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A Survey on Multimedia Protocols in Wireless Network

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ABSTRACT: Multimedia is a combination of audio, video, text, animation and still images. Multimedia networking is popular, but not an easy task. There are many challenges which still exist while transmitting real time multimedia data in wireless environment. The aim of multimedia transport protocols is to transmit multimedia signals from one point to another point. Communication network connects these points employing specific protocols. Communication of Multimedia as opposed to traditional data oriented applications pose special challenges. Hence, in this paper a brief survey is done on the multimedia protocols and also on the real-time challenge of multimedia networking in the wireless network.

KEYWORDS: Multimedia; Wireless Network; Multimedia transport protocol; Multimedia Challenges; Multimedia networking.

I. INTRODUCTION

In order to transmit multimedia signals from one point to another we need multimedia transport protocol. Communication network connects these points employing specific protocols. Initially, original multimedia signals are encoded. This encoding is done to reduce bit rate. Once encoded stream is sent to another location in the network, transport protocols is accountable for packetization and delivery of the encoded stream. On the other hand, reconstruction of the encoded multimedia stream is done from the stream of delivered packets and then decoded in order to produce a useful multimedia signal [7] that is to be played back or stored for further use.

To exchange data over networks, Internet Protocol(IP) is used, which is a packet-based network protocol. It is the base for all network protocols. Transport Control Protocol (TCP) is the most used higher level protocol, which is a reliable transport protocol mainly designed for data transmission and Internet services. Since retransmissions may lead to high delay and which can even cause delay jitter, TCP is not suitable protocol for real-time applications. Delay jitter significantly degrades the quality. Also, multicast is not supported by TCP. Moreover, congestion control mechanisms are not suitable when comes to audio or video media transmission.

User Datagram Protocol is the transport protocol that is generally used for real-time multimedia data transmission. But, UDP does not guarantee the arrival of the packet and it is up to the application or higher level protocols to take care of the sent data. In addition, the basic UDP protocol supports multiplexing and checksum services. Even after being most used protocol for real-time applications, RTP has certain problems. Firstly, no guarantee for the data to reach on time. Also, there is no guarantee for the QoS. In-time delivery requires the help of lower layers. Second point is that, no guarantee for the delivery of packets. Here there can be two possibilities in such cases. Packets may be lost or delivered out of order. In addition, there is no mechanism to recover from packet loss.

II. LITERATURE SURVEY

Elizabeth M. Royer and Chai-Keong Toh [1] gave a review of current routing protocols for ad hoc mobile wireless networks where routing protocols for ad hoc networks are examined and protocols based on a given set of parameters are evaluated. Also an overview of eight different protocols is provided by presenting their functionality, comparison, characteristics and discussion of their respective merits and drawbacks.

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Xiaoqing Zhu and Bernd Girod [2] gave an overview of the technical challenges of video streaming over wireless networks, with a focus on novel cross-layer design solutions for resource allocation. Comparison of performance of various centralized and distributed schemes are also presented, using video streaming over wireless home networks as an application example.

M. Reha Civanlar [3] gave an overview of the existing architectural elements supporting real-time data transmission over the Internet. The protocols directly related to real-time multimedia data transmission over the internet which is classified in four categories as Signaling, Session Control, Transport, Network infrastructure are discussed in detail. Multimedia data stream properties for network use are also mentioned.

Satyajayant Misra et al. [4] presented a Survey of Multimedia Streaming in Wireless Sensor Networks where the requirements of multimedia traffic at each layer of the network protocol stack are categorized and also the mechanisms operating at the application, transport, network, and MAC layers that have been proposed for multimedia streaming in wireless sensor networks at each layer of the stack. Also existing cross-layer approaches are reviewed and a few possible cross-layer solutions to optimize the performance of a given wireless sensor network for multimedia streaming applications are proposed.

Santhosha Rao, Kumara Shama [5] gave overview on cross layer protocols for multimedia transmission in wireless networks. In order to meet the challenging demands on future wireless networks, it may be required to adopt new approaches in which protocols can be designed by violating the reference layered architecture allowing direct communication between protocols in nonadjacent layers such violations of a layered architecture have been termed as cross-layer design (CLD).

III. MULTIMEDIA PROTOCOLS

RTP --- Real-time Transport Protocol: Real-time transport protocol (RTP) is basically an IP-based protocol, that supports transmission of real-time data such as video and audio streams. Services provided by RTP are time reconstruction, loss detection, security and identification of the content. Primarily, RTP is designed for multicast of real-time data, but can be used in unicast also.

One-way transport such as video-on-demand uses RTP. Also RTP is used for interactive services such as Internet telephony. Design of RTP is in such a way that it works in conjunction with RTCP. This is to get feedback on quality of transmitted data and details of participants in the under-way session. RTP alone cannot ensure timely delivery. It needs support from lower layers. i.e. it depends on RSVP to reserve resources and to provide the requested quality of service

RTCP---Real-Time Control Protocol: RTCP is the control protocol designed to work in conjunction with Real-time transport protocol. RTCP extends real-time transport protocol with a control functionality. During RTP session, participants send RTCP packets in regular interval of time to convey feedback on quality of data delivery and information of membership. RTCP provides services like monitoring of QoS, congestion control, identification of source, inter-media synchronization, control information scaling.

