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Tutorial Proposal

**Distributed Media over IP: from IPTV to Wireless
and Mobile Multimedia Networks**

By

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Expected Background Knowledge of the Participants

Audience can be from academia or industry. Any individual with basic knowledge of computer science and engineering can benefit from this tutorial.

Objectives

We are witnessing the Internet applications such as mobile multimedia over IP network technology being destined to play an increasingly important role in communication systems. With the demand for multimedia applications, there will be a growing interest in identifying suitable network architectures and transmitting facilities for this technology. Communication industry has spent considerable effort in designing an IP-based media transport mechanism, voice over IP (VoIP), and multimedia networks that can deliver voice-band telephony with the quality of the telephone networks. The Internet offers phone services less expensive and with numerous additional features such as video conferencing, online directory services, and the Web incorporation.

In this tutorial, we present the fundamentals the latest applications in computer networks. Applications such as Video on Demand (VoD) IPTV, VoIP, and also Multimedia over IP networks schemes. We explain the transportation of real-time signals along with the signaling protocols used in voice over IP (VoIP) telephony and multimedia networking. The tutorial covers the signaling protocols as *H.323 series of protocols*, and *Session Initiation Protocol (SIP)* which are responsible for session signaling. The H.323 protocols interact to provide ideal telephone communication, providing phone numbers to IP address

mapping, handling digitized audio streaming in IP telephony, and providing signaling functions for call setup and call management. The H.323 series support simultaneous voice and data transmission and can transmit binary messages that are encoded using *basic encoding rules*. We also review the *Session Initiation Protocol (SIP)* as one of the most important VoIP signaling protocols operating in the application layer of TCP/IP model. SIP can perform both unicast and multicast sessions and supports user mobility and handles signals and identifies user location, call setup, call termination, and busy signals. SIP can use multicast to support conference calls and uses the *Session Description Protocol (SDP)* to negotiate parameters.

The tutorial further covers Compression of multimedia components such as Digital Voice and Video, focusing on data-compression techniques for voice and video to prepare digital voice and video for multimedia networking will be [resented. The topic starts with the analysis of information-source fundamentals, source coding, and limits of data compression and explains all the steps of the conversion from raw voice to compressed binary form, such as sampling, quantization, and encoding. The discussion also summarizes the limits of compression and explains typical processes of still-image and video-compression techniques, such as JPEG, MPEG, and MP3.

The tutorial will then present real-time transport protocols, such as *Real-Time Transport protocol (RTP)* and the *Real-Time Control Protocol (RTCP)*. The next topic is streaming video in a single server, using *content distribution networks (CDNs)*. Also discussed is the *Stream Control Transmission Protocol (SCTP)*, which provides a general-purpose transport protocol for transporting stream traffic. The tutorial describes detailed streaming source modeling and analysis. In real-time applications, a stream of data is sent at a constant rate. This data must be delivered to the appropriate application on the destination system, using real-time protocols. The most widely applied protocol for real-time transmission is the *Real-Time Transport Protocol (RTP)*, including its companion version: *Real-Time Control Protocol (RTCP)*. UDP cannot provide any timing information. RTP is built on top of the existing UDP stack. Real-time applications may use multicasting for data delivery.

We also cover video streaming that presents a significant challenge to network designers. A video in a single server can be streamed from a video server to a client at the client request. The high bit-rate video streaming must sometimes pass through many Internet service providers, leading to the likelihood of significant delay and loss on video. One practical solution to this challenge is to use *content distribution networks (CDNs)* for distributing stored multimedia content. Video streaming, e-mail, and image packets in the best-effort Internet are mixed in the output queue of the main exit router of a domain. Under such circumstances, a burst of packets, primarily from the image file source, could cause IP video streaming packets to be excessively delayed or lost at the router. One solution in this case is to mark each packet as to which class of traffic it belongs to. This can be done by using the type-of-service (ToS) field in IPv4 packets. As seen in the figure, transmitted packets are first classified in terms of their priorities and are queued in a first in, first out (FIFO) order. The priority of an image file can be equal to or less than the one for video streaming, owing to the arrangement of purchased services.

The Tutorial will then focus on the *Stream Control Transmission Protocol (SCTP)* providing a general-purpose transport protocol for message-oriented applications. SCTP is a reliable transport protocol for transporting stream traffic, can operate on top of unreliable connectionless networks, and offers acknowledged and non-duplicated transmission data on connectionless networks. SCTP has the following features. The protocol is error free. A retransmission scheme is applied to compensate for loss or corruption of the datagram, using checksums and sequence numbers. It has ordered and unordered delivery modes. SCTP has effective methods to avoid flooding congestion and masquerade attacks. This protocol is *multipoint* and allows several streams within a connection. In TCP, a stream is a sequence of bytes; in SCTP, a sequence of variable-sized messages. SCTP services are placed at the same layer as TCP or UDP services. Streaming data is first encapsulated into packets, and each packet carries several

correlated chunks of streaming details. The Tutorial will ultimately present multimedia over general Wireless and WiMAX networks.

