A QoS Real Time Bandwidth Redistribution Transmission Algorithm in WiMAX

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ABSTRACT

WiMAX connectivity uses two components, a Base Station (BS), where the WiMAX signals are broadcast and a Subscriber Station (SS) which is a device, or a group of devices that receives the signals. The SS will request bandwidth in the uplink (UL) from the BS, and the BS will allocate the bandwidth accordingly. As WiMAX can achieve a range of 30 miles with a throughput of 72 Mbps with LOS and 4 miles with NLOS, promoting mobility but maintain effective QoS is difficult. QoS is challenging to achieve due to unpredictable channel conditions such as signal fading and frequency interference. Different types of traffic will require different services from a network, including differing priority status; bandwidth levels and latency tolerances For example, electronic mail is insensitive to delay, but loss of data is its priority, compared to video, which is delay sensitive but data loss insensitive within a certain tolerance. Applications such as interactive graphics are sensitive to both delay and data loss. This paper proposes a QoS bandwidth allocation algorithm that would sample what has been transmitted so far on a periodic basis and compare to the total amount of bandwidth that SS has allocated to it. If there is a significant difference, the bandwidth allocation can be cut down allowing that bandwidth to be relocated to another SS. This would ensure that the bandwidth is directed to the transmissions that actually require it rather on the assumption of bandwidth requirement based on the classification of data being transmitted.

General Terms

Algorithms, Design, Performance

Keywords

Quailty of Service, WiMAX, Bandwidth, Scheduling

1.INTRODUCTION

The IEEE 802.16 standard encompasses both the PHY and MAC layer architecture. Worldwide Interoperability for Microwave Access (WiMAX) can achieve a range of 30 miles with a throughput of 72 Mbps with Line Of Sight (LOS) and 4 miles with Non Line Of Sight (NLOS), (Ohrtman, 2005), promoting mobility but effective Quality of Services (QoS) is challenging to achieve due to unpredictable channel conditions such as signal fading and frequency interference (IEEE 802.16 Broadband Wireless Access Working Group,2002). This work will focus on the MAC layer which provides the interface between the PHY layer and the higher level transmission protocols. Below, figure 1, illustrates the three sub-layers defined in the standard the Service-specific convergence sub-layer (CS), the MAC Common Part Sub-layer (CPS) and the Security Sub-layer. The CS is where other transmission protocols converge into the WiMAX protocol. The Security Sub-layer is where the privacy protocols are handled. The MAC Common Part Sub-layer is where the QoS mechanisms reside.

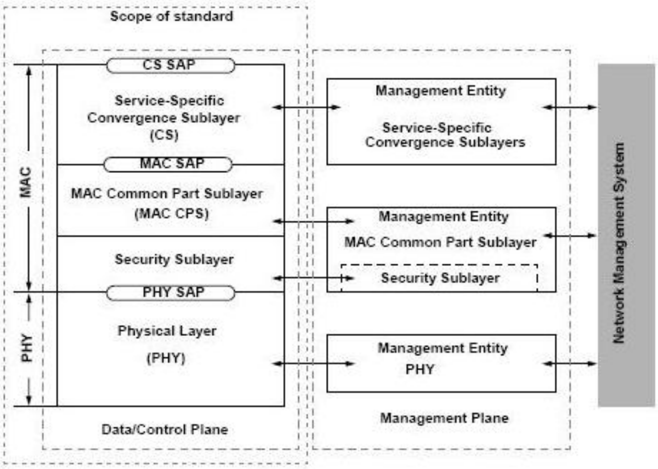

Figure 1. IEEE 802.16 layers and Service Access Points (IEEE, 2004).

WiMAX connectivity uses two components, a Base Station (BS), where the WiMAX signals are broadcast (on the down link) and a Subscriber Station (SS) which is a device, or a group of devices that receives the signals (see figure 2). The SS will request bandwidth in the uplink (UL) from the BS, and then the BS will allocate the bandwidth according to the scheduling policy.

