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DATA HANDOVER USING ENERGY-EFFICIENT MULTICAST SCALABLE STREAMING IN WIMAX NETWORKS

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ABSTRACT:

The Multicast/Broadcast Service (MBS) feature of mobile WiMAX system is a promising knowledge aimed at providing wireless data handover, since it permits the distribution of handover content to large-scale user groups in a cost-efficient method. In this paper, we consider WiMAX networks that convey multiple data handover encoded in accessible manner to moveable receivers by means of the MBS feature. We express and solve the sub stream selection problem to exploit the handover quality, which rises when manifold scalable handover streams are broadcast to mobile receivers with incomplete resources. We proposed a original estimate algorithm for this problem. We proved that our algorithm consumes a small estimate factor of, then it has a time difficulty of, where is the total amount of layers, and is the extreme number of layers in a scalable stream. They applied and validated our algorithm in a imitation setup and deliberate the impact of a wide variety of parameters using manifold data handover. Our reproduction results show that the estimate factor of the proposed procedure is very adjacent to one for applied scenarios. We extend our algorithm to reduce the energy consumption of mobile receivers. This is done by transmitting the selected sub streams in bursts, which permits mobile receivers to go off their wireless interfaces to save energy. We show how our algorithm constructs burst broadcast timetables that reduce energy ingesting without sacrificing the video quality.

Key words: Handover, broadcast, multicast, multicasting, Wimax

1. INTRODUCTION

The request for mobile hypermedia streams has been cumulative in the past insufficient years as indicated by manifold market analysis educations. Multimedia watercourses can be brought to mobile strategies over different wireless networks, counting 3G, WiFi, and WiMAX systems. In this paper, we attention on multimedia flooding over WiMAX networks, which remain quantified by the IEEE 802.16 average. Although the now deployed WiMAX networks are frequently used to provide wireless Internet access to subscribers, the WiMAX average supports various net services. One of these services is the Multicast and Broadcast Service (MBS), which container be used to deliver hypermedia traffic to large-scale operator communities.

In this paper, we discourse two significant problems in handover streaming over WiMAX networks: Exploiting the video excellence and minimizing vigor consumption for mobile receivers. In specific, we consider distribution multiple scalable audiovisual streams to mobile headsets. A scalable handover stream is calm of multiple layers, anywhere each layer recovers the spatial, temporal, or the visual superiority of the rendered handover to the user. Because of their flexibility, scalable handover streams can professionally support varied receivers, adapt to network conditions, and apply the obtainable wireless bandwidth.

We exactly formulate the problematic of choosing the best set of deputize streams from the scalable data handover streams in order to make the most of the quality for moveable receivers. We show that this problematic is NP-Complete. Thus, optimally resolving it in real period may not be computationally possible. We suggest an estimate procedure that produces near-optimal solutions then runs in real period. We logically show that the estimate issue is close to one.

In addition, subsequently many subscribers of the WiMAX handover facilities are expected to be moveable users through energy-constrained plans such as keen phones, minimalizing the energy ingesting of these devices develops an important problematic in order to spread the viewing time. To discourse this problem, we spread our algorithm to diminish the energy ingesting of mobile headphones. The extended procedure first chooses the best sub brooks and then transmits these sub brooks in bursts.

The burst Broadcast of the handover data allows mobile headsets to go off their wireless interfaces for lengthier phases of time in teaching to but energy. Our procedure carefully concepts the burst broadcast schedules that reduce the energy ingesting without forgoing handover quality or presenting any buffer excess or underflow instances.

2. RELATED WORK:

Again at the confirmation phase, here is conversation of keys among the mobile station and improper station, the base position forwards the important to access facility network ASN to get verification. This makes the verification protocol as advanced weighted single and takes at least milliseconds which increases the inexpression in handover also touches the data transfer and message interference. All these subjects of extended verification protocol, induced us to design a new bright weight client based delivery initiation protocol anywhere the handover request will be originated by the movable station MS and will be authentic by the Base station and access service networks [1].

Routing protocol intended for wireless networks are of several kinds that provide to many different needs of researchers and inventors. Multicast routing protocols deliver data after a source to destinations prepared in a multicast group. In the more than a few protocols were proposed to offer multicast facilities for Multihop wireless networks. These protocols remained planned for WiMax networks, directing primarily on network connectivity and by means of the amount of hops (or hop count) as the way selection metric. However, it has been shown that using hop amount as routing metric container result in selecting associations with poor superiority on the path, harmfully impacting the path throughput [2]. Since each of these have a unique set of advantages and disadvantages, it becomes necessary for us to understand which of these might suit a specific scenario best [3].

