A Mobile-aware Adaptive Rate Control Scheme for Improving the User Perceived QoS of Multimedia Streaming Services in Wireless Broadband Networks

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Abstract

Recently, due to the prevalence of various mobile devices and wireless broadband networks, there has been a significant increase in interest and demand for multimedia streaming services such as the mobile IPTV. In such a wireless broadband network, transmitting a continuous stream of multimedia data is difficult to achieve due to mobile stations (MSs) movement. Providing Quality of Service (QoS) for multimedia video streaming applications requires the server and/or client to be network-aware and adaptive. Therefore, in order to deploy a mobile IPTV service in wireless broadband networks, offering users efficient wireless resource utilization and seamlessly offering user perceived QoS are important issues.

In this paper, we propose a new adaptive streaming scheme, called MARC (Mobile-aware Adaptive Rate Control), which adjusts the quality of bit-stream and transmission rate of video streaming based on the wireless channel status and network status. The proposed scheme can control the rate of multimedia streaming to be suitable for the wireless channel status by using awareness information of the wireless channel quality and the mobile station location. The proposed scheme can provide a seamless multimedia playback service in wireless broadband networks in addition to improving the QoS of multimedia streaming services.

The proposed MARC scheme alleviates the discontinuity of multimedia playback and allocates a suitable client buffer to the wireless broadband network. The simulation results demonstrate the effectiveness of our proposed scheme.

Keywords: Congestion control, rate control, multimedia, MARC, QoE

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1. Introduction

Due to the explosive growth of multimedia streaming services, demands for real-time video streaming over wireless broadband networks have increased. Recent advances in high-speed networks have made it feasible to provide real-time video streaming [1]. Pre-4G (4th generation) technologies, such as WiMAX (Worldwide Interoperability for Microwave Access) and LTE (Long Term Evolution), are emerging as a wireless communication system that provides high-data rates as well as a long-range coverage. Achieving higher quality and seamless streaming of video transmission over wireless broadband networks require coping with problems such as channel bandwidth variation, handover and transmission error. Among these problems, channel bandwidth variation and handover due to mobile stations' movement are the most critical problems. Channel bandwidth variation causes network congestion when the transmission rate of video streaming exceeds the channel bandwidth. In case of the mobile WiMAX, the adaptive modulation and coding (AMC) scheme, from half-rate QPSK to 5/6 64-QAM, offers various data rates to the MSs according to the distance between the base station (BS) and the MSs. Accordingly, the continuous streaming transmission of multimedia data is difficult when MSs are moving within wireless broadband networks. In addition, sudden disconnection due to the handover between BSs or sectors leads to errors in several frames because the error that has occurred in one frame would be propagated to the subsequent video frames due to inter mode prediction, which significantly degrades video quality [1][2].

To solve these problems, several methods for video streaming over wireless networks have been proposed. However, those methods have not offered an efficient way to solve user perceived QoS, called QoE (Quality of Experience), degradation because of channel bandwidth fluctuation in wireless network.

In this paper, we propose a new adaptive streaming scheme, called MARC (Mobile-aware Adaptive Rate Control), which adjusts the quality of bit-stream and the transmission rate of video streaming based on wireless channel status, which changes due to the movement of MSs; network status, which is changed due to congestion; and client status, which is the client buffer occupancy ratio. The remainder of this paper is organized as follows. In Section 2, related works are reviewed. Section 3 presents the overall architecture of the proposed MARC system and introduces the new rate control schemes for improving user perceived QoS. Section 4 presents the performance evaluation, and Section 5 presents the conclusion.

2. Related Works

2.1 Adaptive Modulation and Coding (AMC)

Real-time video streaming over any wireless network (e.g., 3G cellular networks, 802.11-based wireless local area networks and 802.16-based WiMAX) poses many challenges, including limited bandwidth, coping with bandwidth fluctuations and lost or corrupted data. Due to the growing popularity of multimedia streaming services over wireless networks, it has been well researched and many solutions have been proposed that combine audio/video processing techniques with the mechanism that are usually dealt with in the data link and physical layer, such as forward error correction (FEC) [1][3].

When a receiver gets a corrupted packet or video frame, it is in no position to correct the errors. However, if some redundant bits in the form of FEC are applied before the transmission, then there is a probability that the receiver would be able to detect and possibly correct the errors. The correction capability of these codes will depend on the kind and the length of the code used. In block codes, M redundancy bits are added to the N information bearing bits. (Note that these extra bits are generated using a generator matrix operation on the bits.) If we consider such MAC packet data units (MPDU), then the resulting bit loss probability is given by [3],

$$b = \sum_{i=M+1}^{M+N} \binom{M+N}{i} b_p^i (1-b_p)^{M+N-i} \frac{i}{M+N} \tag{1}$$

where b_p is the bit loss probability before decoding and b is the decoded bit error probability. The restore probability of such MPDU with payload size N bits and M code bits is given by, $p = (1-b)^{(M+N)}$. It can be argued that if the code M is increased keeping the payload fixed, then the resulting bit loss probability b decreases and packet restore probability p is increased. Thus, the FEC is used in broadband wireless networks such as mobile WiMAX. The FEC has also been successfully used in other wireless environments to support Quality of Service (QoS).

