Harvard University
CSCI S-40, Communication Protocols and Internet Architectures
Reading Assignment for Lecture 12

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Lecture 12 and lecture 13 discuss Voice Over IP (VoIP), SIP, and Quality of Service (QoS).

- 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 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 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 QoS and Differentiated Services will be discussed in Lecture 14. However, these topics are directly related to VoIP and RTP and so we have included the references on them here. These will be **Required Readings for lecture 14.**
  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.imtc.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 -
http://www.cs.columbia.edu/sip/
Historical information on SIP: Prof. Schulzrinne was one of the developers of the protocol.
http://www.cs.columbia.edu/~hgs/rtp/ (Historical information on RTP.)

http://www.h323forum.org/ (Historical information on H.323, which is still used in some commercial video conferencing systems.)
http://www.packetizer.com/ (site is no longer maintained)

OPTIONAL: Readings on WebRTC
http://www.webrtc.org/
https://www.w3.org/TR/2017/CR-webrtc-20171102/
https://webrtchacks.com/
https://webrtc.ventures/2017/06/webrtc-support-in-safari-11/
https://www.html5rocks.com/en/tutorials/webrtc/infrastructure/
https://tools.ietf.org/html/draft-ietf-rtcweb-sdp-08 (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)
https://www.html5rocks.com/en/tutorials/webrtc/basics/#toc-history
https://webrtchacks.com/hangouts-firefox-webrtc/ (Hangouts and WebRTC)
https://webrtchacks.com/safari-webrtc/ (Safari and WebRTC)
https://s3.amazonaws.com/static.andyet.com/webrtc-reports/facetime-report.pdf (as of 11/2017)

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