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A REVIEW ON STREAMING MULTIMEDIA OVER WIRELESS NETWORKS

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Abstract: Wireless multimedia transmission has grown dramatically in recent years. The simplicity, flexibility, and low up-front costs of such systems have not only enabled mobility support for existing multimedia applications but also stimulated the development of new wireless multimedia services. Recent research shows that the wireless network conditions, such as the wireless link layer rate adaptation, contention, and interference can significantly degrade the performance of streaming media applications by incurring re-buffer events and degraded perceptual Quality. This paper reviews the research work related to the streaming in wireless. Existing research areas are covered, streaming multimedia performance, bandwidth estimation techniques and wireless network performance.

Keywords: Wireless, multimedia, streaming, network transmission

I.INTRODUCTION

Wireless networks had been widely deployed over the last few decades. Most of the protocols and applications that were developed for wired networks have been transferred to wireless networks for their actual implementations. However, the wireless characteristics that differ from wired networks may affect the performance of these applications in wireless networks. The network effects on media scaling can be reduced by adjusting streaming data rate by estimating bandwidth or it can be reduced by typical streaming rate selection which is used in media scaling and is based on loss rate and round-trip time. But wireless network conditions and target rates cannot be directly provided by the above measured factors which are affecting media performance [1]. Streaming multimedia quality is impacted by packet delay, jitter and loss due to the network congestion or other changes in network conditions. To mitigate the impact on quality by the network, various techniques have been used to improve streaming media quality, such as buffer optimization, streaming rate selection. This paper reviews the research work in buffering, streaming rate selection, and performance study for streaming multimedia over wireless networks.

II.STREAMING BUFFER

To provide better performance for streaming multimedia over best effort networks, such as the Internet and wireless networks, buffer techniques are often used on the server side, network (caching and proxy), and on the client side [1]. Client side buffering techniques play an important role in streaming multimedia. Generally, client side buffering provides the essential functionality of removing the jitter effects and playback disruption caused by oscillations in the transmission rate at the cost of initial start-up delay. The oscillations in transmission rate may be caused by transport protocols, such as TCP and TFRC that apply the Additive Increase and Multiplicative Decrease (AIMD) based congestion control, the network congestion, or the connection rate adaptation in a wireless network [2,3].

Client side buffers can prevent playback disruptions when the available bandwidth is temporarily below the streaming data rate; unless the buffer is also empty [4]. The number of rebuffer events, or the number of disruptions during playback is a critical performance quality metric. In general, the larger the buffer is, the lower the probability the buffer wills underflow. However, the initial startup delay is also an important quality metric, especially for real time and/or interactive application. For non-realtime or non-interactive streaming applications, buffer overflow is not a critical issue because disk and memory capacity are outpacing the growth in bandwidth available to single stream flow [5]. However, buffer overflow is an issue for mobile devices, such as the PDAs and cellphones, which can still be subject to memory or disk space constraints.

There are a variety of strategies proposed to improve the effectiveness of client side buffering that include slowing down the media playout rate at the client to reduce its consumption rate and help prevent buffer underflow [6]. Most of that research focuses on the minimum buffer size required in a particular streaming environment, while still keeping a low number of playback disruptions. Buffer management is also associated with other research topics, such as the smoothing of Variable Bit Rate (VBR) encoding and VCR like functionality on the client side, such as rewinding or indexing to an arbitrary point, which may require additional buffer space at the client [7].

Buffer Required for Flow Control and Jitter Removing Zimmermann et al. [8] describe buffer underflow and overflow behavior in detail under the ideal network condition for their streaming media system. A simple flow control with stream on/off watermarks are proposed with equations 1 and 2:

$$WMO \leq B - (RN - RC) \times Td \text{ …………(1)}$$

Where WMO is the buffer overflow watermark, B is the buffer size, RN is the streaming send rate, RC is the consumption rate, and Td is the network delay.

$$WMU \geq RC \times Td \text{……………..(2)}$$

Where WMU is buffer underflow watermark. WMO ≥ WMU must hold to make B the minimum buffer size required for the operating environment. However, in Equations 1 and 2, both the RN and RC are assumed as CBR, and all the buffered content is assumed to be playable, which is usually not true in the real world environment.

