AN ENHANCED APPROACH FOR AUGMENTATION OF SYNCHRONIZED BANDWIDTH REQUEST IN WIMAX

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Abstract

This paper carried out a study on the bandwidth request for real-time polling services. In our study, we discovered that although the base station granted the subscriber station to send the bandwidth request, the subscriber station may not be able to allocate the bandwidth request. It is due to processing delay and multicast polling in the subscriber station, which results the bandwidth request being padded unintentionally. The loss of bandwidth requests will cause the degradation of the real-time polling service performance. Therefore, we propose a scheme to overcome this problem. The results of the experiment show that the proposed scheme improves the performance of real-time polling services.

Keywords:*Quality of service, Bandwidth request, Real-time traffic, WiMAX*

1. Introduction

In WiMAX, there are five scheduling services; unsolicited grant service (UGS), extended real-time polling service (ertPS), real-time polling service (rtPS), non real-time polling service (nrtPS) and best effort (BE) service as stated in [1 - 3]. UGS is a service that transmits fixed size data packet on real-time periodic basis. It does not send a bandwidth request before transmitting its data but it uses the reserved bandwidth to perform the data transmission. ErtPS is an improved version of rtPS service. It has reserved bandwidth and it is also allowed to request bandwidth if the reserved bandwidth is insufficient. In contrast, rtPS, nrtPS and BE do not have any reserved bandwidth to transmit data. Hence, bandwidth requests are required in order for rtPS, nrtPs and BE to transmit their data in the next uplink frame. In the scenario where there is sufficient bandwidth to poll each subscriber station (SS) individually; rtPS uses unicast polling to send a bandwidth request to base station (BS). The BS will grant dedicated allocations in periodic basis to each SS in unicast polling. Although the BS granted the SS a dedicated allocation to send the rtPS bandwidth request, SS may not be able to allocate the bandwidth request due to the losses in bandwidth request queuing process and it results the bandwidth allocation being padded unintentionally. Secondly, multicast or broadcast polling may be used when there is insufficient bandwidth to poll each SS individually. Multicast or broadcast polling is a contention-based bandwidth request. Hence, even though rtPS is a higher priority scheduling service, its allocation for bandwidth request is not guaranteed. In our study, “loss of bandwidth request" is referred to the occurrence when the rtPS bandwidth request is not sent but lower priority bandwidth request is sent. The loss of a bandwidth request causes the degradation of the Quality of Service (QoS) priority structure. Subsequently, the next uplink subframe will experience insufficiency of bandwidth allocation when the bandwidth requests are not successfully received by the BS. We propose an alternative scheme to address these issues and to improve the throughput, delay and jitter of real-time traffic.

There are two types of contention-based bandwidth requests in Orthogonal Frequency-division Multiple Access (OFDMA), which are normal contention-based and codedivision- multiple-access (CDMA) contention-based. First, we carry out a comparison on the performance for the two types of contention-based bandwidth requests. Normal contention-based is a mechanism that allows SS to send the bandwidth request during a Region-Full request (REQ). Upon requesting for bandwidth, SS selects a ranging code from the subset with equal probability to allocate the bandwidth request. The ranging code modulates the ranging sub channel and it is transmitted during uplink. The BS provides an uplink allocation for the SS by sending the broadcast connection identifier (CID) with a CDMA_Allocation_IE, which specifies the transmit region and ranging code that should be used by a SS. A SS can determine whether it has been given an allocation by matching the parameters. The comparison of normal contention-based and CDMA contention-based will henceforth be referred to as "Comparison 1". As the result, the better contention-based bandwidth request mechanism; CDMA, which is suitable for the rtPS traffic is selected for further work and it is used to compare with our proposed scheme and known as "Comparison 2".

The rest of the paper is organized as follows. In Section 2, related information is reviewed. An overview of bandwidth request mechanisms is discussed in Section 3. In Section 4, the details of the proposed scheme are explained. Section 5 describes the simulation model and network scenarios. In Section 6, performance evaluation results and analysis are presented. Finally, in Section 7 draws the conclusion.

2. Related Work

[1 - 3] define that service scheduling is a data handling mechanism, which allocates uplink (UL) and downlink (DL) transmission opportunities. Each connection has a set of quality of service (QoS) parameters, which is determined by the scheduling services based on the connection requirements. UGS is to support the transmission of a fixed size of data packed on a real-time periodic basis, i.e. voice over Internet Protocol (VoIP). ErtPS is a new scheduling service that supports delay sensitive real-time applications, i.e. VoIP with silence suppression. rtPS targets for variable size of data packet based on periodic basis, i.e. video streaming while ntrPS is for delay tolerant data traffic. BE is targeted for web services and it does not have specific requirement or QoS guarantee.

UL bandwidth request allocation process is performed by the BS to provide each connected SS an opportunity to request bandwidth for the next UL transmission. The BS scheduler estimates the throughput and latency needs of UL traffic by its scheduling type and QoS parameters. In [4], a multi-hop polling service scheme was proposed. The proposed scheme configures the SSs to send their bandwidth requests to a relay station at respective polling intervals. The relay station generates an aggregate bandwidth request to the BS by accumulating all the bandwidth requests from all SSs. The BS grants bandwidth to the intermediate relay station instead of allocating bandwidth to SSs directly. The relay station will allocate bandwidth to individual SSs later. In [5], the proposed scheme allocates the bandwidth to a zone instead of individual users. Adaptive selection of the zone size is proposed to fit the user's mobility requirement. This paper considered the status of the current relay station and its neighboring relay stations within the zone size in hop count when allocating bandwidth. However, in [6], the rectangular burst construction may render resource wastage as there are unused slots within the burst (internal bandwidth wastage-IBW) or unallocated slot outside the burst (external bandwidth wastage-EBW).

