A RTP-Based Architecture of Multimedia Communications for Wireless Networks[1]

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ABSTRACT

In this paper, we discuss the disadvantages of Real-Time transport protocol (RTP) when deployed in wireless networks. As RTP is used to support audio and video communication over Internet, adaptability over a wide bandwidth range is important to its continued acceptance. We preset a novel architecture of multimedia communications for wireless networks in this paper. RTP can be used in wireless network for multimedia communication by design of a *filter* in the architecture to allow wireless participants to engage in high-bandwidth conferencing sessions.

Keywords: Wireless network; Multimedia communications; Real-Time transport protocol

1. INTRODUCTION

With the explosive growth of the Internet and dramatic increase in wireless access, there is a tremendous demand on multimedia service over wireless Internet. The third generation (3G) wireless networks, foreseen to be the enabling technology for multimedia services with 384kbps to 2Mbps bandwidth, makes it feasible for providing integrated service of data, voice, audio, and video across the wireless link [1]. Continuous media applications, such as audio or video conferencing, Internet radio, on-line seminars, or video-on-demand systems, are becoming commonplace. These applications transmit data at regular intervals and require strict guarantees on maximum delay and minimum bandwidth. When these services are moved in wireless devices with latency over noisy wireless bottlenecks, they must be able to adapt a wide bandwidth range.

Real-Time transport protocol (RTP) is an end-to-end transport protocol. It used to support multimedia traffic on Internet. It may be implemented on top of UDP/IP or ATM, and it takes advantage of the MBONE, which allows bandwidth-efficient distribution of data to many users by eliminating redundant packet transmissions. It is already used to support many video-conferencing tools.

In this paper we study the problem of running RTP over wireless networks. The key issue that we address is the adaptability of the bandwidth used for RTP control messages on wireless link. We highlight some disadvantages of RTP when deployed in wireless networks. We present a framework of multimedia communications for wireless networks to solve those disadvantages.

The remainder of this paper is organized as follows. Section 2 provides background material. Section 3 gives a

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short introduction to RTP and the problem of running RTP over wireless networks is also discussed. Solution of problems and an architecture of multimedia communications for wireless networks are presented in Section 4 and Section 5 concludes the paper and our future work.

2. RELATED WORKS

Today, many researchers work in multimedia communications on RTP and wireless network and have made some progresses. For examples, Qian Zhang addressed the important issues of resource allocation for multiple media transmitted over 3G wireless channel with adaptive QoS (Quality of Service) support, and developed an end-to-end architecture combining the link level and application level QoS adaptation together [2]. Dapeng Wu proposed an adaptive QoS control to increase the robustness of MPEG-4 video communication over wireless channels which consists of optimal mode selection and delay-constrained hybrid automatic repeat request (ARQ) [3]. Richard Han presented a progressively reliable transport protocol, named "Leaky" ARQ, for delivery of delay-sensitive multimedia over Internet connections with wireless access links [4]. K. Brown designed and implemented a real-time control protocol (RTCP) gateway which provides the scaling real-time control bandwidth for wireless networks and does not affect RTP functions on the fixed network [5]. We already presented the guarantee of QoS for wireless multimedia streams based on adaptive session [6]. Our works are partly from above contributions. We can get the characteristics of multimedia communication on wireless networks from reference [2], [3] and [4], and know RTP protocol in wireless networks and its deficiencies from reference [5], and initial ideas and schemes are from our previous works in reference [6].

3. REAL-TIME TRANSPORT PROTOCOL (RTP)

RTP has been designed within the Internet Engineering Task Force (IETF). Note that the moniker 'transport protocol' could be misleading, as it is currently mostly used together with UDP, also designated as a transport protocol. The name emphasizes, however, that RTP is an end-to-end protocol. To avoid misunderstandings, it may help to clear up some of the things that RTP does not attempt to do. RTP has no notion of a connection; it may operate over either connection-oriented or connectionless lower-layer protocols. It has no dependencies on particular address formats, and only requires that framing and segmentation are taken care of by lower layers. RTP offers no reliability mechanisms. It is typically implemented as part of the application and not of the operating system kernel. RTP consists of two parts, a data part and a control part. Continuous media data like audio and video is carried in RTP data packets. The functionality of the control packets is described below. If RTP packets are carried in UDP Datagrams, data and control packets use two consecutive ports, with the data port always being the lower, even numbered one. If other protocol serve underneath RTP (e.g. RTP directly over ATM AAL5), it is possible to carry both in single lower-layer protocol data unit, with control followed by data.

3.1 RTP data packets

RTP data packets consist of a 12-byte header follows by the payload, e.g. a video frame or a sequence of a audio samples. The payload may be wrapped again into an encoding-specific layer. The header contains the following information:

(1) Playload type

A one-byte payload type identifies the kind of pay load contained in the packet, for example JPEG video or GSM audio.

