A Key Issue on Multicast Video Streaming Over WIMAX Network

Chandra. R*, Saravanaselvi. P#

*IIyr M.E Applied Electronics student, Department of Electronics and Communication Engineering
Einstein college of engineering, Tirunelveli-12
rchandrab4u@gmail.com

#Assistant Professor, Department of Electronics and Communication Engineering
Einstein college of engineering, Tirunelveli-12
selvisaramee18@gmail.com

Abstract -The objectives carried out for the multicast video stream over WiMAX network are 1) The method involved to carry out its sub stream formation of video stream 2) The scheduling algorithm 3) Performance evaluation comparison for the existing methods are analysed and its comparative statement is tabulated. The novel idea is proposed for improving performance metrics such as good put, average delay and frame lost ratio to attain Quality of Service in Wimax networks.

Index Terms: WiMAX, Multicast, QoS, Video Stream

I. Introduction

Worldwide Interoperability for Microwave Access (WiMAX) is IEEE 802.16 standards. Wimax Technology works same as Wi-Fi does but more efficient. The data transmission can be routed via Wi-Fi. WiMAX technology provides higher speed connection up to 70 Mbps over the area of 30 miles. There is no need for line of sight connection between subscriber terminals and the base station in WiMAX technology. It will support low latency applications such as voice, video and Internet access at time. WiMAX has the potential to replace a number of existing telecommunications infrastructures. The high data throughput enables efficient data multiplexing and low data latency. Attributes essential to enable broadband data services including data, streaming video and high quality of service (QoS). The performance will enable transparency of quality of service between Mobile WiMAX and broadband wired services such as Cable and DSL.

WiMAX provides fixed, portable or mobile non-line-of sight service from a base station to a subscriber station, also known as customer premise equipment (CPE). This service should deliver approximately 40 megabits per second (Mbps) for fixed and portable access applications.

Mobile WiMAX takes the fixed wireless application and enables it like cell phone applications. For example, mobile WiMAX enables streaming video to be broadcast from a speeding police or other emergency vehicle at over 70MPH. It potentially replaces cell phones and mobile data offerings from cell phone operators. It offers superior building penetration and improved security measures over fixed WiMAX. Mobile WiMAX will be very valuable for emerging services such as mobile TV and gaming.

This paper is organized in such a way that the efficient multimedia broadcast framework over mobile WiMAX networks utilizing the MBS features. The mathematical formulation of problem for selecting the best set of sub streams from the scalable video streams is analyzed to maximize the quality for mobile receivers. An approximation algorithm is proposed to produce near optimal solutions.

The rest of this paper is organized as follows. In Section II, the existing video streaming methods are analyzed. In section III the method is studied to obtain sub stream from the videos. In section IV, the various scheduling algorithm for WiMAX networks is analyzed. In section V, the comparative analysis table is produced.

I. RELATED WORK

A video streaming over WiMAX network is composed of three main entities 1. Source content 2. Wimax Base Station (BS) 3.WiMAX Subscribers Station (SS). Source contents are national TV broadcasters, local broadcaster, internet TV operations and other video broadcast service providers. Multimedia contents are aggregated from different sources and sent to the WiMAX base station. The WiMAX base station constructs a schedule to transmit the incoming data to the subscribers.

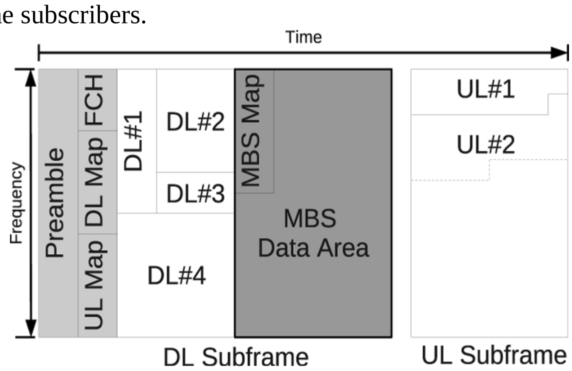

Figure. 1. Frame structure in WiMAX.

In the WiMAX physical layer, data is transmitted over multiple carriers in Time Division Duplex (TDD) frames. As illustrated in Figure 1, each frame contains header information and upload/download maps followed by bursts of user data. Since video dissemination is expected to be a prevalent traffic pattern, the WiMAX standard defines a service called Multicast and Broadcast Service (MBS). To facilitate broadcast and multicast in the MAC layer Using MBS, a certain area in each TDD frame can be set aside for multicast-only or broadcast-only data. The entire frame can also be designated as a download-only broadcast frame. The problem of selecting optimal sub streams of scalable video streams under bandwidth constraints. Solving this problem is important because it enables the network operator to transmit higher quality videos or more number of video streams at the same capacity.