An effective and adaptive bandwidth request scheme is proposed [7] to utilize the remaining bandwidth efficiently for nrtPS. The adaptive bandwidth request scheme selects the contention free scheme when the remaining bandwidth is enough to transmit at least one bandwidth request message. When contention-free-based option is selected, BS chooses a SS to transmit their bandwidth request based on the queue status. Queue status is the information of a packet in the queue of a SS. Using this approach, the collision probability can be reduced due to the decreasing number of contending SS. The proposal enhances the bandwidth efficiency and the data transmission delay of a SS is reduced as compared to the contention-based scheme. A bandwidth recycling scheme has been proposed to allow a SS to utilize the unused bandwidth [8]. As the incoming data for variable bit rate applications is hard to be precisely predicted, the SS may have more amounts of bandwidth than it needed. The proposed scheme tries to utilize the unused bandwidth when it is available and thus more services can be served. Nevertheless, not all bandwidth can be utilized at all times. The author [9] combined Earliest Deadline First (EDF) with Weighted Fair Queuing (WFQ) algorithms and applied this hybrid algorithm in WiMAX's uplink scheduling techniques to alleviate unfairness among the QoS service classes. The proposed scheme claimed the lower priority traffics are not starved while retaining the delay constraint of high priority traffics. While in [10], popularity of the video determined the communication channel uses broadcasting, multicasting or unicasting.

3. Overview of the bandwidth request mechanism in IEEE 802.16

WiMAX system provides a wide-range of QoS control to guarantee the fulfillment of QoS requirements for different service flows. QoS provides information such as maximum latency, maximum sustained rate, minimum reserved traffic rate, jitter and traffic priority. The priority is arranged as UGS > ertPS > rtPS > nrtPS > BE.

The IEEE 802.16 work group defines the standard of WiMAX in [1 - 3]. Generally, bandwidth request is sent to BS by a SS to indicate the needs of an uplink bandwidth allocation for the next transmission frame. The whole process of bandwidth request and allocation can be initialized by SS or BS. Normally, SS will send a request to BS to indicate the needs of an UL bandwidth allocation with a CID. The allocation of bandwidth is controlled by BS and BS schedules the allocation for transmission of the media access control packet data unit (MAC PDU) by checking the QoS requirements. The BS reserves the allocation to a SS if it can satisfy the QoS requirements. Once bandwidth allocation has been reserved, BS will send the uplink map (UL MAP) message to inform the details of the uplink allocation to the SS through broadcast.

Request can be sent in a standalone bandwidth request MAC PDU or a piggybacking bandwidth request on generic MAC PDU [1 - 3]. There are two types of bandwidth requests. They are incremental and aggregate. With an incremental bandwidth request received, BS provides the amount of bandwidth requested instead of the current perception of bandwidth need. In contrast, for an aggregate bandwidth request, BS provides the current perception of bandwidth needed instead the amount requested. Besides that, a SS can make a bandwidth request by using polling. Polling is a mechanism that the BS provides a dedicated UL allocation for a SS to make bandwidth request. Typically, there are two types of polling. The first is unicast, which refers to a SS that is being polled individually and for multicast or broadcast polling, it means multiple or all SSs are polled. If the bandwidth is insufficient to poll each SS independently (unicast), the multicast or broadcast polling in contention-based will be used. However, a SS is not allowed to be inactive during unicast poll. Therefore it may transmit a zero bandwidth request if the SS does not need any bandwidth.

In multicast or broadcast polling, a group or all SSs are assigned with a reserved CID. The polled group sends a bandwidth request during multicast or broadcast polling opportunity. It uses a contention resolution algorithm to minimize the occurrence of collision. A backoff algorithm is usually been applied. In the backoff algorithm, a SS selects a random number with equal probability between 0 and the backoff window. If a collision occurs, the backoff algorithm will reset the process and request it again. Therefore, collision during a contention period causes the bandwidth request delay could not be guaranteed and the packets could not be transmitted immediately. If a SS does not receive any bandwidth allocation within a specific given time, it is assumed that the transmission is unsuccessful. In this case, the SS increases the backoff window by a factor of 2. The PDU is discarded if the maximum numbers of retires for the bandwidth is reached. However, only SS that needs bandwidth shall respond in order to reduce the likelihood collision.

4. Real-Time Bandwidth Request Managers (RBRM)

Even though rtPS is a higher priority scheduling service, the allocation for rtPS cannot be guaranteed because contention-based bandwidth request is used. In addition, SS may not be able to allocate the bandwidth request due to its losses in bandwidth request queuing process. For example, the amount of rtPS bandwidth requests that should be sent in burst 2 and burst 4 of N frame are lost in Fig. 1. Hence, the "loss of bandwidth request slot" was unintentionally being padded. Consequently, BS assumed the amount of rtPS bandwidth request is 0, it causes the burst 2 and burst 4 in UL subframe for N+1 frame to have the insufficient bandwidth to transmit the rtPS data as shown in Fig. 1. The real-time traffic performance is affected in this case.