(2) Timestamp

A 32-bit timestamp describes the generation instant of the data contained in the packet. The timestamp frequency depends upon the payload type.

(3) Sequence number

A 16-bit packet sequence number allows loss detection and sequencing within a series of packets with the same timestamp.

(4) Marker bit

The interpretation of a marker bit depends upon the payload type. For video, it marks the end of a frame, for audio the beginning of a talkspurt.

(5) Synchronization source (SSRC) identifier

A randomly generated 32-bit scalar that uniquely identifies the source within a session.

Some additional bit fields are not described here in the interest of brevity.

3.2 RTP control functionality

RTP offers a control protocol called RTCP that supports the protoc ol functionality. An RTCP message consists of a number of 'stackable' packets, each with its own type code and length indication. Their format is fairly similar to data packets; in particular, the type indication is at the same location. RTCP packets are multicast periodically to the same multicast group as data packets. Thus, they also serve as a liveness indicator of session members, even in the absence of transmitting media data. The functionality of RTCP is described briefly below:

(1) Quality of Service monitoring and congestion control

RTCP packets contain the necessary information for quality-of-service (QoS) monitoring. Since they are multicast, all session members can survey how the other participants are faring. Applications that have recently sent audio or video data generate a sender report. It contains information useful for inter-media synchronization, as well as cumulative counters for packets and bytes sent. These allow receivers to estimate the actual data rate. Session members issue receiver reports for all video or audio sources they have heard from recently. They contain information on the highest sequence number received, the number of packets lost, a measure of the inter-arrival jitter and timestamps needed to compute an estimate of the round-trip delay between sender and receiver issuing the report.

(2) Inter-media synchronization

The RTCP sender reports contain an indication of real-time (wallclock time) and a corresponding RTP timestamp. These two values allow the synchronization of different media, for example, lip-syncing of audio and video.

(3) Identification

RTP data packets identify their origin only through a randomly generated 32-bit identifier. For conferencing application, a bit more context is often desirable. RTCP messages contain an SDES (source description) packet, in turn containing a number of piece of information is the so-called canonical name, a globally unique identifier of the session participant. Other possible SDES items include the user's name, email address, telephone number, application information and alert message.

(4) Session size estimation and scaling

RTCP packets are sent periodically by each session member. The desire for up-to-date control information has to be balanced against of the desire to limit control traffic to small percentage of data traffic, even with sessions consisting of several hundred members. The control traffic load is scaled with the data traffic load so that it makes up a certain percentage of the nominal data rate (5%).

3.3 Disadvantages of RTP in the wireless environment

We identify several disadvantages in RTP adaptability, especially when deployed in the wireless environment. They consist of the following three main problems:

(1) RTP can not dynamically control session bandwidth

In RTP, the session announcement must specify a session bandwidth that is maximal for the entire length of the multicast session. We assume that there are 10 data source in the session and they each may generate data at 128 kbps so that the initiator must specify a session bandwidth of 10*128=1280kbps. If, at some point in the session, there are only one source currently generating data, the bandwidth used for RTCP messages should be 5% of 128kbps (the current bandwidth used) or 6.4kbps, but instead would be calculated to be 5% of 1280kbps (the specified session bandwidth) or 64kbps. Control traffic then accounts for 50% (four times the recommended level) of the total bandwidth used for this session. These excess control messages waste bandwidth on the network and processing time at each RTP participant. This wasted bandwidth is especially a problem in the wireless environment where bandwidth is scarce. We suggest the excess messaging be eliminated by implementing dynamic session bandwidths which change with the actual bandwidth used.

(2) RTCP can not adapt the control bandwidth scalability.

Currently, fixed networks can provide high bandwidth resources, such as 155Mbps ATM and 1Gbps Ethernet etc. But wireless links always provides very low bandwidth capabilities to the end user. So it is commonly necessary to modify the data stream seen on the wireless network to reduce its bandwidth usage. However this can not be achieved by RTCP. For example, consider this common scenario: A multicast session on the 100Mbps Ethernet (fixed networks) is made up of one data source and 10 data sinks (perhaps an on-line lecture series). The data source is transmitting MPEG-1 video at 1.2Mbps. The control bandwidth is set to 5% of 1.2Mbps or 60kbps giving an approximate session bandwidth of 1.2Mbps. If a wireless host joins this conference, the manager must reduce the data bandwidth usage to 128kbps (teleconferencing quality) for support the wireless host. Unless the control bandwidth is also decreased, however, RTCP reports will continue to use 60kbps of bandwidth on the wireless link. Control message would then account for 32% of the total wireless bandwidth used for this session. Clearly, this is unacceptable.

(3) RTCP have not flexible message format sent

Finally, consider again the scenario of 1 data source and 20 receivers. While the data source may be interested in the reception statistics of all the receivers, clearly the receivers themselves would not be. The Receiver Report, however, are multicast to all participants, wasting bandwidth for those participants who do not make use of them. We suggest a mechanism to provide more flexibility in the RTCP packet format, allowing receiver or sender reports to be removed when desired, or replaced with summary information. As wireless hosts tend to be resource-poor, they will commonly act as receivers only and will benefit from the reduced bandwidth usage of the modified RTCP packet.