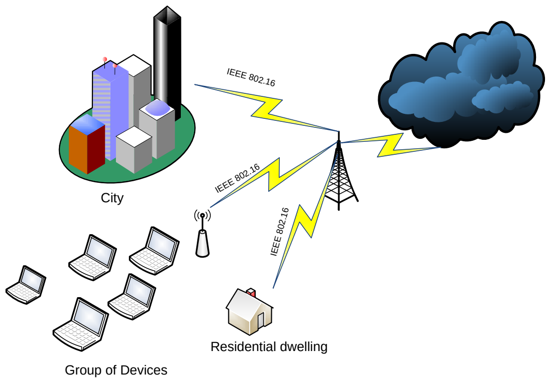

Figure 2 IEEE 802.16 WiMAX Connectivity

The WiMAX 802.16e defined MAC Layer provides differentiation of transmission flows according to the requirements of the user application. The key factors that influence the differentiation of service is delay and bandwidth requirements. Controlling these factors will help to manage QoS that in turn improves the users Quality of Experience.

Different types of application traffic require a variety of qualities from a network. Some types will need a higher priority over others, to support these requirements QoS parameters are utilised. For example, electronic mail is tolerant to delay, but eliminating loss of data is its priority, compared to video, which is delay sensitive but data loss tolerant. Applications such as interactive graphics are sensitive to both delay and data loss (Stallings, 2007). QoS protocols treat these applications differently, providing the best service possible depending on the resources available.

This paper will investigate QoS algorithms that are built into the WiMAX standard. Section 3 will review the Variable Bit Rate Video Service and backhaul connectivity schemes that evidence the efficiency of communication in bandwidth requests which can improve latency and redistribution of resources such as bandwidth. While section 4 will detail the proposed UGS enhancement algorithm which aims to improve the amount of bandwidth that is allocated to via the QOSParamSet Provision but not activated and therefore used by the SS. This will increase the efficiency of distributing a finite resource such as bandwidth.

2. WiMAX Quality of Service

Within a wireless network, effective QoS is challenging to achieve due to "unpredictable channel conditions" (Mellouk, 2009), such as signal fading and frequency interference. WiMAX utilizes a set of parameters to provision the required resources if they are available, then to admit them before activating when required. The provisioned resources are a subset of the admitted resource and the activated resources are a subset of the activated resources (Ohrtman, 2005). Therefore there is a certain amount of resources that are reserved for a SS but not being used.

When using WiMAX, many users will be allocated bandwidth based on the bandwidth requests of the SS. Service Providers, can provide more bandwidth, but due to the number of simultaneous users, it will always be constrained and it will also incur a cost (Ghazal & Ben-Othman, 2009). Delay in transmitting the data, sometimes causing data loss, can be generated by a number of factors, these are; interference between wireless channels; the number of users that share the bandwidth and propagation time. This causes problems in delay sensitive applications such as Voice-over IP (VoIP), (Ghazal & Ben-Othman, 2009). Interference caused by adjacent channels can result in errors within data packets, and even data loss. Fading can be caused by something as small as a minor change in the path between the transmitter and the receiver, and path loss can occur due to obstacles, such as buildings and trees that the signal may encounter (Ghazal & Ben-Othman, 2009). QoS in WiMAX the details of traffic scheduling for either uplink or downlink transmission have not been explicitly defined. The QoS in WiMAX will classify the data traffic into one of five classes based on the the type of traffic being transmitted, these are Unsolicited Grant Service (UGS), Real-Time Polling Service (rtPS), Extended Real – Time

Polling Service (ertPS), non-Real Time Polling Service (nrtPS), Best Effort (BE).

2.1 Unsolicited Grant Service (UGS)

This service is predominantly used for video or multimedia as it provides a maximum traffic rate, efficient latency, resulting in tolerated jitter. The efficiencies are aided by only having a transmit requests rather than commencing with a request for bandwidth the bandwidth allocated will be sufficient to transmit video files. The disadvantage of this is that a standard amount of bandwidth is allocated which if the user does not require it a large percentage of the bandwidth could be unused. This is used for real time services such as VoIP without silence suppression with fixed sized packets (Ghazal & Ben-Othman, 2009, Dawood, 2007). Within the standard this class can request additional bandwidth but not decrease the amount of bandwidth it requires if it is unsed.