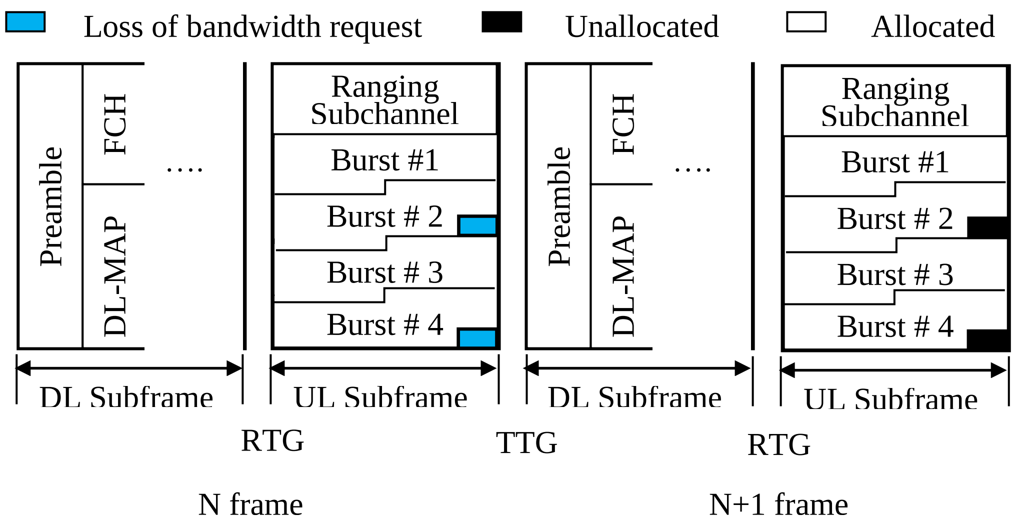

Fig. 1 "Loss of bandwidth request" incident

Therefore, we propose a scheme called real-time Bandwidth Request Manager (RBRM) to overcome this issue and to ensure that the amount of rtPS can be received by the BS. In such scenario, when the amount of rtPS bandwidth requests that should be sent is lost, our proposed RBRM sends the bandwidth amount in an aggregate way by using the opportunities from other service classes (nrtPS and BE). Even though the BS assumed the amount of rtPS bandwidth request was 0, the BS still allocates the amount of loss of rtPS bandwidth request by using the opportunity of nrtPS or BE. For example, burst 2 and burst

4 have sufficient bandwidth to transmit the data in UL subframe for N+1 frame as depicted in Fig. 2. Through RBRM, the priority structure will be maintained and insufficient bandwidth allocation that is caused by the loss of bandwidth requests will be minimized. In this way, rtPS performance can be improved.

RBRM enhanced the QoS by giving compensation to loss of bandwidth requests for rtPS traffic flow. The key design of our RBRM scheme is to request the amount in aggregated way by accumulating the lower priority bandwidth with the rtPS bandwidth. As long as a bandwidth request is successfully received by the BS, the amount of rtPS bandwidth requests is guaranteed. Therefore, it reduced the probability of insufficient bandwidth and helped to increase the network throughput for rtPS. However, when more than one bandwidth requests are delivered, redundancy of the bandwidth occurred. In [6], if the redundancy bandwidth was not utilized, it is known as internal bandwidth wastage (IBW). IBW issue will cause the degradation of the network performance indeed.

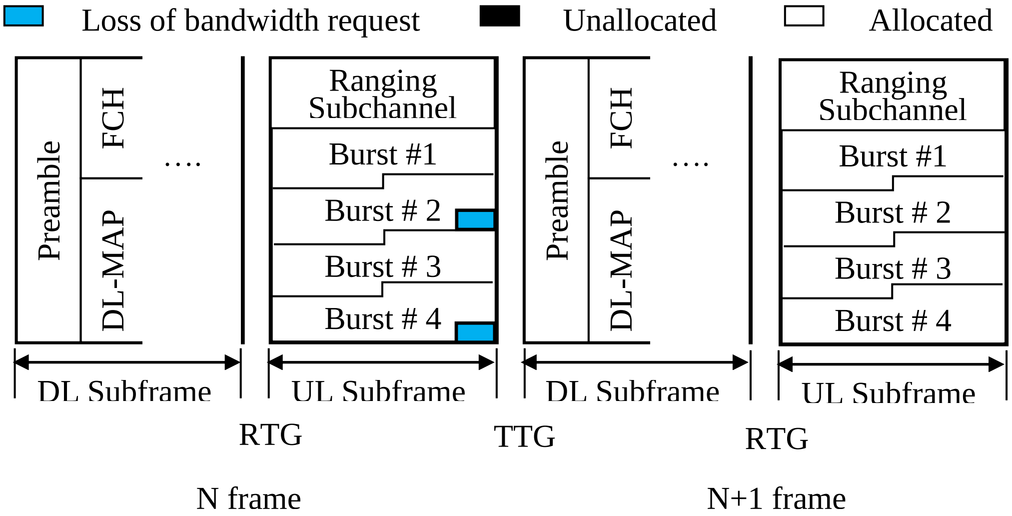

Fig. 2 Recovery of bandwidth requests with RBRM approach

The disadvantageous situation of IBW was inverted to become a beneficial redundant bandwidth by using a related method in [7] in our study. If redundancy of bandwidth occurred in the proposed scheme, the redundancy will be utilized by the current packets and the packets are able to be transmitted immediately. It does not only prevent the redundancy of bandwidth from being wasted but also helps to improve the throughputs, delays and jitter.