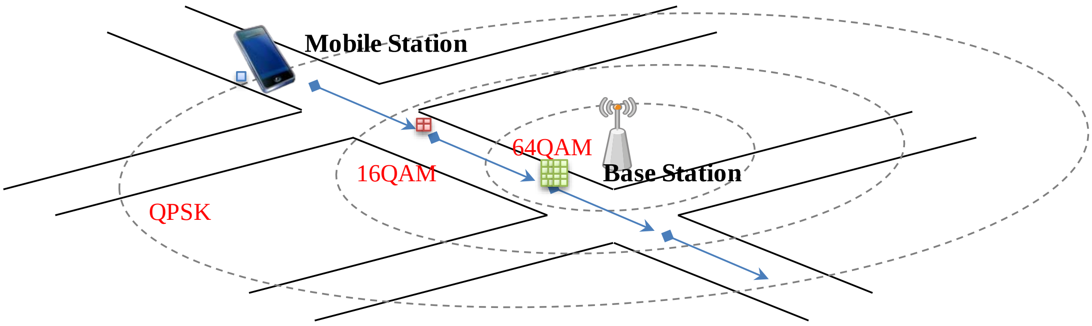

Fig. 1. The adaptive modulation and coding (AMC) scheme in the mobile WiMAX network

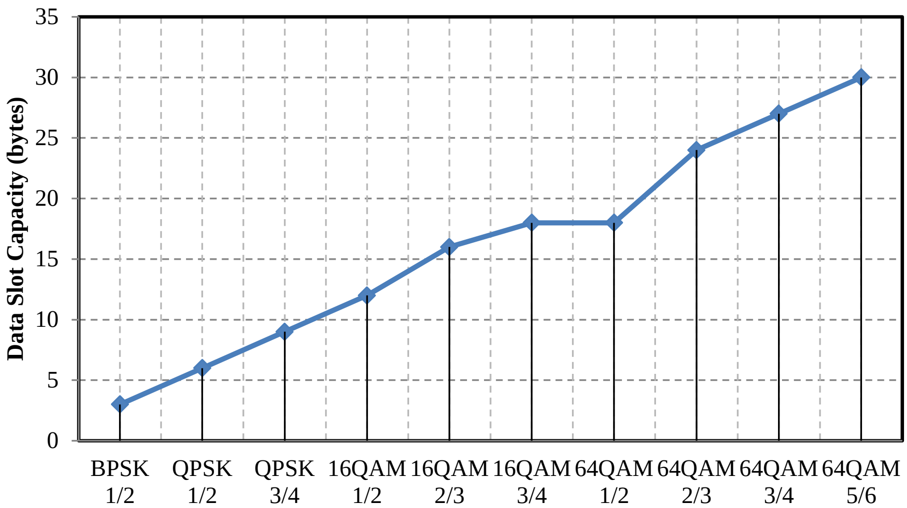

Fig. 2. Data slot capacity in mobile WiMAX using the AMC scheme (BW=10Mhz, (DL:UL)=(28:18), PUSC)

In case of the mobile WiMAX, as shown in **Fig. 1**, the adaptive modulation and coding (AMC) scheme from half-rate QPSK to 5/6 64-QAM offers various data rates to the MS according to the distance between the BS and the MS. For error-prone channels, this scheme provides high reliability. However, the AMC can make various physical resource allocations for video streaming sessions. As shown in **Fig. 2**, the data slot capacity, which is able to carry data sizes in a slot, is different when the FEC in the MCS (Modulation and Coding Scheme) level is changed. If the connection uses a higher FEC scheme to keep the transmission rate of streaming video, then this connection requires more physical resource slots than the previous one; furthermore, those slots are limited in the physical layer. For this reason, the AMC scheme can make it difficult to allocate the channel bandwidth.

Accordingly, the continuous streaming transmission of multimedia data is difficult when the MSs are moving in wireless networks. The AMC scheme is not an efficient way to improve the user perceived QoS of multimedia streaming services in wireless networks. The adaptive rate control scheme based on wireless network status should be considered to improve user perceived QoS.

2.2 TCP-friendly Rate Control

Excessive loss and delay have a devastating effect on video presentation quality, and they are usually caused due to network congestion. Thus, congestion control mechanisms at end systems are necessary to help reduce packet loss probability and delay. Typically, for video streaming, congestion control takes the form of rate control [4][5]. Rate control attempts to minimize the possibility of network congestion by adjusting the video stream rate according to the available network bandwidth. The feedback based rate control scheme is a widely used approach by multimedia streaming services. In feedback schemes, the client continuously monitors stream quality, such as the received data rate or data loss ratio, and sends feedback messages back to the server, which adjusts the data transmission rate accordingly.

A typical example of the TCP-friendly rate control algorithms is TFRC (TCP-Friendly Rate Control) for unicast streaming applications, proposed in [6], which explicitly adjusts its sending rate as a function of the measured rate of loss events based on the TCP throughput modeling equation developed in [7]. TFRC is a rate regulation algorithm to have a non-TCP connection behave similarly to, but more stable than, a TCP connection that traverses the same path. This means that a TFRC connection reacts to the network conditions, which is typically congestion indicated by packet losses. For this purpose, a TFRC sender estimates the network conditions by exchanging control packets between the end systems to collect feedback information. The sender transmits one or more control packets in one RTT. Upon receiving the control packet, the receiver returns the feedback information required for calculating RTT and packet loss probability. The sender then derives the estimated throughput of a TCP connection that competes for bandwidth on the path that the TFRC connection traverses. Finally, the TFRC sender adjusts its data sending rate to the estimated TCP throughput. According to [7], this approximation is reasonable for reaching TCP-friendliness. For this reason, many TCP-friendly rate control schemes use this analytical model. With these algorithms, the streaming server can scale video quality based on estimating available bandwidth. However, these adaptation schemes are only designed for particular congestion control mechanisms.