RTSP---Real-Time Streaming Protocol: Multimedia data is usually sent across the network in the form of streams. This is better instead of storing large multimedia files and then playing back. Streaming breaks data into packets. The real-time data flows through the transmission, decompression and then playing back pipeline just like a flow of water stream. When client play the first packet, second packet is decompressed, while receiving third packet.

RTSP is an application-level protocol designed to work with lower-level protocols like RTP, RSVP, which will provide a complete streaming service over internet. It also enables to select delivery channels and delivery mechanisms based upon RTP. RTSP, the Real Time Streaming Protocol, provides "VCR-style" remote control functionality for audio and video streams. Sources of data include both live stored clips and data feeds.

RSVP --- Resource ReSerVation Protocol: RSVP is the network control protocol that allows data receiver to request a special end-to-end quality of service for its data flows. RSVP is a main component of future Integrated Services Internet which can provide both best-effort and real-time service. RSVP is also used by real-time applications to reserve necessary resources at routers along the transmission paths so that the requested bandwidth can be available when the transmission actually takes place. Protocol stack for multimedia is shown in fig 1.

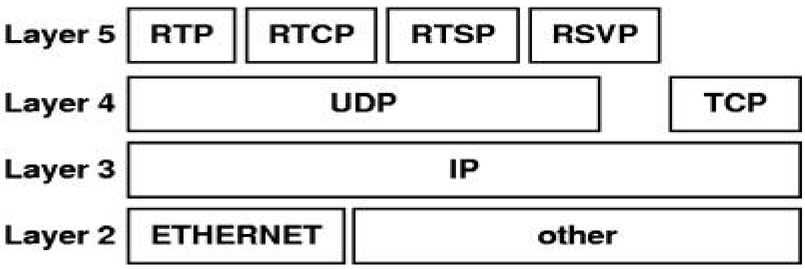

Fig. 1.Multimedia protocol stack.

IV. REAL-TIME CHALLENGES OF MULTIMEDIA DATA TRANSMISSION

Today, multimedia which is a current hot research topic has become an indispensable feature on the internet. Multimedia is difficult for many reasons. It is not an easy task to overcome multimedia challenges. There are many challenges which still exist while transmitting real time multimedia data in wireless environment. It faces mainly 3 challenges.

1. Multimedia applications require much higher bandwidth when compared with traditional textual applications. Ex: 25 second 320*240 quick time movie could actually take 2-3 MB, which is approximately about 1000 screens of textual data. In the olden days, this was unimaginable, where only textual data is transmitted on the net.
2. real-time traffic requirement for multimedia applications.
3. Multimedia data stream is usually bursty i,e occurring at short episodes.

Some other challenges of multimedia data transmission[6] over wireless networks are:
a) **Packet Loss:** Packets are sent across several routers from source to destination. Each router may receive packet stream from several sources simultaneously. Few packets will be dropped, when the incoming packets fills the buffer of any of the routers. In such case, the network is said to be congested. Loss of packet is a great destruction to the received multimedia signal.
The packet loss problem exists for video also. Some portions or blocks of the image cannot be decoded and displayed which results in severe degradation of the reconstructed signal quality. The problem becomes more critical when the signal is compressed too much. In general, natural or original signal contains too much of redundant information. When there is loss, redundant information can be interpolated by human brain. However, to reduce the bit rate and also to remove the redundancy, the signal is compressed and thus, packet loss becomes more infuriating.

b) **Network Jitter:** queuing and propagation delays leads to network jitter. A jitter buffer is used in order to reduce jitter. The first packet in the buffer is held for a while by the receiver before playing it out. The size of the jitter buffer depends on the duration of hold time. Ex: hold time 10ms means the size of jitter buffer is 10ms.
Network Jitter effects are similar to that of packet loss. Implementation of de jittering buffer reduces the effect of jitter and from the application point of view, jitter effect can be translated as extra network losses. The packets which arrive after expiration time are considered to be lost. Lastly, when comes to interactive applications, consideration of the echo effect is must and long with that cancellation mechanism or echo suppression must be implemented.

c) **Packet Delay:**In the case of voice transmission, timing is very important attribute of voice. We have several possibilities, to reduce the delay effect: (i) serialization delay has to be reduced, (ii) using codecs, packet delay cane be reduced which run in real time without giving too much delay (iii) speed of the routers has to be increased in order to

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minimize the queuing delay or some kind of Differentiated Services must be used to reduce packet delay, (iv) the length of the physical material must be reduced to minimize the propagation delay. This can be done by choosing shortest pass from source to destination, in the network layer. Further, the effect of delay is not too much relevant in the case of video streaming, which is a one-way sessions.

V. CONCLUSION

Multimedia protocols are used to transfer multimedia signals from one point to another point. To support Multimedia transport on networks, Multimedia networking is used which build the hardware and software infrastructure, so that users can communicate in multimedia. However, networking of multimedia is not an easy task. A thorough understanding of multimedia protocols used for the transmission of signals in real-time, is necessary for the effective implementation. So a brief survey is done on protocols for real-time multimedia data transmission, and also challenges of multimedia transmission are studied in this paper.

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BIOGRAPHY

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