4. ARCHITECTURE FOR MULTIMEDIA COMMUNICATIONS

To overcome disadvantages of RTP in wireless, we suggest to insert a component, *filter*, in wireless host. We continue to use RTP architecture in wired network suggested by ITEF i.e. RTP is on the UDP/IP and bellow the application. Figure 1 shows our architecture for wireless multimedia communications on RTP.

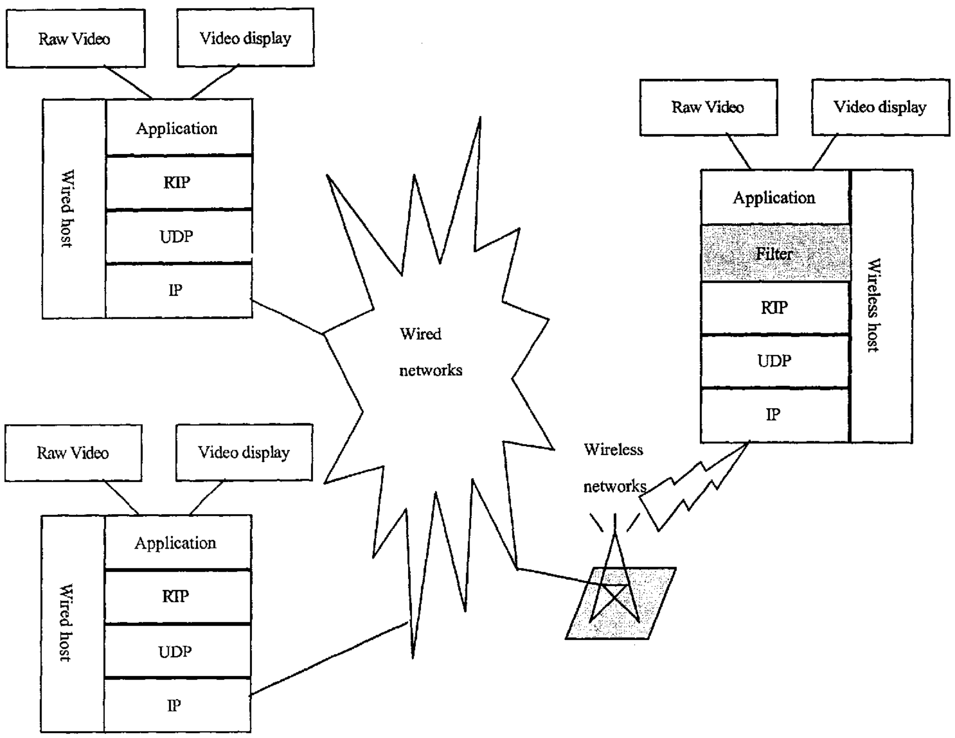

Figure 1 An architecture supporting wireless multimedia communications on RTP

The filter component has 3 functions to solve shortcomings as follows:

(1) Perceive and control session bandwidths

The filter can respond the situations by dynamically modifying the session bandwidth on the wireless link to match the true session bandwidth usage. The filter determines the true session bandwidth by packet feedback in RTCP receiver report (RR). Note that, as part of normal RTCP functionality, each RTCP participant already gets regular sender reports, allowing them to calculate the total bandwidth used for data within some time interval. Therefor each RTP participant could maintain a running estimate of the current session bandwidth used, and modify the rate at which it generates RTCP packet based on the value.

(2) Adapt control bandwidth scaling

The RTP specification gives an algorithm for calculating the frequency that sender and receiver reports should be sent based on the session bandwidth and the number of participants. If RTCP packets are not received from a participant at the required rate, the participant will be marked inactive and eventually dropped from the bandwidth calculation. If the filter filtrates out RTCP packets received from participants, these participants will be marked inactive by other participants. Thus, filter can adapt the control bandwidth by these marker, and effectively track the session bandwidth usage.

(3) Flexibly send RTCP message

The filter uses statistics receiver reports to conduct these participants that not needed multicast. Then it can reduce control bandwidth. The filter can periodically check these, and modify their requirements. If needed, it can return to send message again.

5. CONCLUSION AND FUTURE WORKS

In this paper, we discuss some disadvantages of Real-Time transport protocol (RTP) when deployed in wireless networks after overview of RTP protocol and present an architecture to deal with these problems. The main deficiencies of RTP in wireless networks are lack of dynamical control session bandwidth, lack of adaptive bandwidth scalability and lack of flexible message format sent. We insert a filter in our architecture on RTP layer to solve these problems. Wireless communications for multimedia is a complex question. In future, we will study the implement of the filter and its guarantees and management of QoS on RTP protocol in wireless networks.

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