As does TCP, TCP-friendly algorithms react to indications of network congestion by reducing the transmission rate. The transmission rates of TCP-friendly algorithms are typically smoother than those of TCP [8]. Nevertheless, because network congestion occurs at multiple time scales, the bandwidth available to TCP-friendly streams typically fluctuates over several time scales [9]. Rate smoothing is not useful for a best effort network since the Internet does not provide any information about the bandwidth evolution in advance. Moreover, a smooth data rate does not always guarantee smooth video streaming [10]. In video streaming applications over the Internet, a quality adaptation mechanism is required to improve video stream quality because a TCP-friendly rate control algorithm is inefficient for providing a better video streaming service.

2.3 Quality Adaptation

Given a time varying bandwidth channel due to congestion control, the server should be able to maximize the perceived quality of the delivered video stream up to the level that the available network bandwidth will permit while preventing frequent changes in quality. This is the essence of quality adaptation. The quality adaptation mechanism maximizes perceptual video quality through minimizing quality variation, while at the same time increasing the usage of available bandwidth. Adaptive video streaming is not a new topic. Much research has been conducted in this area and various approaches have been proposed [4][10][11].

There are several ways to adjust the quality of a pre-encoded stored stream, including adaptive encoding, switching among multiple pre-encoded versions, and hierarchical encoding. One may re-quantize stored encodings on-the-fly based on network feedback. However, since encoding is CPU intensive, a server is unlikely to be able to do this for a large number of clients. Furthermore, once the original data has been compressed and stored, the output rate of most encoders cannot be changed over a wide range. In an alternative approach, the server keeps several versions of each stream with different qualities. As available bandwidth changes, the server plays back streams of higher or lower quality as appropriate.

With hierarchical encoding, the server maintains a layered encoded version of each stream. As more bandwidth becomes available, more layers of the encoding are delivered. If the average bandwidth decreases, the server may then drop some of the layers being transmitted. Layered approaches usually have the decoding constraint that a particular enhancement layer can only be decoded if all the lower quality layers have been received.

There is a duality between the adding or dropping of layers in the layered approach and the switching of streams in the multiply-encoded approach. However, the layered approach is more suitable for caching by a proxy for heterogeneous clients. In addition, this approach requires less storage at the server, and it provides an opportunity for selective repair of the more important information. The design of a layered approach for quality adaptation primarily entails the design of an efficient add and drop mechanism that maximizes quality while minimizing the probability of client buffer underflow. Most of all the quality adaptation mechanisms have adopted a layered approach.

Existing quality adaptation mechanisms use the layered encoding scheme, specifically the FGS (Fine Granularity Scalability) encoding algorithm, to match video stream quality with its determined transmitting rate. It provides an effective way for a streaming server to approximately adjust the quality of a video stream without transcoding the stored video stream [4]. However, with these algorithms, a streaming server changes the quality of the transmitting video stream only based only on estimating the available bandwidth; the video quality will often fluctuate. Above all, while most of the previous works have focused on the essentially hierarchical encoded video streaming, these quality adaptation schemes have only been designed for a particular congestion control mechanism. So it is desirable to have a general adaptive streaming scheme that can be applied in combination with different congestion control algorithms. Moreover, unlike MPEG-4, where the FGS mechanism is specified to use hierarchical coding, the scalable H.264/MPEG4-AVC extension standard [12], a recently proposed video compression scheme, does not contain any specification for this feature.

3. Mobile-aware Adaptive Rate Control (MARC)

This network-aware rate control function in existing streaming systems uses feedback information from the client network layer, which normally ignores the quality of multimedia streaming service and does not guarantee an improved quality of multimedia streaming over wireless networks. Furthermore, it does not support user perceived QoS of multimedia streaming services.

In this paper, we propose a mobile-aware adaptive rate control (MARC) scheme, as shown in **Fig. 3**, which does not only dynamically adjust the video transmission rate based on the channel bandwidth variation, but also adaptively adjusts the video quality level based on the wireless channel status and client buffer status. MARC basically uses RTP (Real-time Transport Protocol)/RTCP (RTP Control Protocol) model which is the standard streaming protocol. RTCP feedback periodically reports channel state, buffer state, and the network status information including RTT, packet loss rate. It uses to adjust the video transmission rate and the video quality level.

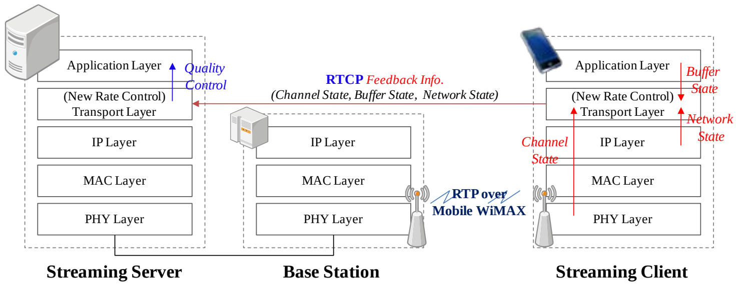

Fig. 3. Concept of the proposed adaptive rate control system

Fig. 4 shows the proposed scheme that consists of the network and client-aware rate control (NCAR) module and wireless channel and location-aware rate control (CLAR) module.

- CLAR module: It distinguishes between the core area and the edge area by using wireless channel status based on SIR (Signal-to-Interference Ratio), information and location of client based on RSSI (Received Signal Strength Indication), and CINR (Carrier to Interference-and-Noise Ratio) with Tx headroom, as shown in **Fig. 4**. It decides the desired maximum/minimum buffering time to adjust the quality of a bit-stream based on the location of mobile.