In addition, there are a variety of strategies proposed to improve the effectiveness of client side buffering by optimizing the buffer size based on jitter removal, such as [9]. To study the buffer size required for removing the jitter in networks, **Yuang et al.**[6], present Markov chain models based on Poisson arrival. However, in wireless networks, the arrival of streaming traffic can not be simply modeled as a Poisson distribution because the capacity changes in wireless networks causes a variance in the streaming traffic arrival rate. Therefore, the jitter removal buffer algorithm are not sufficient to avoid buffer underflow in wireless networks. The buffer optimization approach presented in this thesis is based on estimation of the network condition but does not only consider a TCP-Friendly rate. This allows it to apply to some environments where the buffer underflow is not caused by transmission rate changes of AIMD protocols, such as in some congested wireless networks, where MAC layer rate adaptation dominates the transmission rate changes.

a) Buffer Required for VBR Traffic Smoothing

Another function of the client side buffer is to smooth the VBR media content. Various techniques are used to reduce the traffic bursts of VBR encoded streaming content. Usually, a modest buffering capability is required at the receiver side. For example, a smoothing by temporal multiplexing algorithm is proposed in [10], whereby the VBR traffic is made more uniform by grouping frames and sending them at the average bandwidth. The smoothing algorithm also controls the critical bandwidth, which is defined as the minimum bandwidth required to guarantee the receiver's buffer will not underflow, to provide a smoother bandwidth and a more effective use of buffering.

Salehi et al. [11] propose a work-ahead smoothing algorithm to reduce traffic bursts by modeling the optimal transmission schedule which minimizes both the variance and peak rate at which data is sent to the client for a given buffer size. Alternate flow control algorithms, such as Multi-Threshold Flow Control (MTFC) [12] can also be used to improve buffer management and VBR traffic smoothing. By implementing multiple thresholds in the client side buffer, feedback messages are sent back to the server to adjust the sending rate. By combining the threshold control and a consumption prediction module, the algorithm proposes to achieve fewer rate adjustments and a higher buffer utilization.

Jenkac et al. [13] analyze the behavior of VBR media over VBR channels, such asthe variable bit rate caused by wireless Radio Link Control (RLC) layer retransmissions in cellular networks. By modeling the deterministic and random channel conditions, the algorithm determines appropriate initial delays and buffer sizes for streaming video over variable bit rate wireless channels.

b) Characterization of Streaming over Wireless Networks

With the development of the streaming techniques and wireless networks, more wireless streaming research has been conducted recently. This section reviews related areas of characterization and performance study for streaming multimedia in wireless networks. The research from ***Kuang et al.*** [14] characterizes the Real Media traffic over an IEEE 802.11b wireless network. Multiple network layer data gathered shows that the Real Media performs well for excellent and good channel conditions, and performs poorly for fair and poor channel conditions. The IEEE 802.11b MAC layer retransmission mechanism is able to hide most physical layer burst errors from higher layer protocols, even in the poor channel condition where 67.5% of the media packets sent require at least one retransmission

Furthermore, the Real Media application layer NACK-based (Negative Acknowledgment based) error control is effective in recovering missing packets. However, the network RTT changes and rebuffer events caused by MAC-layer retransmission and rate adaptation are not studied in their research. The stream Windows Media Service video over a wireless campus network and analyzes performance across application, network and wireless link layers. Some of the key findings include:

- Wireless LANs, i.e. IEEE 802.11g, make it difficult for streaming video to gracefully degrade as network performance decreases.
- Video streams with multiple encoding levels can more readily adapt to degraded wireless network conditions than can clips with a single encoding level.
- Under degraded wireless network conditions, TCP streaming can provide higher video frame rates than can UDP streaming, but TCP streaming will often result in significantly longer playout durations than will UDP streaming.
- Current techniques used by streaming media systems to determine effective capacity over wireless LANs are

inadequate, resulting in streaming target bit rates significantly higher than can be effectively supported by the wireless network.