2.2 Real-Time Polling Service (rtPS)

This algorithm was developed for streaming media such as MPEG-4, which produces a variable bit rate output for transmission to cater for complex sections of the video. The algorithm incorporates unicast requests that provide the opportunity for the SS to give more information about the size of grant it requires in the form of bandwidth requirements. This request can be per connection or an aggregated request by the SS. This provides greater efficiency in the use of the bandwidth but with the disadvantage of a greater overhead with the quantity of requests required to achieve this (Ghazal & Ben-Othman, 2009, Dawood, 2007)..

2.3 Extended Real – Time Polling Service (ertPS)

The newest out of the algorithms in WiMAX, which was added when the standard was amended in 2005 to become 802.16e, it is ideal for applications such as VoIP that has a variable data rate during transmission. Bandwidth is allocated according to the user’s status. It reduces the bandwidth wastages found in UGS and reduces the MAC overhead and access delay found in rtPS (Ghazal & Ben-Othman, 2009, Dawood, 2007).

2.4 non-real time Polling Service (nrtPS)

This algorithm is ideal for traffic that has variable data rates but is not a real-time application as in the previous algorithms. This would cater for protocols such as FTP which required guaranteed bandwidth but latency is not a priority. In doing so it offers a unicast poll request that can get through congested traffic (Ghazal & Ben-Othman, 2009, Dawood, 2007).

2.5 Best Effort (BE)

As QoS is not a priority for the user the resources need not be provisioned. Therefore transmission will happen when there is enough resources for the connection. Email or HTTP would generally be in this category (Ghazal & Ben-Othman, 2009, Dawood, 2007). The downside of this class is that it cab suffer from stavation of any resources resulting denial of service and therefore no data being transmitted.

Once the traffic has been classified, the connection is sent into the relevant queue and so per-connection QoS is used (Cho, Song, Kim, & Han, 2005). Three of these classes provide QoS for real time traffic flows, Voice over IP (VoIP) for example, these are UGS, rtPS and ertPS. It is has been found that UGS has the best delay performance, but under this scheduling service the SS is allocated a fixed amount of bandwidth. If the traffic does not use the capacity of the bandwidth that has been allocated during transmission in an uplink (UL) frame, the bandwidth is left unutilized. In rtPS the BS will periodically poll the SS to ask how large the next subframe is, that's queued to be transmitted bandwidth is then allocated accordingly. Polling the SS will require bandwidth; therefore BS will need to assign itself bandwidth also, (IEEE, 2004). The ertPS algorithm builds on the efficiency of the UGS and rtPS classes. The SS can be assigned resources dynamically by the BS, thus resources are used efficiently, (Kim, & Hwang, 2009).

3. CURRENT IMPROVEMENT METHODS

Having a QoS system that is interoperable with all types of traffic efficient bandwidth utilization is a challenging undertaking. This section will look into methods that have been proposed to improve bandwidth use for different types of traffic.

In Chen, Deng, Hsu, & Wang, (2009), investigated and proposed a new method for scheduling video traffic, named the Variable Bit Rate Video Service (VBRVS). It uses two mechanisms, One Level VBRVS algorithm and Two Level VBRVS algorithm. The One Level VBRVS algorithm involves the BS assigning uplink bandwidth to the users using video based on their traffic state transition; when this state is changed, a bandwidth request is required. This provides for the variable bit traffic that may have peaks in the transmission stream which this accommodates. The disadvantage is the bandwidth request could create additional delay in the process. The second mechanism, the Two Level VBRVS allows the bandwidth request process to be reduced, and therefore reducing the bandwidth wastage. This algorithm uses two bits in the MAC header allowing the BS to know the demand of the SS. The BS can then decide if a bandwidth request opportunity should be assigned, or a data transmission opportunity. The bandwidth allocated can then be deterministic. Simulations evidenced that both algorithms in the proposed VBRVS mechanism will have "less bandwidth waste ratio than the conventional uplink scheduling algorithms." Chen, Deng, Hsu, & Wang, (2009).