- NCAR module: It is based on network-aware congestion control and client-aware flow control scheme, as shown in **Fig. 4**. The congestion control adjust sever transmission bit-rate based on the estimated value of network-aware information such as network congestion degree (i.e., packet loss ratio), bandwidth variation, and adjusting interval. The flow control adjust the quality of a bit-stream based on the estimated value of the client-aware information, such as client buffer length, display size and battery state.

In order to adjust the quality of a bit-stream by using SVC (Scalable Video Coding) video extractor, which can conrol the bandwidth usage of the video stream over the wireless broadband network, the extractor of proposed system is controlled by events of CLAR and the flow control in the NCAR module. The transmission bit-rate of RTP packet streaming is only controlled by the congestion control in the NCAR module.

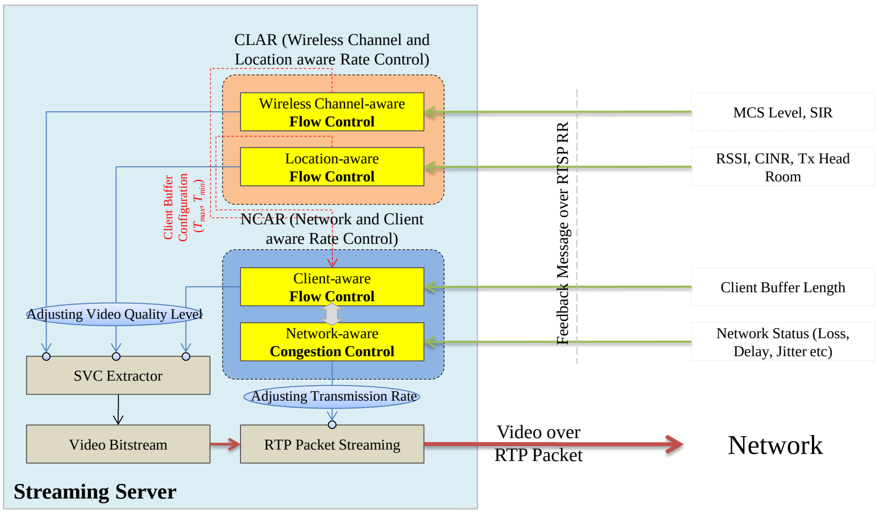

Fig. 4. The proposed network and client-aware rate control (NCAR) and wireless channel and location-aware rate control (CLAR) modules

3.1 Network and Client-aware Rate Control (NCAR)

In [11], we have proposed the NCAR scheme to improve the user perceived QoS of multimedia streaming services. The NCAR scheme has two key functions for rate-control, that is congestion control and flow control. The congestion control function adjusts the server's transmission bit-rate based on the estimated value of network-aware information such as the network congestion degree (i.e., packet loss ratio, delay) and bandwidth variation. The flow control, which prevents the server from overwhelming the client such that in over buffered and under buffered conditions, is the process of managing the bit-rate of media transmissions between the server and client to prevent a fast sender from over-running a slow receiver. In the NCAR, the flow control adjusts the quality level of the transmitting video based on the estimated value of the client-aware information, such as the buffer occupancy ratio.

The rate control of multimedia streaming seeks to achieve smooth playback quality, rather than simply transmit at the highest attainable bandwidth. Therefore, this calls for suitable mechanisms to smoothen the playback rate, which would otherwise end up in oscillation when a stream is sent over a traditional additive-increase multiplicative-decrease (AIMD) algorithm as in TCP. In order to reduce rate oscillation, we propose a new congestion control algorithm. This algorithm follows the additive-increase heuristic-decrease (AIHD) behavior. The consequence of this behavior is less aggressive data rate reduction compared to TCP's AIMD behavior when congestion occurs. The AIHD uses the following equations:

$$I : R_n = R_c + \left(\frac{PacketSize}{RTT}\right) \times U_f \tag{2}$$

$$D : R_n = \beta \times R_t + (1-\beta) \times R_c ; 0 < \beta < 1 \tag{3}$$

Such congestion control is based on the current transmission rate (R_c) that indicates at a time how many bits are allowed to be sent per one second. R_n is the new transmission rate and R_t is the estimated available bandwidth that is calculated using a well-know equation based TCP model to determine its transmission rate at a time and *RTT* is round-trip time. β is the weight parameters of heuristic decrease and U_f is the update frequency that is equal to $\Delta T / RTT$ where ΔT is the update interval time. **Fig. 5** shows the AIHD transmission rate (R) behavior during the additive increase phase. The rate from $R_{\min}$ crosses the maximum sustainable rate R_{st}. Once a congestion signal (i.e., packet drop, ECN bit mark or congestion event) is received, it decreases from $R_{\max}$ to $R_{\min}$ by the heuristic decrease equation, (3).

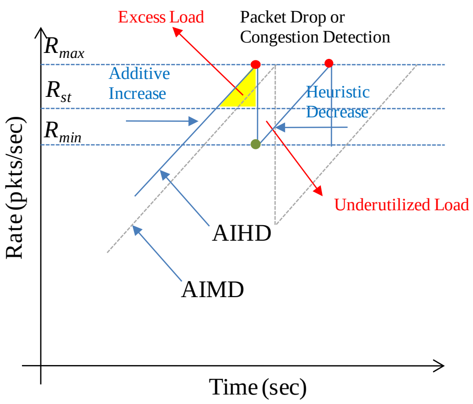

Fig. 5. AIHD rate of behavior (packets/sec)

For stability, we require that the amount of transmitted data when R is above R_{st} (the "excess load" in the **Fig. 5**) be no more than the unexploited transmission opportunity when R is below R_{st} (the "underutilized load" in the **Fig. 5**), i.e., $R_{st} > \frac{R_{\min} + R_{\max}}{2}$. Also, since our heuristic decrease rate is $R_{\min} = \beta \times R_t + (1-\beta) \times R_{\max}$, and we have (if $R_{st} > R_t$):

$$R_{st} = R_{\max} - \beta \times \left(\frac{R_{\max} - R_t}{2} \right) \tag{4}$$