Cohen et al. [1] combined a group of TDD frames together into a super-frame. They describe a cost based scheme where a cost function is associated with each user-channel pair. Three user interaction models are considered: (i) user can either be statically hooked to a channel, (ii) user can choose to listen to a channel, or (iii) the user channel association can keep changing based on the transmission medium conditions but in this paper does not consider the delay requirements which are central to video streaming.

Jiang et al. [2] proposed a scheme to transmit scalable video streams in which two layers of each video are transmitted separately. The base layer is transmitted as one stream over a reliable channel while the enhancement layer is transmitted as a different stream over a less reliable channel. Conceptually, this work implements a rate adaptive multiple description coding. However, it describes only one stream and it does not address the resource management problem arising in multi stream transmission scenarios.

Reguant et al. [3] considered splitting a video stream into two streams and transmitting them over two different broadcast networks. The first stream is transmitted over a DVB-H network at all times while the second stream is transmitted over WiMAX network most of the time. If the user wants to use some other non-video application in parallel, the stream going through WiMAX is degraded to accommodate that application. This ensures a minimum video quality at all times while maintaining the flexibility of using other applications. WiMAX applications and do not utilize MBS. In contrast, our approach considers a multimedia-intensive system with extensive use of MBS.

Shi et al. [4] propose a burst scheduling algorithm for energy minimization on per subscriber basis for unicast data. The algorithm arranges the mobile subscribers in ascending order based on the ratio of the current data arrival rate to the required data rate. If the current rate is significantly higher than the required rate, the mobile subscriber can go to sleep for some interval. After computing the sleep intervals for all mobile subscribers the bursts are scheduled in a longest interval first manner. After transmission of each burst, the algorithm checks to ensure that the data requirements of all mobile subscribers are being satisfied. The work in [4] is designed for unicast streaming of video and does not consider the multicast/broadcast service. Also the algorithm requires maintaining state information of all mobile subscribers served by a base station.

Liao and Lee [5] suggest a scheduling scheme where the uni-cast data is clustered around the multicast data bursts. They assumed that the burst length and positions for a particular stream is the same in all super-frames.

The work in [6] is designed for unicast streaming of video and does not consider the multicast/broadcast service. Also the algorithm requires maintaining state information of all mobile subscribers served by a base station. Then they present an enhancement to the longest virtual buffer first scheduling algorithm proposed by Shi et al. [6] by clustering the unicast data around the multicast data bursts. Their work evaluates the energy efficiency in a multi-class traffic scenario, whereas our work is focused on the energy efficiency of the video broadcast service.

II. SUBSTREAM FORMATION

A. *MULTIPLE DESCRIPTIONS CODING (MDC):*

Multiple descriptions coding (MDC) is a coding technique that fragments a single media stream into n sub streams ($n \geq 2$) referred to as descriptions. The goal of MDC is to create several independent descriptions that can contribute to one or more characteristics of video: spatial or temporal resolution, signal-to-noise ratio, frequency content. Descriptions can have the same importance (balanced MDC schemes) or they can have different importance (unbalanced MDC schemes). The more descriptions received, the higher the quality of decoded video. The quality of a stream can be expected to be roughly proportional to data rate sustained by the receiver. MDC allows for rate-adaptive streaming. Content providers send all descriptions of a stream without checking the download limitations of user. Experiments have shown that Multiple Description is very robust: the delivered quality is acceptable even at high loss rates.