Initial network entry is one of the most significant processes in Mobile WiMAX network because the initial network entry process is the first gate to establish a connection to Mobile WiMAX. Thus, many physical parameters, performance factors, and security contexts between SS and BS are determined during the process. However, specifically, the SBC negotiation parameters and PKM security contexts do not have any security measures to keep their confidentiality [4, 5].

The last notable delay component in an initial entry is the ranging delay. Much larger delay is observed at the indoor measurement than at the outdoor measurement, thus implying that the ranging delay strongly depends on the channel quality. The ranging delay performance should be affected by the vendor-specific algorithms such as initial power setting. Here, we explain the detailed ranging procedure. After a terminal transmits a code division multiple access (CDMA) ranging code, the terminal waits for a response [6]. If the terminal cannot receive a response, a timer expires, and the terminal selects a new CDMA code, and then sends it after a random back-off [7, 8]. Transmission power is generally increased for each attempt. The terminal should repeat this operation till the terminal receives message from the BS or transmission power of the terminal finally reaches the maximum value defined by the BS. As a result, the ranging delay is influenced by the channel condition because more attempts are generally required before a ranging success in bad channel environments [10].

3. PROPOSED SYSTEM:

In this paper a framework for multicasting scalable data handover over moveable WiMAX networks. We exactly examined the difficult of selecting the greatest sub streams of mountable data streams under bandwidth restraints. Solving this tricky is significant because it allows the network worker to convey higher excellence data handover or more number of handover streams at the same capacity. We presented that the deputize stream assortment problem in attendance of bandwidth limitation is NP-Complete. We proposed a novel estimate algorithm for this problem. We proved that our algorithm has a minor estimate factor and it consumes a time difficulty. The total number of deposits and the supreme number of layers in a scalable stream. We applied and validated our procedure in a imitation setup and deliberate the influence. The wide range of limits using multiple video traces. Our imitation results demonstration that the estimate factor of the proposed algorithm is very close to one for applied scenarios. We show that the planned algorithm:

1) Effectual in terms of execution time
2) Attains high radio reserve utilization
3) The exploits the received video quality and
4) The reduces the energy ingesting for mobile receivers.

3.1 BUFFER & ENERGY EFFICIENT SCHEDULING

That equivalent number of moveable stations is getting each stream. For assessing the energy saving, we usage the Average Energy Competence metric. This is straight linked to the detail that the number of substituting in a schedule partakes a high impact on vitality savings.

Which reduces the total quantity of bursts needed, the schedulable data dimensions of a scheduling opening be data rates of the preferred sub streams, the total records sent in to the streams.