When R_t is less than $R_{\max}$ ($R_{\max} > R_t$), to reduce the rate oscillation, we require $R_{st}(AIHD) > R_{st}(AIMD)$, or $\frac{R_{\max}}{4} > \frac{\beta}{2} \times (R_{\max} - R_t)$ where $1 < 2\beta / (2\beta - 1)$. In order to reduce the rate oscillation, β is given by the following conditions:

$$\beta > 0.5 \tag{5}$$

The client monitors the network conditions and gathers related information, and the server changes its transmission rate according to the available network bandwidth estimated from the packet loss ratio, RTT, and retransmission timeout (RTO) values. To estimate the available network bandwidth, the server uses one formulation of the TCP response functions described in (6) [4]:

$$T = \frac{PacketSize}{RTT \times \sqrt{\frac{2p}{3}} + t_{RTO} \times \left(3\sqrt{\frac{3p}{8}}\right) \times p \times (1 + 32p^2)} \tag{6}$$

Eq. (6) gives an upper boundary on the sending rate T in bytes/sec, as a function of the packet size $PacketSize$, round-trip time RTT, steady-state loss event rate p, and the TCP retransmit timeout value t_{RTO}. After estimating the available network bandwidth ($Rate_{ABW}$), as shown in **Fig. 6**, the server compares the available network bandwidth with its current transmission rate ($Rate_{Current}$) to determine whether or not network congestion has occurred.

```
Calculating Rate_ABW

if ( Rate_ABW > Rate_Current ) {
        Slow Start - Increase (I)
        if ( qlen_c > qmax_th)
                min { Quality_SVC (Level_Max), Quality_SVC (Level + 1)}
} else  if (Rate_ABW < Rate_Current ) {
        Congestion Control - Decrease (D)
        if (qlen_c < qmin_th)
                Max {Quality_SVC (Level_min), Quality_SVC (Level - 1)}
} else
        Steady State
```

Fig. 6. Pseudo code for the NCAR scheme

The NCAR has the flow control scheme that adjusts the quality level of the transmitting video by using the SVC bit-stream extractor. Through this adjustment, media playback discontinuity is alleviated and a suitable receiver buffer is allocated, resulting in low playback delay. The client monitors the client buffer conditions and gathers current decoding information, and the server changes the quality level of the transmitting video according to the resource status of the client buffer estimated from the current client buffer length ($qlen_c$), the quality level of the transmitting video bit-stream, and the minimum/maximum thresholds. To estimate the resource status of the client buffer, the server uses the minimum threshold ($qmin_{th}$) and the maximum threshold ($qmax_{th}$), which change based on the quality level of the transmitting bit-stream, as in the following equation:

$$q\max_{th} = \sum_{k=t_0}^{t_0+T_{\max}} \frac{R_{SVC}(k)}{PacketSize} \tag{7}$$

$$q\min_{th} = \sum_{k=t_0}^{t_0+T_{\min}} \frac{R_{SVC}(k)}{PacketSize} \tag{8}$$

These give upper and lower boundaries on the thresholds, as functions of the desired maximum buffering time ($T_{\max}$) such as maximum delay bounds, the desired minimum buffering time ($T_{\min}$) such as video pre-fetching time, bit-rate of SVC bit-stream at a time (t_0) $R_{SVC}(t)$. After estimating the resource status of the client buffer, as shown in **Fig. 6**, the server compares $qlen_c$ with $qmin_{th}$ and $qmax_{th}$ to determine whether the client buffer state is under buffered or over buffered. When $qlen_c > qmax_{th}$ and $Rate_{ABW} > Rate_{Current}$, the server increases the quality level of video transmission ($Quality_{SVC}(level)$). On the other hand, when $qlen_c < qmin_{th}$ and $Rate_{ABW} < Rate_{Current}$, the server decreases the quality level of video transmission.

3.2 Wireless Channel and Location-aware Rate Control (CLAR)

In wireless networks, the channel bandwidth variation and handover due to client movement are the most critical problems. Moreover, if the BS uses AMC (Adaptive Modulation and Coding) scheme, then the MS can have another problem, which is of the physical resource allocation variation. When the MS recedes from the BS, changed MCS (Modulation and Coding Scheme) levels can make it difficult to allocate channel bandwidth because the physical resource is limited in the BS. These problems are barely considered in the core area. Because the core area has a lower probability of the handover and can have a better channel condition than the edge area; however, in the edge area, these should be considered. In order to solve these problems, CLAR distinguishes between the core area and the edge area by using wireless channel status based on SIR (Signal-to-Interference Ratio), information and location of client based on RSSI (Received Signal Strength Indication), and CINR (Carrier to Interference-and-Noise Ratio) with Tx headroom, as is shown in **Fig. 7**.