Another problem occurs when the current aggregate bandwidth is accumulated with the previous aggregate bandwidth throughout the time. The amount of redundant bandwidth would increase exponentially. In order to overcome the problem, our proposed scheme reinitializes the aggregate bandwidth to zero at the beginning of each uplink frame. The proposed scheme is operated as follows:

1. **When** each SS is activated:
   Initialize *Totalbandwidth* = 0
2. **When** uplink frame begin:
   Initialize *Totalbandwidth* = 0

3. **Sort** bwReq according to precedence // rtPS->nrtPS->BE
4. **While** *bwReq* need to be sent
   Set *Totalbandwidth = Totalbandwidth + bwReq(amount)*
   Create *bwReqHeader*
   Set *bwReqHeaderType* = 1
   Set *Cid*
   Set the *bwReq(amount) = Totalbandwidth*
   Enqueue the *bwReqHeader onto QUEUE*
5. **While** *bwReqHeader* in *QUEUE* is not Empty
   Get *bwReqHeader* **Pass** to *PHY* for transmission
6. **If** (All packets Checked**)** or (No current packets) //When redundancy of bandwidth occurs
      Return
   Else
      Allocate the packet to the *bandwidth redundancy*

whereby *TotalBandwidth* is the amount of the aggregate bandwidth, *bwReq* is bandwidth request. *bwReq(amount)* is the amount of a specific service bandwidth request, *bwReqHeader* is the bandwidth request MAC PDU, *QUEUE* is the queue for bandwidth request and *PHY* is the physical layer.

5. Simulation Model

5.1 Uplink Request Manager (URM)

In order to distinguish the advantages of the new RBRM design, we have simulated another bandwidth request mechanism that is referred as Uplink Request Manager (URM). URM scheme merely implements the standard specifications as in [1 - 3]. The key properties of URM include the following:

1. When there is sufficient bandwidth to poll each SS individually, rtPS uses unicast polling to send bandwidth requests to BS.
2. When there is insufficient bandwidth to poll each SS individually, contention-based multicast or broadcast polling is used.
3. Bandwidth Requests for each service class are independent and not consolidate.

5.2 Simulation Environment

The experimental environment that used in this research is the point-to-multipoint mode and it is referenced to [8] and [11]. The setup is 1 BS and the numbers of SSs are ranged from 10 to 70 with an incremental of 10 SSs in each scenario. The BS is directly surrounded by the SSs in a circular mode with a distance of 100m. Physical layer and MAC layer parameters are configured according to Table 1 for both BS and SSs. However, the BS has extra MAC layer configurations, which are described in Table 2.

Parameter	**Value**
Frame duration	20ms
Uplink frame duration	10ms
Downlink frame duration	10ms
Modulation scheme	64 QAM 3/4
Transmission power (dBm)	25
Antenna height	15m
FFT size	2048

Contention-based bandwidth request type	Normal / CDMA
Wait DCD timeout interval	25s
Wait UCD timeout interval	25s

Table 1 Configurations and parameters

Simulation Environment for Comparison 1

The objective of the experiment is to evaluate the performance of the normal contention-based and CDMA contention-based bandwidth requests that deployed in WiMAX. The experiment is conducted by using rtPS traffic. rtPS is categorized as real-time traffic in standard [1 - 3] with the performance metrics of throughput and latency. A comparison between the normal approach and CDMA approach is intended to be observed in this scenario. Each SS is associated with 1 uplink connection. The connection parameters are shown in Table 3.

DCD	5s
UCD	5s
TTG	10 US
RTG	10US
SSTG	4US
Bandwidth request minimal backoff value	2
Bandwidth request maximum backoff value	15
Ranging minimal backoff value	3
Ranging maximum backoff value	15

Table 2 Extra MAC layer configurations for BS

Application	**Real-time Multimedia Application**
Traffic type	VBR
Scheduling class	rtPS
Start time	300
End time	3600
Mean bit rate	512kbps
Distribution	Exponential

Table 3 Connection parameters

Simulation Environment for Comparison 2

Simulation environment for *Comparison 2* is to evaluate and analyze our proposed scheme, RBRM. Simulation parameters are defined same as *Comparison 1*, except for only CDMA is used to contend bandwidth. The main reason of choosing CDMA is its better results in *Comparison 1* and similar reviews had also been done in [12]. In the simulation environment for *Comparison 2*, there are 3 types of traffic being generated; they are rtPS, nrtPS and BE traffics. Each SS is associated with 1 rtPS, 1 nrtPS and 1 BE connection. These connections are used to simulate users' activities in [8]. The parameters of the traffic are shown in Table 4.

Simulation Environment for Comparison 3

Simulation environment for *Comparison 3* is to evaluate and analyze RBRM and URM in variations of modulation scheme and coding. There are no changes in simulation parameters. In the simulation environment for *Comparison 3*, 20% of the total SSs are placed 50% further of their positions away from BS as compared to *Comparison 2*. These placements cause the SSs having different modulation scheme and coding. In other words, BS and 80% of the SSs are located in 64 QAM while 20% of the SSs are resided in 16 QAM modulated network.

Application	Real-time Video	FTP	HTTP
Traffic Type	VBR	VBR	VBR
Scheduling class	rtPS	nrtPS	BE
Start time	15s	15s	15s
End time	75s	75s	75s
Mean bit rate	2Mbps	51Mpbs	2kbps
Distribution	Exponential	Exponential	Exponential

Table 4 Traffic parameter for comparison 2

6. Performance evaluations

Comparison 1

Fig. 3 shows the total throughput in bit per second for the normal contention-based and CDMA contention-based. It is observed that total throughput for normal contention-based is higher as compared to the CDMA contention-based after the number of SSs increased to 30. There is only 0.005% when the SS number is less than 30. Fig. 4 shows that the average of total end-to-end delay for normal contention based increases dramatically as compared to the CDMA contention-based after 40 SSs. In Fig. 5, the total end-to-end jitter in seconds for normal contention-based is higher as compared to the CDMA contention-based after 50 SSs.