Existing researches shows that the media scaling performance is limited when the optimal streaming rate is not correctly selected over wireless networks. Typical streaming typical rate selection used in media scaling is based on loss rate, round-trip time or a bandwidth estimate to adjust the streaming data rate to reduce the network impact on media performance. However, these measurements do not always provide clear indications of wireless network conditions and target rates for adaptation. For example, research [15] show that retransmissions and rate adaptation at the wireless MAC layer may reduce the loss rate while increasing the round-trip times measured by the applications. Thus, wireless network conditions hidden from the application can cause bad media scaling decisions. Moreover, media scaling action is usually taken during degraded performance and therefore is not effective in avoiding performance degradation.

III. STREAMING PERFORMANCE IMPROVEMENT OVER WIRELESS NETWORKS

This section reviews the related approaches to improve streaming performance for wireless networks, including novel transport layer protocols, cross-layer approaches, and bandwidth estimation approaches.

a) Bandwidth Estimation

Bandwidth estimation refers to the end-to-end measurement of bandwidth-related metrics, such as capacity, available bandwidth and bulk TCP transfer capacity, performed by the end hosts of a path without requiring administrative access to intermediate routers along the path. Several applications can benefit from knowing the bandwidth characteristics of their network paths. For example, peer-to-peer applications, overlay networks, Content Distribution Networks (CDN), intelligent routing systems, end-to-end admission control, and multimedia streaming applications can all benefit from bandwidth estimation techniques [16].

Streaming multimedia applications usually prefer to be rate-based, and often use UDP with higher layer congestion control mechanisms. Traditional congestion control mechanisms, which use a measured increase of packets/frames lost and/or delay as the indicators of congestion in the network are not sufficient for the streaming applications that require explicit rate based congestion control mechanisms. Loss rate and RTT only provide some indicators of congestion, but do not provide the clear extent of the congestion in the network. For example, the applications are not able to know the amount of traffic that can be sent out without causing congestion. Therefore, it is hard for the streaming applications to make the proper congestion control decisions. Moreover, for streaming applications in wireless networks, the loss rate and delay may not be caused by network congestion, thus this may misinform the streaming systems. Having the knowledge of capacity and available bandwidth can make the congestion control of streaming applications more efficient and accurate. Bandwidth estimation related metrics, techniques, taxonomy and evaluations by extending the bandwidth estimation survey to more recent and wider areas.

b) Bandwidth Related Metrics

The term bandwidth is often imprecisely applied to a variety of network throughput related concepts, such as capacity, available bandwidth, bulk transfer capacity and achievable throughput. Applications are usually concerned with different bandwidth related metrics. Therefore differentiating these concepts is important for the developing, evaluating and applying bandwidth estimation tools.

Capacity: Capacity is defined as the maximum possible bandwidth that a link or end-to-end path can deliver [16]. At the link layer, the transmission rate of each segment is usually fixed and constrained by the physical layer medium and the propagation delay. At the IP layer, each hop, which could be multiple link layer segments, delivers data at a rate lower than its nominal transmission rate due to the overhead of link layer encapsulation and framing. Prasad et al. [16] define the IP layer capacity by Equation 3:

$$C_{L3} = C_{L2} \frac{1}{1+\frac{H_{L2}}{L_{L2}}} \quad \ldots\ldots\ldots\ldots(3)$$

Where CL3 is the IP layer capacity, CL2 is the link layer capacity, HL2 is the total link layer overhead, and LL3 is the size of an IP packet.

Given a certain type of link layer network, the authors assume that the link layer overhead is fixed. Therefore, the IP layer capacity of a hop is defined based on the Maximum Transmission Unit (MTU) of the IP layer network.

Furthermore, the authors also define the end-to-end capacity C as

$$C=\min_{i=1\ldots H} C_i \quad \ldots\ldots\ldots\ldots(4)$$

where Ci is the capacity of i-th hop, and H is the number of hops in the end-to-end path. The hop with the minimum capacity is the narrow link on the path.