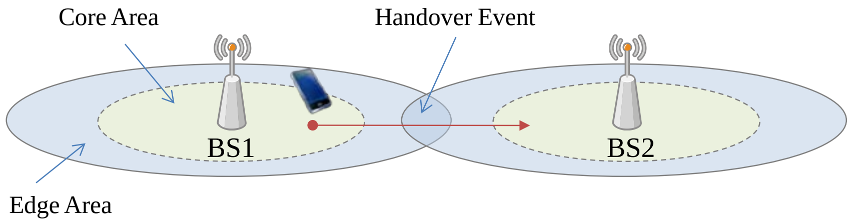

Fig. 7. Distinguishing between the core area and the edge area

The wireless channel status has already been used to adjust the MCS level in BS, which uses AMC (Adaptive Modulation and Coding) scheme. The CLAR scheme uses the MCS level to distinguish the client area, as is shown in **Fig. 8**. If the client area is the core area and the current MCS level ($MCSLevel_{current}$) is smaller than the area section thresholds of MCS level ($MCSLevel\_Section_{th}$), then the server changes into the edge area. The server decreases the quality level of video transmission and increases the $T_{\max}$ (the desired maximum buffering time) and $T_{\min}$ (the desired minimum buffering time) values, which are insensitive to changes in the client buffer. The server can transmit many video frames that have smaller bit-stream size than the previous bit-stream size to playback a video because, due to becoming insensitive, the quality level is rapidly decreasing and the quality level is slowly increasing. These operations can achieve media playback continuity in the edge area; however, the video quality will degrade more than the core area. If the client area is the edge area and $MCSLevel_{current}$ is larger than $MCSLevel\_Section_{th}$, then the server changes the client area into the core area. The server only adjusts the $T_{\max}$ and $T_{\min}$ to be more sensitive than the edge area. The quality level of video transmission will be changed via the NCAR scheme.

```
if (MCSLevel Changed) {
        if (MCSLevel_current > MCSLevel_Section_th and Edge Area)
            { Area Section: Core Area
             T_max = T_max/1.5 , T_min = T_min/1.5
        } else if (MCSLevel_current < MCSLevel_Section_th and Core Area)
            { Area Section: Edge Area
             Max { Quality_SVC (Level_min), Quality_SVC (Level - 1 )}
             T_max = 1.5 x T_max , T_min = 1.5 x T_min
        }
}
if (RSSI going down && Core Area) {
        if (CINR_current < CINR_Section_th and TxHeadRoom_current < TxHeadRoom_Section_th )
            { Area Section: Edge Area
             Max { Quality_SVC (Level_min), Quality_SVC (Level - 1 )}
             T_max = 1.5 x T_max , T_min = 1.5 x T_min
        }
} else if (Edge Area) {
        if (CINR_current > CINR_Section_th and TxHeadRoom_current > TxHeadRoom_Section_th )
            { Area Section: Core Area
             T_max = T_max/1.5 , T_min = T_min/1.5
        }
}
```

Fig. 8. The CLAR scheme pseudo code

When the client is located in the core area, sometimes the client can have lower channel conditions due to channel interferences. Therefore, as shown in **Fig. 8**, CLAR uses also the implicit location information; the server can estimate the location of the client by using RSSI, CINR and Tx headroom values. Because RSSI, CINR and Tx headroom values are always getting smaller when the MS recedes from the BS and, in another case, RSSI and TX headroom values are getting larger, except CINR value.

In the core area, the server adjusts the quality of a bit-stream and the transmission rate of the streaming video to provide both video quality improvement and media playback continuity by using the NCAR scheme. However, in the edge area, the server focused on media playback continuity. For example, as shown in **Fig. 9**, the server immediately decreases the quality level of video transmission the moment the client is located within the edge area. And, to be insensitive to changes in the client buffer, $T_{\max}$ and $T_{\min}$ are increased 1.5 times. The server can transmit more video frame which are low quality bit-streams than the previous; it can cause the client buffer to have enough video data to overcome congestion and handover in wireless networks.

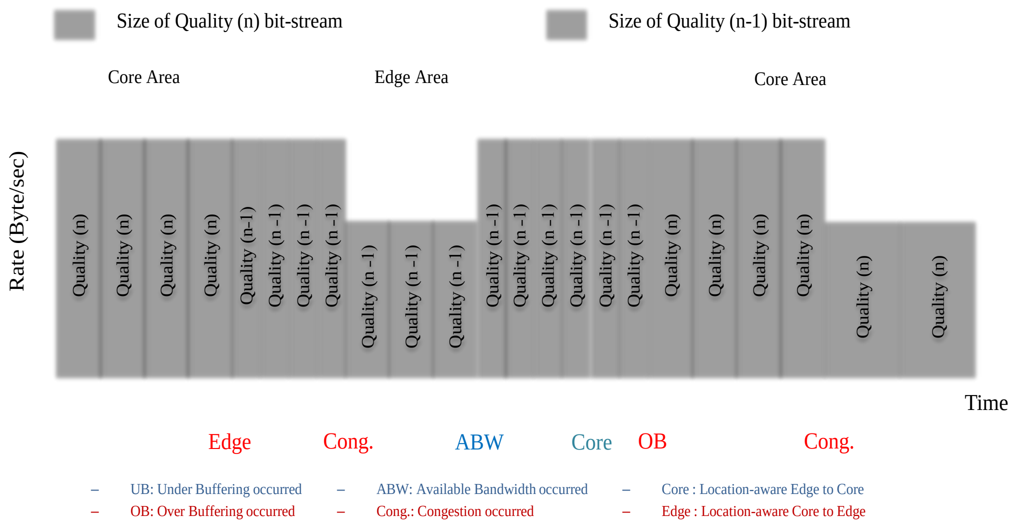

Fig. 9. Illustration of CLAR and NCAR operation

The proposed MARC, which consists of the NCAR and CLAR schemes, not only can improve the reliability and fairness of streaming data, but also alleviate the media playback discontinuity and rate oscillation for streaming video over wireless networks.

4. Simulation Results

This section presents the simulation results of MARC performance in a wireless network. In order to evaluate the performance of MARC, a number of experiments have been performed on the basis of the NS (Network Simulator) [13].