Real-time traffic support in the WiMAX IEEE 802.16 based backhaul networks involve several base stations to be connected together using a microwave link (Dai, & Zhao, 2007), enabling connections to be established whilst moving. Dai & Zhao (2007) suggests a "simple enhancement to the bandwidth request mechanism in IEEE 802.16 for supporting packet voice traffic". Each scheduling scheme will require a different amount of information to be included in the bandwidth request. In Dai & Zhao's scheme the bandwidth requests are combined to create an aggregate bandwidth request, the aim is to make the bandwidth request process more efficient by cutting the amount of time and bandwidth required by both the SS and BS. These aggregate bandwidth requests will include information regarding the latency requirements of buffered real-time packets from the SS, this will help the BS to make informed resource allocation decisions. Through simulations it was evidenced that the system provided "satisfactory real-time performance for the voice traffic" (Dai & Zhao, 2007). Results showed that "there is an optimum amount of information to be transmitted in the bandwidth requests in order to achieve good voice packet transmission performance" (Dai & Zhao, 2007).

4. PROPOSED UGS BANDWIDTH REDISTRUBUTION

As stated in section 2.1 when the scheduling algorithm UGS, used for real time applications such as VoIP, is being used, the BS periodically allocates a fixed amount of bandwidth to the SS. This can lead to a waste of bandwidth when the packets the SS transmits are too small to fill an uplink sub frame. This section will propose an enhancement to the UGS algorithm that will endeavor to improve the utilization of bandwidth.

By building intelligence into the algorithm, the aim is to cut down the allocated bandwidth when it is found that it is not needed. The BS would sample what has been transmitted so far on a periodic basis and compare this to the total amount of bandwidth that SS has actually allocated to it. If there is a significant difference, the bandwidth allocation can be cut down allowing that bandwidth to be reallocated to another SS. If it is found that the SS begins to use more of the bandwidth, more can then be reallocated as this is built into the standard.

$$\sum^{n}_{i=1} (B_{ai} - B_{ui}) + B_t \qquad (1)$$

In (1) above n is the number of SS using the total available bandwidth, i is each instance of the connection, B_a represents bandwidth allocated to the SS, B_u is the bandwidth that is actually being used by the SS and B_t is the total bandwidth available. This will calculate the total bandwidth that is not currently being used and add it to the total bandwidth available ready to be redistributed to a SS that required more bandwidth. Bandwidth is described as the amount of data that can be transmitted within a given time.

$$\sum^{n}_{i=1} (B_{ai} - B_{ui}) \leq B_{bi}, \qquad (2)$$

If a SS transmitting real-time traffic that falls into the UGS class is sending packets through the uplink traffic that uses less than 80% of the allocated bandwidth (B_b)(2), under this algorithm that wasted bandwidth should be reassigned. Similarly more of the wasted bandwidth should be reassigned when utilization is less than 60%, 40% and 20%. Alternatively, if it is then found to be using over 90% of the allocated bandwidth, the station should be allocated more bandwidth.

4.1 Simulations

To simulate the proposed bandwidth redistribution enhancement a Visual Studio C# program was created. To model the data flow two timers were used, one to represent the packets being created by the user and another to represent the VoIP packets being successfully conveyed. The utilization of the bandwidth is calculated by measuring the data flow over a fixed interval, facilitated by a third timer. These timers call an attached subroutine at selected intervals. Each time this third timer expires, a calculation is carried out to evaluate the average use of the bandwidth by the user. If the result of this calculation shows that less than 80% of the bandwidth is being utilized, the allocated bandwidth is reduced, and the third timer is restarted. If the transmitted packets increase in size, and more of the allocated bandwidth is then utilized to over 90%, more bandwidth is reserved. If the utilization percentage grows less than 5% the interval is slowed down to lower the allocated bandwidth. At each stage the average calculation, and the amount of bandwidth allocated is written to a text file.