Some link layer technologies, such as IEEE 802.11 WLAN as described in Section 2.2.2, do not operate with a constant transmission rate. The capacity definitions in Equation 3 and 4 can be only used for those techniques during the time intervals in which the capacity remains constant.

Available Bandwidth: Available Bandwidth is defined as the maximum unused bandwidth at a link or end-to-end path in a network, which depends on not only the link capacity, but also the traffic load, and is typically a time-varying metric [16]. In the available bandwidth Ai of a hop i of a end-to-end link over a certain time interval is given by the unutilized fraction of capacity:

$$A_i = (1-U_i)C_i \quad \ldots\ldots\ldots\ldots(5)$$

Where ui is the average utilization of hop i in the given time interval, and the Ci is the capacity of hop i. By extending the available bandwidth definition to an H-hop path, the authors define the available bandwidth of the end-to-end path, A, as the minimum available bandwidth of all H hops:

$$A=\min_{i=1.....H} A_i \quad \text{.................(6)}$$

The hop with the minimum available bandwidth is called the tight link of the end-to-end path. The narrow and the tight link are both indicate the bottleneck of a network. However, based on the definitions, they are not necessarily at the same hop.

Bulk Transfer Capacity: Bulk Transport Capacity (BTC) [17] defines a metric that represents a network's ability to transfer significant quantities of data with a single congestion-aware transport connection (e.g., TCP). Thus, the BTC is the maximum long term average throughput obtainable by a single flow of an ideal TCP implementation on an end-to-end network path. The ideal TCP implementation here means that the TCP must implement all standard congestion control algorithms specified in IETF RFC 2581. However, RFC 2581 leaves several implementation details open, so different implementations on these details will yield non comparable measures of BTC. Therefore, any BTC measurement must specify the details about the congestion control algorithms that are not specified in RFC 2581. In RFC 3148, the BTC of a end-to-end path is defined as:

BTC = Data sent/Elapsed time

where data sent represents the unique data bits transfered, which does not include header bits or emulated header bits or retransmitted data, and elapsed time is the measurement interval.

BTC is different from available bandwidth in term of bandwidth metrics. BTC is TCP-specific, but available bandwidth does not depends on a specific transport protocol. Furthermore, BTC takes bandwidth sharing with other TCPs into the consideration, but available bandwidth assumes the average traffic load is constant and estimates the bandwidth available to the additional traffic [16].

Achievable Throughput: Achievable throughput is defined as the throughput of a host-to-host path under a completely specified set of conditions, such as transmission protocol, end host hardware, operating system, tuning method and parameters, etc.[18]. Achievable throughput is extremely application specific, and thereby represents the throughput that an application in this specific setting might achieve

Achievable throughput is different from the available bandwidth because the bottleneck could be in an end host, so achievable throughput may or may not correlate with available bandwidth. In addition, achievable throughput only takes into account the portion of the capacity that can be used by the specific application. For instance, a TCP friendly protocol may yield a lower achievable throughput than the available bandwidth as discussed above, while a UDP application may yield a higher achievable throughput than available bandwidth if it aggressively takes the bandwidth from other TCP-based applications. The definition of achievable throughput is close to the BTC measurement from the bandwidth sharing point of view. However, achievable throughput allows the use of parallel connections and could apply to both transport protocols with or without congestion control, i.e. TCP and UDP protocols.

The general relationship between the terminologies used as bandwidth related metrics can be presented as:

Capacity > Achievable Throughput ≥ Available Bandwidth > BTC

Figure 1 illustrates the relationship with the presence of crossing traffic. The capacity is the maximum possible bandwidth that includes the volume used by the crossing traffic. The achievable throughput could get a higher volume than the Available Bandwidth, which is the Capacity−Crossing Traffic. The reason is that Achievable Throughput considers the bandwidth that may be aggressively taken from responsive crossing traffic, such as TCP. Furthermore, the BTC only considers TCP traffic, which will take the congestion control into consideration. Therefore, the BTC volume is likely to be smaller than Available Bandwidth since it needs to maintain a TCP-Friendly share with the crossing traffic in the network.

