Voice over IP – TCOM 631/Sec 001
Spring 2023

COURSE INFORMATION

This subject will introduce you to Voice over IP concepts, theory, and practice. Voice over IP, in short VoIP, refers to the process that analyzes, digitizes, and sends voice information/discrete packets over IP-based (Internet Protocol) systems. It is a technology comprised of applications, systems, networks, and rules that govern its implementation and operation.
Voice is a crucial part of human communication and takes a special place within the digital arena. Voice is the instrument we all play. It must be processed and transmitted efficiently to sound right.
In this course, we will start analyzing building blocks that comprise digitized voice and continue with the discovery of how these combine into modern IP Telephony solutions.

INSTRUCTOR

Instructor: Dragan Hrnjez
E-mail: dhrnjez@gmu.edu
Location: remote/offsite
Office Hours: Tuesdays, 6:30 PM to 7:00 PM, Room 3707
(Please schedule office hour no later than 24 hours in advance). Alternate arrangements can be made. We will discuss these in our first meet up.

TIME AND PLACE

Time: Tuesdays, 7:20 pm - 10:00 pm
Place: TBD

CALENDAR AND SCHEDULE

Class calendar and proposed schedule will be available via Blackboard and at the URL listed below. Any potential changes will be announced and discussed during the semester.

ALTERNATE WEBSITE

http://mason.gmu.edu/~dhrnjez
user id: tcom631, password: voiceIP

COURSE GOALS

- Develop confidence and skills needed to understand VoIP technology.
- Enhance and correlate the knowledge you are going to construct with other telecommunication topics and aspects of the digital world we live in.
- Incorporate your knowledge to real-world scenarios.

- Master technical communication and information exchange.
- Enjoy the ability to learn, interact with your peers and have a good time.

TEXTBOOKs

There are no required books for this course. The following books are optional (used during previous semesters):

Carrier Grade Voice over IP, *McGraw-Hill Education; Richard Swale and Daniel Collins; 3rd Edition; October 2013*
ISBN: 978-0-07-182771-3
This book will be available for purchase in the GMU Bookstore.

Voice over IP Fundamentals; *Cisco Press; Davidson, Peters, Gracely, Bhatia, Kalidindi, Mukherjee; 2nd Edition; July 2006*
ISBN: 1-58705-257-1

IP Telephony: Deploying VoIP Protocols and IMS Infrastructure; *John Wiley & Sons Ltd.; Oliver Hersent; 2011*
ISBN: 978-0-470-66584-8

RTP: Audio and Video for the Internet; *Addison Wesley; Colin Perkins; June 2003*
ISBN: 978-0-470-02359-4

Supplemental Readings: Additional materials will be distributed on an ongoing basis.

GRADING POLICY

Grading*	
Midterm	12%
Labs	12%
Homewoks	12%
Discussion boards	6%
Project	34%
Final exam	24%

*Final grade determined by a weighted average

Midterm
Option 1: in-class, closed-book/closed-notes, MC/TF questions, simple problems and essay questions.
Option 2: remote and timed, MC/TF questions, simple problems.
The decision will be made during the semester.

Labs
We will have three labs and the facilitation for these will be determined during the semester.
We will discuss and explain lab activities in more detail as we kick in this semester.

Homeworks
Few homework assignments will be spread throughout the semester. These will be designed to help comprehend and keep up with the presented material.
They are due the following week after they are assigned.

Return hard or softcopy in a single file with the student name visible in the file name.

Discussion boards
We will have few discussion boards throughout the semester.
These are mandatory and they will be graded based on involvement and effort.

Project
The project will be assigned second week and is due last week of semester.
This is a group project (3-4 students).

Final Exam (take home – last week of the semester)
Comprehensive MC/TF questions, more complex engineering problems and essay questions.

Honor Code:

All assignments are conducted under the rules and regulations of the GMU Honor Code Policy

Proposed Topics

Topics
Introduction to Voice and Voice Transmission Technologies: Voice characteristics, digitalization and encoding. Traditional circuit switched equipment and networks used in telephony. Signaling basics. Potential use-cases for VoIP deployments (benefits/challenges). Enterprise/Campus and Commercial Telephony. Typical VoIP Connection Strategies.
Concept of Transporting Voice over a Packet Switched Network: Internet Protocol (IP) introduction. Real time protocols: RTP, RTCP, RTSP, SCTP, UDP-Lite/Liter - packet formats, functionality and features. Real-time media synchronization.
Voice over IP Decomposition: Human voice and coding techniques, compression. Factors that affect VoIP quality: delay, jitter, packet loss, echo. Performance and quality metrics for VoIP: MOS, R-Factor, PESQ. VoIP performance measurement and monitoring tools.
Intro to VoIP Signaling Protocols: Overview. H.323 signaling protocol: format and inter-workings.
SIP Signaling Protocol: Architecture, format and inter-workings.
SS7 Signaling Protocol: Architecture, format and inter-workings.
The Softswitch Architecture: Interoperability of different signaling protocols (H.323, SIP, SS7) using Softswitches, Applications of Softswitches in a carrier grade VoIP environment. (SS7 signaling over IP-based networks). VoIP – PSTN migration and integration strategies.
Voice over IP Network Planning and Design: Traffic analysis and forecasting (advanced), numbering and dial plans, number routing, vendor selection criteria for LAN and WAN deployments. E.911, CALEA.
VoIP Quality and QoS: A thorough explanation of QoS components, protocols and trade-offs. RSVP, Diffserv, MPLS and 801.2q protocols are covered in detail in terms packet format, features and functions and their pros and cons.
VoIP Security: Vulnerabilities, security requirements and protection technologies. NAT/Firewall considerations. VoIP encryption analysis.
NextGen VoIP: VoIP Mobility. VoIP Equipment: Adapters, soft phones, WiFi phones, mobile phones. Collaboration and presence. Billing and Mediation. VoiceXML. IP Multimedia Subsystem (IMS).
Future of VoIP – Challenges, Concerns, Way Ahead. Open Source, Use Cases – Skype, Vonage, etc.