For the experiments, SOCCER_352x288_30_oring_02_yuv video clip was encoded into SVC profile layers using reference software JSVM (Joint Scalable Video Model) [14], as is shown in **Table 1**.

Table 1. Bit-stream characteristics of the SVC profile layers

JSVM Layer	Resolution	Frame Rate	Bitrate (kbps)	Spatial Level	Temporal Level	Quality Level	Y-PSNR (dB)	U-PSNR (dB)	V-PSNR (dB)
20	352x288	30	594.8	1	4	0	31.0413	39.7593	41.7566
33	352x288	30	813.5	1	4	1	32.4077	40.8846	42.9231
34	352x288	30	955.3	1	4	2	33.4442	41.3644	43.3295
35	352x288	30	1,083.40	1	4	3	34.7454	41.795	43.1284

4.1 Comparison of Performance in Wireless Network

We performed NS simulation for MARC, NCAR [11] and RAP (Rate Adaptation Protocol) [5]; however, there was no rate control (no RC) method, which is UDP based RTP streaming, to compare the performance of streaming quality in the wireless network. BS configuration in the simulation network is shown in **Table 2**.

Table 2. BS configuration in the simulation network

	MCS Level	Data Slot Capacity	Required Data Slot for 2Mbps	Channel BW for 47 slot/frame
Area 1	64QAM 3/4	27 bytes	47 slots	2 Mbps
Area 2	16QAM 3/4	18 bytes	70 slots	1.35 Mbps
Area 3	QPSK 3/4	9 bytes	140 slots	0.67 Mbps
Area 4	BPSK 1/2	3 bytes	417 slots	0.22 Mbps

The connection link, which is between the BS and the MS, used video streaming with the maximum capacity of 2Mbps at 3/4 64QAM; the maximum data slots of the BS downlink were limited to 47 slots. RTP was used for video streaming. RTCP messages were used for feedback information; it was periodically reported to the server at 0.5 sec intervals.

The simulation time was set to 200 sec. From 0 to 200th sec, the video server sent a video stream in Area 1. Beginning at the 15th sec, the MS moved to Area 2. After the 10th sec, the MS moved to Area 3, and the MS then returned to Area 2 after the 15th sec. After the 5th sec, the MS moved to Area 1. This scenario was repeated two times during simulation time. The client buffer size was set to 1,232 Kbytes; it can keep 800 packets in the buffer. The initial buffering time in the client was set to 2 sec, and the $MCSLevel\_Section_{th}$ was set to 3/4 16QAM.

Fig. 10 shows the discontinuity count and quality level of SVC bit-stream for MARC, NCAR, RAP and no RC. The RAP and non RC schemes do not change the quality level of the bit-stream because they do not have the quality adaptation scheme. Moreover, to adapt quality for channel variation, the MARC and NCAR schemes change the quality level during MS movement. However, RAP, NCAR and no RC schemes cause media playback discontinuity problems. The proposed MARC does not have the discontinuity count during video streaming.

As shown in **Fig. 11**, the resulting average PSNR of MARC and NCAR is similar to those of the RAP and no RC. The RAP scheme needs a large client buffer to playback without overflow, as is shown in **Fig. 11**. If the client buffer size is not enough to keep packets, then the RAP scheme causes an overflow problem. The no RC scheme cannot well utilize the client buffer. However, the MARC and NCAR schemes can allocate a more suitable client buffer than those by RAP and non RC.

As shown in **Fig. 12**, the network based rate control schemes, such as MARC, NCAR and RAP, which have congestion control to provide reliable and fair multimedia streaming, provide higher link utilization than the no RC scheme. Moreover, MARC and NCAR, which have flow control based on client buffer status, can alleviate underflow and overflow problems by using quality adaptation. MARC and NCAR can provide a seamless video playback service in wireless networks.

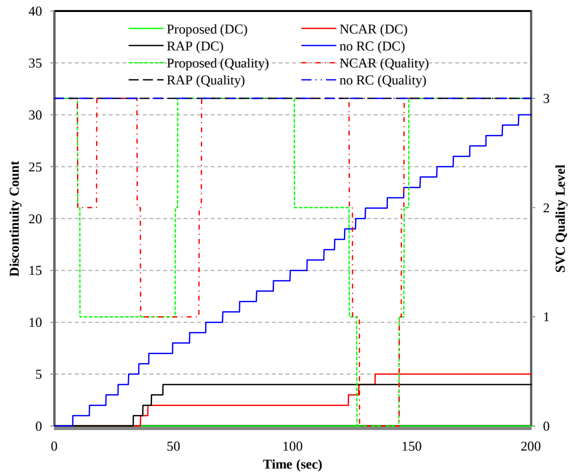

Fig. 10. Comparison of the discontinuity count and quality level of SVC bit-stream

