari-webrtc/
https://tools.ietf.org/html/draft-ietf-rtcweb-sdp-11 (SDP for WebRTC, good SDP diagram)
https://tools.ietf.org/html/rfc7657 (Diffserv and RTC)
https://tools.ietf.org/html/draft-ietf-rtcweb-rtp-usage-26 (RTP and WebRTC)
https://tools.ietf.org/html/draft-ietf-tsvwg-rtcweb-qos-18 (DSCP for WebRTC)