The University of Jordan
King Abdullah II School for Information Technology (KASIT)
Computer Information Systems Department (CIS)
First Semester 2013/2014

Course Title: Special Topics – VoIP Technology and Challenges
Course Code: 1902768
Prerequisite: None
Providing Department: Computer Information Systems (CIS)
Instructor: Dr.Mousa AL-Akhras (mousa.akhras@ju.edu.jo)

Online Course Site:
Site address: blackboard.ju.edu.jo
User Name: 1902768_1_std
Password: 1902768_1_std

Course Description:
The course aims to provide students with knowledge of VoIP technology. At the beginning of the course students are introduced to basic steps in conducting research. Then VoIP technology is presented in details. Topics covered include: Public Switched Telephone Network vs. Voice over IP technology, applications, advantages and challenges of VoIP, Quality assessment techniques for VoIP applications. Speech coding technology and packet loss concealment techniques are also presented. At the end of the course, students' projects will be discussed and presented.

Assessment Criteria:
1. Exams: Midterm exam (30 points).
2. Research project + Research presentation + Joint Discussion (30 Points). **Project Deadline 15/12/2010**

   Deliverables:
   - **Week 5:** Proposal + short presentation (5 Minutes)
   - **Week 7**: Critical summary of previous literature & Find A Niche
   - **Week 9:** Progress Report + Formulation of your Novelty **(How your method is different from previous work)** - Presentation (10 Minutes)
   - **Week 12:** Interim Report + Initial Results (10-15 Minutes)
   - **Week 16:** Final Report + Presentation (30 Minutes)

3. Final exam (40 points)

Intended Grading Scale:

90 – 100	A
80 – 89	B+
70 – 79	B
50 – 78	C+
<50	C

Course Contents:
- Introduction to Research Methods – How to conduct research, research output and publishing process.
- VoIP:
  - VoIP vs. Traditional Telephony
  - VoIP Technology
  - VoIP Advantages, Applications & Protocols
  - VoIP Challenges Before, During and After VoIP session
  - Main Challenges (Delay, Packet Loss and Jitter)
  - VoIP speech Quality measurement Methods
    - Subjective Quality assessment
    - Objective Quality Assessment (Intrusive and Non-Intrusive)
  - Speech Coding Technology

- o Packet Loss Concealment Technology
- Student Research and Presentation

Intended Learning Outcomes:

Successful completion of this module should lead to the following learning outcomes:

A-Knowledge and Understanding (students should)

(A1) have some understanding of the basic concepts and techniques of VoIP

(A2) have some understanding of

- Differences between traditional telephony and VoIP
- Advantages and disadvantages of traditional telephony and VoIP
- Applications of VoIP
- VoIP Protocols
- Challenges that faces VoIP technology with focus on Delay, Packet Loss and Jitter
- Quality Assessment methods in VoIP, including: Subjective (MOS) and Objective (Intrusive (PESQ) and Non-Intrusive(E-Model))
- Speech Coding Technology

B-Intellectual skills-with ability to

(B1) Appreciate the subtleties related to different VoIP applications and Protocols (H.323 vs. SIP)

(B2) Analyze different factors that may affect speech quality

(B3) Differentiate between Subjective and Objective speech quality assessment methods

(B4) Differentiate between simulation techniques and environments

(B5) Packet Loss simulation using N-state Markov Models

(B6) Analyze and design a solution for the problem of speech quality assessment

C- Practical Skills-With ability to

(C1) Represent and implement different impairment factors that may affect VoIP quality.

(C2) Measure the speech quality of a VoIP traffic.

(C3) Compare the speech quality you obtained with those obtained by previous researchers.

D-Transferable Skills-With ability to

(D1) Deploy communication skills.

(D2) Deploy research skills.

(D2) Work effectively within a group.

(D3) To work to tight deadlines

(D4) effectively present the final work in a demo.

References:

1. Mousa AL-Akhras and Iman ALMomani, Internet Skills in Scientific Research, ISBN 978-9957-72-040-7, ZAMZAM publisher and distributor, Amman, Jordan, 2011.
2. M. AL-Akhras. Quality of Media Traffic over Lossy Internet Protocol Networks: Measurement and Improvement. PhD thesis, Faculty of Technology, De Montfort University, U.K., August 2007.
3. Mousa AL-Akhras, Measuring the Quality of VoIP Traffic, ISBN-13 978-3-639-30581-4, ISBN-10: 3639305817, VDM Publishing House Ltd., 2010.
4. Mousa AL-Akhras and Iman ALMomani, VoIP Quality Assessment Technologies - Book: VoIP Technologies, edited by: Shigeru Kashihara, ISBN: 978-953-307-549-5, Publisher: InTech, Vienna, Austria. Publishing date: February 2011. Download from http://www.intechopen.com/books/show/title/voip-technologies.
5. R. Arora. Voice over IP: Protocols and Standards. Technical report, Washington University in St. Louis.
6. Cisco.com. Understanding Delay in Packet Voice Networks.
7. Daniel Collins. Carrier Grade Voice over IP. McGraw-Hill Companies, 2nd edition, 2003.
8. Goodman. Internet Telephony and Modem Delay. IEEE Network, 13(3):8{16, May-June 1999.
9. IEC. Voice and Fax over Internet Protocol (V/FoIP). International Engineering Consortium.