B. *LAYERED CODING:*

The majority of codecs uses Single Description (SD) video coding. This approach does not partition any data at all. MDC is a form of data partitioning, thus comparable to layered coding as it is used in MPEG-2 and MPEG-4. In contrast to MDC, layered coding mechanisms generate a base layer and n enhancement layers. The base layer is necessary for the media stream to be decoded; enhancement layers are applied to improve stream quality. However, the first enhancement layer depends on the base layer and each enhancement layer n + 1 depends on its subordinate layer n, thus can only be applied if n was already applied. Hence, media streams using the layered approach are interrupted whenever the base layer is missing and as a consequence the data of the respective enhancement layers is rendered useless. The same applies for missing enhancement layers. Many different techniques exist to generate a layered video bit stream. The most common types are known as temporal layering, spatial layering and signal-to-noise-ratio (SNR) layering. Temporal layering is achieved by distributing the individual images of a video sequence over a set of layers. Thus, the more temporal layers that are used in the decoding process, the higher the frame rate of the video. In spatial layering, a multi resolution representation is used to split each image into set of layers. In this case an increased number of reconstruction layers correspond to higher spatial resolution of the individual images of the video. In SNR layering, the amount of lossy compression applied through quantization is progressively adjusted. All three types of layering are highly appropriate both for videoconferencing applications and streaming applications. Temporal layering has the lowest complexity and is the easiest to implement, since it precludes manipulation of individual images. However, temporal layering affects the design of the inter-frame compression scheme of the video coder, since inter-frame dependencies imposed by the temporal prediction must be resolvable by a decoder that only receives a subset of the temporal layers. Spatial layering is highly desirable since it makes it possible to decode the video at different spatial resolutions.

C. Scalable Video Coding (SVC):

The objective of the SVC standardization has been to enable the encoding of a high-quality video bit stream that contains one or more subset bit streams that can themselves be decoded with a complexity and reconstruction quality similar to that achieved using the existing H.264/MPEG-4 AVC design with the same quantity of data as in the subset bit stream. The subset bit stream is derived by dropping packets from the larger bit stream. A subset bit stream can represent a lower spatial resolution, or a lower temporal resolution, or a lower quality video signal (each separately or in combination) compared to the bit stream it is derived from. The following modalities are possible:

1. *Temporal (frame rate) scalability*:

   The motion compensation dependencies are structured so that complete pictures (i.e. their associated packets) can be dropped from the bit stream. (Temporal scalability is already enabled by H.264/MPEG-4 AVC. SVC has only provided supplemental enhancement information to improve its usage.)

2. *Spatial (picture size) scalability*:

   Video is coded at multiple spatial resolutions. The data and decoded samples of lower resolutions can be used to predict data or samples of higher resolutions in order to reduce the bit rate to code the higher resolutions.

3. *SNR/Quality/Fidelity scalability:*

   Video is coded at a single spatial resolution but at different qualities. The data and decoded samples of lower qualities can be used to predict data or samples of higher qualities in order to reduce the bit rate to code the higher qualities.

4. *Combined scalability:*

   A combination of the 3 scalability modalities described above. SVC enables forward compatibility for older hardware: the same bit stream can be consumed by basic hardware which can only decode a low-resolution subset (i.e. 720p or 1080i), while more advanced hard9ware will be able decode high quality video stream (1080P)

IV. SCHEDULING ALGORITHM

A. *Deficit Round Robin (DRR):*

If the packet size is less than the quantum assumed it is accepted. If the packet size exceeds the quantum then, only quantum size is accepted from connection and a deficit counter is set with a value equal to difference between quantum and packet size. At the end of each Round, if there are any bytes remaining in deficit counter, the amount of bytes is carried over to the next round.

B. ***WEIGHTED ROUND ROBIN(WRR):***

Weighted round robin scheduling algorithm the packets are transmitted in first in first out (FIFO) queues. Each queue in the input buffer has its own weight. Based on the weight the packet will be transmitted. The WRR scheduler operates in rounds. In each round the scheduler visits each queue in a round robin fashion starting with queue one. During each visit of a queue one or more packets may be serviced.

C. ***VIRTUAL CLOCK SCHEDULING ALGORITHM(VC):***

Virtual clock scheduling Algorithms used for scheduling play a major role in **providing** quality of service in video streaming. Virtual clock control average data rate of data transmission. The algorithm stamps the packets with virtual clock values and transmits the packets in the ascending order of their stamps. According to the flows specified transmission rates the virtual clock algorithm determines which packets should be forwarded.

D. ***RATE BASED(RB) METHOD:***

The video play back service is divided into levels depends on client buffer level& all clients within the same level of service. RB method is used to minimize the probability of request drops from handoff clients. to reduce the request drop probability. When a request is generated, each request specifies its desirable bandwidth *M* and minimum bandwidth *m* to the system. The difference between *M* and *m* is called bandwidth loss tolerance (BLT). A fraction of BLT, called actual borrowable bandwidth (ABB).

If the cell does not have enough bandwidth, the existing request may temporarily give up a certain amount of bandwidth by moving down to a lower service level. As soon as bandwidth becomes available, the borrowed bandwidth will be returned to the degraded request. The important feature is that no request will be served below its minimum bandwidth requirement once it is admitted.