In conclusion, the CDMA has a lower total throughput when the number of SSs increases, but for the total delay and jitter, CDMA contention-based is significantly better. Since the rtPS traffic is a delay sensitive traffic, CDMA contention-based approach is useful for the comparison of original scheme and the proposed scheme.

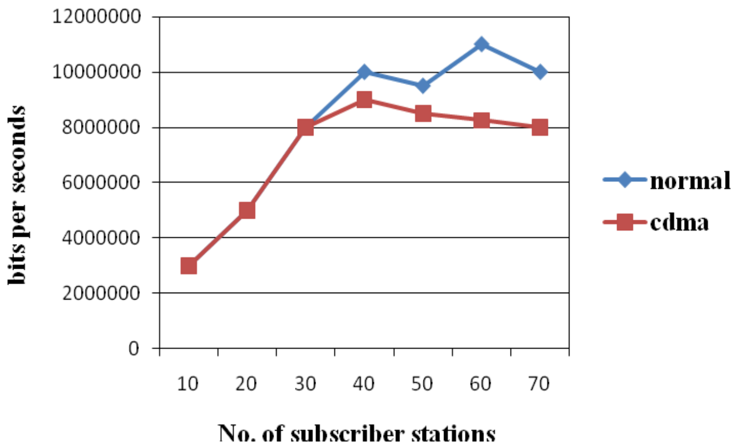

Fig. 3 Total throughput for Comparison 1

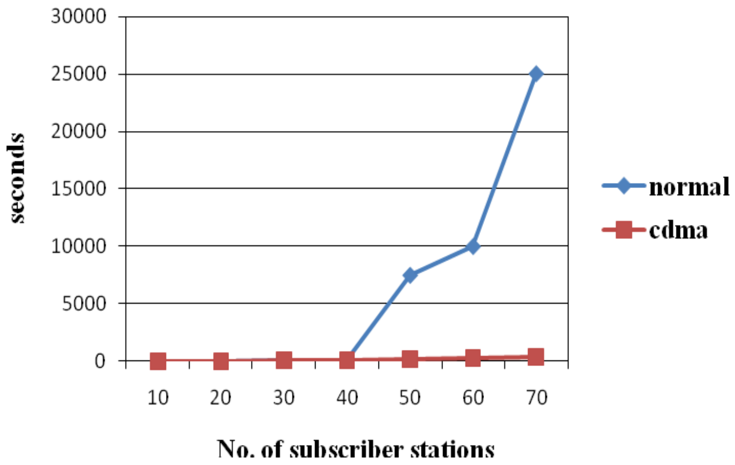

Fig. 4 Total end-to-end delay for Comparison 1

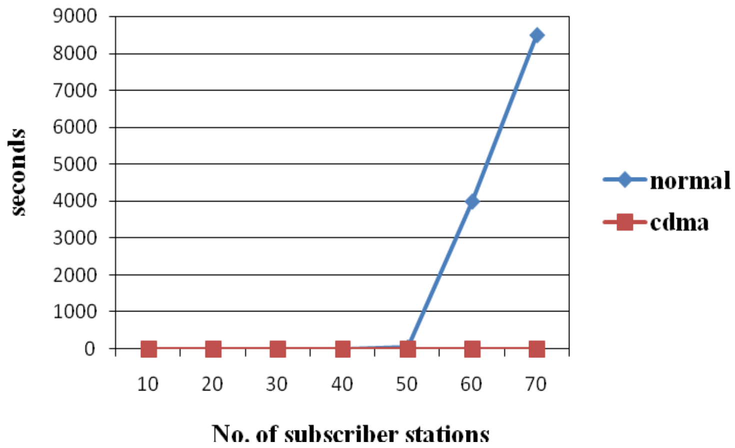

Fig. 5 Total end-to-end jitter for Comparison 1

Comparison 2

In Scenario *Comparison 2,* only rtPS and nrtPS traffic are being evaluated. Total throughput for rtPS and nrtPS are presented in Fig. 6 and Fig. 9 respectively. It is observed that average of total throughput for rtPS in RBRM scheme is higher as compared to URM scheme. The average rtPS improvement of RBRM scheme in throughput is 11.3% while the degradation of throughput for nrtPS is 2.6%. Fig. 7 and Fig. 8 show that the average of total end-to-end delay and total end-to-end jitter for rtPS RBRM scheme is lower compared to the URM scheme. The improvement of end-to-end delay and end-to-end jitter for rtPS in RBRM scheme is 4.8% and 7.4% respectively.

Fig. 9 shows the total throughput for nrtPS. The average of total throughput for nrtPS in URM scheme is higher as compared to the RBRM scheme. Fig. 10 and Fig. 11 show that the average of total end-to-end delay and total end- to-end jitter for nrtPS in URM scheme is lower compared to the RBRM scheme. The total end-to-end delay and total end-to-end jitter for nrtPS in RBRM scheme are 0.06% higher compared to the URM scheme.