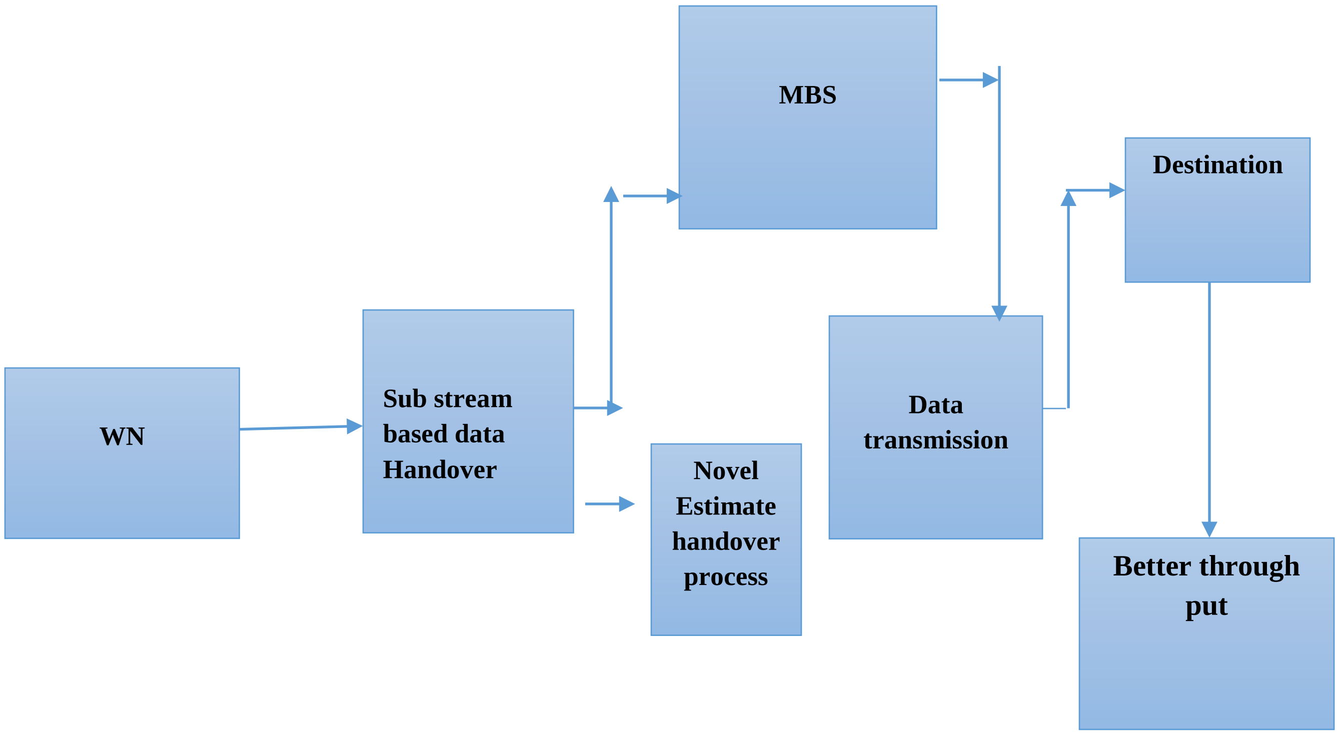

Fig 1. Basic data handover process Diagram

3.2 SUB STREAM SELECTION ALGORITHM:

Start:

Input: sub stream MBS Capacity, Frame duration Scheduling window size

Output: Data burst allocation in the MBS area of the current scheduling window

For each enhancement layer I across All stream do

Compute $\rho i = Ts - Rs \; and \; \emptyset i = (Qs - MBS)$

Select K largest $\frac{\emptyset_i}{\rho_i}$ such that $\sum_{i \in K} \rho_i < PF - \sum rs - 1$

Determine lower bound $Q_{0=} \sum_{T}^{i \in K} \varphi_i + \sum qS1$

Compute scale factor K = $\in Q0/S$

Scale the quality values such that $q''_{sl} = \frac{q_{sl}}{K}$

For q=1 to 2Q do

For s = 1 to S do

If S is I, Compute $R(s, q)$

Else, Compute MBS

Back track the table $R(s, q)$ to find the sub streams S*

Arrange sub stream in ascending order of B_s/T_s

Allocate $\sigma_s = \min \frac{B_s}{T_s} frame\ to\ stream\ S$

Update $B_s = B_s + \sigma_s * F - \sigma_s TTs$

If no valid Allocation found do

Find Sub stream (I, s) such that $q_{si} = \min s \in s, i \epsilon l, \{qsi\}$

Discard Sub stream (l, s)

Stop

The Sub stream Selection Algorithm (SSA) algorithm causestal any buffer overflow or underflow**.** The estimate parameter happening the quality of answer and also on the time competence of this procedure. The estimate parameter can be assumed of as a knob for modification the trade-off between explanation quality and answer computation spell. We first vary the estimate parameter average excellence of the conventional data's damages as the estimate parameter increases.

3.3 Data handover process based on MBS:

In this paper, we adopt the MBS definition of network. It is based on the transit delay of successive packets between the entry and the exit network nodes. Let T_j represents the delay experienced by the packet going through a queue. The difference of transit time between two consecutive packets of a tagged flow can be written as

$$J_j = T_{j+1} - T_j$$

Which can be positive or negative. The average end-to-end delay jitter is then given by the expected absolute value of this random variable

$$\mathrm{J} = E[\,T_{j+1} - T_j]$$

We consider a single node with infinite buffer and a FCFS discipline. There are different streams of packets arriving to this node. Define,

• The total arrival rate

• The link bandwidth

• The total traffic load

In we have shown and validated by simulation that the end-to-end jitter of a tagged flow produced by a single node can be approximated by the following formula:

$$J = \frac{1}{C-\gamma}(1-\varepsilon^{\frac{1-\rho}{\rho}}\left(\frac{1-\rho}{\rho}+\varepsilon^{\frac{1-\rho}{\rho}}\right))$$

Note that this formula depends on the traffic parameters only through the bandwidth, the arrival rate, and the traffic load. On the other hand, the relation that connects the loss probability BT, the average overall throughput XT of the resource, and the arrival rate is given in [2]

$$B_T = \frac{\gamma - X_T}{\gamma}$$

From equations, we conclude the relationship between the arrival rate, the throughput and the loss rate:

1- When the overall throughput increases then the jitter decreases.

2- When the overall throughput increases then the loss rate decreases.

3- When the overall packet loss increases then the jitter increases.