Based on a VoIP transmission using the G.711 codec, including overheads, packets of size 1744 bits will be generated at a rate of 50 per second, (*Voice Over IP - Per Call Bandwidth Consumption*, 2006). Based on this information initial bandwidth allocation is set to 200Kbps to ensure complete accommodation of the resource requirements. Transmissions are averaged at a time interval of 200ms. Also included in this simulation are transmissions with packets of size 1104 and 624 bits.

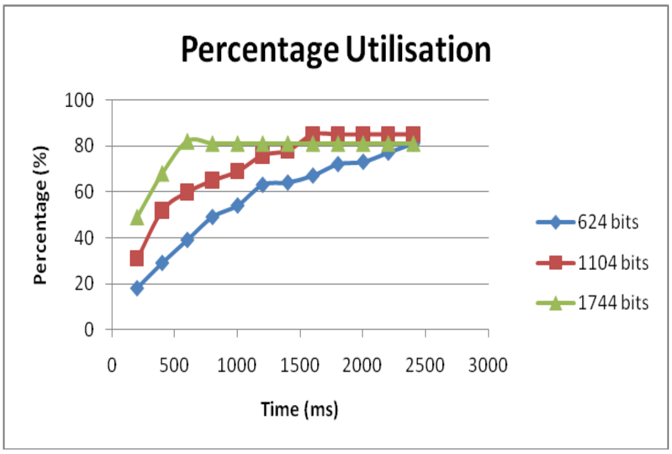

Figure 3: Bandwidth utilisation

Figure 3, illustrates that the bandwidth utilization increased over time, this coincides with the graph in Figure 4 which illustrates the allocated bandwidth decreasing over the same period of time. It can be seen that the transmissions including smaller sized packets have less bandwidth allocated to them throughout the sample time increasing the utilization they would otherwise be producing. For a constant transmission at this rate for ten minutes, this method would free up 45Mb of data capacity for reallocation when sending 1744 bit packets

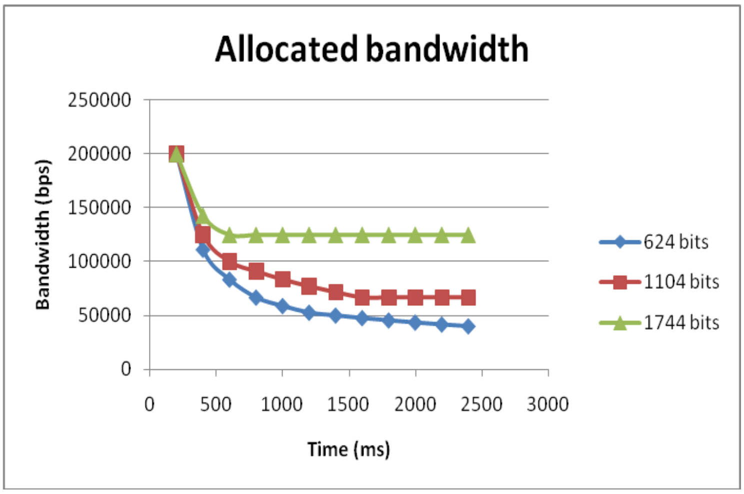

Figure 4, Allocated bandwidth

5. CONCLUSIONS

Quality of Service plays a large role in the performance of a network. WiMAX is an emerging technology that is currently being widely researched, but maintaining QoS poses a challenge. This paper has proposed a bandwidth redistribution algorithm building on the current QoS class scheme aimed at improving bandwidth utilization for real time traffic with packets of a fixed size such as VoIP. This enhancement worked on cutting down allocated bandwidth until sufficient balance of resources was achieved. Through means of a program written to simulate the proposal it was evidenced that specified amounts of reserved bandwidth were successfully removed until the bandwidth was being used to at least 80%. Overall a reasonable amount of bandwidth was redistributed for other requests. As in a WiMAX network, bandwidth is shared among many users; this improvement to the QoS will improve bandwidth efficiency sufficiently, allowing greater utilization of finite resources available.

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