E. ***BUFFER SENSITIVE BANDWIDTH ALLOCATION:***

The playback of a video will start only after a buffer level, called preferred buffer level (PBL). Based on the play rate of a video, the system determines the buffer playback duration (BPD) of the video frames at the buffer. BSBA method is to allocate the limited mobile bandwidth at the base station to the concurrent request in the cell based on their playback buffer duration.

The allocation of bandwidth is divided into 2 phase. The bandwidth allocated to serve a client request is the minimum of the bandwidth that is equally divided from the available bandwidth.

$$BW\_Adi = min\left(\frac{BW\_available}{n}, \overline{BW\_consumei}\right)$$

Where i, n=number of concurrent requests

i) More bandwidth will be allocated to the client whose buffer playback duration is shorter.

$$BW_{ASi} = \left(\frac{(buffer\ playback\ duration)^{-1}}{(\sum_{i=1}^{n} buffer\ playback\ duration)^{-1}} \times (1\_BW\_AdFR) \right)$$

Where

BW_ASi =total bandwidth allocated at the first phase

F. *BUFFER SENSITIVE RATE-BASED (BSRB):*

The buffer level at a client can be easily calculated by the server based on the playback time of the last frame transmitted to the client. Based on the buffer level, the buffer playback duration (BPD) at a client is determined. If BPD is large, it can tolerate a longer period of transient overloading.

The system divide the service into level based on the minimum & expected workload requirement of each request. The entire request in the same cell will be served at the same level. then actual amount of bandwidth to be allocated to a request I at service level n by using

$$bandwidth\ for\ request\ I = n - \frac{buffer\ level\ at\ client\ I}{Banwidth\ adjustment\ period}$$

III. PERFORMANCE ANALYSIS

In p. sankar [6] the simulation has been carried out for video with and without network load. The performance parameters Throughput, loss rate and delay-jitter are evaluated at the Destination. The throughput, loss rate and delay Jitter was measured for 1sec for the varying size frames. Actual throughput in bits per sec for the three algorithms is given in Table. I. It was observed that DRR and VC gives a better throughput when compared to WRR. the VC algorithm drop rate is lowest in comparison with WRR and DRR. It is concluded from the result that the dropping starts only when the routers are having insufficient buffer capacity. Delay jitter is a performance measure that shows the delay variation between arrival times of consecutive packets. VC gives less jitter characteristics due to time stamping as compared to other algorithms As shown in Table I.

algorithm	**Throughput (Bits/sec)**	**Loss Rate (Bits/sec)**	**Delay Jitter**
Weighted Round Robin(WRR)	2,46,724	37,853	low
Deficit Round Robin(DRR)	2,58,016	50,399	High
Virtual clock scheduling algorithm(VC)	2,73,368	37,853	Lowest

Table I: performance

In order to simplify the simulator, Kam-Yiu Lam1[7] only simulate a single cell with a base station and a number of mobile clients. The set of mobile clients in the cell are divided into three groups and each group of clients has different buffer sizes and request videos with different workload characteristics. The performance parameters Frame lost Rate, Request Drop Rate and Normalized mean service level was calculated for Rate based(RB) method , Buffer Sensitive bandwidth allocation(BSBA) and Buffer Sensitive Rate Based(BSRA) algorithm. The RB method have higher frame lost rate than Buffer sensitive Bandwidth allocation(BSBA), Buffer Sensitive Rate Based(BSRA) as shown in table II.

algorithm	**Frame lost rate**	**Request Drop rate**	**Normalized mean service level**
Rate based method(RB)	98%	0.09	0.089
Buffer sensitive Bandwidth allocation(BSBA)	35%	0.125	-
Buffer Sensitive Rate Based(BSRA)	34%	0.859	0.001

Table II: performance

IV. CONCLUSIONS

In this work the key issues involved in the multicasting of video stream over wimax is carry out. The mathematical solution is analysed for selecting the optimal sub streams of scalable video streams under bandwidth constraints to maximize the quality for mobile receivers.

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AUTHORS

Chandra.R –.chandra.R, PG *Applied Electronics* student, Einstein college of engineering and rchandrab4u@gmail.com.

Saravanaselvi. P – Saravanaselvi. P, *Assistant Professor, Department of Electronics and Communication Engineering* Einstein college of engineering, Tirunelveli-12and email address and *selvisaramee18@gmail.com*