These relationships allow understanding and explaining some phenomena, observed in live measurements, which can highly affect the end-to-end network performance. Typically, we can conclude that the jitter has a behavior opposite that of the throughput, and the same behavior as the loss rate. Running time of our procedures by changing the problematic size in two ways. First we keep the preparation window dimensions fixed and increase the amount of streams. Next scheduling the quantity of streams fixed and proliferations the preparation window. We compare the implementation times of our procedures to that of the optimum.

4. RESULT DISCUSSION:

A SSA to obtain the stationary if the forward/non-forward location of both node is strong on the static opinion only; if not, it is lively. The stationary broadcast procedure is a singular case of the active one. The difference is that the advancing node set resulting from static views container be used in any diffusion while the one resulting from dynamic interpretations is in general used in a detailed dissemination.

4.1 Throughput Performance

This remains the production of entire number of usual data packets separated by total number of sent data packets.

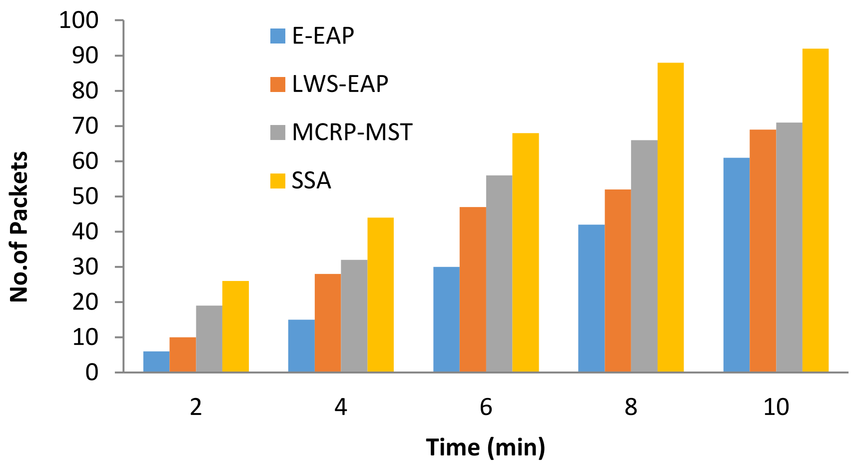

Fig1. Performance of throughput

This metric stretches an approximation of how well-organized a routing procedure is, since the amount of routing packages sent per statistics packet gives an idea of how well the protocol keeps the routing in order updated. The developed the Usual Sending Load metric is, the developed the upstairs of routing packets and so the lower the competence of the protocol.

4.2 The Data Delivery Fraction:-

The packets are transported from source to destination on their network. It is intended by dividing the number of data conventional by ending state finished the quantity set originated from initial point on network.

$$PDF = (Pr/Ps)*100$$

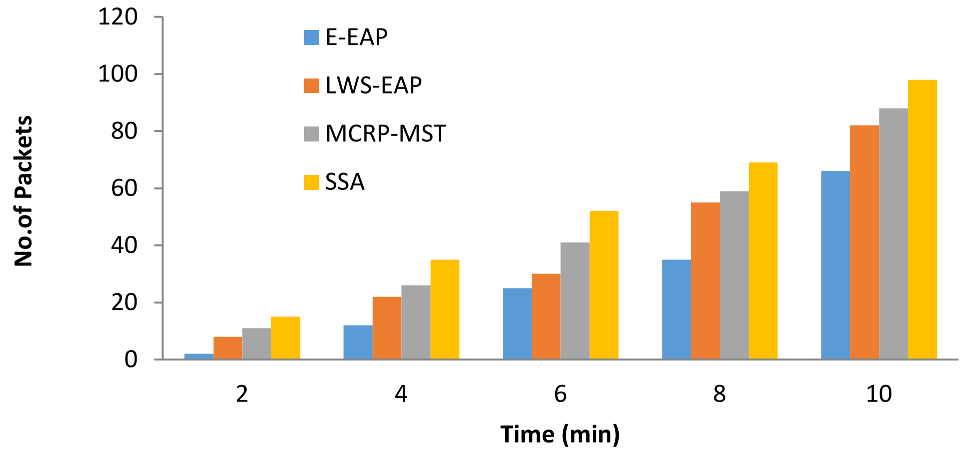

Fig 2. Performance of Delivery ratio

Where Pr is total Data received & Ps is the total data sending on their network.

4.3 HANDOVER DELAY PERFORMANCE:

The handover expansion initiated by the portable station in network. When the mobile posting initiates entrust request the ASN demand and confirms the collection important and then initiates the trust module. The entrust module achieves the three way handclasp procedure with the portable location in network. After the procedure of pre verification the handover module sends a process acknowledgement. By receiving this acknowledgement the mobile station directs the node handover request to the handover module.