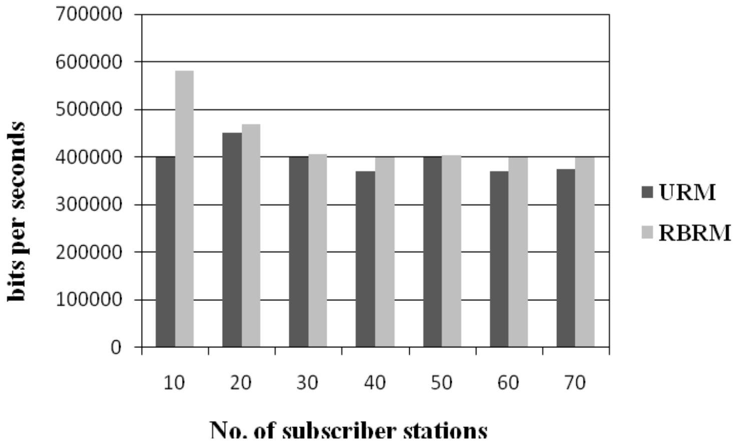

Fig. 6 Total throughput for rtPS

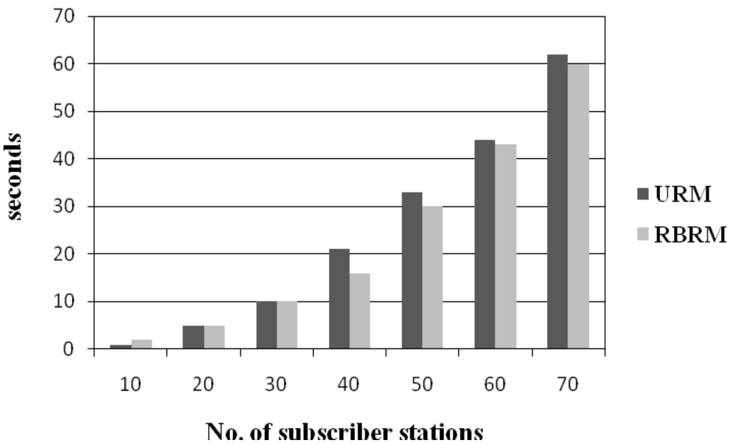

Fig. 7 Total end-to-end delay for rtPS

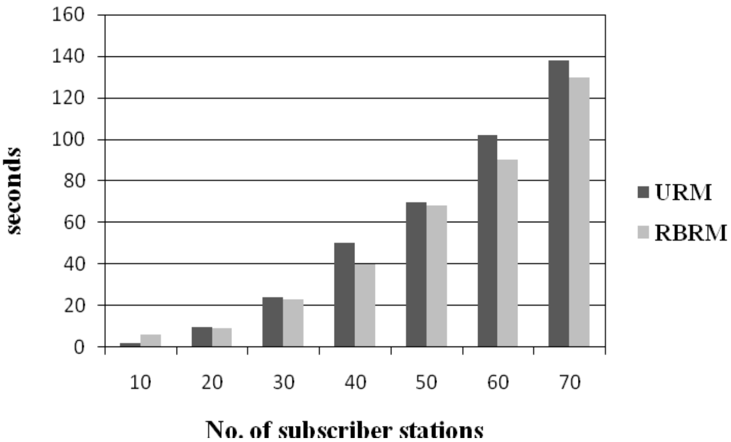

Fig. 8 Total end-to-end jitter for rtPS

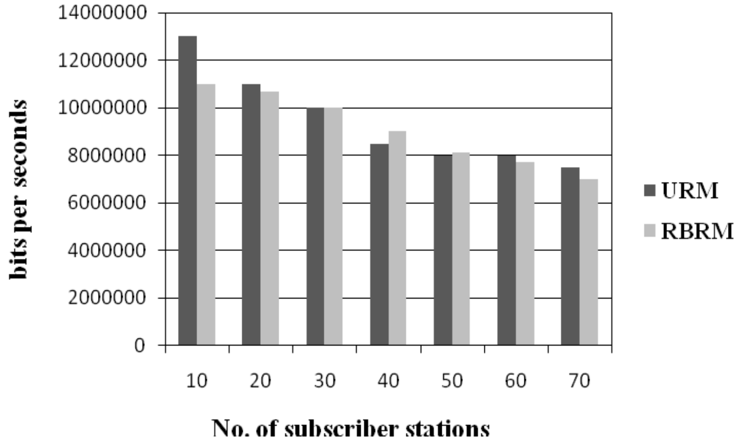

Fig. 9 Total throughput for nrtPS

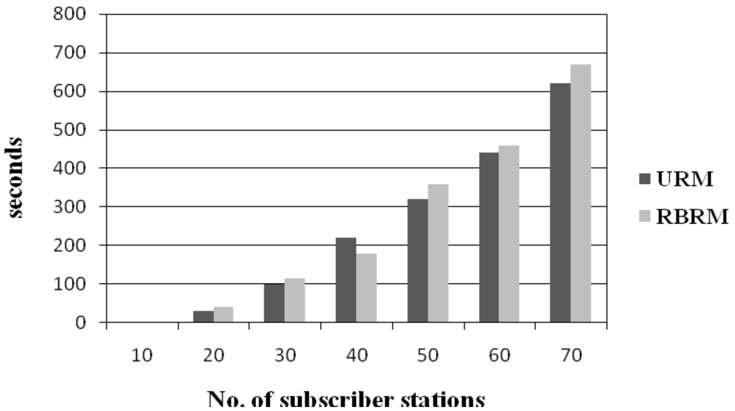

Fig. 10 Total end-to-end delay for nrtPS

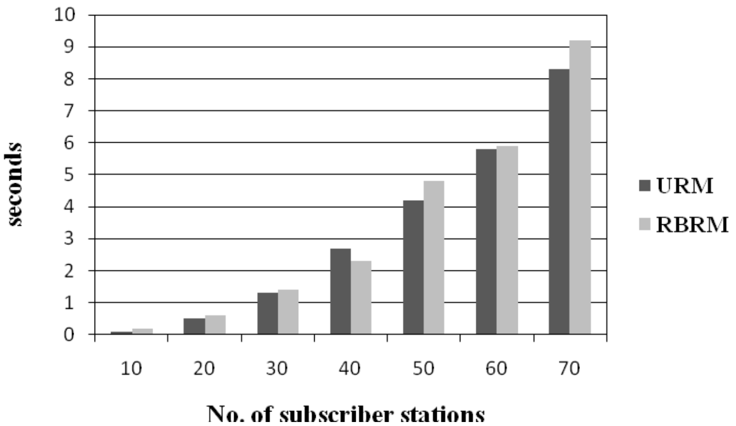

Fig. 11 Total end-to-end jitter for nrtPS

From the analysis, the rtPS performance improved substantially compared to the degradation of nrtPS. Although the throughput of nrtPS dropped, but it is acceptable since the QoS priority structure is UGS > ertPS > rtPS > nrtPS > BE. Furthermore, the improvement in rtPS is more than 4 times of the degradation in nrtPS. On the average, rtPS throughput performance improved by 11.3% while delay and jitter improved by 4.8% and 7.4% respectively. Contrary, a 2.6% degradation of throughput for nrtPS was detected. The summary of the analysis is presented in Table 5.

Real-time polling services traffic	
Attribute	Improvement of performance
Total throughput	10.6%
Total delay	2.2%
Total jitter	5.5%
Non real-time polling service traffic	
Attribute	Degradation of performance
Total throughput	1.6%

Table 5 Summary of the result analysis for comparison 3

Comparison 3

