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Tutorial Proposal

Overview of VoIP, and Multimedia over Broadband IP Networks

By

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Title:

Overview of VoIP, and Multimedia over Broadband IP Networks

Abstract:

In this tutorial, we present the fundamentals of VoIP and also Multimedia over IP network schemes. We explain signaling protocols and media exchange protocols for transportation of real-time voice over IP (VoIP) telephony and multimedia of IP networking. The tutorial reviews the control signaling protocols of H.323 series of protocols, and Session Initiation Protocol (SIP) which are responsible for session signaling. 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 such as sampling, quantization, and encoding will be presented. 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). The tutorial describes detailed streaming source modeling and analysis. We also cover video streaming over from a video server to a client at the client request. The talk will cover type of-service (ToS) field in IPv4 packets and similar solution in IPv6. The Tutorial will then focus on the Stream Control Transmission Protocol (SCTP) providing a general-purpose transport protocol for message-oriented applications. 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.

Detailed Outline of the Topic Covered

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. In this tutorial, we first present the fundamentals of 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 presented. 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)*. 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. We explain how 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. 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. We explain a similar solution for IPv6 technology.

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. The tutorial will describe the SCTP's unique features such as being error free, having ordered and unordered delivery modes, having effective methods to avoid flooding congestion and masquerade attacks, *multipoint*, and

the fact that it allows several streams within a connection. The Tutorial will ultimately present multimedia over general Wireless and WiMAX networks. The breakup of the talk is as follows:

Overview of IP Telephony (0.5 hour)

- VoIP Signaling Protocols
- H.323 Protocols
- Session Initiation Protocol (SIP)
- Softswitch and MGCP

Digital Voice and Compression (1 hour)

- Signal Sampling
- Quantization and Distortion
- Still Images and JPEG Compression
- Raw-Image Sampling and DCT
- Quantization and Encoding
- Moving Images & 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
- Lempel-Ziv Encoding
- FAX Compression for Transmission

Video Streaming 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 & Batch Arrival Models
- Content Distribution Networks (CDNs)
- CDN Interactions with DNS
- Multimedia Security
- Providing QoS to Streaming

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

Contact Information

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Biography of Speaker

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 a networking MSE Program 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** (since 2005). 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 is especially the **recipient of several Outstanding Tutorial Presentation awards**.

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.

My History of Tutorial Presentations

1. **N. Mir**, *Outstanding Performance Tutorial Award*, "Mobile IP and Computing, and WiMAX Technology," *The 20th International Conference on Parallel and Distributed Computing Systems (PDCS'08)*, Orlando, Florida, November 16-18, 2008.
2. **N. Mir**, *Accomplished Tutorial Award*, "Wireless Networks, Mesh Networks, Mobile IP, and WiMAX Technology," *International Conference on Communication Systems and Networks (CSN'08)*, Langkawi, Malaysia, April 2-4, 2008.
3. **N. Mir**, *Outstanding Tutorial Performance Award*, "Wireless Mobile Ad-Hoc Network Technology," Invited Tutorial Talk, *International Conference on Communications, Internet and Information Technology (CIIT'05)*, Cambridge, Massachusetts, Oct30-November 3, 2005.
4. **N. Mir**, *Outstanding Performance Tutorial Award*, "Wireless Sensor Network Technology," *International Conference on Communications, Internet and Information Technology (CIIT'04)*, St. Thomas, Virgin Islands, November 22-24, 2004, Status.