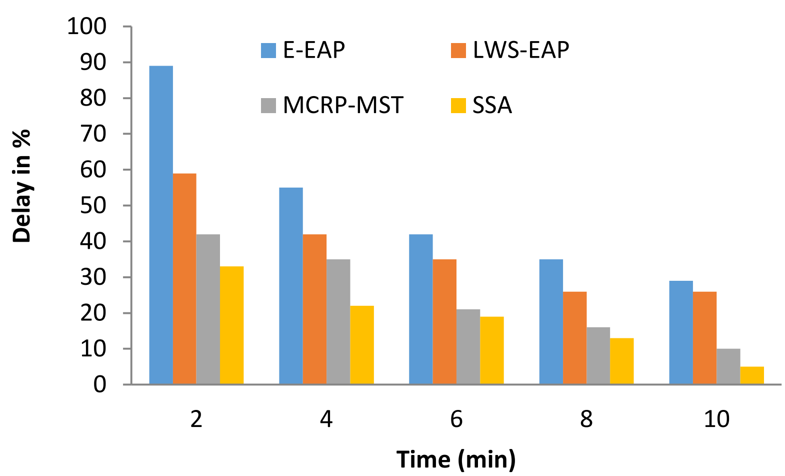

Fig 3. Performance of handover

A difficulty is with the determination of unsuccessful the call container be for the time being interrupt or even ended strangely. Technologies which make use of hard handovers, more often than not have events which can reinstate the association to the foundation cell if the association to the objective cell cannot be finished. However re-establishing this suggestion may not for all time be possible and level at what time likely the operation may cause a temporary break to the describe.

4.4 The End-to-End delay:-

They consume compute a regular number of delay on network, it comprises all conceivable delay produced by protecting through route uncovering latency, line up at the border queue, retransmission delay on intermediate node control, spread and move time.

$$D = (Tr - Ts)$$

Where Tr is receive Time and Ts is sent Time.

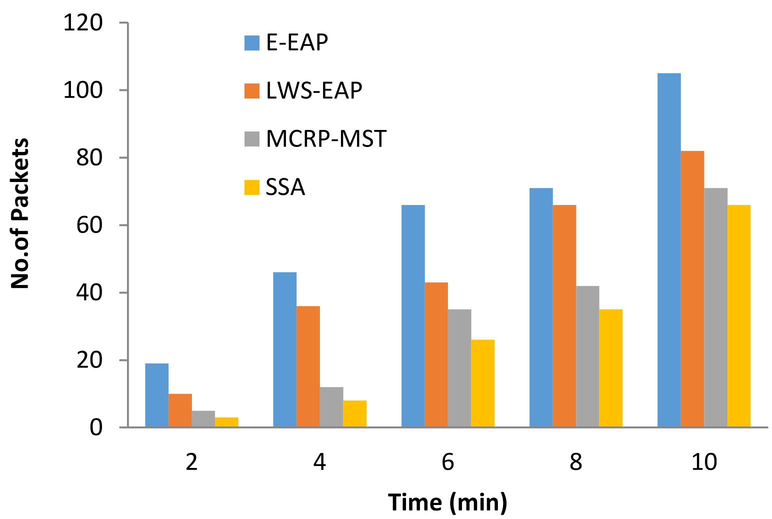

Fig4. Performance of delay ratio

That time taken a data packet to be across a WiMax network since twitch to termination point on the network. To minimize the liveliness ingesting by means of different algorithms. These procedures proposition a decent solution, then they select the enlargement with the advanced remaining energy in the group as the collection data for the next round. However, this does not promise the maximum continuation of the general network lifetime. Therefore, if the node through the uppermost outstanding energy is a node situated at the lateral of the network this container lead other nodes to spend substantial amounts of energy to spread that node, which cannot be liveliness well-organized for the complete network.

5. CONCLUSION:

The addition of SSA based handover to be a talented approach due to their same nature and complementary features. In this paper, we examine several significant issues for the interworking of Mobile WiMAX. We discourse a tightly coupled interworking construction. Further, a unified and proactive perpendicular handoff scheme is intended based on the building with aims to provide always the greatest SSA for users. Both the presentation of applications and system conditions are measured in the handoff process. Moreover, we originate evaluation procedures to estimate the situations of both WiMAX and SSA to develop in terms of available bandwidth and packet delay. A replication study has confirmed that the proposed structures can keep positions always being best associated.

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