In *Comparison 3*, the average of total throughput for rtPS in RBRM scheme has improved by 13% as compared to URM scheme while the degradation of throughput for nrtPS is 8.4%. In Fig. 12, RBRM always has a higher throughput than URM. However, it is shown that the difference dropped from 48% to only 6% at 10 and 70 SSs respectively. On the other hand, nrtPS looses about 36% at the beginning and regains back approximate 1% at the 70 SSs scenario in RBRM approach.

At the same time, Fig. 14 and Fig. 15 show that the average of the total end-to-end delay and total end-to-end jitter for rtPS in RBRM scheme is minimized by 6.7% and 11.8%, respectively as compared to the URM scheme. These improvements are critical to real-time applications because the latency is one of the QoS parameters required by [1 - 3]. At 70 SSs scenario, RBRM scheme still manages to have 3% lower delay and 9% lower in jitter. Meanwhile, Fig. 16 and Fig. 17 represent the total end-to-end delay and total end-to-end jitter for nrtPS in RBRM scheme. Since nrtPS is a delay tolerable traffic, the degradations are acceptable. The delay and jitter variances between RBRM and URM for nrtPS are too small, only 0.01%.

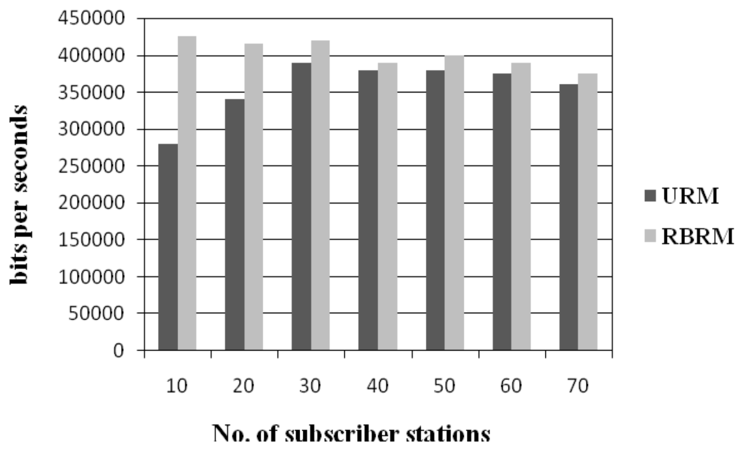

Fig. 12 Total throughput for rtPS in different modulation scheme and coding

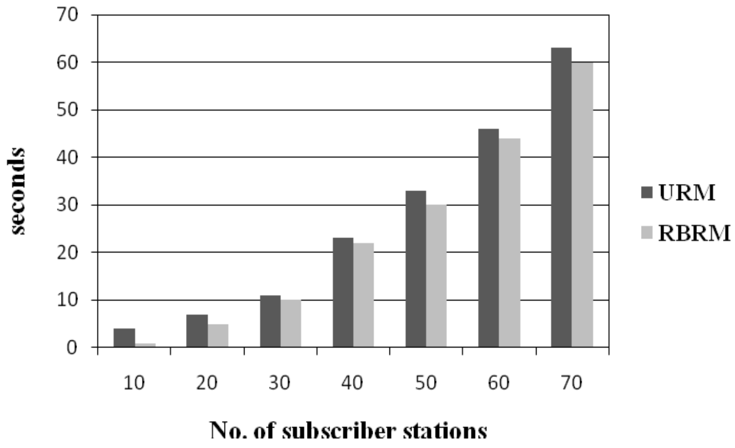

Fig. 13 Total end-to-end delay for rtPS in different modulation scheme and coding