Time Allocations for the Major Course Topics (3 hours) and Tutorial High-Lights

Overview of Digital Voice/Video and Compression (1 hour)

- Overview of VoIP Signaling Protocols
- H.323 Protocols
- Session Initiation Protocol (SIP)
- Signal Sampling
- Quantization and Distortion
- Still Images and JPEG Compression
- Raw-Image Sampling and DCT
- Quantization and Encoding
- Moving Images and MPEG Compression
- MP3 and Streaming Audio
- Limits of Compression with Loss
- Basics of Information Theory
- Entropy of Information
- Shannon's Coding Theorem
- Compression Methods Without Loss
- Run-Length Encoding
- Huffman Encoding

Video Streaming Applications and Real-Time Media Transport Protocols (1 hour)

- Real-Time Transport Protocol (RTP)
- Real-Time Control Protocol (RTCP)
- Estimation of Jitter in Real-Time Traffic
- Distributed Multimedia Networking
- Stream Control Transmission Protocol (SCTP)
- SCTP Packet Structure
- Self-Similarity and Non-Markovian Streaming Analysis
- Self-Similarity with Batch Arrival Models
- Content Distribution Networks (CDNs)
- CDN Interactions with DNS
- Multimedia Security
- Providing QoS to Streaming
- IPTV
- Video on Demand (VOD) Technology

Mobile multimedia and Voice and Video Streaming over Wireless Networks (1 hour)

- Introduction to WiMAX technology
- Mobile Transport Protocols
- Mobile Computing and Mobile IP

- TCP and UDP for Mobility
- Protocols for Voice over Mobile IP
- Protocols for Video Streaming over Mobile IP
- Mobile IPTV
- Video on Demand (VOD) Technology

Reference Text

1. Nader F. Mir, "Computer and Communication Networks," By Prentice-Hall Publishing Co., ISBN: 0131747991, 1st edition Nov. 2006
2. "Carrier Grade Voice Over IP," 2nd Ed., 2003, Daniel Collins, ISBN 0-07-140634-4, McGraw-Hill Networking Professional.

Qualifications of the Instructor

Nader F. Mir is currently a Professor, and served as the Associate Chairman, at the Electrical Engineering Department of San Jose State University, California. He is also the Director of MSE Program in optical sensors and networks for Lockheed-Martin Space Systems Corporation. He is also an expert in intellectual property development and a consultant for patent litigation cases in the areas of communications, telecommunications and computer networks both in protocol and hardware levels. Prior to his current position, he was an associate professor at his current school, and assistant professor at University of Kentucky in Lexington. From 1994 to 1996, he was a research scientist at the Advanced Telecommunications Institute, Stevens Institute of Technology in New Jersey working on design of advanced telecommunication networks. From 1990 to 1994 he was with the Computer and Communications Research Center at Washington University in St. Louis and worked as a research assistant on design and analysis of high-speed switching systems project. From 1985 to 1988, he was with Telecommunication Research & Development Center (TRDC), Surrey, and as a telecommunications system research & development engineer participated in design of a high-speed digital telephone Private Branch Exchange" (PBX), and received the best "design/idea" award. Professor Mir is internationally known through his research, journal editorial positions, and his books in the areas of: Analysis of Computer Communication Networks, Internet Applications of Voice over IP and Multimedia Networks, Design and Analysis of Networking Devices and High-Speed Routers, Wireless and Mobile Networks and Wireless Sensor Networks, Applications of Integrated Circuits in Computer Communications. He is a senior member of the IEEE and has served as the member of technical program committee and steering committee for a number of major IEEE communications and networking conferences such as WCNC, GLOBECOM, and ICC and IASTED. He has several journal editorial positions such as: the Editor of Journal of Computing and Information Technology, a Guest Editor for Computer Networking at CIT Journal (in 2008), the Editorial Board Member of the International Journal of Internet Technology and Secured Transactions (in 2005), and the technical **Editor of IEEE Communication Magazine**. He was granted a successful U.S. Patent in 2006, claiming an invention related to hardware/protocol for use in high-speed computer communication networks. Dr. Mir has received a number of prestigious national and university awards and research grants. He is also the recipient of a university teaching recognition award and a research excellence award. He is also the recipient of a number of outstanding presentation awards from leading international conferences. **He has presented several successful tutorials at international conferences such as: CIIT04, CIIT05, CSN08, PDCS08, PDCN10, IARIA-ICN11.** Dr. Mir has published two successful books, one of which is a world-wide adopted university text-book entitled **"Computer & Communication Networks" published by Prentice Hall Publishing Co.** He has also

published numerous refereed technical journal and conference papers all in the field of communications and networking. He received the B.Sc. degree (with honors) in electrical engineering in 1985, and the M.Sc. and Ph.D. degrees both in electrical engineering from Washington University in St. Louis, MO, in 1990 and 1995 respectively.