10. IETF. SIP: Session Initiation Protocol. Internet Engineering Task Force (IETF), May 2002.
11. ITU-T. Recommendation G.107 - The E-model, a Computational Model for use in Transmission Planning. International Telecommunication Union- Telecommunication Standardization Sector (ITU-T), March 2005.
12. ITU-T. Recommendation G.109 - Definition of Categories of Speech Transmission Quality. International Telecommunication Union-Telecommunication Standardization Sector (ITU-T), September 1999.
13. ITU-T. Recommendation G.114 - One-Way Transmission Time. International Telecommunication Union-Telecommunication Standardization Sector (ITUT), May 2003.
14. ITU-T. Recommendation G.114 Appendix II - Guidance on One-Way Delay for Voice over IP. International Telecommunication Union-Telecommunication Standardization Sector (ITU-T), September 2003.
15. ITU-T. Recommendation G.165 - Echo Cancellers. International Telecommunication Union-Telecommunication Standardization Sector (ITU-T), March 1993.
16. ITU-T. Recommendation G.711 - Pulse Code Modulation (PCM) of Voice Frequencies. International Telecommunication Union-Telecommunication Standardization Sector (ITU-T), November 1988.
17. ITU-T. Recommendation G.723.1 - Dual Rate Speech Coder for Multimedia Communications Transmitting at 5.3 and 6.3 kbit/s. International Telecommunication Union-Telecommunication Standardization Sector (ITU-T), May 2006.
18. ITU-T. Recommendation G.726 - 40, 32, 24, 16 kbit/s Adaptive Differential Pulse Code Modulation (ADPCM). International Telecommunication Union-Telecommunication Standardization Sector (ITU-T), December 1990.
19. ITU-T. Recommendation G.728 - Coding of Speech at 16 kbit/s Using Low-Delay Code Excited Linear Prediction. International Telecommunication Union-Telecommunication Standardization Sector (ITU-T), September 1992.
20. ITU-T. Recommendation G.729 - Coding of Speech at 8 kbit/s Using Conjugate-Structure Algebraic-Code-Excited Linear-Prediction (CS-ACELP). International Telecommunication Union-Telecommunication Standardization Sector (ITU-T), March 1996.
21. ITU-T. Recommendation P.800 - The E-model, a Computational Model for use in Transmission Planning. International Telecommunication Union-Telecommunication Standardization Sector (ITU-T), March 2005.
22. ITU-T. Recommendation P.800.1 - Mean Opinion Score (MOS) Terminology. International Telecommunication Union-Telecommunication Standardization Sector (ITU-T), March 2003.
23. ITU-T. Recommendation P.861 - Objective Quality Measurement of Telephoneband (300-3400 Hz) Speech Codecs. International Telecommunication Union-Telecommunication Standardization Sector (ITU-T), February 1998.
24. ITU-T. Recommendation P.862 - The E-model, a Computational Model for use in Transmission Planning. International Telecommunication Union-Telecommunication Standardization Sector (ITU-T), March 2005.
25. ITU-T. Recommendation P.862.1-Mapping Function for Transforming P.862 Raw Result Scores to MOS-LQO. International Telecommunication Union-Telecommunication Standardization Sector (ITU-T), March 2005.
26. A. Kansal and A. Karandikar. Adaptive Delay Estimation for Low Jitter Audio over Internet. In IEEE Global Telecommunications Conference, 2001. GLOBECOM '01., volume 4, pages 2591-2595, 25-29 November 2001.
27. N. Laoutaris and I. Stavrakakis. Adaptive Play out Strategies for Packet Video Receivers with Finite Buffer Capacity. In IEEE International Conference on Communications, 2001. ICC 2001., volume 3, pages 969-973, 11-14 June 2001.
28. Hong Liu and P. Mouchtaris. Voice over IP Signaling: H.323 and Beyond. IEEE Communications Magazine, 38(10):142-148, October 2000.
29. Nachiappan Nachiappan and Fredrik Sjoqvist. Survey of Voice over IP (VoIP). Technical report, Standford University, 2004.
30. M. Narbutt and L. Murphy. Improving Voice over IP Subjective Call Quality. IEEE Communications Letters, 8(5):308{310, May 2004.
31. R. Ramjee, J. Kurose, D. Towsley, and H. Schulzrinne. Adaptive Playout Mechanisms for Packetized Audio Applications in Wide-Area Networks. In

INFOCOM '94. Networking for Global Communications. 13th Proceedings IEEE, pages 680{688, 12-16 June 1994.
32. L. Sun. Speech Quality Prediction for Voice over Internet Protocol Networks. PhD thesis, School of Computing, Communications and Electronics, University of Plymouth, U.K., January 2004.
33. Kuo-Kun Tseng, Yuan-Cheng Lai, and Ying-Dar Lin. Perceptual Codec and Interaction Aware Playout Algorithms and Quality Measurement for VoIP Systems. IEEE Transactions on Consumer Electronics, 50(1):297-305, Feb 2004.
34. S. Zeadally and F. Siddiqui. Design and Implementation of a SIP-Based VoIP Architecture. In 18th International Conference on Advanced Information Networking and Applications, volume 2, Los Alamitos, 2004. IEEE COMPUTER SOC.
35. Yuan Zhang. SIP-Based VoIP Network and its Interworking with the PSTN. In Electronics & Communication Engineering Journal, volume 14, pages 273-282, December 2002.
36. E.E. Zurek, J. Le_ew, and W.A. Moreno. Objective Evaluation of Voice Clarity Measurements for VoIP Compression Algorithms. In Proceedings of the Fourth IEEE International Caracas Conference on Devices, Circuits and Systems, 2002., pages T033-1-T033{6, 17-19 April 2002.

We may provide some notes. However, it is very important to note that they are supplementary and not a substitute to the recommended books

Students are encouraged to make heavy use of the library, E-LIBRARY http://ezlibrary.ju.edu.jo/login or from within the university using (http://e-library) and internet resources such as:

We will be glad to discuss with you the relevance of any material that you may intend to read. We are willing to discuss (and/or give you pointers to), during office hours and/or at any possible time agreed upon, any issue or advanced topic in artificial intelligence.