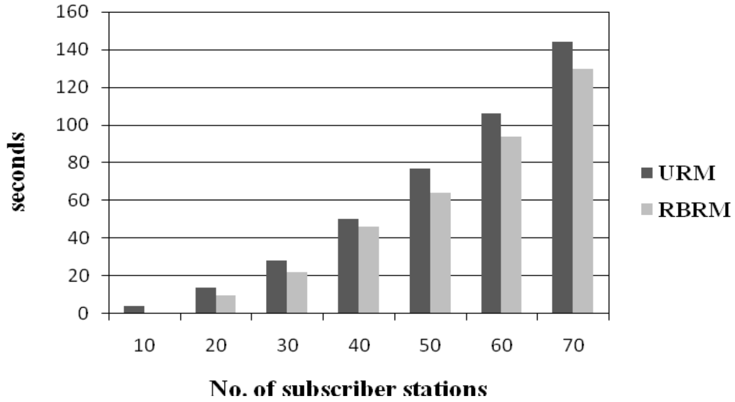

Fig. 14 Total end-to-end jitter for rtPS in different modulation scheme and coding

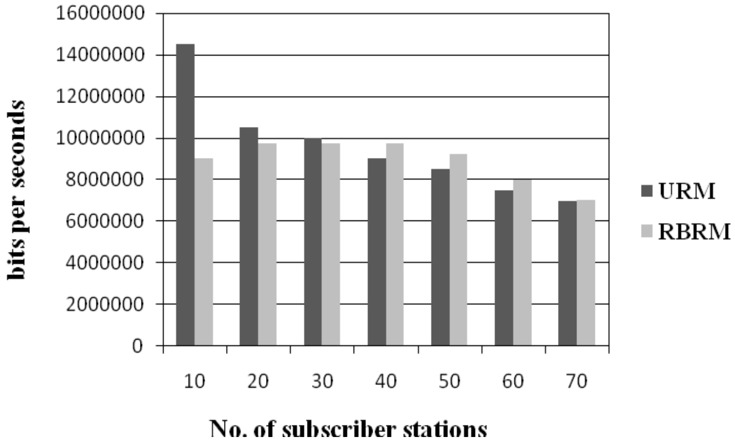

Fig. 15 Total throughput for nrtPS in different modulation scheme and coding

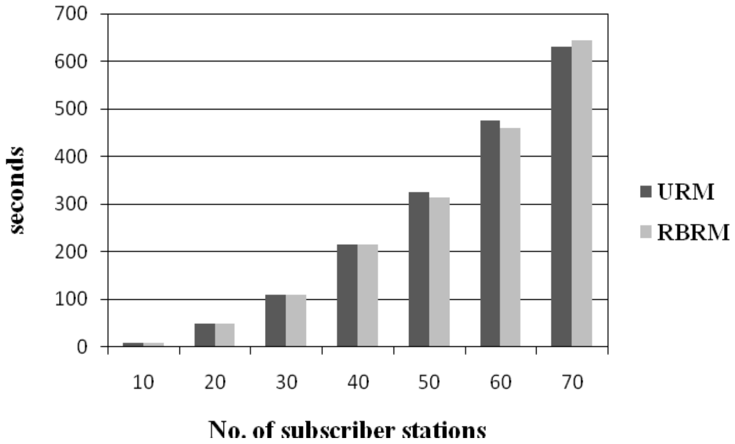

Fig. 16 Total end-to-end delay for nrtPS in different modulation scheme and coding

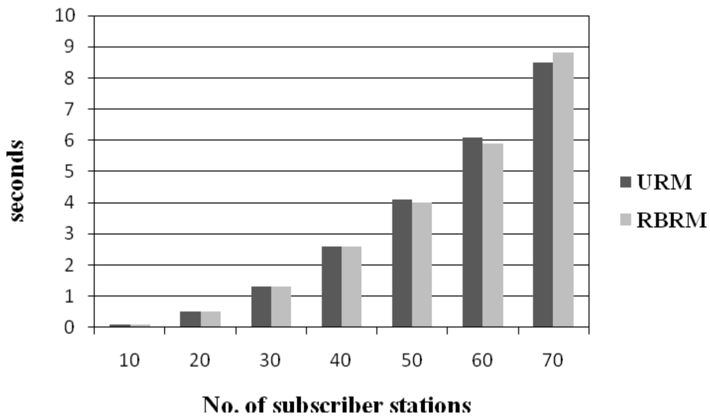

Fig. 17 Total end-to-end jitter for nrtPS in different modulation scheme and coding

From the results, the rtPS performance improved more compared to the degradation of nrtPS. The rtPS throughput performance improved by 13% while delay and jitter improved by 6.7% and 11.8% respectively. It is observed that the RBRM scheme improves the rtPS performance even with variation of modulation scheme and coding applied. Hence, we made a conclusion that our proposed scheme could be applied to any other QoS centralized wireless networks, i.e. Long Term Evolution (LTE). The summaries of the result are shown in Table 6.

Real-time polling services traffic	
Attribute	Improvement of performance
Total throughput	13%
Total delay	6.7%
Total jitter	11.8%
Non real-time polling service traffic	
Attribute	Degradation of performance
Total throughput	8.4%

Table 6 Summary of the result analysis for comparison 3

7. Conclusion

In conclusion, RBRM scheme improved the performance of rtPS with the trade-off of nrtPS performance. The results should be considered acceptable since the QoS priority structure is UGS > ertPS > rtPS > nrtPS > BE and the improvement in rtPS is 6 times more the degradation in nrtPS. RBRM scheme requested the bandwidth amount in an aggregate way in order to compromise the loss of real-time service bandwidth request has reduced the probability of insufficient bandwidth for the real-time service flow. RBRM also utilizes the redundancy of bandwidth from not being wasted. However, there may still be internal bandwidth wastage in the redundant bandwidth caused by under utilizations.

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