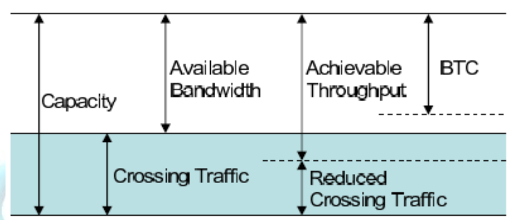

Figure 1: Bandwidth Related Terminology

c) Bandwidth Estimation Techniques

Based on the taxonomy and detailed review of current available bandwidth estimation techniques from the survey paper [16], and other related research, this section discusses four major active bandwidth estimation techniques: Variable Packet Size (VPS) probing, Packet Dispersion, Self-loading Probing, Probe Gap Model (PGM) and other related methodologies.

Variable Packet Size (VPS) Probing: VPS measures the capacity of each hop along an end-to-end path. VPS was first proposed by Bellovin [19] and first implemented in pathchar [20] by Jacobson in 1997. The RTT to each hop in the network can be approximated by the summary of three delay components: serialization delays, propagation delays, and queuing delays. VPS makes the following assumptions to utilize the RTT information to estimate the capacity of each hop:

- VPS assumes that each hop of a path increases the one-way delay of a packet by a serialization latency given by L/C, where L is the packet size and C is the hop's capacity.
- By sending multiple packets of the same size to each hop of the network, VPS assumes at least one packet will not encounter any queuing delay.

- Propagation delays are independent of the packet size and are constant for each hop.

The required RTTs for different packet sizes can be measured by sending ICMP messages to the network. VPS sends out ICMP messages with the Time-To-Live (TTL) field of IP header set to force the packets to expire at a particular hop. The router at that hop will discard the expired packet and return a Time Exceeded ICMP messages to the sender, which can be used to measure the RTT to a particular hop.

The VPS model has some advantages compared to other related bandwidth estimation techniques. First, VPS is able to measure the network capacity in an uncooperative environment, meaning it does not need special software on both the source and destination. Additionally, the VPS technique can measure the entire network path at each hop along the path. Finally, because VPS sends a large number of probing packets and records the minimum traversal times, it can mitigate the effects caused by crossing traffic.

Packets Dispersion: Packet dispersion techniques, such as packet pair or packet train probing, measure the end-to-end capacity of a network path. The packet pair dispersion techniques were first introduced in [21]. Packet pair dispersion sends two packets with the same size back-to-back into the network. After the packets traverse the narrow link, the time dispersion between the two packets is linearly related to the narrow link capacity. Packet train dispersion probing extends packet pair probing by using multiple back-to-back probing packets, however, the concepts are similar to that with of single pair.

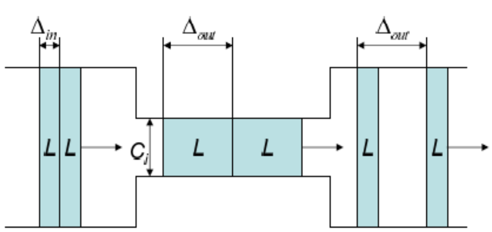

Figure 2: Packets Dispersion

Figure 2 illustrates the basic concept of packet dispersion. The most important assumption of packet dispersion techniques is that there is not crossing traffic during the packet pair probing. When packets of size L with initial dispersion in go through the link of capacity Ci, the dispersion after the link out becomes [16]. Compared to other bandwidth estimation techniques, packet dispersion techniques usually have a faster measurement time, and induce less stress on the network path. However, the effects caused by crossing traffic may significantly degrade the accuracy of the link capacity measurement.

IV.CONCLUSION

In this review paper we have discussed various research related to multimedia streaming buffer, Buffer Required for VBR Traffic Smoothing, Characterization of Streaming over Wireless Networks, Streaming Performance Improvement over Wireless Networks. This paper also reviews the related approaches to improve streaming performance for wireless networks, including novel transport layer protocols, cross-layer approaches, and bandwidth estimation approaches.

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