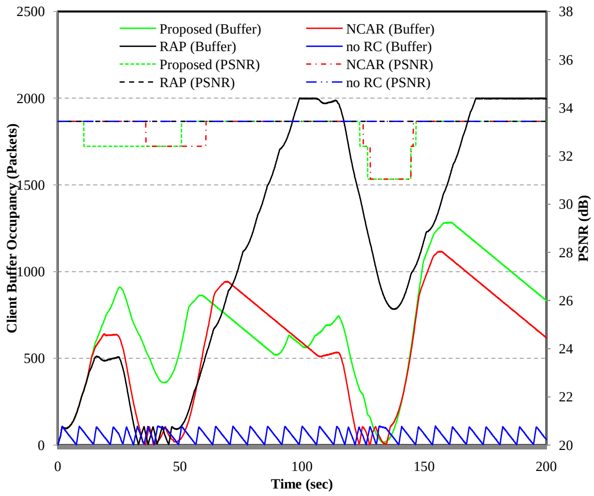

Fig. 11. Comparison of PSNR and the client buffer occupancy

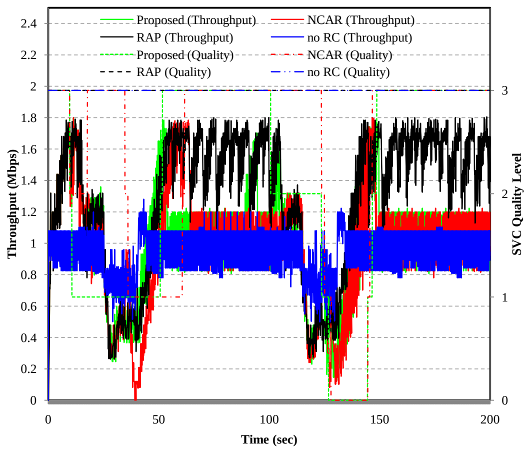

Fig. 12. Comparison of the throughput and quality level of SVC bit-stream

4.2 Performance Comparisons of Different Client Buffer Sizes

The proposed MARC adjusts the quality level of a bit-stream enough to be stored as a video bit-stream in the client buffer by being aware the wireless channel status. The stored video bit-stream is used to playback video during MS movement. This behavior seems sensitive for the client buffer size. To evaluate MARC performance with different client buffer sizes, a simulation was performed for the MARC and NCAR schemes to compare their performance in terms of various client buffer sizes, from 1,050 Kbytes to 1,950 Kbytes. The simulation configuration was used as shown in **Table 2**, and the scenario of section A was used. However, the initial start time of MS movement was changed from the 15th sec to the 35th sec to allow enough time for storing the video bit-stream in the client buffer. **Fig. 13** and **Fig. 14** show the discontinuity times and the packet loss rate based on client buffer size.

As shown in **Fig. 13**, the MARC scheme can alleviate media playback discontinuity. NCAR that does not have flow control based on the wireless channel status, and it has shown more discontinuity times than MARC that has CLAR. Moreover, the result shows that NCAR discontinuity times are inversely proportional to the size of the client buffer. For the loss rate, MARC has shown good performance, as is shown in **Fig. 14**. That is, before the packets are dropped in the client buffer due to buffer overflow, MARC can adjust the bit-stream quality level in advance by using CLAR based on the wireless channel status.

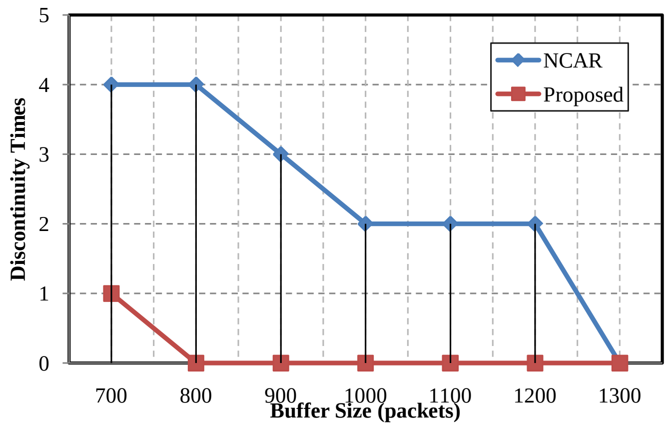

Fig. 13. Comparison of the discontinuity times of MARC and NCAR

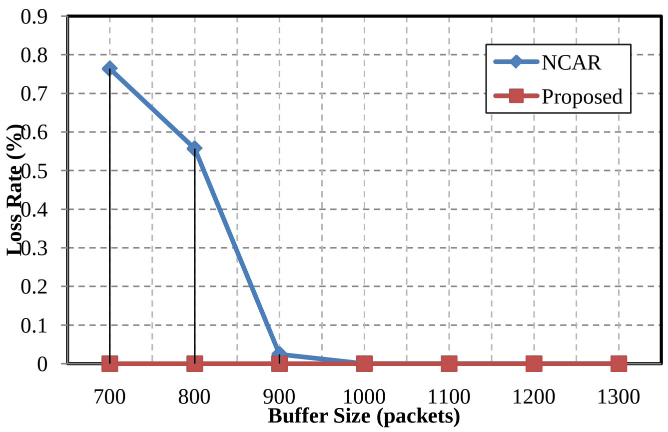

Fig. 14. Comparison of the packet loss rates of MARC and NCAR

5. Conclusion

In order to deploy mobile IPTV services in wireless broadband networks, it is important to achieve efficient wireless resource utilization and to seamlessly offer QoE (Quality of Experience) to users. Therefore, we have presented a new rate control scheme to improve the QoE of multimedia streaming services in broadband wireless networks, called MARC (Mobile-aware Adaptive Rate Control). The proposed MARC, which consists of the network and client-aware rate control (NCAR) scheme and wireless channel and location-aware rate control (CLAR) scheme, not only can improve the reliability and fairness of multimedia streaming, but also alleviate media playback discontinuity and rate oscillation.

Our simulation results reveal that the proposed MARC can appropriately control the transmission rates of video streaming depending on the MS status in the wireless network. The proposed MARC can also provide real time playback without media discontinuity during client movement in the wireless network. As the mobile Internet environment becomes more saturated with information and more advanced types of service flows, the proposed MARC will help to ensure the continued development of mobile Internet systems through helping to guarantee a smooth multimedia streaming within the system. We will further improve the stability of the proposed rate control scheme in a variety of wireless network environments and extend the MARC to real applications and systems.

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